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AUDIO NOISE REDUCTION APPLICATION

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Abstract – "Abstract – Noise reduction is the process of removing the noise from any kind of audio signal, in order to get a much cleaner and clearer sound of the audio that has been recorded. There are various noise removal or noise reduction algorithms that have been created for the sole purpose of getting a clean audio and not a disaster in the form of audio. According to our studies Kalman Filter provides comparatively better results than other filters so we have used it in our application. Also, Kalman Filter has much less execution time and it has better optimization. In this project, we will be using an input audio signal, and this software will help us analyze this audio signal and differentiate between the required audio and the noise. In order for us to separate the noise from this signal, we use various filtering algorithms that will help us get the output and also show us the actual amount of noise present in the audio. The algorithm that we are going to use are GAUSSIAN FILTER and KALMAN FILTER

INTRODUCTION.

Noise reduction from an audio file, in this project we are reducing the extra noise present in the audio. We are using different filters and algorithm for this task. This is the application which can be a website or mobile application in future. Getting clean audio recordings can be really hard, especially in noisy environments. Sometimes there's nothing you can do to avoid it. So, what can you do? We are facing many problems while recording an audio in a crowded place, or we are also not able to hear the voice of a caller properly when we are at a traffic signal. So, we record that audio and can reduce the noise from that audio file, and we can get the appropriate thing which we lost during the ongoing recording or on phone call. Sometimes we also not able to hear properly as speaker voice is very low so we can use this application for our convenience. Noises are classified based on the sources of noise or distortions and they include: i. An electronic noise was resembling thermal noise and shot noise, ii. Acoustic noise is emanating from moving, vibrating or colliding sources resembling revolving Machines, keyboard clicks, moving vehicles, rain and wind, iii. The electromagnetic noise that may interfere with the transmission and reception of image, voice, and knowledge over the spectrum, iv. With the presence of a voltage, Electrostatic noise generated, v. Fading and communicating distortion and vi. Lost data packets and Quantization noise due to network congestion. Signal distortion is that the term typically accustomed describes a systematic undesirable change in an exceedingly signal and refers to changes in an exceedingly signal from the non-ideal characteristics of the communicating, echo, signal fading reverberations, multipath reflections and missing samples.

GOALS AND OBJECTIVE

The goal of this project is to create an application that is capable of removing noises from recorded audio files and to alter the audio files according to one's desire.

It consists of four main body: -

- Record audio: - In this process we can record the audio live and we can proceed it into next process, where we will use the to improve the deteriorated audio file.
- Gaussian filter: - This filter is used to reduce the frequency of an audio file. As some audio have Abrasive sound and we cannot able to hear that properly so for that we use this filter for this task.
- Kalman filter: - This is again a filter used to enhance the frequency of an audio file. There are some audio which have very less sound and we cannot able to hear that so we use this for better audio.
- Speech recognition: - This part of the project is to analyze the audio and with the help of this software it gives out the emotions of the audio file.

The various goals of our project are listed below:

- Noise Reduction – One of the most important features of our application is to reduce noises from recorded audio files.

• Recording – The application will provide its users the feature of live recording through a microphone.

• The objective is dependent upon the data length in our noise reduction system. • Study of noise cancellation techniques.

A. Definition and Features

KALMAN FILTERING:

Kalman filter is a mathematical process that uses consecutive or continuous data inputs and also uses sets of equations to correctly predict the right position, value, velocity, etc. or any other value that one wants to measure. It is quite useful when random values, errors or unpredicted values are present in the measured values.

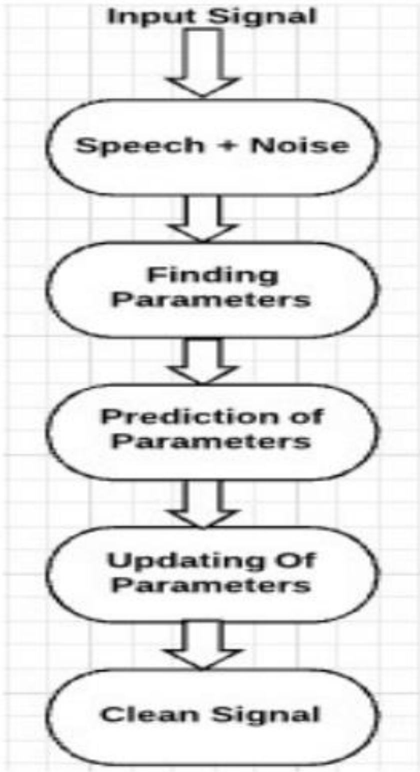

Fig. Data Flow Diagram

As shown in the figure above, First input signal is given to Kalman Filter which contains clean speech and noise, then in the next step parameters are searched by predicting them. By using these parameters input signal is the updated and clean signal is finally then given as output.

