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AN EFFICIENT SPEECH DENOISING METHOD USING WAVELET TRANSFORM FOR PREPROCESSED SIGNAL TO GOOGLE TRANSLATOR

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Abstract— *The goal of this research is to see how well the wavelet transform performs while de-noising a spoken signal. Wavelets are frequently employed in digital voice processing, particularly in speech signal coding, augmentation, and noise removal. Because of the background noise of genuine speech, it might be difficult to recognize it in many situations. A voice denoising algorithm's purpose is to recover the original speech signal by reducing noise with the least amount of distortion possible. There are a number of techniques that may be used to assist recover speech that has been distorted by noise. Many of the commonly used denoising techniques do so in the frequency domain, where the noisy signal's power spectral density (PSD) function can be analyzed in a short time period. Then, for each frame of the noisy data, the short-time spectral frequency and amplitude of clean speech are computed. As a result of the limits of methodologies, estimate mistakes are introduced. For decades, many spectrum estimating strategies have been studied in order to decrease estimation mistakes.*

The discrete wavelet transform technique is employed in this work to denoise an input noisy voice stream. Different wavelet filters, such as Daubechies, Symlets, and Coiflets, are used to evaluate the performance of discrete wavelet transforms. MATLAB software was used to do the analysis. Different sorts of environmental background noises were studied as input noisy speech signals, such as babbling noise (crowd of people) or noisy talks with various types of background vehicle noises (cars, train, plane etc.). With hard orsoft thresholding approaches, the input noisy speech signal was decomposed by applying four alternative threshold selections to the wavelet coefficient: sqtwolog, heursure, rigrsure,and minimaxi thresholding. The signal-to-noise ratio (SNR) and MSE values between noisy and output signals were used to compare reconstructed speech to the original speech signal. Detailed comparisons of several wavelet families' capabilities against various background noise types are included, as well as the discovery of an efficient approach (Maximal overlap DWT-MODWT) for denoising noisy voice signals.

I. INTRODUCTION

In terms of both signal-to-noise ratio and audible quality, the suggested technique delivers state-of-the-art denoising performance. Based on the assumption that noise and distortion are the primary factors limiting data transmission capacity in telecommunication. These have an impact on the precision of the measurements. In communication and signal processing, removing noise and distortion is a theoretical and practical consideration. Another key issue is that noise reduction and distortion removal are serious issues in many applications, including voice recognition, cellular mobile communication, image processing, radar, sonar, and medical signal processing, where desired signals cannot be isolated from noise and distortion.

Generally, noise in communication channels disrupts communication, and recovering the original signals from the path without noise is a difficult undertaking [1]. Denoising techniques, which remove noise from a digital stream, are used to do this. For the removal of disturbances from digital audio streams, a variety of denoising techniques have been developed. However, the effectiveness of those strategies is limited[2]. An audio denoising technique based on wavelet transformation is presented in this paper. Denoising is done in the transformation domain, and better denoising is accomplished by grouping closer blocks together[3]. The strategy highlights all of the wonderful elements given by the collection of blocks while also safeguarding the essential aspects of each individual block. The strategy highlights all of the wonderful elements given by the collection of blocks while also safeguarding the essential aspects of each individual block[4]. The blocks are filtered and re-positioned in their original locations. The grouped blocks overlap each other, resulting in a vastly different estimation for each constituent. A approach based on this denoising strategies explained in detail, as well as its efficient implementation. The findings of the implementation show that while the use of wavelets in the field of de-noising audio signals is still relatively young, their use has been growing over the last 20 years[5].

One way to think about wavelets is that they are similar to how our eyes identify the environment at different distances. 3 As we approach even lower scales, we can come up with new details that we haven't seen previously. On the other hand, we'd be completely frustrated if we tried to accomplish the same thing with an image. We would only be able to view a blurred tree image if we expanded the photo "closer" to a tree; we would not be able to see the limb, leaf, or dew drop. The camera can only display one scale of resolution at a time, despite the fact that human eyes can see on many different degrees of detail.

