

High-frequency Noise Removal of Audio Files using Daubechies Wavelet Transform

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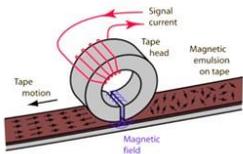
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1



A compact cassette comprises two small spools, between which a magnetic tape is threaded and wound. In tape recorders and cassette players, a tape head is employed to both record and play audio tracks on these magnetic tapes. This is achieved by converting the electric signal into a magnetic fluctuation and vice versa.

2



Over prolonged periods of storage within the Earth's magnetic field, these cassettes undergo an impact where the magnetic particles alter their orientation, leading to distortion in the recorded audio signal. Consequently, this distortion gives rise to high-frequency noise within the audio signal, causing significant disruption.

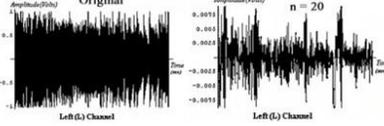
3

$$W_{\phi}(j_0, k) = \frac{1}{\sqrt{M}} \sum_x f(x) \phi_{j_0, k}(x)$$

Wavelet transform was used to obtain the frequency domain representation of the distorted audio signal to identify high-frequency noise.

4

The localized variable "n" was defined in the Wolfram Mathematica source code to use a trial-and-error method to remove high-frequency noise (where "n" is the Daubechies wavelet level).



5

$$f(x) = \frac{1}{\sqrt{M}} \sum_k W_{\phi}(j_0, k) \phi_{j_0, k}(x) + \frac{1}{\sqrt{M}} \sum_{j=j_0}^{\infty} \sum_k W_{\psi}(j, k) \psi_{j, k}(x)$$

Inverse Wavelet transform was used to convert the audio signal back to its time domain form to follow a trial-and-error approach to obtain the suitable "n" value by listening to the output.

6

This technique can be used to conserve vintage songs that were originally stored in compact cassettes and spools.



Abstract

In general, audio signals are contaminated with various types of noise. This paper presents a novel signal processing method developed for high-frequency noise elimination using wavelet transforms. As a continuation of a previous study that used Fourier transform for noise removal in audio files, in this study Daubechies wavelets were used to reduce computational complexity and achieve better noise reduction performances. Compared to the Fourier transform, the Daubechies wavelet transform method removes the noise in each signal while preserving its vital characteristics. The suitable level of the Daubechies wavelet for noise removal in each channel was obtained using a trial-and-error approach. It was identified that the ideal range for the level of the Daubechies wavelets for noise removal is between 17 and 20. Moreover, unlike the Fourier transform, the Daubechies wavelet transform demonstrates a proficient capacity in eliminating noise from data point that lies completely outside the rest in the audio data set. Wolfram Mathematica 12.3 software package was used to complete this research. This method can be applied to conserve vintage audio recordings originally recorded in cassettes and spools.

Keywords: Digital Signal Processing, Wavelet Transforms, Daubechies Wavelet Transform, Fourier Transforms, Noise Removal

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1. Introduction:

Several forms of noise can be found in a vintage audio recording. Major forms of such noises are random noise, pop noise and localized random noise (Abraham, 2017). These noises heavily affect the quality of original audio signals. Therefore, it is of vital importance to use digital signal processing techniques to minimize or eliminate the noises from audio signals to conserve vintage audio recordings as well as to produce high-quality audio files such as music recordings. This can be achieved using various forms of digital signal processing techniques including filtering, noise reduction algorithms and adaptive noise cancellation.

Digital signal processing (DSP) is the application of signal-processing techniques in digital devices. DSP enables the processing of various types of digital signals, such as audio, sonar and radar, sensor readings, and biomedical signals (Oppenheim, 1978; Vaseghi, 2008). These signals can be analyzed and manipulated using different domains, including the time domain (1D signals), spatial domain (multidimensional signals), frequency domain, autocorrelation domain, and wavelet domain. Numerous methodologies are available for the mitigation of noise in signals.

One of the techniques employed for the elimination of noise in audio files involves the utilization of Fourier transform (Rabiner & Gold, 1975, Smith, 2008; Ashwin & Manoharan, 2018; Zhang, 2019; Fernando & Kularathne, 2023). As a continuation of the study by Fernando & Kularathne, 2023, in which Fourier transform was used for noise removal, this paper discusses a novel method developed using Wolfram Mathematica to apply wavelet transform to reduce noise in vintage audio recordings. Wavelet transform generates a representation in both frequency and time domain. This allows efficient analysis of more localized details in each signal. Generally, discrete wavelet transform and inverse discrete wavelet transform can be represented in equations 1 and 2, respectively.

