A REVIEW: AUDIO NOISE REDUCTION AND VARIOUS TECHNIQUES

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Abstract- During the transmission, storage process mostly occurred noise in audio. De-noising techniques are applied to reduce unwanted signal form audio. In this paper MATLAB is used to perform the algorithms of filters.In this paper DWT and filters are used to minimize the unwanted noise. To reduce the noise from audio is difficult task, so, Butterworth filter and elliptical filters, Chebyshev filters are used with low-pass, high-pass, Band-reject. We present an overview to reduce the noise efficient and give much better results for future research

Index Terms- Butterworth filter, Chebyshev filter, Elliptical filter, Gaussian noise.

I. INTRODUCTION

Audio noise reduction system is that kind of system which is helpful to remove the unwanted noise from speech signals. Audio noise reduction can be classified into two kinds. Complementary type and Non complementary type. Complementary type involves the compression of audio signal and proper way before recorded. Non Complementary Type (single ended type) is an efficient technique to reduce the noise level which is present in source material already [4]. Both analogue and digital devices have particular quality that make them prone to noise. There is an active noise control (ANC) which is also called noise cancellation or active noise reduction (ANR) is technique for reducing the unnecessary sound and that sound which is not processed, by addition of a second sound, specifically designed to cancel the existing one [7]. sound is an analog signal that works on frequency, which comprises of compression phase and rarefaction phase.

II. TYPES OF NOISE

There are many types and sources of noise or distortions and they include:

1. Electronic noise such as thermal noise and shot noise.
2. Acoustic noise emanating from moving, vibrating or colliding sources such as revolving Machines, moving vehicles, keyboard clicks, wind and rain.
3. Electromagnetic noise that can interfere with the transmission and reception of voice.

Communication channel distortion and fading.

6. Quantization noise and lost data packets due to network congestion.

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 communication channel, signal fading reverberations, echo, and multipath reflections and missing samples [10]. Depending on its frequency, spectrum or time characteristics, a noise process is further classified into several categories:

1. White noise: purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.

2. Narrow band noise: It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.

3. Coloured noise: It is non-white noise or any wideband noise whose spectrum has a non flat shape. Examples are pink noise, brown noise and autoregressive noise.

4. Impulsive noise: Consists of short-duration pulses of random amplitude, time of occurrence and duration.
5. Transient noise pulses: Consist of relatively long duration noise pulses such as clicks, burst noise etc.

III. FILTERS

The main function of a filter can be elaborated by testing the frequency dependent nature of the impedance of inductors and capacitors [3].As the frequency changed the values of both reactive impedance changes, voltage divider ratio also changes respectively. This operation produces the change in i/o transfer function on the depended of frequency, It is known as frequency response [11].

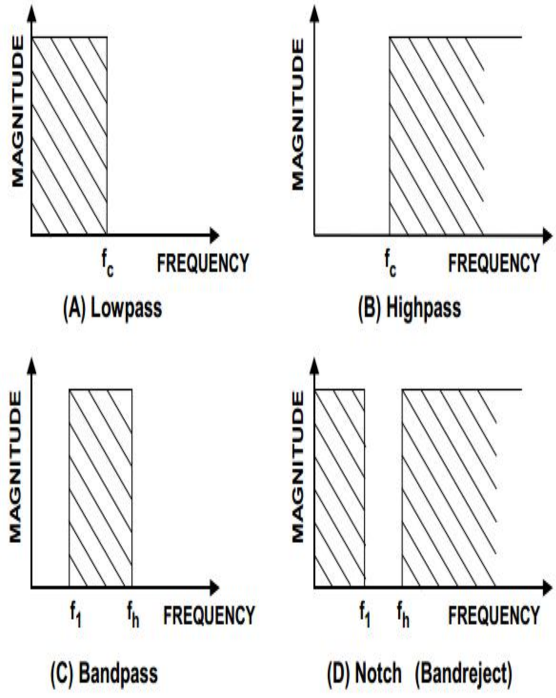

Fig.1.1 Types of filters [1]

The functional complement to the low-pass filter is the high-pass filter. Here, the low frequencies are in the stop-band, and the high frequencies are in the pass band. Figure shows the idealized high-pass filter, low-pass filter, band-pass filter, band-reject filter [1].

IV. METHODS OF FILTERS

(a) Butterworth filter
(b) Chebyshev filter
(c) Elliptical filter

a) BUTTERWORTH FILTER

The Butterworth filter is the best compromise between attenuation and phase response. It has no ripple in the pass band or the stop band, and because of this is sometimes called a maximally flat filter [1].The Butterworth filter achieves its flatness at the expense of a relatively wide transition region from pass band to stop band, with average transient characteristics. The normalized poles of the Butterworth filter fall on the unit circle [7].