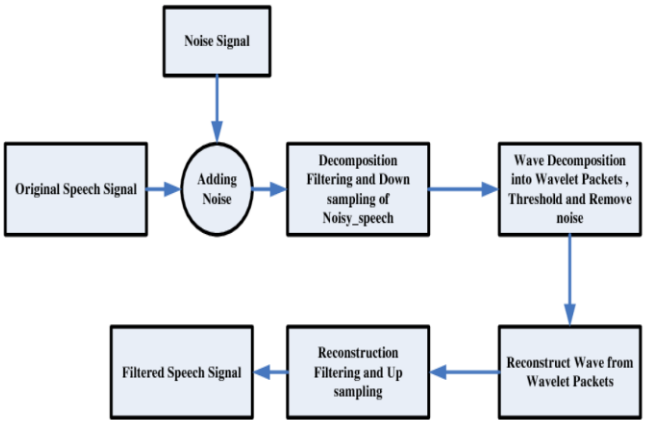
In this chapter, we'll look at how to use the wavelet transform to remove noise in an audio stream. Using the wavelet tool in MATLAB, we build a technique. When compared to Weiner filters and other approaches, this wavelet can provide better results. By lowering the low pass and high pass signals in audio sources, the wavelet can offer an integrated output[6]. When compared to other tools, wavelet gives superior accuracy when utilizing the MATLAB tool. The basic wavelet function and wavelet family are introduced first, followed by wavelet signal analysis in audio signals. In a model system with Gaussian noise injected into an audio stream, we will demonstrate the utility of wavelets in reducing noise. In the parts that follow, we'll show you how to reduce noise in a sinusoidal signal that was generated in the previous section.

II.PROPOSED SYSTEM

To remove noise in the speech we are using MATLAB, the technique which we are used is wavelet transform it is used to enhance the speech and to remove the noise in this case we are using threshold the frequencies which are satisfying threshold condition those frequencies are considering as noises then that frequencies are removed by using wavelet transform.

Wavelet transforms were devised to cope with the Short Time Fourier Transform in order to solve the problem that the Fourier Transform had. In time-frequency decomposition, the short time Fourier transform (STFT) is commonly employed to extract audio features. There is a minor distinction between STFT and FT. The signal is fragmented into small segments, as defined by STFT. It's reasonable to suppose that the little parts will remain stationary. The 'W' window function is utilised. Only stationary time points are allowed, and the window's breadth is equal to the number of segments in the signal. The window function is called at t=0.

Assume the window width is 'T,' the window function overlaps at T/2 seconds, and the signal and the window function are multiplied. The primary window is referred as the mother of the wavelet. One of the models for creating window functions is the mother wavelet. Consider a voice signal. There is some background noise in the signal, and we are adding gaussian noise to it. We are considering gaussian noise instead of a large number of sounds since we can easily calculate how much percent noise is decreased mathematically utilising Gaussian noise.
There exists the good amount of advantages over the previously existing models such as background noise is removed effectively by using wavelet transform, capable of removing noises without creating damage to the non- stationary signals and all kinds of noises like white noise, gaussian noise and other kinds of noises can be reduced in this method.



Block diagram of speech denoising using wavelet

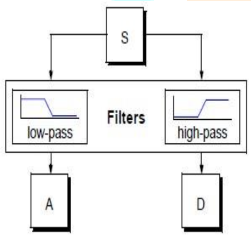
Speech is a very important way for individuals to convey data from sender to receiver utilizing routers that mimic the emotion of a human voice. Speech is usually used to transmit information. Such speeches can be displayed as a channel that responds to excitement waveform. The geometry of the vocal tract causes particular frequencies to be excited. This will be enhanced, while other frequencies will be attenuated. The excitation emerges as semi-regular bursts of air, giving the yield speech a regular appearance. Speech can be divided into two types: voiced and unvoiced. The range of voiced communication is broad and energetic. Because wavelet analysis is based on simulating the front-end sound-related fringe, efforts have been made to use this signal processing tool for speech enhancement.

The most often used method is based on non-linear wavelet coefficient thresholding, which connects multi-resolution analysis and non-linear filtering. The Thresholding method is a denoising method. The wavelet transform divides the loud speech signal into two segment coefficients: approximation (low pass) and spots of interest (high pass). Each and every approximation or detail. The length of sections is half that of the original voice signal. As a result of the concentration of speech signal energy in the approximation component, noise has a little influence on the estimate segment while having a significant impact on the detail segment. In the event that the detail is subjected to a thresholding technique. It completely eliminates noise while keeping the signal energy unchanged.