$$W_{\varphi}(j_0, k) = \frac{1}{\sqrt{M}} \sum_x f(x)\varphi_{j_0,k}(x) \dots\dots\dots (1)$$

$$f(x) = \frac{1}{\sqrt{M}} \sum_k W_{\varphi}(j_0, k)\varphi_{j_0,k}(x) + \frac{1}{\sqrt{M}} \sum_{j=j_0}^{\infty} \sum_k W_{\psi}(j, k)\psi_{j,k}(x) \dots\dots\dots (2)$$

where $f(x)$, $\varphi_{j_0,k}(x)$, and $\psi_{j,k}(x)$ are functions of the discrete variable $x = 0,1,2,\dots,M-1$. Normally we let $j_0 = 0$ and select M to be a power of 2 (i.e., $M = 2^j$) so that the summations in Equations (1) and (2) are performed over $x = 0,1,2,\dots, M-1$, $j = 0,1,2,\dots, J-1$, and $k = 0,1,2,\dots,2^j- 1$ (Gonzalez & Woods, 2002).

Wavelet transform is an infinite set of various transforms, depending on the merit function used for its computation. Wavelet transform can be sorted in several ways including the wavelet orthogonality. Orthogonal wavelets are used for discrete wavelet transform development and non-orthogonal wavelets for continuous wavelet transform development.

The discrete wavelet transform (DWT) returns a data vector of the same length as the input. Usually, even in this vector, many data are almost zero. This corresponds to the fact that it decomposes into a set of wavelets (functions) that are orthogonal to its translations and scaling. Haar, Daubechies, Coiflet, and Bio-orthogonal are some examples of discrete wavelets (Singh & Sharma, 2012; Rowe & Abbott, 1995).

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Among the above-mentioned wavelet transform techniques, Daubechies wavelet is one of the widely used methods in digital signal processing. The Daubechies wavelet concept was developed by the renowned Belgian mathematician, Ingrid Daubechies. In the realm of discrete wavelet transform (DWT) algorithms, Daubechies wavelets find widespread application. These algorithms process signals by breaking them down into distinct frequency sub-bands, each representing varying levels of detail within the signal.

Several characteristics render Daubechies wavelets well-suited for signal processing endeavors. One such attribute is their compact support, where these wavelets remain non-zero over finite intervals. This compact support feature enhances computational efficiency and facilitates the localization of signal properties. Additionally, Daubechies wavelets exhibit a high degree of regularity (Figure 1), enabling the generation of smooth and precisely controlled frequency responses. This combination of properties contributes to the effectiveness of Daubechies wavelets in various signal-processing applications (Rowe & Abbott, 1995; Mandala et al., 2023; Ben-Israel, 2001).

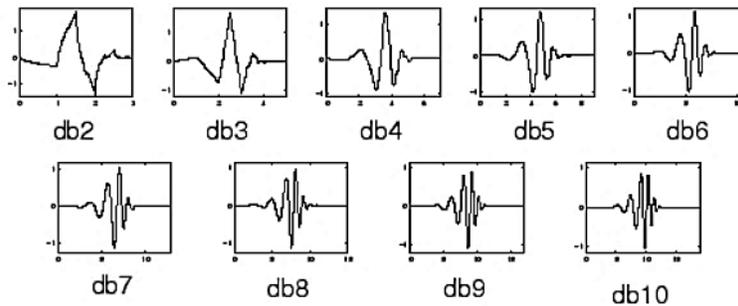


Figure 1: Shape of Daubechies wavelets from level 2(db2) to 10(db10).

The number associated with the Daubechies wavelets, such as Daubechies-2 (D2), Daubechies-4 (D4), Daubechies-6 (D6),...etc., represents the number of vanishing moments. Vanishing moments determine the ability of a wavelet to represent and capture different levels of smoothness in a signal. Higher vanishing moments provide a better approximation of signals with more regularity. The index number refers to the number of coefficients. Each wavelet has several zero moments or vanishing moments equal to half the number of coefficients. For example, D2 has one vanishing moment, D4 has two,...etc. A vanishing moment limits the wavelets’ ability to represent polynomial behavior or information in a signal (Popov et al., 2018; Vonesch et al., 2007).

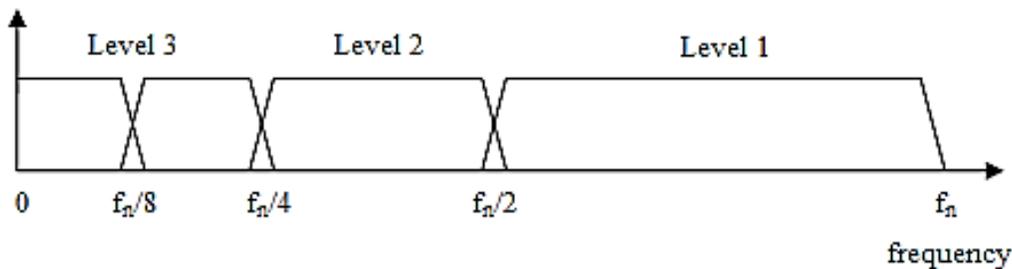


Figure 2: Frequency domain representation of the Discrete Wavelet Transformation.