The poles are spaced equidistant on the unit circle, which means the angles between the poles are equal. Given the pole locations, $H_0\omega_0$, andα_0 (or Q) can be determined. These values can then be use to determine the component values of the filter. The design tables for passive filters use frequency and impedance normalized filters. The Butterworth filter is normalized for a -3 dB response at $H_0\omega_0$ = 1[8].

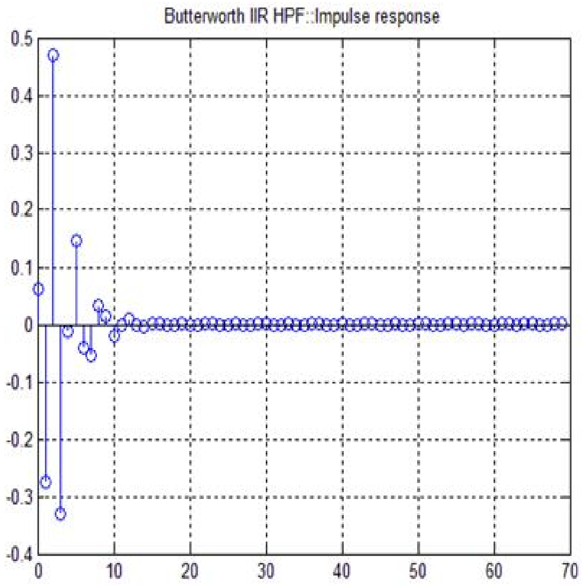

Fig.1.2. Butterworth frequency response[6] [7]

b) CHEBYSHEV FILTER

Chebyshev1 filters have a narrower transition region between the pass band and the stop band[1]. The sharp transition between the pass band and the stop band of a chebyshev filter produces smaller absolute errors and faster execution speeds than a Butterworth filter.

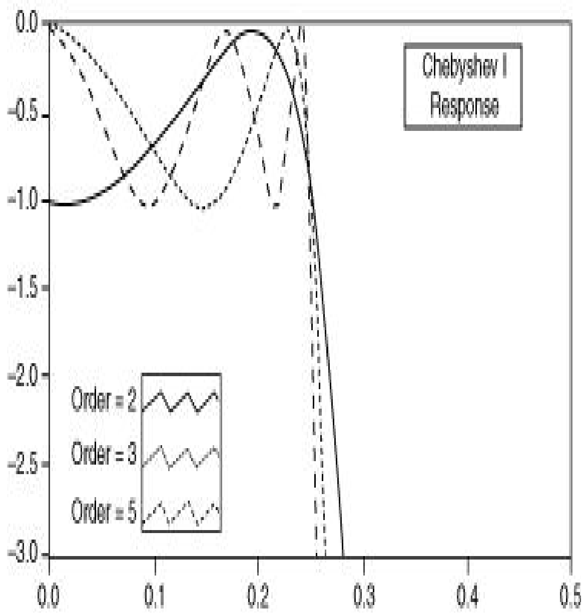

Fig.1.3 Chebyshev frequency response [6]

c) ELLIPTICAL FILTER

Butterworth And Chebyshev filters are all-pole designs, which mean that the zeros of the transfer function (roots of the numerator) are at one of the two extremes of the frequency range (0 or ∞). For a low-pass filter, the zeros are at f =∞ [12]. If finite frequency transfer function zeros are added to poles an Elliptical filter.

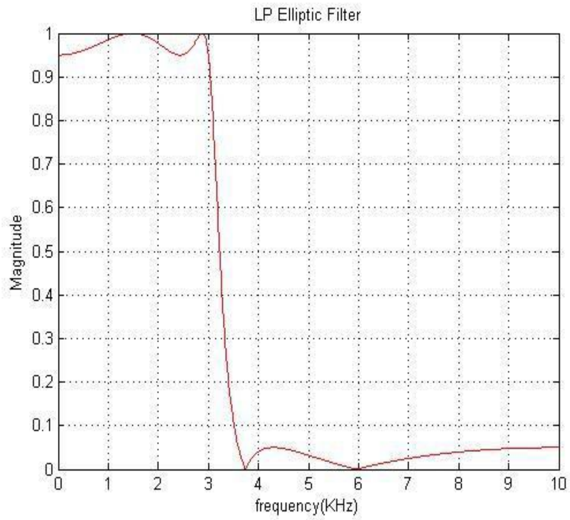

Fig. 1.4 Step and impulse response [13]

