STATISTICAL AND SALIENT FEATURES ANALYSIS ON ACOUSTIC SIGNAL FOR MEDICAL AND EDUCATIONAL APPLICATIONS.

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APPROVAL

This Thesis titled "STATISTICAL AND SALIENT FEATURES ANALYSIS ON ACOUSTIC SIGNAL FOR MEDICAL AND EDUCATIONAL APPLICATIONS", submitted by Muhammad Rakib Prodhania ID: 171-19-1945, Md Emon Chowdhury ID: 181-19-2022, Md Nayeem Shah ID: 181-19-2017 to the Department of Electronics and Telecommunication Engineering, Daffodil International University has been accepted as satisfactory for the partial fulfilment of the requirements for the degree of B.Sc. in Electronics and Telecommunication Engineering (E.T.E) and approved as to its style and contents. The presentation has been held in August 2022.

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We hereby declare that this thesis has been done by us under the supervision of **Professor Dr A K M Fazlul Haque, Professor,** Department of **E.T.E.** We also declare that neither this thesis nor any part of this thesis has been submitted elsewhere for the award of any degree or diploma.

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We would like to express our gratitude to him for his kind help and unwavering support in completing our thesis and also thank to other faculty member's department of Electronics and Telecommunication Engineering.

DEDICATION

A fire and related explosions at a container depot in Sitakunda Upazila, Chittagong District, Bangladesh killed at least 47 persons and injured about 450 others on the night of June 4, 2022. The event occurred at the BM Container Depot in Sitakunda Upazila's Kadamrasul region. After a fire broke out in the loading area at around 21:00 BST, a major explosion erupted at around 23:45 BST, setting off a chain reaction of explosions that spread throughout the depot due to the igniting of chemicals housed in the containers. This thesis is dedicated to their departed souls. May Allah rest them in peace.

ABSTRACT

In this report, statistical and salient features have been extracted for acoustic signals in medical and educational applications. In recent 2 to 3 years, the whole world has witnessed a lockdown in the COVID-19 situation. The world's communication was down physically during this period, but they had connected through online communication. However, the main reason is that in this COVID-19 situation, online education is becoming much more popular. Now, most people are familiar with online education. However, most learners faced a common problem. The instructor's voice is not clear. When the instructor recorded the audio or live online education, the audio signal is mixed with unwanted noise like instrumental, environmental, etc. Telemedicine has become a major part also in this pandemic and doctors can communicate with the patient over voice and instruct them what to do in critical situations when a patient is unable to go to the hospital or the circumstance is not letting the patient go to the hospital. In both cases, students and patients or doctors face the same kind of problem, as a result, both medical and educational sides are being focused. In this work, especially the notch filter which has been used provides a better result as a solution and it eliminates the 60 Hz instrumented signal. The graphs are used for comparison and it is being clarified why the notch filter is used and how it performs better.

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CHAPTER 01: INTRODUCTION

1.1 Overview

Audio signals which are received or any audio which is heard via many of the receivers of an audio signal is not noise free. So, when that audio signal is heard, audio distortion and other noises are mixed with the signal and the real audio signal can't be heard. There are many cases where noise-free audio is required but when the audio is recorded, the recorder itself or the surroundings create an extra noise which is mixed with the real audio signal and it makes the real audio signal weak or distorted. The exact result which is expected can't be found in these cases. Moreover, during the pandemic, there are so many online classes and meetings have been held via multiple sources like Google Meet, Zoom, Microsoft Teams etc. where the voices needed to be clear but there was noise also mixed with it as various people are accessing the internet from various region with a higher pitch of sound density. To solve this problem. Audio de-noising or making it free from the noises is the main purpose of this paper. There are many ways like Fourier Transform and wavelet transform which can be used to analyze and recover the audio signal. Also, there is another application of this work is in telemedicine. This is an emerging thing in Bangladesh. When doctors cannot go to the patient physically or vice versa and they communicate via audio calls over the mobile phone or the internet. In this case, audio is playing a big role. In this paper, wavelet transformation is used and if the signal has too much noise, wavelet transformation is done 2 or 3 times to make sure the best quality of audio.

Signals like audio and non-stationary signals can be analyzed and spectral properties along with temporal properties can be analyzed via the wavelet method. Making noise-free audio signals is not an easy task. Since there are so many sources of noise present, effectiveness is a very big ask in this type of consequence.

1.2 Aims and Objective

This thesis aims to find out some solutions like

- To find a solution for the de-noising signal at the maximum level.

- Maintain SNR as low as possible.
- Scale the components in an assigned frequency range.
- To gain noise-less signal for voice signals.
- Extract exact voices in the pandemic and make sure the best quality of noise-free signal for Telemedicine, online classes and meetings during the pandemic.
- In the Medical sector to provide a noise-free signal.
- Earthquake signal analyzer to get the actual result.

1.3 Motivation

There are many ways to de-noise an audio signal but most of them are done with Fourier Transform, but this does not provide a good result. As a result, the wavelet transform is considered an alternative to the Fourier Transform and this is more reliable than the Fourier Transform. This process will offer a noiseless audio signal to the audience. MATLAB is used as the tool to determine the level of noise. The process will be carried out multiple times to get an actual noise-free signal. In the other works which have been done, only Fourier Transformation or Wavelet Transformation has been used. None has shown the effectiveness of the potential of a notch filter in de-noising [1]. In this work, with wavelet transform notch filter is also discussed and how the notch filter is better has been proved via the data, statistics and graph.

1.4 Expected Outcome

- Offer noiseless audio signal to the audience in terms of class to the students and in terms of telemedicine to the patients.
- Carry out better voices for online meetings and classes on various platforms.
- The solutions will lead to a better and more effective solution for fellow researchers.

1.5 Report Layout

In the 1st chapter, the aims and objectives, motivation and the expected outcome have been discussed. In the 2nd chapter, the discussion is about the circumstances and details of related works and papers on this topic. The third chapter is all about the findings and solutions of the following topic of the thesis and what exactly is trying to be done. The 4th chapter analyzes the overall scenario and how the signal will be made noise-free and the comparison with noised signal and noise-free signal. The last chapter is a conclusion. This chapter describes the summary and the future scope will indicate where further investigation can be done and fellow researchers can take it to the next level.

Chapter 2: Background

2.1 Finding the problem

A recent scenario saw that the audio demand is increasing day by day in terms of video. In recent 2 to 3 years, the whole world lockdown the COVID-19 situation. The world's communication was down physically during this period, but they had connected through online communication. When people communicate via the Internet, they find a problem: noisy audio. However, the main reason is that in this COVID-19 situation, online education is becoming much more popular [2]. Now, most people are familiar with online education. However, most learners faced a common problem. The instructor's voice is not clear. When the instructor recorded the audio or live online education, the audio signal is mixed with unwanted noise like instrumental, environmental, etc. So, this problem is focused and the solution is based on this.

2.2 Related Works

Noise reduction on acoustic signals is a type of technology that can be used to decrease noise or undesired signals from the original signals. For the removal of disturbances from digital audio streams, a variety of noise reduction approaches have been proposed. However, the efficiency of those methods is limited. The usage of wavelet transform is explored in this work as part of a discussion of noise reduction strategies for acoustic signals. It presents an overview of how to minimize undesired noise and produces significantly better results for future research projects. In addition, the wavelet-based de-noising approach is used to put the experimental results into practice [3].

Some works have been researched and it denotes that there were some scopes to improve. Priyanka Khattar et. al. [4] illustrated that audio denoising is very tough and they used Daubechies and Haar; forms of the wavelet transform. It also states that the Fourier transform considers the audio signal as stationary, while actually, it is non-stationary. So, it tells that wavelet transform is the more efficient one. In another research, AKM Fazlul Haque et. al. [5] stated that ECG is very sensitive to audio. ECG is a very weak signal and it uses an amplifier. The power line noise needs to be eliminated to get a better result. These two works states which de-noise policy is better and why the ECG signal needs to be very accurate. But no work has integrated these two matters. In this thesis, these two parts have been integrated and how medical applications like ECG or other sensitive audios can be de-noised is stated and to eradicate the power line noise of 60Hz a notch filter has been used. Wavelet transform performs better when a notch filter is added to it and this is the outcome of this work.

2.3 Necessity of wavelet transformation over Fourier transformation

Wavelet transform (WT) is very powerful compared to Fourier transform (FT) because of its ability to describe any signals both in the time and frequency domain simultaneously while FT, describes a signal from the time domain to the frequency domain.

Chapter 3: Methodology

3.1 Finding the problems

In this thesis the problem statement is that Fourier transform is not reliable in all the cases and to get rid of the noises created at the recording or sound generating end is continuous and the sources of noise and their depth are different, so to get the exact clear voice is the main target and it has been focused in throughout the thesis.

