NOISE CANCELLATION

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Abstract—

One of the fundamental challenges affecting the performance of communication systems is the undesired impact of noise on a signal. Noise distorts the signal and originates due to several sources including, system non-linearity and noise interference from adjacent environment. Conventional communication systems use filters to cancel noise in a received signal. This paper consists of how to remove unwanted ambient sounds and proposes a technique for reducing noise from a signal's time series using adsp algorithms. Here we removed the noise in an audio file using digital filters.

I. INTRODUCTION

Noise cancellation (NC), or active noise reduction (ANR), is a method for reducing unwanted sound by the addition of a second sound specifically designed to cancel the first. The concept was first developed in the late 1930s, later developmental work that began in the 1950s eventually resulted in commercial airline headsets with the technology becoming available in the late 1980s. The technology is also used in road vehicles and in mobile telephones.

Noise cancellation is one of the main challenges in communication systems and has been a focus of study for many years. Techniques and technologies for noise cancellation have emerged from the need to mitigate unwanted noise present in the desired signal. Noise distorts the received signal in a random manner and occurs because of several sources. In addition, there are several other factors affecting the received signal such as crosstalk and electromagnetic interference.

II. THEORY

A. Explaination

Sound is a pressure wave, which consists of alternating periods of compression and rarefaction. A noise-cancellation speaker emits a sound wave with the same amplitude but with inverted phase (also known as antiphase) to the original sound. The waves combine to form a new wave, in a process called interference, and effectively cancel each other out – an effect which is called destructive interference.

Modern active noise control is generally achieved through the use of analog circuits or digital signal processing. Adaptive algorithms are designed to analyze the waveform of the background aural or nonaural noise, then based on the specific algorithm generate a signal that will either phase shift or invert the polarity of the original signal. This inverted signal (in antiphase) is then amplified and a transducer creates a sound wave directly proportional to the amplitude of the original waveform, creating destructive interference. This effectively reduces the volume of the perceivable noise.

A noise-cancellation speaker may be co-located with the sound source to be attenuated. In this case it must have the same audio power level as the source of the unwanted sound in order to cancel the noise. Alternatively, the transducer emitting the cancellation signal may be located at the location where sound attenuation is wanted (e.g. the user's ear). This requires a much lower power level for cancellation but is effective only for a single user. Noise cancellation at other locations is more difficult as the three-dimensional wavefronts of the unwanted sound and the cancellation signal could match and create alternating zones of constructive and destructive interference, reducing noise in some spots while doubling noise in others. In small enclosed spaces (e.g. the passenger compartment of a car) global noise reduction can be achieved via multiple speakers and feedback microphones, and measurement of the modal responses of the enclosure.

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B. Applications

Applications can be "1-dimensional" or 3-dimensional, depending on the type of zone to protect. Periodic sounds, even complex ones, are easier to cancel than random sounds due to the repetition in the wave form.

The Protection of a "1-dimension zone" is easier and requires only one or two microphones and speakers to be effective. Several commercial applications have been successful: noise-cancelling headphones, active mufflers, anti-snoring devices, vocal or center channel extraction for karaoke machines, and the control of noise in air conditioning ducts. The term "1-dimension" refers to a simple pistonic relationship between the noise and the active speaker (mechanical noise reduction) or between the active speaker and the listener (headphones).

In some cases, noise can be controlled by employing active vibration control. This approach is appropriate when vibration of a structure produces unwanted noise by coupling the vibration into the surrounding air or water.

C. *Basic Theory*

We have used the butterworth filters to filter the noises from an input audio signal and we also used fft, fftshift to plot amplitude spectrum for finding the noise frequencies and plotting.

1. Butterworth filter

A Butterworth filter is a type of signal processing filter designed to have a frequency response as flat as possible in the passband. Hence the Butterworth filter is also known as "maximally flat magnitude filter". It was invented in 1930 by the British engineer and physicist Stephen Butterworth in his paper titled "*On the Theory of Filter Amplifiers*".

The frequency response of the Butterworth filter is flat in the passband (i.e. a bandpass filter) and roll-offs towards zero in the stopband. The rate of roll-off response depends on the order of the filter. The number of reactive elements used in the filter circuit will decide the order of the filter.