Kalman Gain (KG)–

$$KG = \frac{Eest}{(Eest + Mea)}$$

0 <= KG <= 1

Here,

KG=Kalman Gain

Eest=Error I Estimation

EMea=Measurement Error

Current Estimate –

$$ETt = ETt - 1 + KG(Mea - ESTt - 1)$$

Here ETt = Current Estimate

ETt-1 = Previous Estimate

Mea = Measurement

New Error Estimate –

$$\text{EESTt} = \frac{(EMea)(EESTt - 1)}{(EMea) + (EESTt - 1)}$$

Here EESTt = New Error Estimate.

According to our observations Kalman filter is best filter for removing noises as it works perfectly for removing static as well as dynamic noises.

SNR values of Kalman filter are as follows for White noise(db), Random Noise(db), Color Noise(db)

Kalman Filter	0.02227	-5.0952	-16.4817
	17.8932	5.45473	12.2185

Measuring & Updating – The Kalman filter uses these two filters State prediction and Measurement update over and over again to get the correct value.

Fig.DATA FLOW DIAGRAM

KNN CLASSIFIER:

KNN can be accomplished by the steps mentioned below:

Step-1: Load the data set for pre-processing and then applies the KNN algorithm for classification.

Step-2: Select the value of k which will define the number of neighbors.

Step-3: Compute the Euclidean distance of every point of neighbors.

Step-4: Take the k number of neighbors, according to their distance in ascending order of its Euclidean distance.

Step-5: Among the selected k number of neighbors, look for the feature at each and every classification follows.

Step-6: Fit the new data point to the group which consist the maximum number of neighbors Step-7: The algorithm returns the predicted outcome, try testing algorithm's accuracy. believe that resources are infinite and can be accessed at any time.

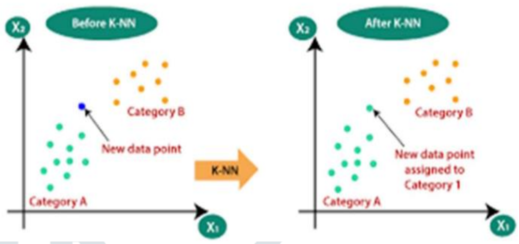

Fig. KNN CLUSTERING GRAPH

The above Fig Displays the graphical interface for KNN classification. One can load an image for classification from the dataset by clicking on the button 'Load Image'. Once the input image is loaded, press 'identify cancer' button to classify the image whether malignant or benign. The GUI also illustrates the features considered for classification, in total 13 features were extracted.

APPROACH TO DESIGN

Looking up the UI, we start to test the application. Firstly, record the live audio and see how it sounds. Then we will save that audio in the database by giving it a name (.wav) format. Now, we need to analyze that which filter we should use for reducing the noise. Hence it is very difficult to find a middle path which continues the application without any error.

The UI looks like, the figure below:

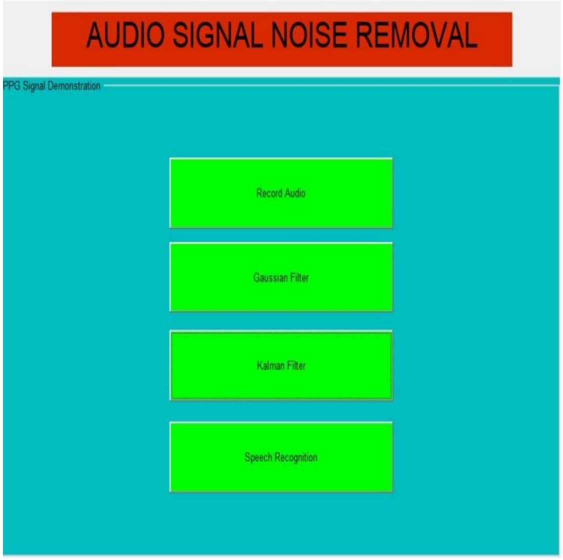

Fig. USER INTERFACE

This includes 4 buttons, which are as follows

1) **A record button-** this will help us record an audio file, and save it in database, as (.wav) file 2) **Gaussian filter button-** this will help us to select a file and remove noise from it using the gaussian filter algorithm and save the file in the (.wav) format in database

3) **Kalman filter button-** this will help us to select a file and remove noise from it using the Kalman filter algorithm and save the file in the (.wav) format in database

4) **Speech recognition-** this will help us classify among the selected audio file, to find which sentiment it belongs to.

