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AUDIO NOISE REDUCTION USING DIFFERENT FILTERS

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Abstract: Speech signal analysis is one of the important areas of research in multimedia applications. Digital filters effectively reduce the unwanted higher or lower order frequency components in a speech signal. 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. There are many kinds of broadband noise, which may include heat noise, wind noise, quantization noise, and all kinds of random noise such as white noise and pink noise. Statistical relationship between the noise and speech; i.e. uncorrelated or even independent noise, and correlated noise (such as echo and reverberation). In acoustics applications, noise from the surrounding environment severely reduces the quality of speech and audio signals. Therefore, basic linear filters are used to denoise the audio signals and enhance speech and audio signal quality. Our objective is of a noise reduction system with heavily dependent on the specific context and application as to increase the intelligibility or improve the overall speech perception quality which aimed to reduce unwanted ambient sound by implementing through different filters. We have studied and analyzed the results of various filters using different types of parameters like SNR, PSNR, MSE and the Time to reduce the noise for noisy signals.

Keywords: Butterworth filter; Chebyshev filter; Impulsive keyboard noise; Speech enhancement.

I. INTRODUCTION

A. Audio Noise

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 complementary expansion of the audio signal which restores the original dynamic range, at the same time has the effect of pushing the reproduced 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 specifically for the reduction of audible noise in virtually any audio source. Noise reduction is the process of removing noise from a signal. All recording devices, either analog or digital, have traits which make them susceptible to noise. Noise can be random or white noise with no coherence, or coherent noise introduced by the device's mechanism or processing algorithms [5]. 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 speaker emits a sound wave with the same amplitude but with inverted phase (also known as anti phase) to the original sound [1]. The waves combine to form a new wave, in a process called interference, and effectively cancel each other out - an effect which is called cancellation. Modern active noise control is generally achieved through the use of analog circuits or digital signal processing. Adaptive algorithms are designed to analyze the waveform of the background neural noise, then based on the specific algorithm generate a signal that will either phase shift or invert the polarity of the original signal [6]. 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 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, image and data over the radio-frequency spectrum.
4. Electrostatic noise generated by the presence of a voltage.
5. 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 [8].

Depending on its frequency, spectrum or time characteristics, a noise process is further classified into several categories:
A.White noise: purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.
B. Band-limited white noise: Similar to white noise, this is a noise with a flat power spectrum and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest. The autocorrelation of this noise is sink-shaped.
C. Narrowband noise: It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.
Colored 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.
D. Impulsive noise: Consists of short-duration pulses of random amplitude, time of occurrence and duration.
Transient noise pulses: Consist of relatively long duration noise pulses such as clicks, burst noise etc.

III. FILTERS

Filters are networks that process signals in a frequency-dependent manner. The basic concept of a filter can be explained by examining the frequency dependent nature of the impedance of capacitors and inductors. Consider a voltage divider where the shunt leg is reactive impedance. As the frequency is changed, the value of the reactive impedance changes and the voltage divider ratio changes. This mechanism yields the frequency dependent change in the input/output transfer function that is defined as the frequency response [2]. Filters have many practical applications. A simple, single-pole, low-pass filter (the integrator) is often used to stabilize amplifiers by rolling off the gain at higher frequencies where excessive phase shift may cause oscillations. A simple, single-pole, high-pass filter can be used to block dc offset in high gain amplifiers or single supply circuits. Filters can be used to separate signals, passing those of interest, and attenuating the unwanted frequencies.

IV. BASIC LINEAR DESIGN

Low Pass filter passes the frequency from zero up to some designated frequency, called as cut-off frequency. After cut-off frequency; it will not allow any signal to pass through it. The low frequencies are in the pass-band and the high frequencies in the stop-band. Figure shows the idealized low pass filter. 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. If a high-pass filter and a low-pass filter are cascaded, a *band pass* filter is created. The band pass filter passes a band of frequencies between a lower cutoff frequency, f l, and an upper cutoff frequency, f h. Frequencies below f l and above f h are in the stop band. An idealized band pass filter is shown in figure. A complement to the band pass filter is the band-reject, or notch filter. The idealized filters defined above, unfortunately, cannot be easily built. The transition from pass band to stop band will not be instantaneous, but instead there will be a transition region. Stop band attenuation will not be infinite.