The inductor and capacitor are reactive elements used in filters. But in the case of Butterworth filter only capacitors are used. So, the number of capacitors will decide the order of the filter.

$$f_{cutoff} = \frac{1}{2\pi RC}$$

The generalized form of frequency response for nth-order Butterworth low-pass filter is;

$$H(j\omega) = \frac{1}{\sqrt{1+\varepsilon^2(\frac{\omega}{\omega_C})^{2n}}}$$

Where,
n = order of the filter,
ω = operating frequency (passband frequency) of circuit
ω_C = Cut-off frequency
ε = maximum passband gain = A_{max}

The below equation is used to find the value of ε.

$$H_1 = \frac{H_0}{\sqrt{1+\varepsilon^2}}$$

Where,
H_1 = minimum passband gain
H_0 = maximum passband gain

2. FFT and FFTSHIFT

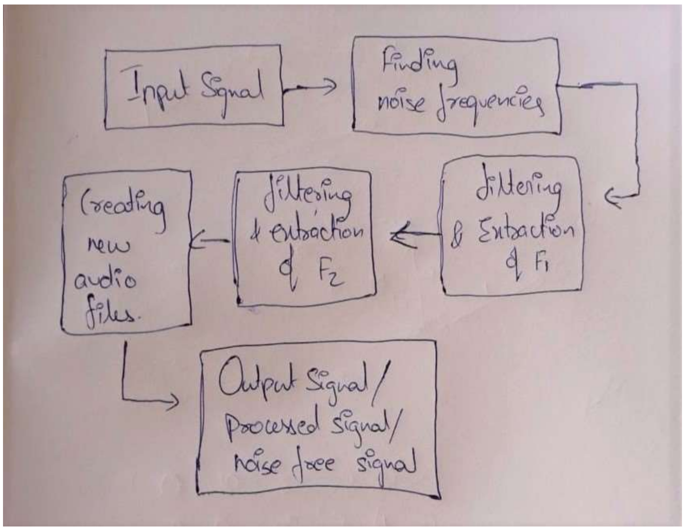
The fft function in MATLAB uses a fast Fourier transform algorithm to compute the Fourier transform of data. Consider a sinusoidal signal x that is a function of time t with frequency components of 15 Hz and 20 Hz. Use a time vector sampled in increments of 1 50 of a second over a period of 10 seconds.

fftshift is useful for visualizing a Fourier transform with the zero-frequency component in the middle of the spectrum. For vectors, fftshift(X) swaps the left and right halves of X. For matrices, fftshift(X) swaps the first quadrant with the third and the second quadrant with the fourth

III. METHODOLOGY

At first, the input audio signal is read by the command 'audio read' and then we take frequency and sampling frequency from the given audio signal then we calculate the small parameters like length, its time matrix and its size now we plot the original one-sided amplitude spectrum audio for finding the noises in the input audio signal.

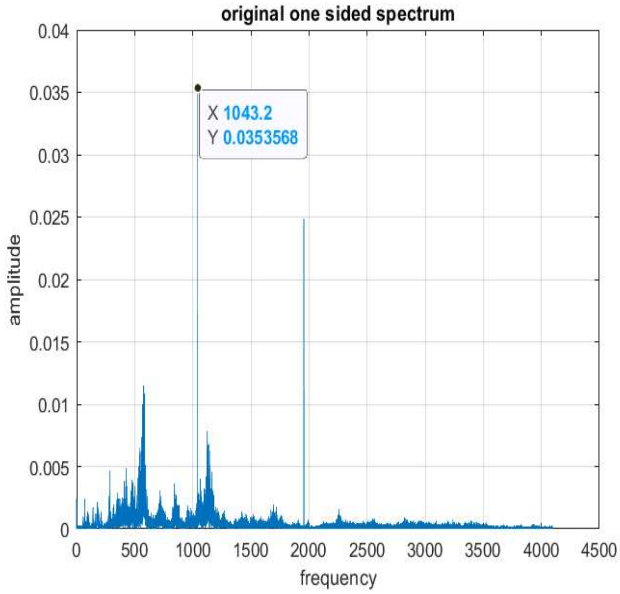
Now we start by removing the noises one by one from the graph of one side amplitude spectrum we can find the range of each noise we remove the noise using butterworth band-stop filter and then we filter to get output of signal after removing the one noise we extracted the noise using butterworth band pass filter and then we filter to get the extracted noise. In this way, we repeated the same procedure for other noises and we finally get the processed audio signal after filtering all the noises. We plotted graph at every stage of this process.



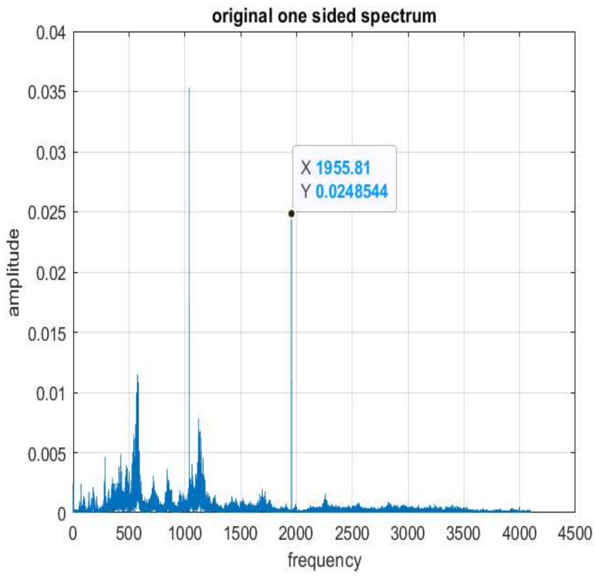
The above indicates the block diagram of our algorithm

