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SPEECH SIGNAL PROCESSING WITH WAVELET TRANSFORM

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Abstract

In numerous speech preparing applications, speech must be handled in the presence of undesirable background noise like white Gaussian noise, colored noise, vehicle inside noise and numerous other genuine noises. It is fundamental to improve the first speech signal using noise reduction strategies to dispose of any undesirable background noise. Speech enhancement intends to improve the speech quality for human audience or improve the speech signal for other speech preparing calculations. A wavelet based speech enhancement calculation is implemented in programming. A different calculation, called wavelet packet decomposition, is implemented as opposed to the most widely recognized method wavelet pyramid decomposition. The following part focuses on carrying out wavelet packet decomposition method for speech enhancement on genuine stages.

Keywords: speech, wavelet, noise.

1. Introduction:

One of the imperative movement of people is communication through speech [1]. Individuals obtain different approaches to get data from the outside world or to speak with others [2]. Speech, picture and composed content are the three most important beginnings of data. Speech can be highlighted as the best and agreeable one from numerous points of view [3]. Speech does send linguistic substance, yet additionally delivers other helpful data like the sensations of the speaker. Speech handling offers practical and hypothetical understandings of how human speech can be processed using both sign preparing methods and knowledge from hearing sciences, phonetics, linguistics, and psychology [4]. The rapid advancement in digital sign preparing makes the speed of addressing, putting away, recovering and handling speech data simpler and quicker. Additionally, the development of speech preparing methods started new and different application regions like speech enhancement, speech union, speech coding and speech acknowledgment [5].

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2. Methodology:

Wavelets are mathematical functions that separate data into different recurrence components and then investigate every part with a resolution matched to its scale. In contrast to the traditional Fourier methods, with the wavelet approach, different frequencies are analyzed using different resolutions, depending on the signal sort instead of using the same window (consequently same resolution) for each spectral component. In the Fourier theory (FT), a signal is considered as a sum of hypothetically endless sines and cosines, which makes the FT useful fo r limitless and periodic signal analysis. The FT dominated the signal-processing field for a long time because it works well in providing the recurrence data of the signal being analyzed. Despite the fact that it provides recurrence data, lamentably, it fails to give any data about the event time, which led the scientists to suggest new methods such as the Short-Time Fourier Transform (STFT). This methodology cuts the objective signal in several parts and then break down each part separately, which creates another issue: How to cut the signal? This issue is solved by the wavelet transform, which provides a completely scalable modulated window, requiring thus no signal cutting.

The wavelet transform is described as a mathematical method in which the objective signal is analyzed in the time domain by using different versions of a dilated and translated basis work called the mother wavelet. The Haar wavelet, the first wavelet transform, was introduced toward the start of the 20 th century by a German scientist, and then named after him. The Haar wavelet basis work has reduced support and number coefficients and was subsequently used in physics to study Brownian motion. From that point on, several studies have been carried out either in the development of the wavelet theory or its applications in different fields. In the field of signal processing, the extraordinary accomplishments achieved by other researchers, A wide scope of successful wavelet-based applications. Hence researchers developed the first non-inconsequential wavelets after the work carried out by Mallat on the relationships between the Quadrature Mirror Filters (QMF), pyramid algorithms and orthonormal wavelet basis. Inspired by Mallat's work, another significant work was developed by Ingrid Daubechies. Daubechies achieved the cornerstone of numerous applications by constructing a set of wavelet orthonormal basis functions. The same creator in a joint effort with others introduced a set of wavelet biorthogonal basis

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functions which are used in different applications such as picture coding, compression and example recognition. The figure 1 shows the processed and noise free wavelet representation of speech signal.

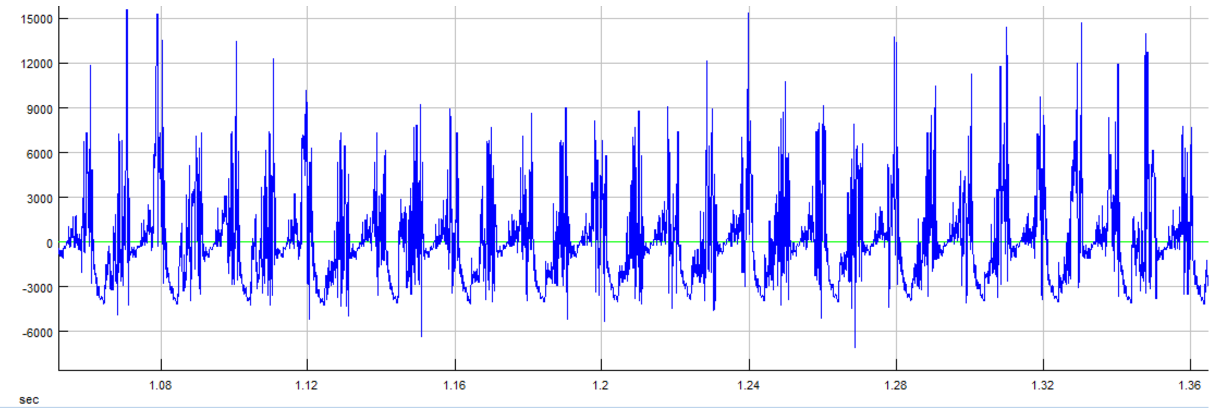

Figure 1: Wavelet representation of speech signal.

3. Conclusion:

The first piece introduces some significant concepts in connection with the subject of this research work. In this specific situation, is dedicated to the characteristics of speech signals. It includes the theory behind speech production and discernment as well as a description of an assortment of auditory models. It also, significantly, highlights the noise chain in an audio system (acquisition stage and play back stage). Part 3 basically audit some existing techniques used in speech enhancement or denoising. It includes different techniques such as brush filters, adaptive enhancement methods, spectral subtraction, and Wiener filtering. These methods are then compared and their advantages and disadvantages are presented. Part 4 consists of a careful audit of the theory on the wavelet transform. The definition of the transform is introduced first, at that point, concepts such as the continuous wavelet transform and Mallat's multiresolution analysis are presented. The concepts of filters banks and the discrete wavelet transform are introduced in the second piece of the section. Filters banks are vital while carrying out a wavelet based system. Different aspects like wavelet families (orthonormal and biorthogonal) and wavelet types (Daubechies, Coiflet and Symlet) are also presented. To supplement part 3, some existing work on speech enhancement (denoising) using the wavelet transform is reviewed toward the end of this section.

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