SIMULATION

The project consists of 4 buttons, which are as follows

1) A record button- Depending upon your computer hardware, this will help us record an audio file, and save it in database, as (.wav) file, the time limit completely depends upon your computer RAM, as it can record a 3 second audio per 4gb of ram.
2) Gaussian filter button- this will help us to select a file and remove noise from it using the gaussian filter algorithm and save the file in the (.wav) format in database, gaussian filter algorithm works to find out and smoothen out the noise, provided there is noise present, but if no noise is available, then it will just dull the audio output incase of clear audio input.
3) Kalman filter button- this will help us to select a file and remove noise from it using the Kalman filter algorithm and save the file in the (.wav) format in database, this will also help us remove a lot of noise from audio, provided the audio noise is present in the input file, but if clear input audio is available then, it will have noise in the audio output file.
4) Speech recognition- this will help us classify among the selected audio file, to find which sentiment it belongs to. For example we have included sentiments like happiness, anger, lie in our audio database, when we upload an input file it will help in the recognition of the audio by matching it through the database and give us an output.

This project is simulated in such a way which is user friendly and requires at-least 4 Gb of RAM in you devices, and also this created on the MATLAB version 2017b, and windows 10 version. Now depending upon the size of audio input you use to filter, the more time it will take to analyze and give the output in graphical and (.wav) audio output format.

DISCUSSION OF RESULTS

Code implementation follows the following steps –

- First, we record the audio speech or takes already present speech and gives it as input.
- Then an instruction to play noisy audio at 0 SNR
- Then following things are calculated from the given data 1

1. Length of Signal
2. Kalman Gain
3. Predicted state error

4. Estimated error sequence
5. Output of Desired signal

To start the process of audio enhancement or audio noise reduction first we take the audio signal as input by recording it or by taking it from the database. In the audio signal we can record a person talking or any other signal that you want to clean. We calculated the LPC coefficients of the original noisy speech signal and calculate the Kalman gain for each loop for updating of the next state. Looping is done as the past samples have an influence over the future samples. So noisy signal is estimated by looping and then the estimated value of noise is removed from the audio signal given as input. Then clear signal is given as output from the application.

Here is the image of User Interface of the application: -

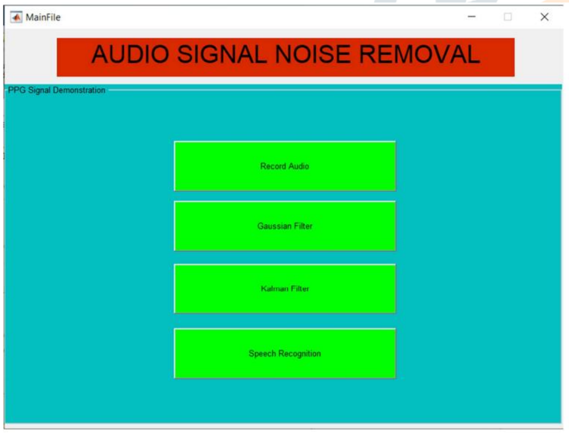

Fig.UI of the Applicaton

STEP1- RECORD THE AUDIO BY CLICKING ON THE RECORD AUDIO BUTTON, IN THE UI. THE AUDIO RECORDING WILL START AND THE SYSTEM WILL RECORD AUDIO FOR 3 SECONDS, USING THE INTERNAL/EXTERNAL MICROPHONE. WE CAN ALSO CHANGE THE AMOUNT OF TIME FOR THE RECORDING, DEPENDING UPON THE CONFIGURATION OF THE SYSTEM, PROGRAM IS BEING RUN ON. (WARNINGTHE LONGER THE AUDIO, MORE TIME WILL BE TAKEN TO GET THE OUTPUT)

STEP2- NOW AFTER THE AUDIO IS RECORDED, IT WILL SHOW YOU THE GRAPH OF THE RECORDED AUDIO, SAVE THE AUDIO FILE AS (NAME.WAV).