III.SYSTEM DESIGN

Wavelet Analysis of a Signal

Low and high frequency components make up a signal. The most essential aspect of signals is their low frequency content; high frequency stuff is noisy. The voice sound will change if the high frequency parts are removed, but you will still be able to hear what is being said. We acquire approximations using the wavelet transform, which are high frequency segments, while details are in low frequency segments.[1]



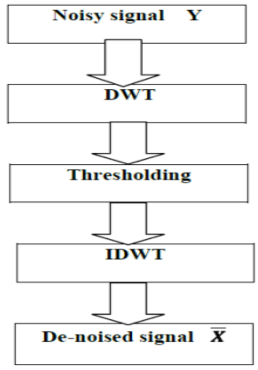
Approximation of detailed coefficient

ER/UML Diagram

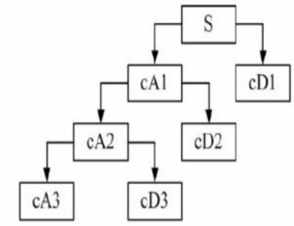
The wavelet transform is a noise reduction method. Johnstone and Donohoe have proposed it. We can increase performance by eliminating noise with the wavelet transform. The steps are as follows:

Input noisy Signal: In this thesis study, we applied wavelet method for denoising of speech signals. For the noisy input speech signal, we have added gaussian noise because it is effective in mathematical environment.

- Apply DWT: To this signal we have applied discrete wavelet transform. The primary phase of the DWT algorithm breaks down the signal into sets of coefficients.
- Thresholding and IDWT: The input noisy speech signal was decomposed by applying threshold selection to the wavelet coefficient: sqtwolog. After this step, we have applied inverse discrete wavelet transform to these coefficients.
- De-noised Signal: In the final step of this wavelet transform algorithm the original speech signal is reconstructed and recovered which is free from noise.



Module Design and Organisation



Wavelet Decomposition Tree

The above Wavelet Decomposition Tree explains how the input noisy signal is divided into two categories. It separates signal into two separate low pass and high pass sections. Low pass section contains main information regarding the audio signal whereas, high pass section contains the noise present in the signal. So, we will not consider the high pass section as it contains the noise. We mainly focus on the low pass section where the signal is again divides into separate parts until the noise is completely removed from the signal [2].

We shall utilize wavelet transform instead of Fourier transform in non-station signals since it is more efficient and also works with non-periodic audio signals with varied transient values. We can't apply Fourier analysis due to its disadvantage. We utilize frequency domain in lower analysis. As a result, we employ wavelet analysis. Wavelet transforms a signal into several different forms, then threshold is applied to each level to eliminate noise. By separating the signal into low and high frequencies as it comprises of noise, low frequencies will stay and high frequencies will be erased.

IV.IMPLEMENTATION AND RESULTS

MATLAB is a high-performance language for technical computing .it integrates computation, visualization, and programming in an easy-to-use environment where problems and solutions are expressed in familiar mathematical notation.

MATLAB stands for matrix laboratory, and was written originally to provide easy access to matrix software developed by LINPACK (linear system package) and EISPACK (Eigen system package) projects.

Although MATLAB is encoded in C, C++ AND Java, it is lot easier to implement than these three languages. For example, unlike the other three, no header files need to be initialized in the beginning of the document and for declaring a variable. The data type need not be provided. It provides an easier alternative for vector operations. They can be performed using one command instead of multiple statements in a for or while loop.