3.2 Algorithm and Flow chart of the process

Here the algorithm and flow chart of a de-noising signal has been discussed. First, look at the flowchart:

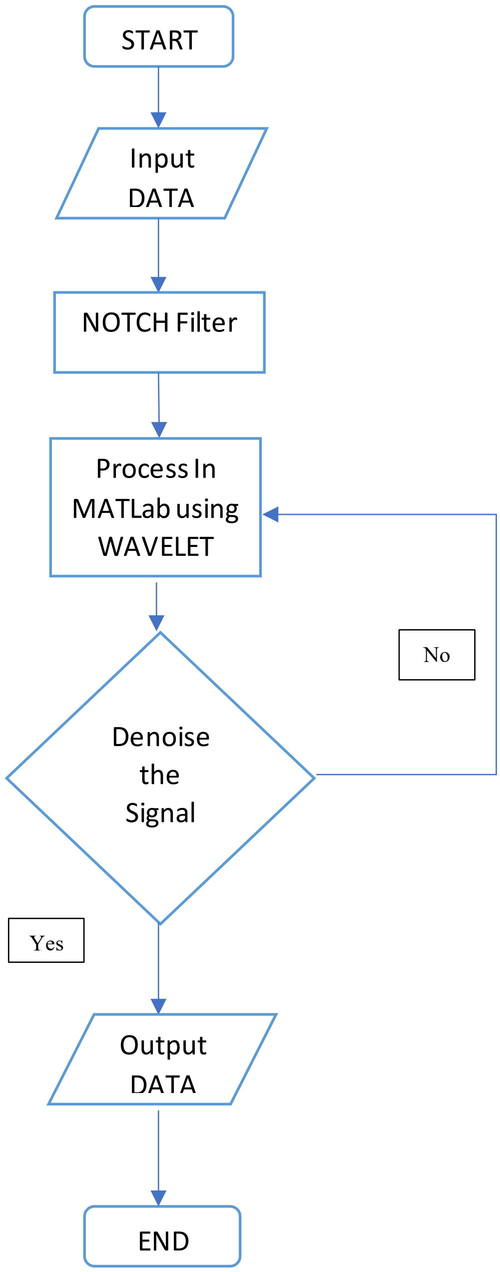

Fig: 3.1 Steps of De-noising.

So, the steps include-

1. The process starts.
2. Data will be given as an input (Here data means signal which needs to be de-noised).
3. The data will be fed to the Notch filter.
4. Then data will be fed to MATLAB as a wavelet transform process.
5. If the noise is totally nullified it will end the process by giving proper output.
6. If the noise is not nullified or has not reached a satisfactory level the process will be carried out one more time.
7. This will continue until nullifying the noise level to a certain acceptable stage.
8. Thus, it will carry out the output and end the process.

3.3 Experimental Result

The experiment was held in MATLAB using WAVELET Transform. The detailed method is described here.

MATLAB is used here as software and the audio signal which needed to be made noise free is given as an input and the expected output will be available after going through this code. At first, an audio signal is given as an input into MATLAB and converted audio signal into .Mat file by using code. Then it is needed to go to the MATLAB Wave menu Simulator and then WAVELET transforms available in MATLAB and loads the signal.

3.4 Input Signal

Here two sets of data have been taken, male data 1, female data 1, male data 2 and female data 2. The first set of data has been taken in a high noisy area and the second set of data has been taken in a moderately low noisy environment. This will help to understand the difference and how wavelet transform and the notch filters perform in a noisy environment and a less noisy environment.

Here firstly input audio signal is taken and the graph of the input audio signal looks like above in the MATLAB.

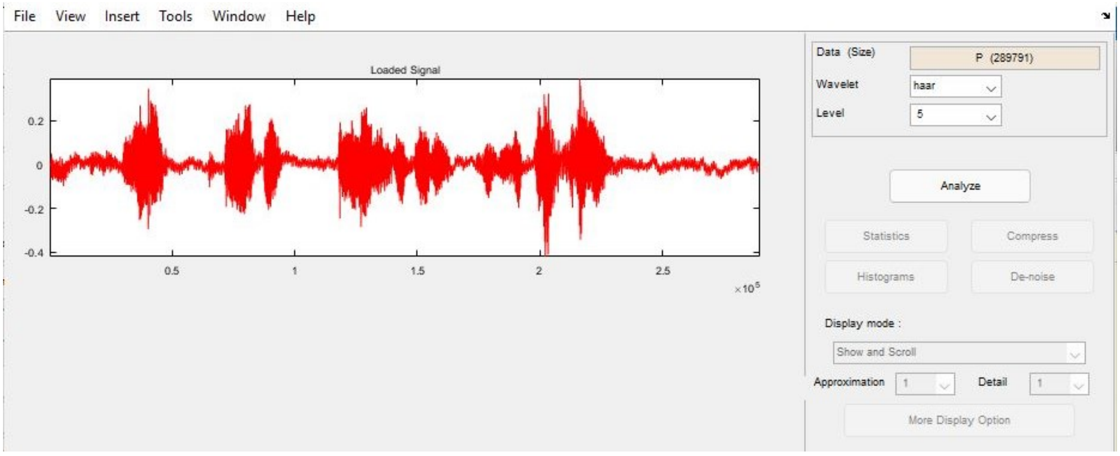

Fig 3.4.1: Experimental result input code

Then “dB wavelet transforms” at levels 1,2,3,4,5 is performed to analyze the signal. After analyzing the signal, the output will be like the below-

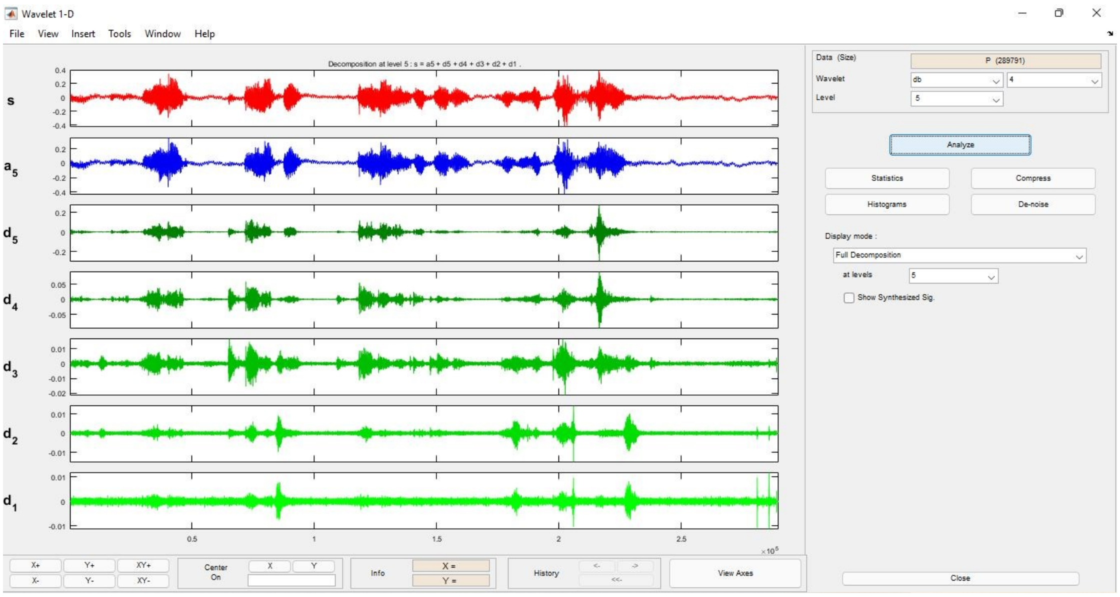

Fig 3.4.2: Analyzing signals in 5 different levels

The signal which was taken is loaded in MATLAB and five levels of analysis are done here. There are denoted as d1, d2, d3, d4, and d5. Four wavelets are present and they have been performed dB wavelet transforms at 5 levels. Full decomposition mode is selected to show the whole data set.