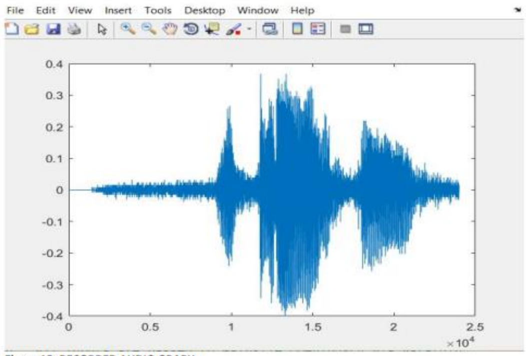

Figure 15- RECORDED AUDIO GRAPH

Figure 16- SAVING NAME OF RECORDED AUDIO

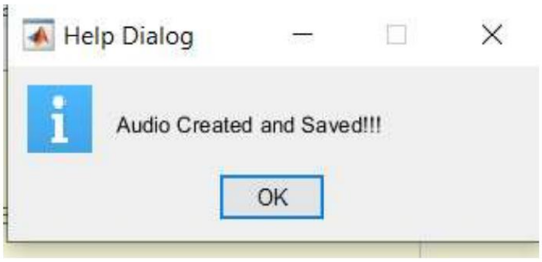

Figure 17- SAVE

STEP3- NOW CLICK ON, THE SPECIFIC FILTER THAT YOU WANT TO USE, THAT IS GAUSSIAN FILTER OR KALMAN FILTER.

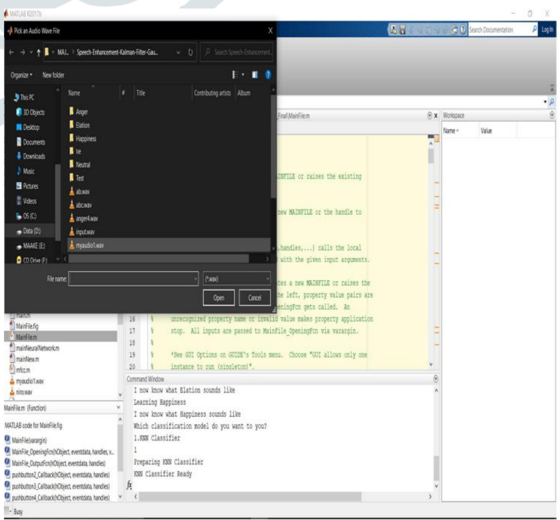

Figure 18- CHOOSE AUDIO FILE

STEP4- SELECT THE AUDIO FILE THAT WAS RECORDED.

STEP5- THE PROGRAM WILL ANALYZE THE AUDIO FILE AND THEN GIVE THE OUTPUT IN GRAPH AND ALSO SAVE THE CLEANED AUDIO OUTPUT.

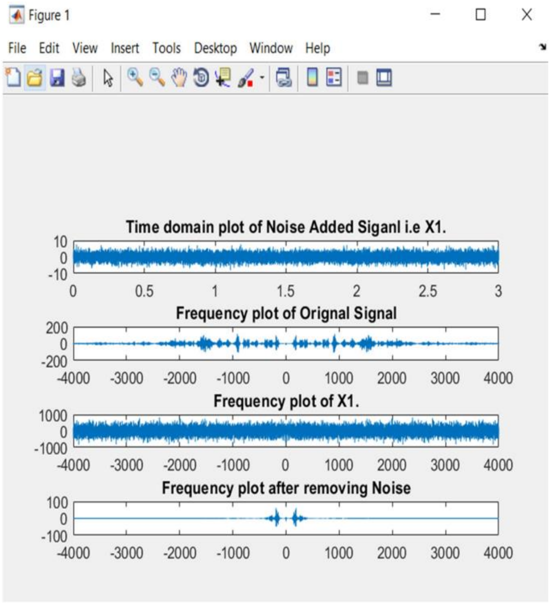

Figure 19- OUPUT FOR GAUSSIAN ALGORITHM

CONCLUSION AND FUTURE WORKS

This concludes that we have been able to identify the algorithms that we will be requiring for the analysis of the sound and cleaning the audio that will help us remove distortions or any disturbance from the audio uploaded via our software. These algorithms have been tested in real world scenarios and have been developed accordingly and are ready to be used for various applications.

Noise Reduction – One of the most important features of our application is to reduce noises from recorded audio files.

Recording – The application will provide its users the feature of live recording through a microphone.

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