2. Methodology

The distorted song was obtained in the “wav” file format. These files were generated by connecting the output of an audio cassette player to a computer using a 3.5 mm AUX cable. Then the output signal was recorded in the computer as a “wav” file. Since the processing time for an audio track with a duration of over three minutes is high, the “Audio Trim” function in Wolfram Mathematica was used to trim and extract 10 s samples from the audio file to apply digital signal processing techniques. First, the sample “wav” file was read into Wolfram Mathematica 12.3 and converted into a data file using the inbuilt function “AudioData”. The accuracy of digitized data in Wolfram Mathematica was verified using the open-access SoX (Sound Exchange) software. Since there are two separate channels in audio waves used in this experiment, channel 1 and channel 2 were separated in the form of 2D arrays to remove the noise components.

Then Daubechies wavelet transform of each channel was obtained to identify and remove high-frequency noise from the audio file. To obtain the level of the Daubechies wavelet level, a Wolfram Mathematica algorithm was developed with the help of some in-built functions. This procedure was repeated for each channel of the given audio sample while varying the level of the Daubechies wavelet to obtain the suitable value.

After obtaining the suitable threshold value for a 10 s sample, it was applied to the entire audio track. The accuracy of this method was examined for a diverse collection of songs, and a few of those samples and their corresponding threshold values are mentioned in Table 1.

3. Results

Table 1: Adequate threshold values obtained for each channel of the selected 10 songs in the experiment.

	Song	Channel - 1	Channel - 2
1	“Sandagalathenna” tele drama theme song by Amarasiri Peris and Nelu Adikari	19	19
2	“Mage Mandri Nam” by Amarasiri Peiris	20	20
3	“Aradhana” by W D Amaradeva	19	19
4	“Teri Bindiya Re” by Lata Mangeshkar and Mohammed Rafi	18	18
5	“Indunil” by Indrachapa Bulathsinghala	20	20
6	“Pasak Kota Ethi” by Gunadasa Kapuge	20	20
7	“Obama Maada Viya Maada Obama Viya” by Malani Bulathsinghala	18	18
8	“Yedho oru Pattu” by Unidathil Ennai Koduthen and Karthik	17	17
9	“Enna Mada Nale” by Shanthi Geethadeva	20	20
10	“Hello” by Lionel Richie	17	17