3.5 Statistics from the data

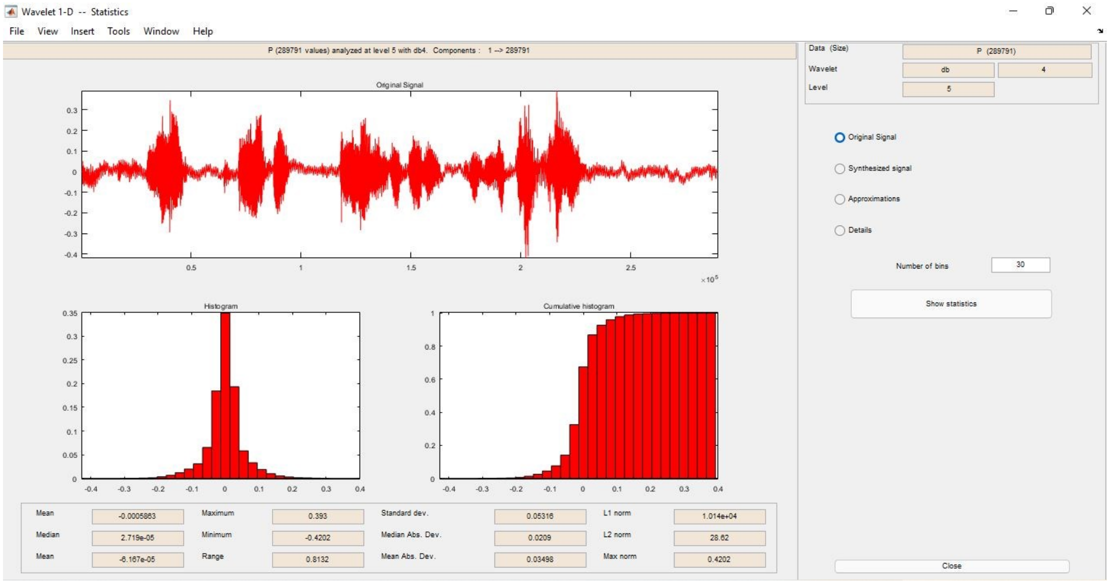

Fig 3.5: Statistics from the data

In this figure, statistics are shown and two histograms are shown, one is a histogram and the other one is a cumulative histogram.

3.6 Denoising method

Denoising is now performed and from MATLAB Fixed from threshold is selected and noise structure is selected. Now denoising is done by selecting the thresholding method "Fixed from threshold" and selecting noise structure "soft or hard" according to input signal noise structure.

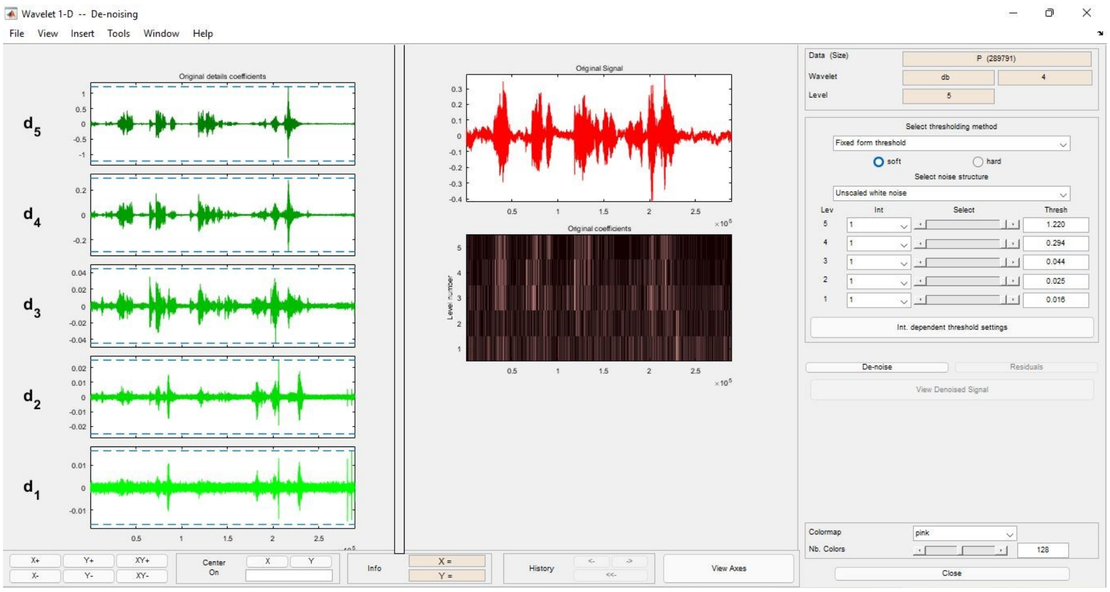

Fig 3.6.1: Original and de-noised signal

After selecting the thresholding method, we get to perform denoise the signal and get the output. Original signal and original co-efficient can be seen through the pictures.

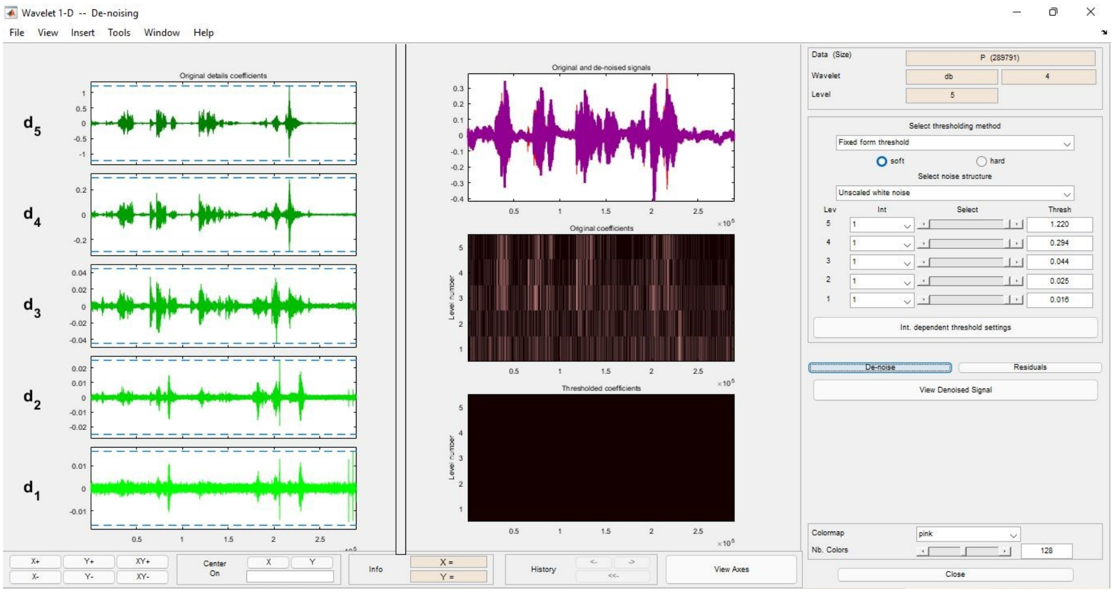

Fig 3.6.2: Original and de-noised signal with parameters

Here different types of noise structures are present – Unscaled white noise, Scaled white noise, and Non-white noise [6]. It needs to be selected from them according to signal noise structure. In the picture threshold co-efficient can be seen also.

Now the procedure is performed to view the De-noise signal-

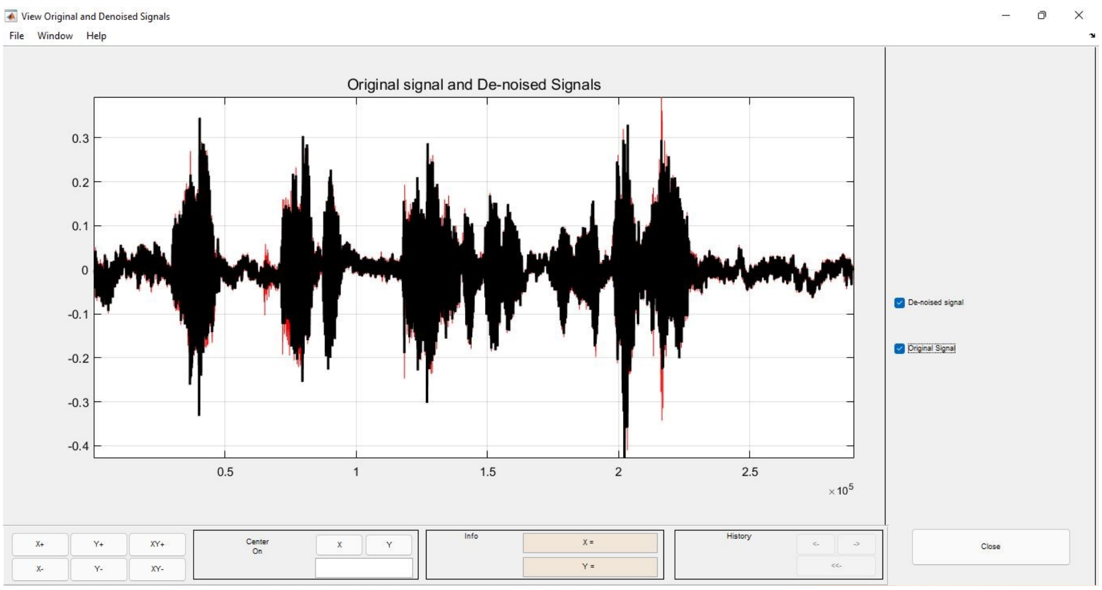

Fig 3.6.3: Comparing Original and de-noised signal

This show the denoised signal, still the signal has some noise, Here the red mark is the input signal and the black mark is the denoise signal.