IV. RESULTS

We got two noises from an input audio signal through the single-sided spectrum. In single-sided spectrum we get the noise frequencies as shown in below graph

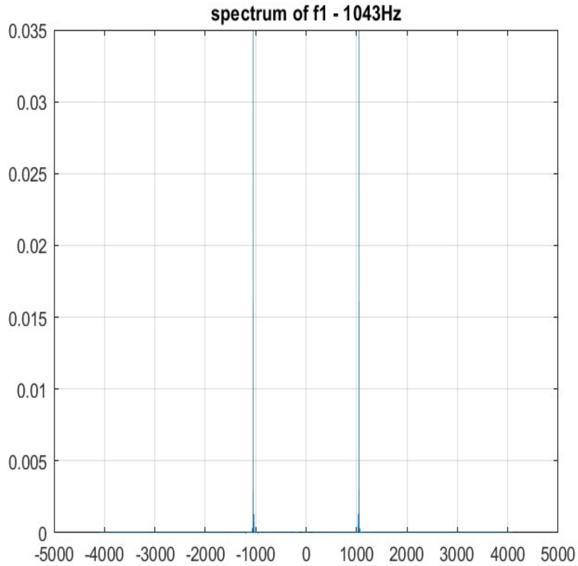


In this graph we can clearly see that the frequency of first noise is 1043 HZ so we kept the range of frequency for the filtering process as 1040-1050 HZ

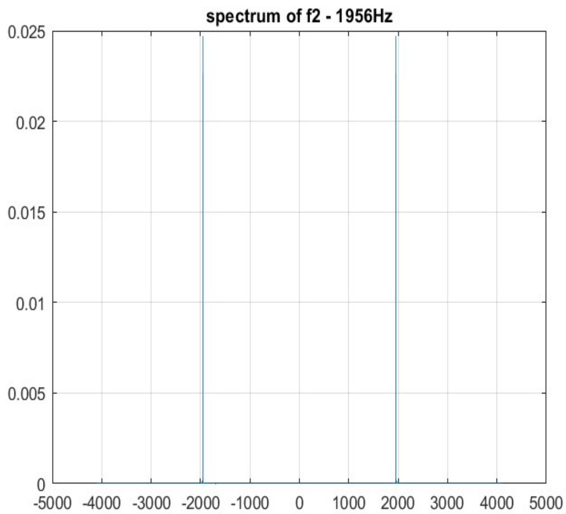


In this graph we can clearly see that the frequency of second noise is 1956 HZ (approx) so we kept the range of frequency for the filtering process as 1950-1960 HZ

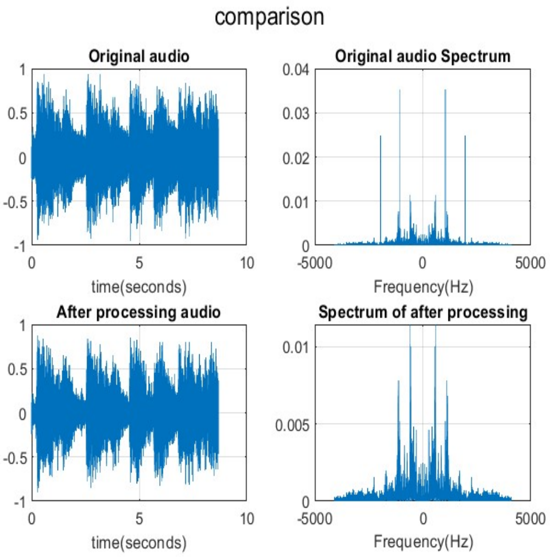
The spectrum of extracted noises is as follows



Here we can clearly see that accurate frequency of noise f1 that is 1043 HZ.



Here also we can clearly see that accurate frequency of noise f2 that is 1956 HZ



In above we have the comparison graph of input audio signal that is the original audio and the output obtained by implemented our algorithm. In above graph we also have the comparison between the amplitude spectrum of input and output audio signals.

We can clearly see that the noise frequencies in the original spectrum which are of more amplitude after the implementation of our matlab algorithm we can see that there are no noise frequencies in the after-processing spectrum that is there is no high amplitude samples in the graph.

V. CONCLUSION

In this paper we have discussed the importance of noise cancellation, application, it's implementation of matlab algorithm. We implemented the algorithm by using butterworth bandpass and stop filters with that the noises we found are stopped and filtered to get the noise free processed audio signal.

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