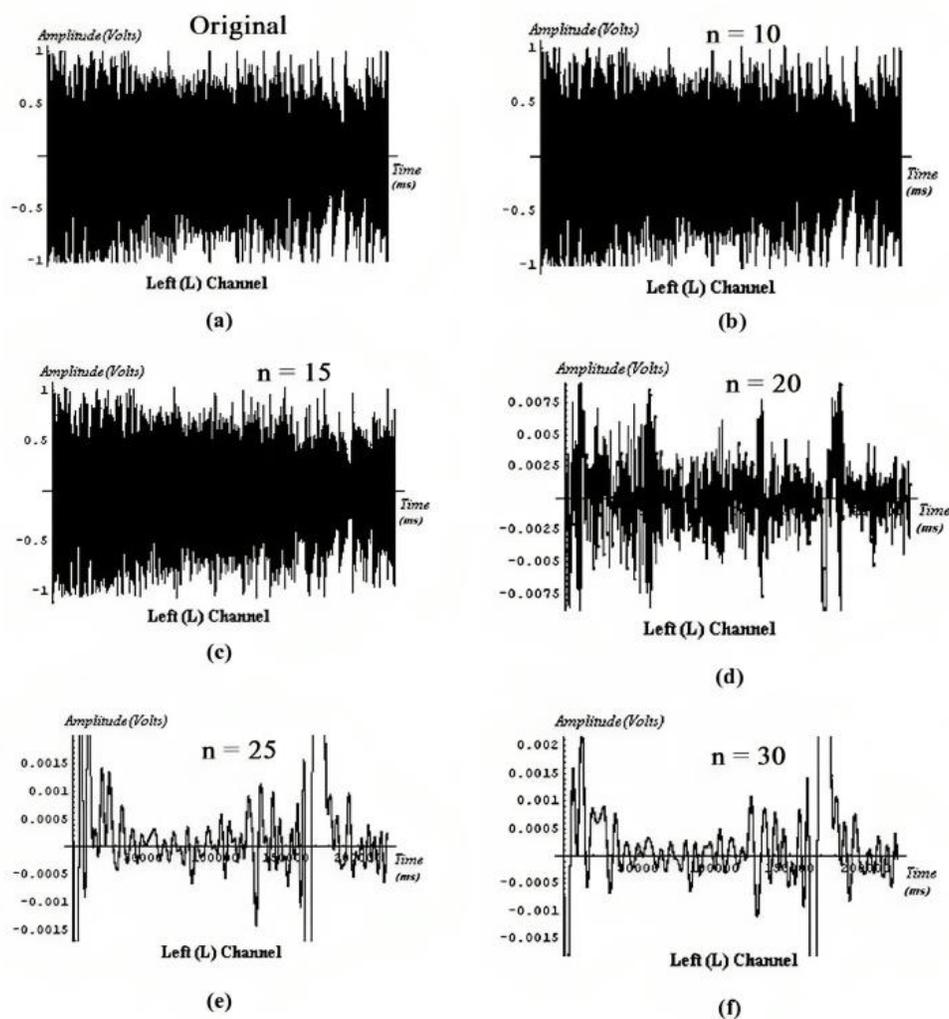


Figure 3: (a) Original Song, (b) $n=10$, (c) $n=15$, (d) $n=20$, (e) $n=25$ and (f) $n=30$; where “ n ” is the applied Daubechies wavelet level.

4. Discussion

This research was a continuation of a similar study conducted using the Fourier transform to remove high-frequency noise from vintage songs recorded in audio cassettes and spools (Fernando & Kularathne, 2023). Even though the Fourier transform technique successfully eliminated the high-frequency noise from a given audio file, it had some negative impacts on the original audio file as the high-frequency components of the original song were also affected during the noise elimination process.

Therefore, to minimize this effect, the Daubechies wavelets were used in this research to eliminate noise from audio files. In addition to that the Haar wavelet transform and Coiflet wavelet transform were also considered in this project, but it was identified that neither one of those transforms is suitable for this application. The Haar wavelet transform was not suitable because it always requires 2^n data points, which a dataset generated by an audio recording rarely satisfies. The Coiflet wavelet transform failed to generate

considerable results. Comparison between these methods showed that Daubechies wavelets are the most convenient form of wavelet transform for noise reduction in audio files.

Wolfram Mathematica 12.3 was used for this study while the results generated in different stages of the developed algorithm were verified using globally recognized open-source software packages used in the audio engineering industry. The SoX software was then used to compare the accuracy of the digitized dataset of the audio file using Wolfram Mathematica. It was observed that the two datasets were equal up to four decimal points.

Compared to the algorithm developed to apply the Fourier transform in Fernando & Kularathne, 2023, the application of Daubechies wavelets for noise removal using Wolfram Mathematica was easier as there are several inbuilt functions to simplify the work. Therefore, to obtain the suitable level of the Daubechies wavelet for noise removal in each channel, the inbuilt command “DaubechiesWavelet” was used. The adequate level was obtained by following a trial-and-error method where generated output after applying Daubechies wavelets was converted into its time-domain representation using the inbuilt command “InverseWaveletTransform” and listened using high-quality speakers and earphones.

The results obtained for several songs indicated that the suitable level of the Daubechies wavelet for noise removal is between 17 and 20. Furthermore, it was observed that in many samples used in this study including the results shown in Table 1, the noise from both channels of a given song was eliminated using the same Daubechies wavelet level. This was a significant improvement when focusing on the simplicity of noise removal compared to the Fourier transforms method.

In the study, 10 s samples from original audio files were used to limit the processing time because the trial-and-error approach taken in the study requires the algorithm to run multiple times for each channel of an audio file to obtain the suitable level of the Daubechies wavelet.

Compared to the results obtained in the previous study in which the Fourier transform was used to remove noise from audio files (Fernando & Kularathne, 2023), the negative impacts on the high-frequency components of the original audio sample were extremely reduced when Daubechies wavelet method was used. Therefore, this technique is ideal to remove the added noise and conserve vintage audio recordings while minimizing the impact on the high-frequency components in the original audio recording.

5. Conclusion

The digital signal processing algorithm developed using Daubechies wavelets transform in this study can be used to remove high-frequency noise from vintage audio recordings and conserve them. Compared to the Fourier transform, the negative impacts on the original audio signal during the noise removal can be minimized by using this method. Results of this study indicate that the high-frequency noise can be successfully eliminated from a given audio file using the Daubechies wavelet level between 17 and 20. This method can be used to conserve vintage audio recordings originally recorded on cassettes and spools.

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