As the signal isn't fully denoised, the procedure needs to be followed again. And generate the last denoise signal into .Mat file. Then again load the signal.

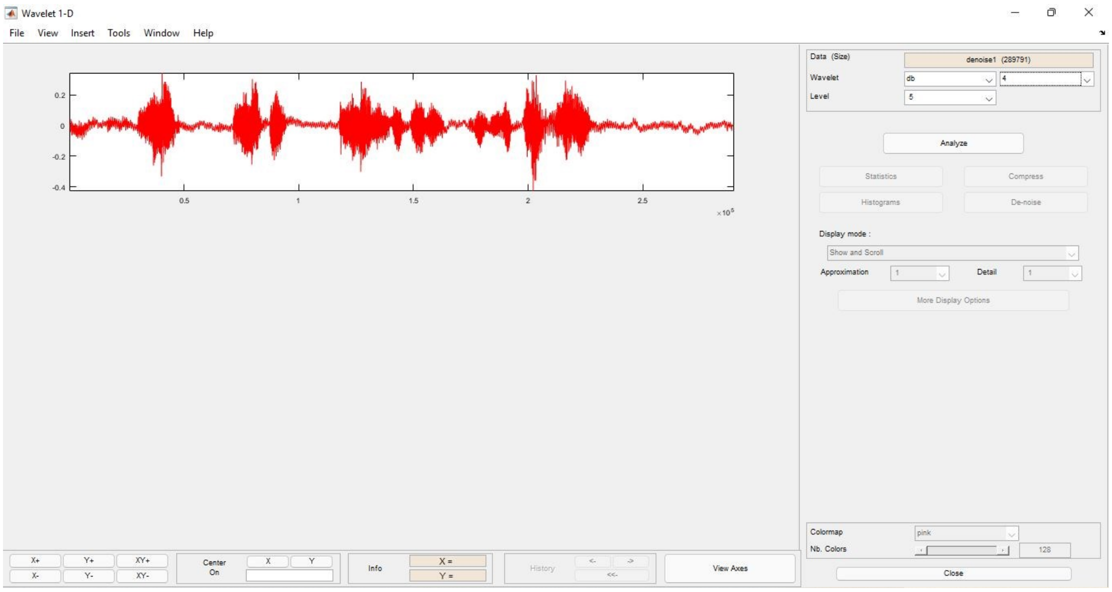

Fig 3.6.4: 2nd-time signal which shall be loaded

Then again perform dB wavelet transform is performed at levels 1,2,3,4,5 to analyze the signal. After analysing the signal the out will be like this-

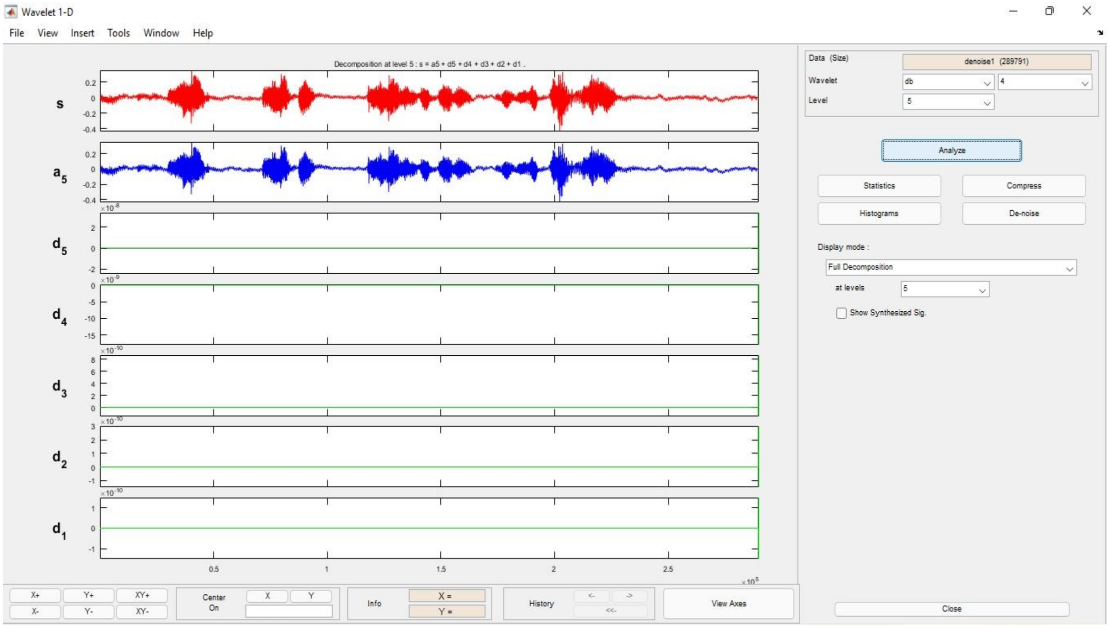

Fig 3.6.5: dB wavelet transforms in 5 levels

After analysing, the loaded signal again the statistics are gotten. Here some differences between the first loaded signal can be seen.

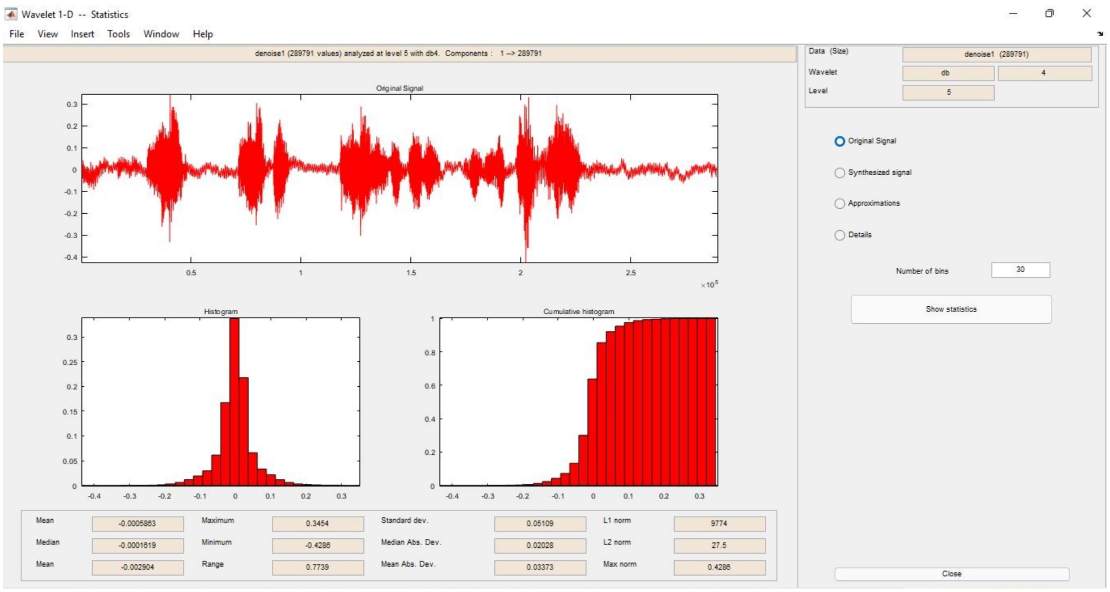

Fig 3.6.6: Histogram and Cumulative Data

Now denoising is done by selecting the thresholding method “Fixed from threshold” and selecting noise structure “soft or hard” according to input signal noise structure. After selecting the thresholding method, it is needed to perform denoise the signal and get the output.

