



Review: Audio Noise Reduction using DWT Technique with Hard and Soft Threshold

Nishan Singh

Guru Kashi University (GKU), Talwandi Sabo, Bathinda, Punjab, India

Dr. Vijay Laxmi,

Associate Professor

Guru Kashi University (GKU), Talwandi Sabo, Bathinda, Punjab, India

Abstract: Speech signal analysis is one of the important areas of research in multimedia applications. In paper the speech enhancement is performed using DWT with thresholding and types of wavelet are used to denoised the audio signals and enhance speech and audio signal quality. Our main objective is to reduce noise from system which is heavily dependent on the specific context and application. As, we want to increase the intelligibility or improve the overall speech perception quality. After studying and analyzing, we have concluded that Noise reduction technology is aimed at reducing unwanted ambient sound, and is implemented throughout two different methods with parameters such as SNR, PSNR, MSE and the Time to reduce the noise for noisy signals for removing noise. We can analyze the denoised signal by signal to noise ratio (SNR), mean square error (MSE), Threshold values and elapsed time analysis. In the DWT Coif wavelet with soft threshold is best as compared to coif hard threshold, in DWT soft threshold results are has been best as compared to hard threshold.

Keywords: MSE, SNR, PSNR, DWT denoisng, Coif.

I. INTRODUCTION

Audio noise reduction system is the system that is used to remove the noise from the audio signals. Audio noise reduction systems can be divided into two basic approaches. The first approach is the complementary type which involves compressing the audio signal in some well-defined manner before it is recorded (primarily on tape). On playback, the subsequent balancing extension of the audio signal which restores the original dynamic range, at the same time has the effect of approaching the reproduce tape noise (added during recording) farther below the peak signal level—and hopefully below the threshold of hearing. The second approach is the single-ended or non-complementary type which utilizes techniques to reduce the noise level already present in the source material—in essence a playback only noise reduction system [4]. This approach is used by the LM1894 integrated circuit, designed specially for the decline of audible noise in virtually any audio source. Noise reduction is the process of removing noise from a signal. All soundtrack devices, both analogue or digital, have traits which make them vulnerable to noise. Noise can be random or white noise through no reliability, or consistent noise introduced by the device's mechanism or processing algorithms. There is a Active noise control (ANC), also known as noise cancellation, or active noise reduction (ANR), is a method for reducing unwanted and unprocessed sound by the addition of a second sound specifically designed to cancel the first [7]. Sound is a pressure wave or we can say sound is the analog signals that are processed according to their frequency, which consists of a compression phase and a rarefaction phase. A noise-cancellation amplifier emits a sound wave with the same amplitude but with inverted phase (also known as anti phase) to the original sound. The waves unite to form a new wave, in a process called interfering, and efficiently cancel each other out - an effect which is called termination. New active noise control is generally achieved through the use of analog circuits or digital signal processing. Adaptive algorithms are planned to analyze the waveform of the background no neural noise, then based on the specific algorithm generate a signal that will either phase shift or invert the polarity of the original signal. This anti phase is then amplified and a transducer creates a sound wave directly proportional to the amplitude of the original waveform, creating destructive interference [3]. This effectively reduces the volume of the perceivable noise. The transducer emitting the noise cancellation signal may be located at the location where sound attenuation is wanted (e.g. the user's ear/any music/headphone sound). This requires a much lower power level for cancellation but is effective only for a single user.

II. TYPES OF NOISES

There are numerous type and source of noise or distortions and they include:

Signal distortion is the term often used to describe a systematic undesirable change in a signal and refers to changes in a signal from the non-ideal characteristics of the contact channel, signal fading reverberations, echo,

and multipath reflection and lost samples [10]. Depending on its frequency, spectrum or time characteristics, a noise process is classified into several categories:

1. **White noise:** purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise in theory contains all frequencies in equal power.
2. **Band-limited white noise:** Similar to white noise, this is a noise with a flat influence band and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest. The auto correlation of this noise is sink-shaped.
3. **Narrowband noise:** It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.
4. **Colored noise:** It is non-white noise or any wideband noise whose spectrum has a non flat shape. Such as pink noise, brown noise and auto regressive noise.
5. **Impulsive noise:** Consists of short-duration pulses of random amplitude, time of occurrence and duration.
6. **Transient noise pulses:** Consist of relatively long duration noise pulses such as click, burst noise etc.

III. TECHNIQUES WORKED OUT

A. Discrete Wavelet Transform

Discrete Wavelet Transform (DWT) is introduced to overcome the redundancy problem of CWT. The approach is to scale and translate the wavelets in discrete steps as given in equation (1.6).

$$DWT(\tau_0, s_0) = \frac{1}{\sqrt{s_0^f}} \int_{-\infty}^{\infty} f(t) \psi\left(\frac{t - k\tau_0 s_0^f}{s_0^f}\right) dt$$

Where s_0^f is the scaling factor, τ_0 is the translating factor, k and j are just integers. Subsequently, we can represent the mother wavelet in term of scaling and translation of a dyadic transform as

$$\psi_{j,k}(t) = 2^{-f/2} \psi(2^{-f}t - k)$$

Replacing equation, the coefficients of DWT can be represented as [10]:

$$C_{f,k} = 2^{-f/2} \int_{-\infty}^{\infty} f(t) \psi(2^{-f}t - k) dx$$

The Discrete Wavelet Transform is identical to a hierarchical sub band system where the sub bands are logarithmically spaced in frequency and represent octave-band decomposition [8]. By applying DWT, the image is actually divided i.e., decomposed into four sub-bands and critically sub sampled as shown in Figure 1:

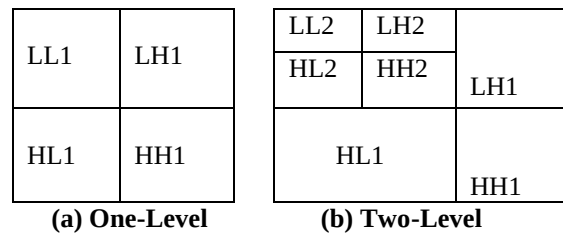


Figure 1: Image Decomposition

These four sub bands arise from separable applications of vertical and horizontal analysis filters for wavelet decomposition as shown in Figure 2.

