Audio Noise Reduction using Butter worth Filter

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Abstract: *Digital filters effectively reduce the unwanted higher or lower order frequency components in a speech signal. In this paper the speech enhancement is performed using different digital filters .In this real noisy environment is taken into consideration in the form of Gaussian noise. The Time domain as well as frequency domain representation of the signal spectra is performed using Fast Fourier transformation technique. MATLAB in built functions are used to carry out the simulation. Gaussian type noise is added using in-built function randn () and keyboard noise is added as a second speech file to the original speech signal. The filters remove the lower frequency components of noise and recover the original speech signal. It is also observed that keyboard noise is typical to remove as compared to Gaussian type but these filters performed well to get sharper spectra of original speech signal.*

Keywords: *Butterworth filter; Gaussian noise; Impulsive keyboard noise; Speech enhancement.*

I. INTRODUCTION

1.1. 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, both analogue 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. Their 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 speaker emits a sound wave with the same amplitude but with inverted phase (also known as anti phase) to the original sound. 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 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 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 and

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. 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 sinc-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. Examples are pink noise, brown noise and autoregressive 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 clicks, burst noise etc.

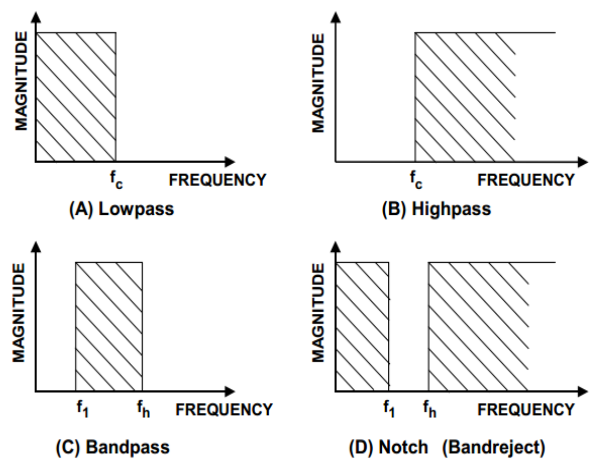
III. FILTER

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[12].

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

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. 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.



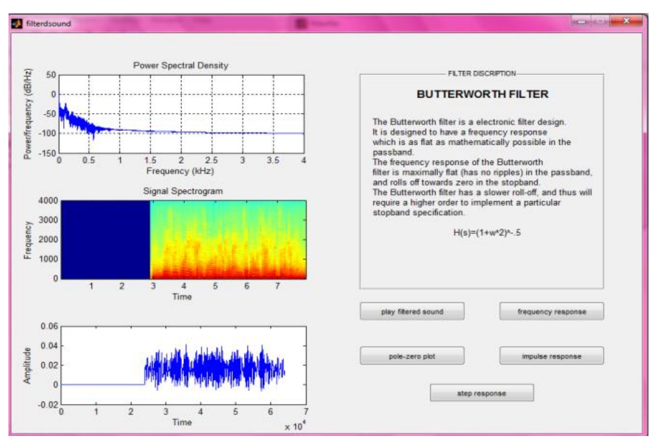


Figure 1(a) . Types of filter

V. PROBLEM DEFINITION

The Current applications include noise propagation problem in industrial air handling systems, noise in aircrafts and tonal noise from electric power, as well as isolation of vibration from noise is one kind of sound that is unexpected or undesired . The noise related problem that I have studied can be divided into non-additive noise and additive noise. 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 that I have studied in literature survey. 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. In acoustics applications, noise from the surrounding environment severely reduces the quality of speech and audio signals.