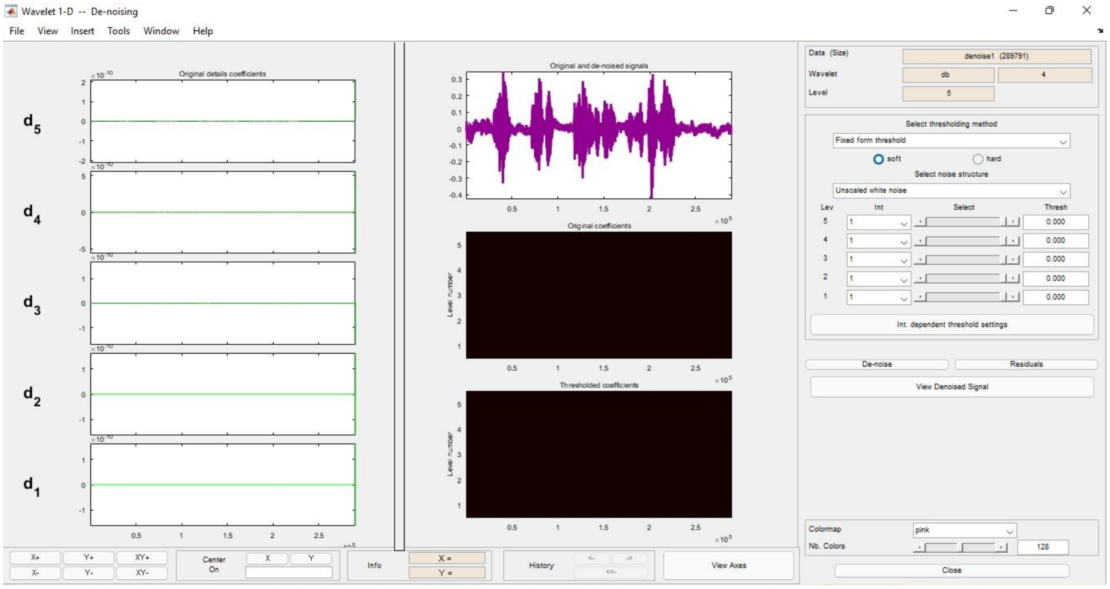

Fig 3.6.7: dB wavelet transforms with co-efficient

This is the penultimate stage to get the final denoised signal. Now the fully denoise signal is present to view-

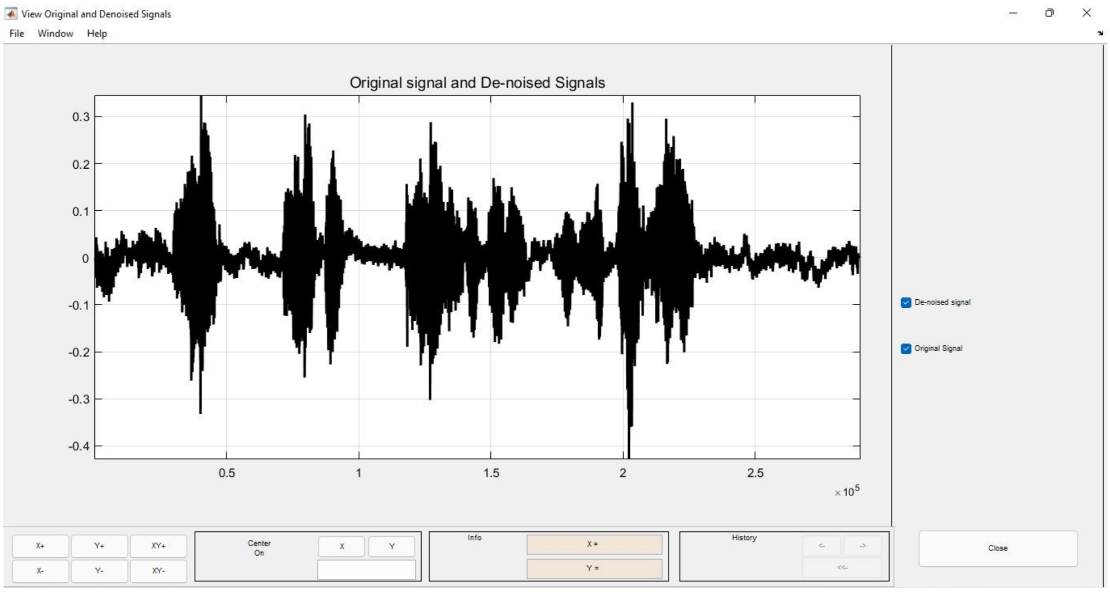

Fig 3.6.8: Final output signal after 2nd phase

This is the fully denoised signal. After repeating the procedure 2 times, an acceptable result has been found, so no more repetition is needed and the signal can be considered noiseless.

Here some comparison between the input signal and after denoising signal is shown and also comparing the male and female audio signal-

Two sets of data are taken, male voice and female voice which are denoted as Male data 1 and Female Data 1. In terms of the Male data set, it can be seen that denoising is done in two steps. It is denoted as Denoise 1 and fully denoise and Female Data 1 is fully denoised at level 1.

Table 3.6.1: Table of Male Data 1

Data Type	Input	Denoise 1	Fully Denoise
Mean	-0.0005863	6.519e-10	5.543e-10
Median	2.719e-05	0	-5.864e-06
Mean	-6.167e-05	-0.001597	-0.003689
Maximum	0.393	0.2937	0.1389
Minimum	-0.4202	-0.2778	-0.1203
Range	0.8132	0.5715	0.2592
Standard dev.	0.05316	0.1788	0.01014
Median Abs. Dev.	0.0209	0.002977	0.001511
Mean Abs. Dev.	0.03498	0.008879	0.004785
L1 norm	1.014e+04	2573	1387
L2 norm	28.62	9.625	5.46
Max norm	0.4202	0.2937	0.1389

From the table, it can be seen that it needed to be denoised in two steps.

For the first set, another piece of data is taken which is Female Data 1

Table 3.6.2: Table of Female Data 1

Data Type	Input	Fully Denoise
Mean	-0.0002002	-0.0002002
Median	0.0001689	0.0001711
Mean	-0.001969	0.01171
Maximum	0.3482	0.3881
Minimum	-0.3763	-0.3906
Range	0.7245	0.7787
Standard dev.	0.04579	0.04133
Median Abs. Dev.	0.01704	0.01646
Mean Abs. Dev.	0.02837	0.02598
L1 norm	6796	6226
L2 norm	22.42	20.23
Max norm	0.3763	0.3906

In this table, it can be seen that after only going through a one-time signal is denoised and it has reached an acceptable range.

To make the data more reliable another set of data is taken for test, which is denoted as Male data 2 and Female data 2. In terms of the Male data set, it can be seen that denoising is done in two steps. It is denoted as Denoise 1 and fully denoise and Female Data 1 is fully denoised at level 1.

Table 3.6.3: Table of Male Data 2

Data Type	Input	Fully Denoise
Mean	0.001034	0.001034
Median	0.0009376	0.0007578
Mean	0.0163	0.008486
Maximum	0.5209	0.5258
Minimum	-0.5231	-0.5445
Range	1.044	1.07
Standard dev.	0.08225	0.08022
Median Abs. Dev.	0.02118	0.02082
Mean Abs. Dev.	0.04877	0.04759
L1 norm	1.139e+04	1.139e+04
L2 norm	39.74	38.76
Max norm	0.5231	0.5445

From this data set, it can be seen that though the previously male voice had to go through two times completing the process, this time is done only once and it has reached an acceptable range.

Table 3.6.4: Table of Female Data 2

Data Type	Input	Denoise 1	Fully Denoise
Mean	0.0007703	0.0007703	0.0007703
Median	-0.005776	-0.005258	-0.005247
Mean	-0.03063	0.01398	0.01398
Maximum	1.048	1.278	1.278
Minimum	-1.04	-1.02	-1.02
Range	2.088	2.298	2.298
Standard dev.	0.2073	0.1241	0.1235
Median Abs. Dev.	0.03617	0.02988	0.02986
Mean Abs. Dev.	0.1024	0.06696	0.06662
L1 norm	2.662e+04	1.739e+04	1.739e+04
L2 norm	105.7	63.3	62.98
Max norm	1.048	1.278	1.278

This voice had to go through the process two times though the previous female voice had to complete the process only once. Two sets of data bring reliability and acceptance of the work and process which is done and the voice frequency changes from person to person and the noise which was along with it also can be different in various situations.

3.7 Notch Filter

The notch filter is used generally to remove a specific frequency of the signal or the narrow band of signals. In terms of the audio system, a notch filter is used to nullify the noise which has been created through the power grid line or the source itself [7]. In the previous wavelet transform method, the noise was eliminated but the notch filter gives a way better result. The procedure is the same as before but the output is more reliable than before. It can be compared and also it will be shown later. The basics are shown below pictures.