V. INTRODUCTION TO WAVELET TRANSFORM

Wavelet transform consists of a set of basis functions that can be used to analyze signals in both time and frequency domains simultaneously. This analysis is accomplished by the use of a scalable window to cover the time-frequency plane, providing a convenient means for the analyzing of non-stationary signal that is often found in most application [8].
Wavelet analysis adopts a wavelet prototype function known as the mother wavelet given as:

$$\psi(\tau, s) = \frac{1}{\sqrt{s}} \psi\left(\frac{t-\tau}{s}\right) \qquad (1.1)$$

This mother wavelet in turns generates a set of basis functions known as child wavelets through recursive scaling and translation.
Where, s reflects width of a basis function,
τ, is translation that specifies its translated position on time axis,
$\psi\left(\frac{t-\tau}{s}\right)$ is mother wavelet.

CONTINUOUS WAVELET TRANSFORM

CWT analyzes the signal through the continuous shifts of a scalable function over a time plane. This technique results in redundancy and it is numerically impossible to analysis at infinite number of wavelet sets [10].

SOME STUDIES ON AUDIO NOISE REDUCTION SURVEY

Shankar and Duraiswamy, 2012 proposed the noises present in communication channels are disturbing and the recovery of the original signals from the path without any noise is very difficult task. This is achieved by de-noising techniques that remove noises from a digital signal. Many de-noising technique have been proposed for the removal of noises from the digital audio Signals.

Charles et al., 2013 presented a new adaptive filter whose coefficients are dynamically changing with an evolutionary computation algorithm and hence reducing the noise. This algorithm gives a relationship between the update rate and the minimum error which automatically adjusts the update rate.

Jebastine and Rani, 2012 described the development of an adaptive noise cancellation algorithm for effective recognition of speech signal and also to improve SNR for an adaptive step size input. An adaptive filter with Fast Block Least Mean square Algorithm is designed for noise free audio (speech/music) signals. The signal input used is a audio speech signal which could be in the form of a recorded voice.

Obulesu and kumar, 2013 studied the audio signals are synthetic signals, in which music or speech, are often corrupted by noise during recording and transmission. Audio de-noising procedures are designed to attenuate the noise and retain the signal of interest. This Block thresholding method eliminates "musical noise" by grouping Time-frequency coefficients in blocks before being attenuated.

Sampat and Vithalani, 2012 presented the de-noising of one dimensional signal using threshold is one of the major applications of wavelet transform. Determination of threshold type and threshold value is one of the important tasks in threshold based de-noising techniques. De-noising of audio signal is a subjective matter and remains a valid challenge. Eight different de-noised files are generated. Various parameters are measured and compared.

Niedzwiecki and Ciolek, 2013 presented the digital videos are often corrupted by a noise during the acquisition process, storage and transmission. It made the video in ugly appearance .It is applied with Two Dimensional Fast Discrete Wavelet Transform (2-D FDWT), Three Dimensional Fast Discrete Wavelet Transform (3-D FDWT), Two Dimensional Double Density Wavelet Transform (2-D DDWT) and Three Dimensional Double Density Wavelet Transform (3-D DDWT).

Raghavendra and PremPyara, 2013 studied a robust DWPT based adaptive bock algorithm with modified threshold for de-noising the sounds of musical instruments shehnai, dafli and flute is proposed. Hence, the optimal wavelet and level of decomposition may be different for each signal. The obtained de-noised signal with this algorithm is close to the original signal.

Ramli et al., 2012 presented a new adaptive filter whose coefficients are dynamically changing with an evolutionary computation algorithm and hence reducing the noise. This algorithm gives a relationship between the update rate and the minimum error which automatically adjusts the update rate. Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal.

CONCLUDING REMARKS

The different filters are studied to reduce the noise from the digital signals. The non-additive noise includes multiplier noise and convolution noise, which can be transformed into additive noise through homomorphism transform. The additive noise includes periodical noise, pulse noise, and broadband noise related problems. The noise generated by the engine is one kind of periodical noise while the one generated from explosion, bump, or discharge is pulse noise problem. Butterworth filter, that is used to reduce the noise from the signals with the different frequency and ripple factor. So order of the filters for different center frequencies was investigated for different filters. And it can be concluded that for different center frequencies, order of the filter always remains same. In the future the signal related noise is reduced with the help of Chebyshev filter and DWT technique and produce the different SNR and thersholding values and their pole etc

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