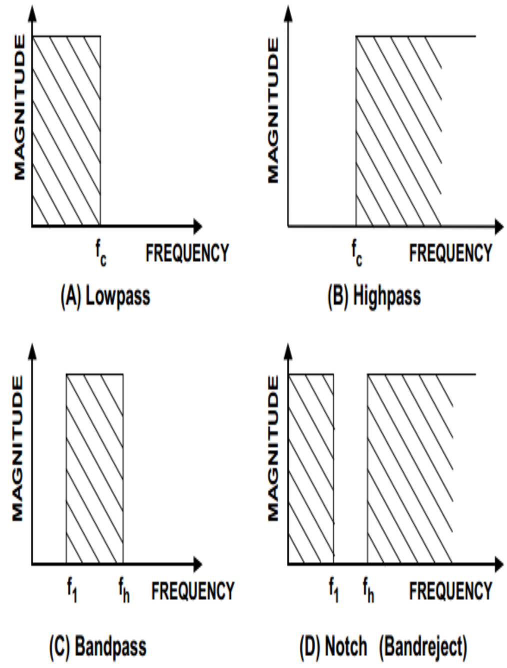

Figure 1. TYPES OF FILTER

V. RESULTS

PSNR value for different Filter

Name of Signal	Butterworth Filter (8th order)	Chebyshev Type-1 Filter (8th order)	Elliptic Filter (8th order)
1N.wav	36.2673	33.3970	33.4022
2N.wav	32.2801	30.8981	30.8981
3N.wav	35.8952	34.4867	34.4881

SNR value for different Filter

Name of Signal	Butterworth Filter (8th order)	Chebyshev Type-1 Filter (8th order)	Elliptic Filter (8th order)
1N.wav	-286.642	0.3853	-0.1369
2N.wav	-138.194	0.0033	-0.0094
3N.wav	-139.506	1.3363	1.1963

MSE value for different Filter

Name of Signal	Butterworth Filter (8th order)	Chebyshev Type-1 Filter (8th order)	Elliptic Filter (8th order)
1N.wav	2722.56	1405.89	1407.58
2N.wav	1664.17	1210.60	1210.58
3N.wav	94.2837	68.1701	68.1921

Total Time Elapsed value for different Filter

Name of Signal	Butterworth Filter (8th order)	Chebyshev Type-1 Filter (8th order)	Elliptic Filter (8th order)
1N.wav	0.0052090	0.0044865	0.0045901
2N.wav	0.0019367	0.0012391	0.0019504
3N.wav	0.0043975	0.0043692	0.0043706

VI. CONCLUSION & FUTURE WORK

The different filters with different frequencies are used to remove noise. It can be concluded that for different center frequencies, order of the filter always remains same. By changing its center frequencies filters are being tuned to different frequencies. We have designed Butterworth filter, Chebyshev filter type-1 and Elliptic filter to remove the noise. We have designed these filters with different type's i.e. low pass, high pass, band pass and band reject and the order of these filters are same. By using this we can get the better results of de-noising, especially for low level noise. It is generally not possible to filter out all the noise without affecting the original signal. We can analyze the denoised signal by signal to noise ratio (SNR), mean square error (MSE), peak signal to noise ratio (PSNR) and elapsed time analysis. From the above results the Chebyshev Type-1 Filter (8th order) is the best as compared to the other filters because in Chebyshev Type-1 Filter (8th order) SNR is high and ellapsed time is less as compared to other filters but the PSNR is high and SNR is negative in case of Butterworth filter.

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