VI. RESULTS & ANALYSIS

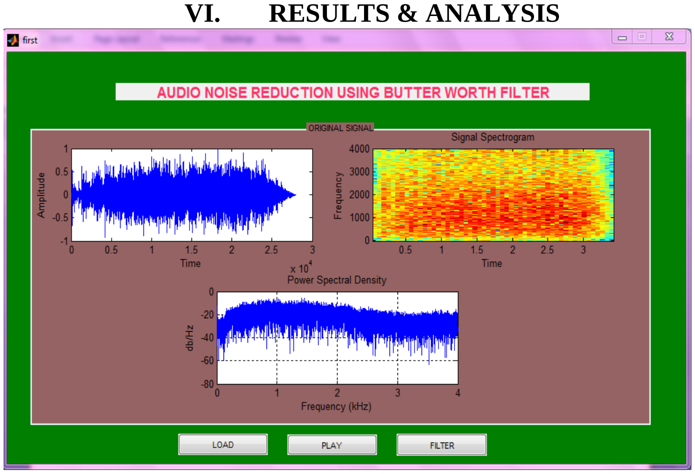


Figure 1(b). Upload Audio Signal File

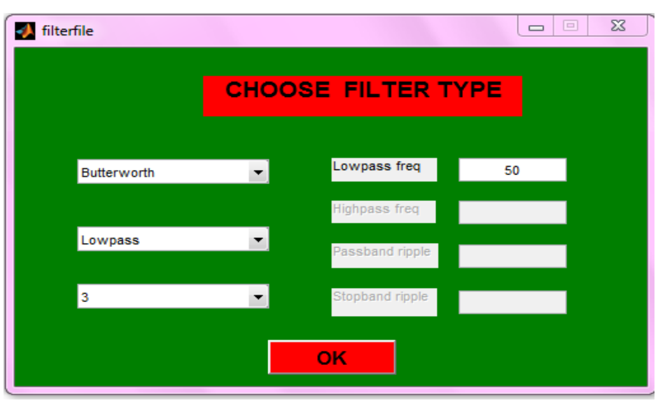


Figure 1(c). Choose Filter Type

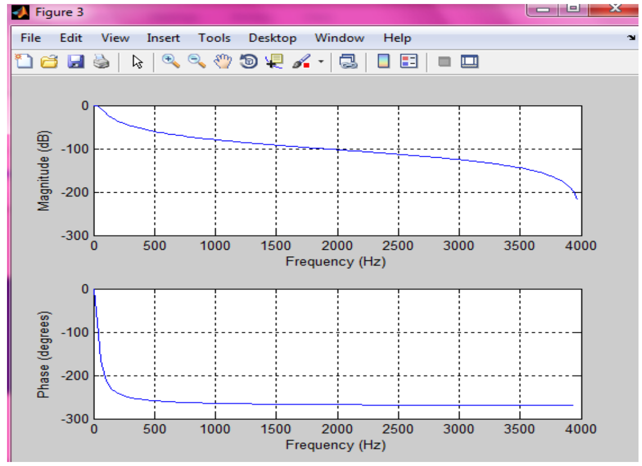


Figure 1 (d) Butter worth Filter result

Figure 1(e) Frequency Response of Butterworth Filter

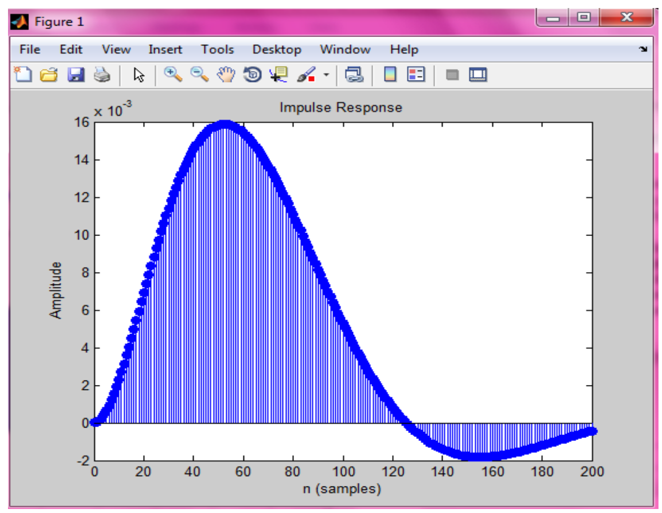


Figure 1(f) Impulse Response of Butterworth Filter

VII. CONCLUSION & FUTURE WORK

In This Paper we concluded the Butterworth filter that is used to reduce the noise from the signals with the different frequency and ripple factor. The order of different frequency produces the different results that are shown in the result and analysis section. Their are different types of frequency and impulse graph produced according to audio signals. In the future the signal related noise is reduced with the help of Chebyshev filter and DWT technique and produce the different SNR and thresholding values and their pole etc.

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