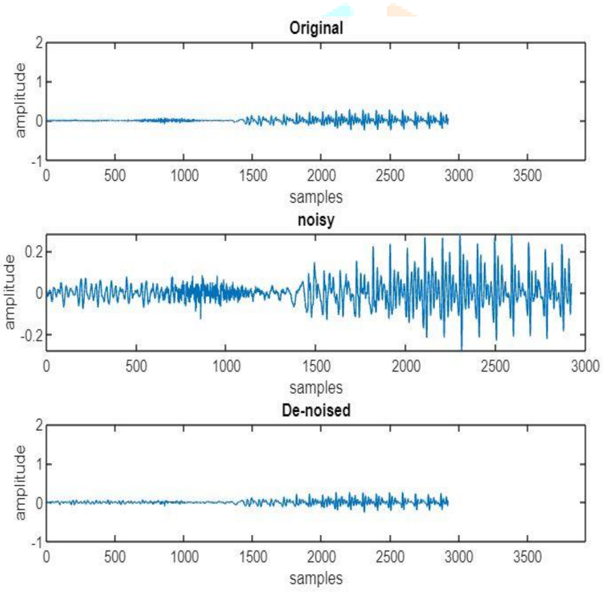
We will upload the audio signal and noise to MATLAB. To get denoised output, we used the audioread function to record audio files, then applied the wavelet transform to produce clean signal based on the threshold value. Finally, we compared the output audio noise audio and output audio to get clean output audio and plotted graphs for better understanding.

Implementation of Key Functions

- audio read ()
- imnoise ()
- length ()
- DWT ()
- IDWT ()
- wavedec ()
- sqrt ()
- wdencamp ()
- wrcoef ()
- subplot ()
- plot ()

V. RESULT AND DISCUSSIONS

Comparison of Noisy signal with De-noised signal



VI. TESTING AND VALIDATION

Design of test cases and scenarios

The following steps are used to compute signal denoising using the wavelet transform. We must first calculate the Gaussian noise numerically, taking into account the mean and variance values. The signal's quality is then assessed using the signal to noise ratio.

Ratio of signal to noise:

One of the most often used methods for determining signal quality is the signal to noise ratio. SNR is determined by dividing the square of the clean signal by the square of the difference between the clean and noisy signals. The signal length is used to calculate the total, which is referred to as the global signal to noise ratio. The signal-to-noise ratio (SNR) is expressed in decibels (dB).

$$SNR = 10\log_{10}\frac{\sum_n s^2(n)}{\sum_n[s(n)-\hat{s}(n)]^2}$$

The psnr (clean signal, noise signal) function in MATLAB may be used to determine the SNR value.

Threshold value: To eliminate Gaussian random noise, a thresholding rule is needed. There are two different sorts of thresholding techniques. Soft thresholding and hard thresholding are two types of thresholding. The sqtwolog technique is used in this work. It is one of the gentle thresholding types.

Noise level = median(abs(detailed coefficient))/0.675 noise-----(1)

The formula below is used to compute the sqtwolog threshold value.

Noise level = sqrt(2*log(length(noisy signal))*threshold -------(2)

Validation

The accuracy of this model is analyzed by using psnr function.

Accuracy of input and noise signal (input, noise) is 22.345 i.e., consider 100%

Accuracy of noise with output signal without processing is 32.52 i.e., 75%

Accuracy of output signal without noise (input, sig denoise) is 22.058 i.e., 99%

- To calculate the accuracy of different signals we have used psnr function. In first step consider the psnr of input signal and noise signal and calculate accuracy.
- In second step calculate accuracy of noise with output and this is analyzed before applying the wavelet and after applying the wavelet transform.
- At last, we apply wavelet transform and thus accuracy of input signal and output signal by removing noise with the help of wavelet transform is calculated and we got same accuracy for both input and output signal.

VII. CONCLUSION

When it comes to eliminating background noise from audio signals, the suggested approaches demonstrated to be more accurate. Wavelet transform is a new approach for improving audio signals. We tested the effectiveness of several types of wavelet families using varied noisy audio signals in this study. The major purpose of this research is to minimise background noise by evaluating the input audio signals and using various threshold approaches. SNR and MSE values were used to calculate the efficiency/performance of signals in this investigation. Then, for input noisy audio signals, we used several thresholding approaches. When compared to other thresholding approaches, the sqtwolog threshold produced a greater signal-to-noise ratio.

After analyzing the data, it was discovered that the sqtwolog threshold provides the optimum SNR values for a variety of noisy audio input sources. This threshold approach for removing background noise in audio signals is a novel method not previously documented in the literature. This approach performed admirably on a variety of noisy input signals. It may be concluded from the obtained findings that the sqtwolog approach can be used to create high quality speech augmentation of noisy audio sources. We measured the effectiveness of wavelet families for various types of noisy signals using all of the collected data.

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