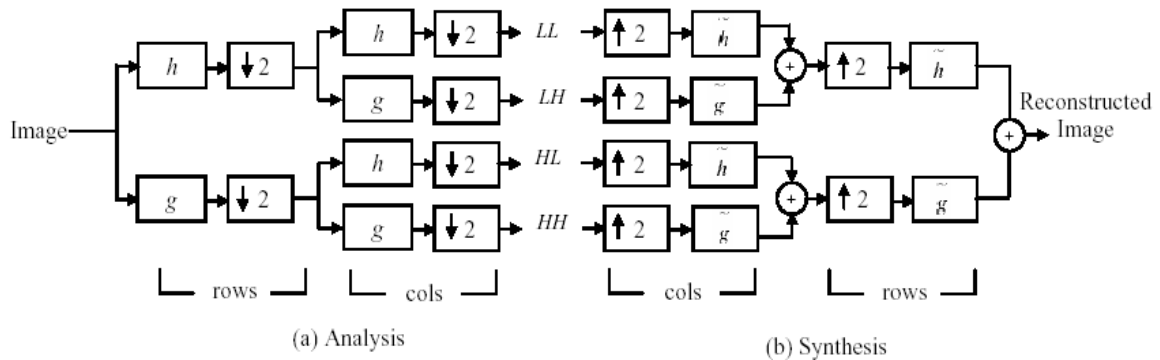


Figure 2: One level filter bank for computation of 2-DWT and Inverse DWT

The filters h and g have shown in Figure 1 is one-dimensional Low Pass Filter (LPF) and High Pass Filter (HPF) respectively. Thus, decomposition provides sub bands corresponding to different resolution levels and orientation. These sub bands labeled LH1, HL1 and HH1 represent the finest scale wavelet coefficients i.e., detail images while the sub band LL1 corresponds to coarse level coefficients i.e., approximation image[10]. To obtain the next coarse level of wavelet coefficients, the sub band LL1 alone is further decomposed and critically sampled using similar filter bank shown in Figure 2 (a). This results in two-level wavelet decomposition as shown in Figure 2(b). Similarly, to obtain further decomposition, LL2 will be used. This process continues until some final scale is reached. The decomposed image can be reconstructed using a reconstruction (i.e., Inverse DWT) or synthesis filter as shown in Figure 2(b). Here, the filters g and h represents low pass and high pass reconstruction filters respectively.

B. Rows transformation

Wavelet decomposition in two dimensions begins with the one dimensional wavelet transform on each row of the image $f(x,y)$. The decomposition process begins with convoluting the rows of $f(x,y)$ with low pass filter coefficients to obtain $L(x)$ and down sampled the wavelet coefficients to retain only the even indexed rows of $f(x,y)$. Next, repeat the process for the rows with high pass filter coefficients to obtain $H(x)$ and similarly retain only the even indexed rows off $H(x,y)$. The need for down sampling by 2 helps to reduce the size of the wavelet coefficients to the original size of $f(x,y)$ as shown in Figure 3[10].

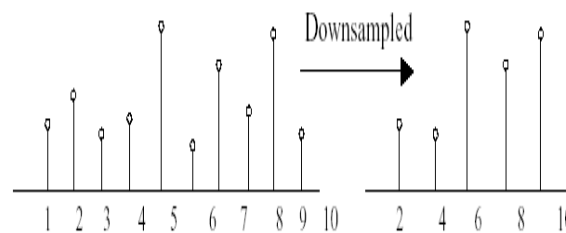


Figure 3: Down Sampling Process

C. Columns transformation

After completing the rows transformation, we perform the decomposition on each column of the image $f(x,y)$. The columns of $f(x,y)$ will convolute with low pass filter coefficients to obtain $L(x)$ and down sampled the wavelet coefficients to retain the even indexed columns of $fL(x,y)$. Repeat the same process for the high pass filter coefficients to $H(x)$ and retains only the even indexed column of $fH(x,y)$ [10].

IV. CONCLUSION & FUTURE WORK

From the above results the DWT Coif wavlet with hard threshold and soft threshold and Sym4 hard and soft threshold is implemented and compared with each others. In this Coif wavelet with soft threshold is best as compared to coif hard threshold and Sym4 wavelet with hard and soft threshold. In DWT soft threshold results are has been best as compared to hard threshold. Future work might involve a real time implementation of the system so that the maximum noise is reduced form the audio signals and videos. In the future anybody can extent the order of the different filters and works on higher amplitude signals. They can calculate the efficiency of the filters that they have to implement. In the DWT we are using coif and sym4 with hard and soft threshold but in the future different types of wavelet is implemented with different types of thresholding techniques or hybrid techniques is designed with the help of filters and wavelets and thresholding techniques. Other things in future the results may be improved in the filters and DWT techniques.

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