This is the input signal-

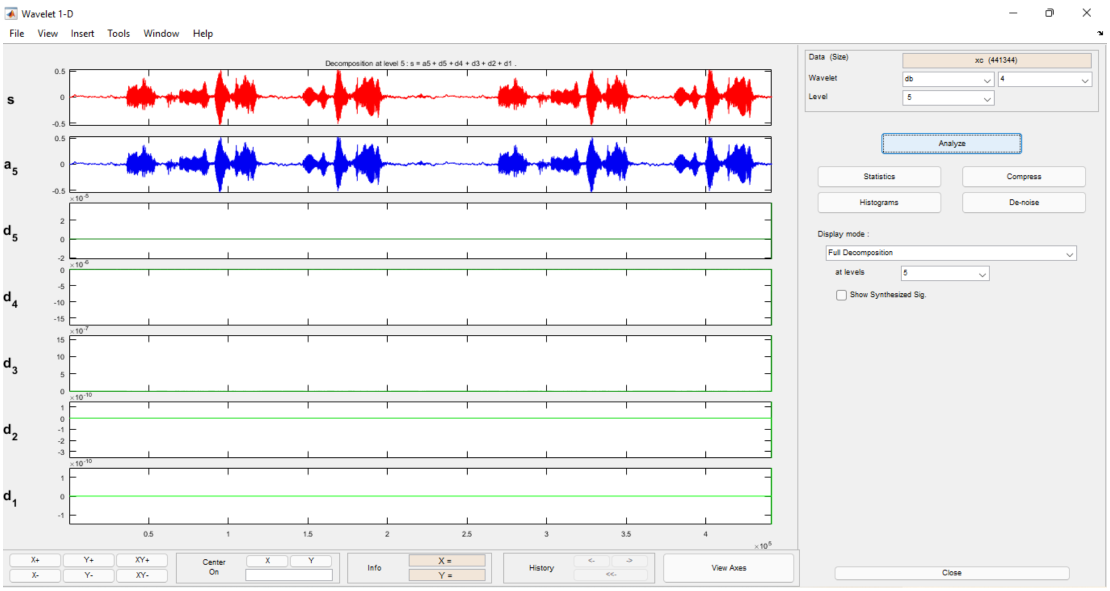

Fig 3.7.1: dB wavelet transforms at 5 levels for notch filter

Then again perform dB wavelet transform is performed at levels 1,2,3,4,5 to analyze the signal. After analysing the signal the out will be like this

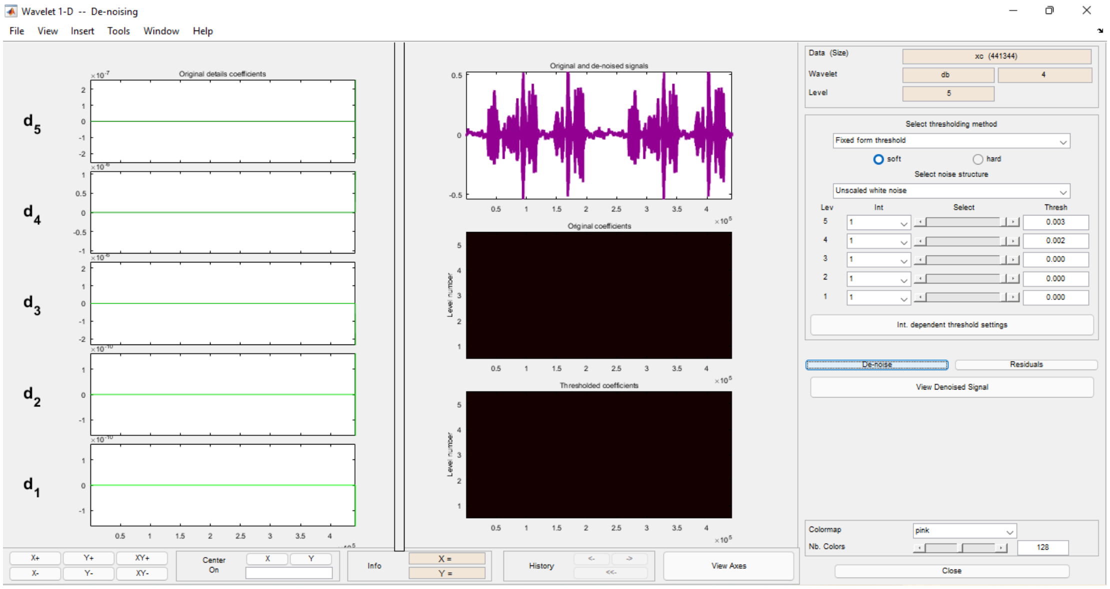

Fig 3.7.2: dB wavelet transforms with co-efficient for a notch filter

After analysing the loaded signal again the statistics are gotten.

After selecting the thresholding method, the denoise signal is performed and gets the output. Original signal and original co-efficient can be seen through the pictures.

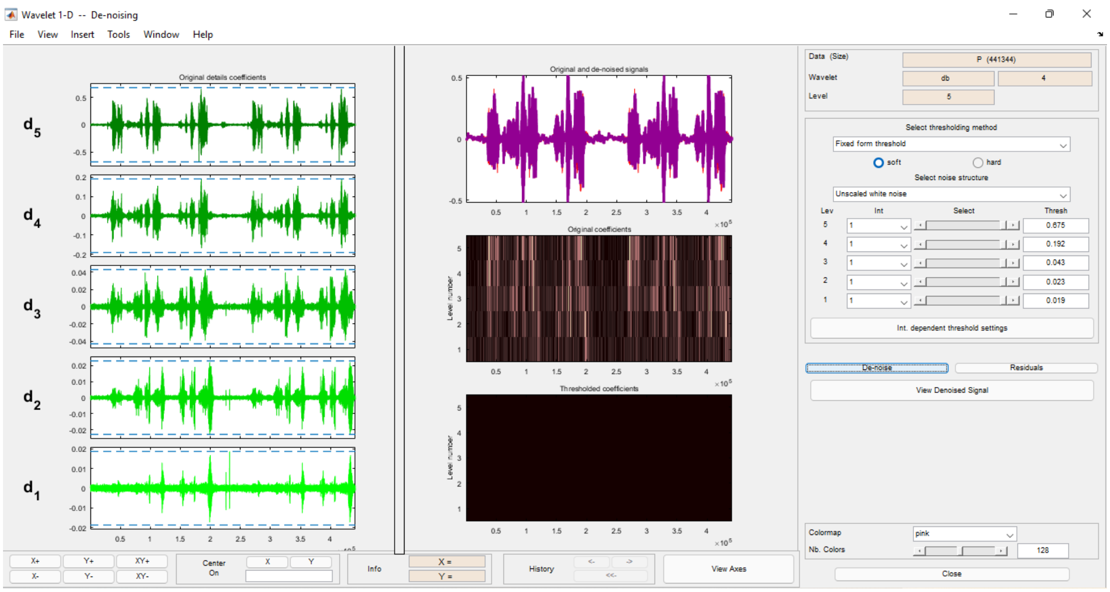

Fig 3.7.3: dB wavelet transforms with co-efficient and threshold for the notch filter

Here different types of noise structures are present - Unscaled white noise, Scaled white noise, and Non-white noise. It needs to be selected from them according to signal noise structure. In the picture threshold co-efficient can be seen also.

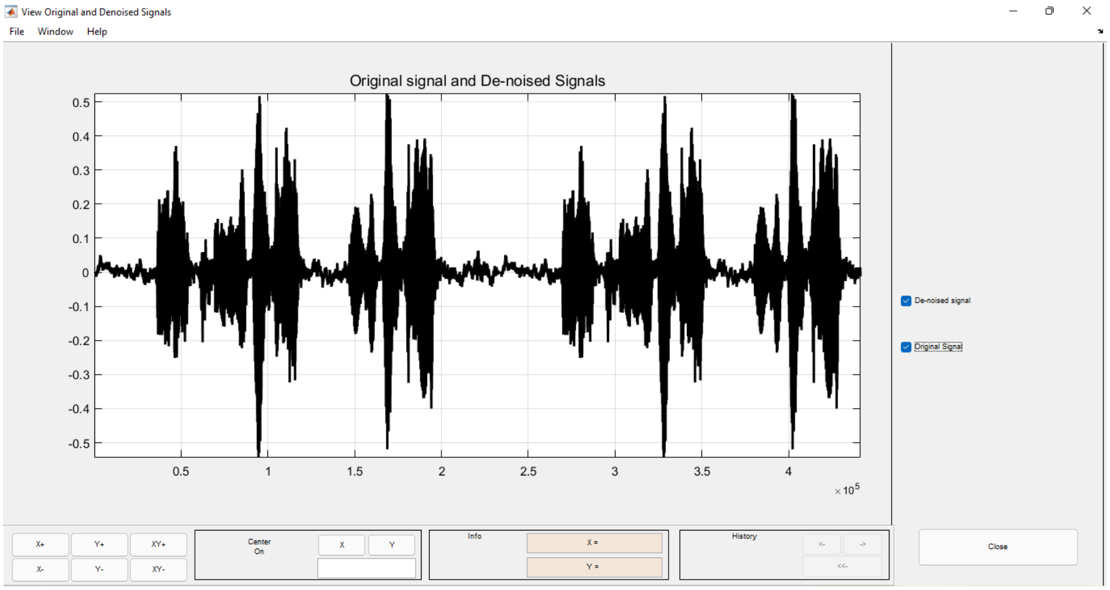

Fig 3.7.4: Original and de-noised signal output for notch filter

Here notch filter is used to eradicate the noise which has been caused by the power grid and it looks a bit better result than the previously performed system. This show the denoised signal, the signal has a very low noise which can be tolerated and accepted. This level of voice clarity is needed to communicate comfortably between the doctor and the patient to understand each other and as telemedicine is totally based on voice communication and internet-based the more clarity you have in the communication, the more success you can achieve in the diagnosis [8]. In medical applications, the noise must be controlled and reduced in a careful way. In the operation theatre or in the ECG room or other sound-sensitive medical tests must be done in a well-soundproofed room, so the noise which is generated at 60 Hz should be eliminated and a notch filter this work to eliminate this range of sound. It is known that if it is needed to permit a certain range of frequency or eliminate a certain frequency notch filter does this work to meet this criterion. So, to meet this criterion notch filter has been used.

Now two acoustic or audio files have been named multimedia file 1 and multimedia file 2, from the spectrum analyzer the graph will make things clear.

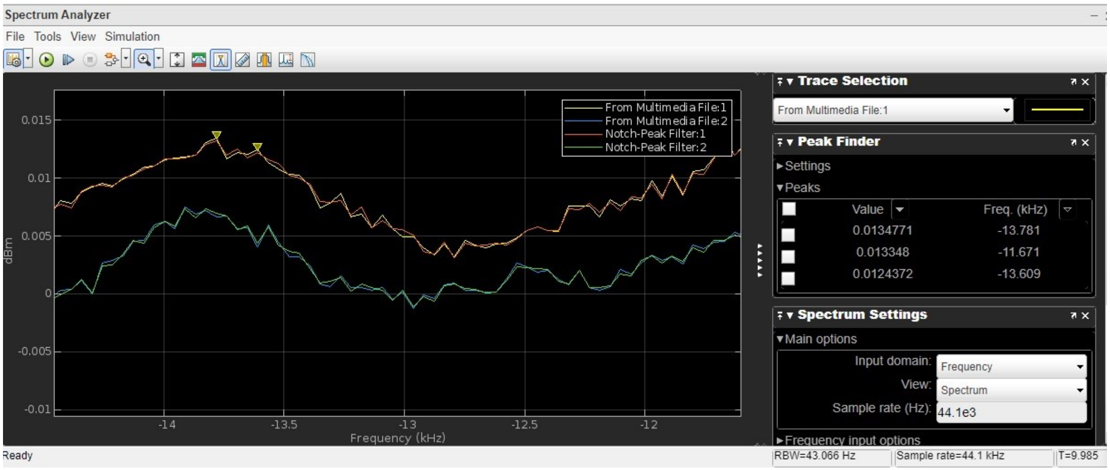

Fig 3.7.5: Notch Filter output

It can be seen that from multimedia file 1 and multimedia file 2, the resultant audio has been got after going through a notch filter, one has been denoted as notch peak filter 1 and the other has been indicated as notch peak filter 2. The graph says that the noise has been reduced and the audio is now in very good condition.

Chapter 4: Experimental Data Analysis

4.1 The data collection

The data which are shown in this section are practical data which have been gathered from the procedures followed to get the desired result via wavelet transform, Notch filter and the data sets which have been taken from two different male voices and two different female voices. MATLAB tool has been used as the performing software and data has been compared from some of the parameters achieved from the performed procedure.

4.2 Comparing the data

The method here which have been applied is wavelet transform. The notch filter nullifies the noise which has been created due to power line noise. Comparison will be shown in the later part and it will describe how it has to stand out the procedure and if the value is reliable or not also the threshold values will be discussed where necessary.

4.3 Comparative Data Analysis

Here two data will be compared one is from the regular wavelet transform and another one is the notch filter using wavelet transform. Which one clarifies the voice more is the main concern and it will be validated via the data which have been achieved through the MATLAB function, values and the graphs will also make a visual representation.

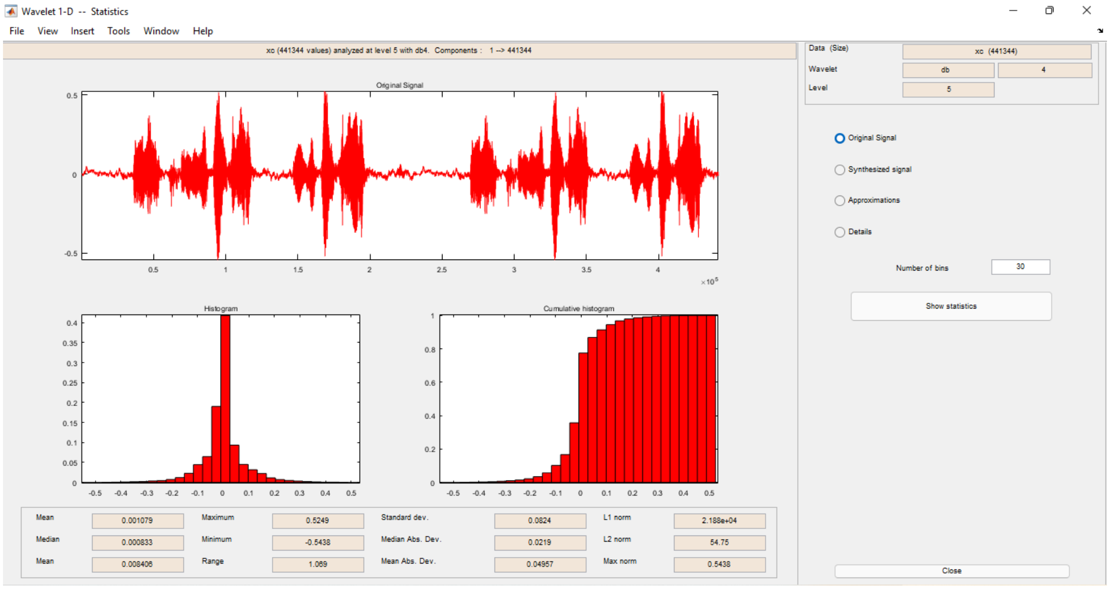

Fig 4.3.1: Notch filter data analysis

To validate the data here the previous data which have been shown in chapter 3 has been again shown as a reference.

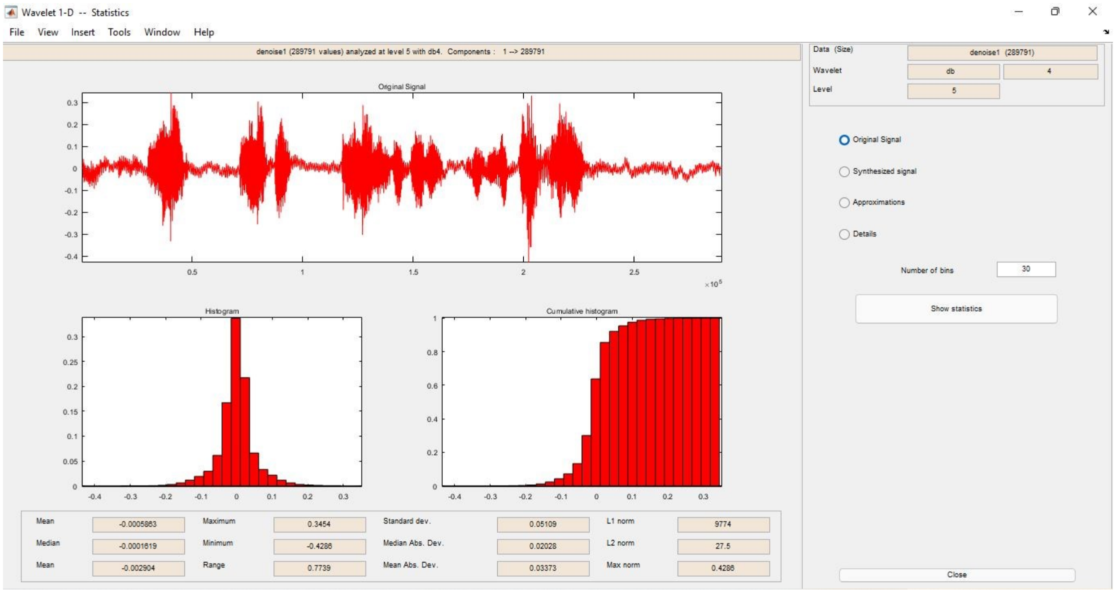

Fig 4.3.2: wavelet transform data analysis

A comparison table has been made below:

Table This clearly indicates the mean, median maximum, minimum, ranges and deviations in the two different series.

Table 4.3.1: Wavelet Transform vs Notch filter data analysis

Data Type	Wavelet Transform	Notch Filter
Mean	-0.0005863	0.001079
Median	-0.0001619	0.000833
Mean	-0.02904	0.008406
Maximum	0.3453	0.5249
Minimum	-0.4286	0.5438
Range	0.7739	1.069
Standard dev.	0.05109	0.0824
Median Abs. Dev.	0.02028	0.0219
Mean Abs. Dev.	0.03371	0.04957
L1 norm	9774	2.188e+04
L2 norm	28.62	54.75
Max norm	0.4286	0.5438

This clearly indicates the mean, median maximum, minimum, ranges and deviations in the two different series.

It also indicates that the notch filter has better output in terms of de-noising the signal. Notch filter use can definitely make the voice clearer so the communication between doctor-patient or the recorded online classes can be heard clearly. Covid-19 has made the classes online where teachers record classes online, also doctors who are in remote places and sometimes where doctors want to encounter a severe type of patient or where doctors are not able to reach only telemedicine can help and telemedicine is dependent over voice and internet and where the place is remote generally the network condition is poor, so there is already some noises or network issue causes distortions [2]. As a result, the notch filter can be used and very useful audio can be found via these procedures.

It also can be figured out via the graphs in these two ways -

4.4 Graph from wavelet transform

This graph shows the output which has been got from the signal which was given as the input at first from the male voice data.

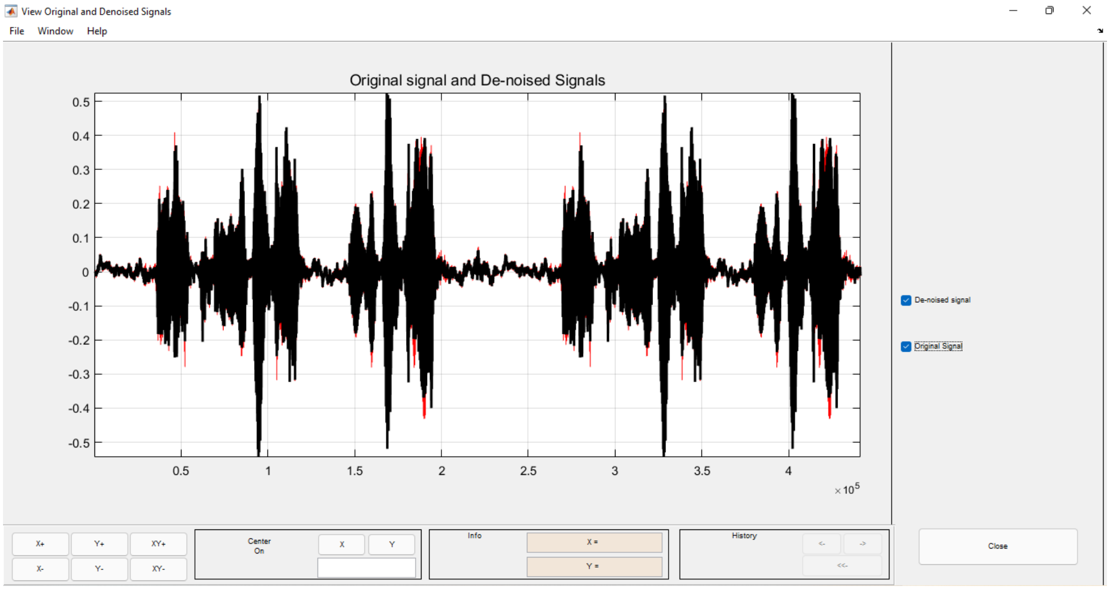

Fig 4.4.1: wavelet transform original and de-noised signal

This is what the graph looks like after the wavelet transform. It indicates after the procedure is repeated after 2 times this is the result.

Secondly, this is the graph we get after the notch filtered audio de-noising, the difference between the main audio signal and the output signal has been pointed out with different colours.

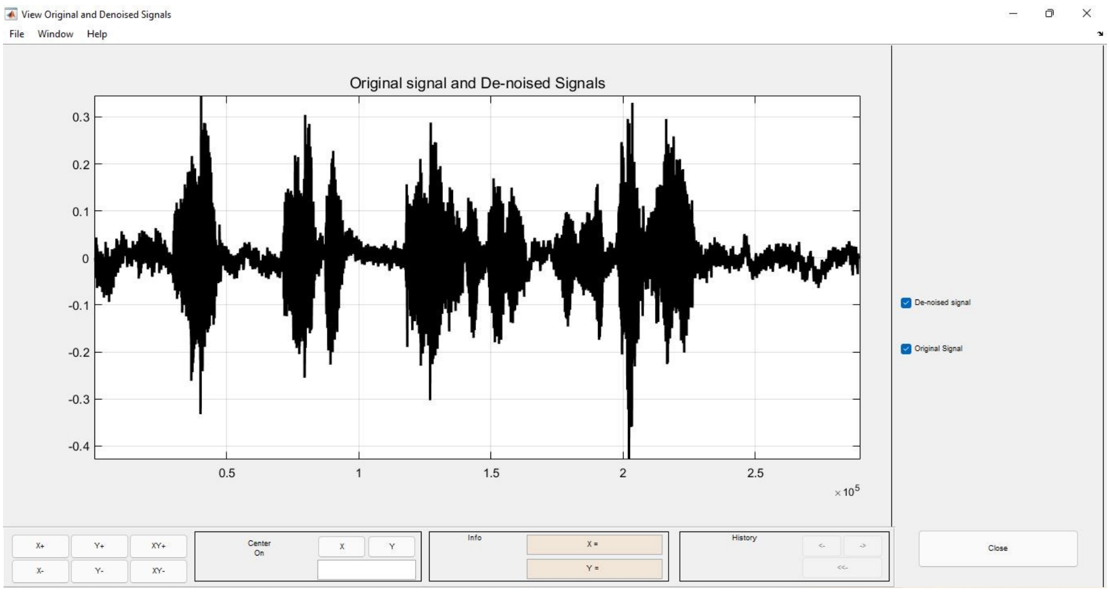

Fig 4.4.2: Notch filter original and de-noised signal

Here the input and output of both signals can be seen, the original signal has been marked as red colour signal and the de-noised signal is the black one.

If the figure is seen carefully, it can be seen that there are some red marks seen drew over the black marks. It indicates that the original signal had noise and some parts don't have black marks on them, so the resultant signal is de-noised in that part. Specifically, the noise in those certain points has been removed due to the notch filter [7]. But in the first picture, the resultant picture is not as clear as this one. So, this proves the fact that power line noise and regular noises like white noise and other distortions can be removed successfully in this method.

4.5 Analysis

In this context, two sets of different existing methods have been considered to compare with the proposed method. If data types and the values are considered, it can be noticed that the dB wavelet with a notch filter has a lower value in the data set.

Table 4.5.1: Comparing data of Haar and dB wavelet transform with a notch filter

Data Type	Haar Wavelet	Db Wavelet with Notch Filter
Mean	0.001079	0.001079
Median	0.0008676	0.000833
Mean	0.009387	0.008406
Maximum	0.6273	0.5249
Minimum	-0.4498	0.5438
Range	1.8771	1.069
Standard dev.	0.08545	0.0824
Median Abs. Dev.	0.02149	0.0219
Mean Abs. Dev.	0.04959	0.04957
L1 norm	2.253e+04	2.188e+04
L2 norm	55.13	54.75
Max norm	0.6498	0.5438

From this analysis, it is found that the clarification of wavelet transformation with a notch filter performs better than the other existing methods.

Chapter 05: Conclusion

In conclusion, it can be expressed that wavelet transformation is superior to Fourier transform and that de-noising an audio signal by using a notch filter in wavelet transform works very well. Both male and female voices were used in the sample audio, which were recorded in two environments- one noisy and the other not so noisy. Data was first put through a notch filter before being transformed using wavelets. Five stages of wavelet modification were applied until the signal was denoised. The results demonstrate that wavelet transformation is superior, and since human speech, in particular, has fluctuating frequency levels and the Fourier transform views audio as a stationary signal, wavelet transformation has been applied. Here wavelet tool of MatLab has been used to analyze the experimental data. A notch filter was also employed to improve the outcome because it removed the 60Hz AC power line noise.

APPENDIX

ACRONYMS

COVID-19	: Coronavirus Disease 2019
QoE	: Quality of Experience
ECG	: Electrocardiogram
dB	: Decibel
SNR	: Signal to noise ratio
WT	: Wavelet Transform
FT	: Fourier Transform

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