Review Paper on Frequency Based Audio Noise Reduction Using Different Filters

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Abstract : **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.**

I. INTRODUCTION:

The intelligibility of human speech plays an important part in communication. It is both a measure of comfort and comprehension. The quality and intelligibility of the speech are not only determined by physical characteristics of the speech itself but also by communication conditions and information capacity, the ability to get the information from context, mimics and gestures. When discussing intelligibility it is important to understand the difference between a real and recorded speech. During a real conversation a person can recognize the surrounding sounds and concentrate on the speech of another person thus filtering the desired information out of various audio environments. Therefore the ability of a human to recognize and filter sounds significantly increases the intelligibility and comprehension of the speech even if a communication takes place in a noisy environment, situation or condition. Listening to recorded speech is a different situation. The recording equipment doesn't focus on certain audio streams (unless it is a specialized shotgun microphone) and impartially record everything that happens in the audio spectrum. As a result we receive a “flat picture” of all recorded sounds which often makes the speech unintelligible, quiet and buried in the noises. Additional reasons why speech recordings may be indistinct and distorted can be due to technical limitations of recording equipment, poorly placed or defective microphones and objective difficulties to record high quality "clean" sound. Generally all audio hindrances are divided into two main categories: noises and distortions. If we consider an original human speech in a recording as a useful signal all the additional information which decreases the quality of a useful signal are noises. Everything that changes the original useful signal itself are considered distortions.

Noises are mostly characterized by time and frequency (domains). In time domain noises can be:

Continuous, slowly changing noises, like the sound of an office, industrial equipment, sound of the wind, traffic, hiss of an old record or a bad phone line.

Discontinuous, repeated, usually tonal noises like honks, beeps or bells.

Pulse like, abrupt, usually unharmonious and sometimes loud noises like clicks, taps of the steps, gunshots, bangs and thumps. In frequency domains noises can be:

Broad band noises which present at many frequencies like background hiss or fizzing sounds.

Narrow band noises which represent a set of certain frequencies, fairly stable tonal sine waves (sinusoid): drones, power-supply hums, equipment hindrances (drills, chainsaws) machinery engine noises.

Distortions are modifications of the useful speech signal that decrease its quality. When distortions occur, parts or the whole speech signal changes and become new and sometimes can sound unacceptable. Typical distortions at acoustical level are reverberation and echo effects.

II. Active vs. passive noise control:

Noise control is an active or passive means of reducing sound emissions, often for personal comfort, environmental considerations or legal compliance. Active noise control is sound reduction using a power source. Passive noise control is sound reduction by noise-isolating materials such as insulation, sound-absorbing tiles, or a muffler rather than a power source [1].

Active noise canceling is best suited for low frequencies. For higher frequencies, the spacing

ISSN: 2278 – 7798

requirements for free space and zone of silence techniques become prohibitive. In acoustic cavity and duct based systems, the number of modes grows rapidly with increasing frequency, which quickly makes active noise control techniques unmanageable. Passive treatments become more effective at higher frequencies and often provide an adequate solution without the need for active control [3].

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. LITERATURE SURVEY:

Romain Serizel, Marc Moonen [2010] – has presented the combined active noise control and noise reduction schemes for hearing aids to tackle secondary path effects and effects of noise leakage through an open fitting. Such leakage contributions affect the noise signals. The result of these signals appears to have a non-negligible impact on the final signal-to-noise ratio. The author studied a noise-reduction algorithm and an active noise control system in cascade may be efficient as long as the causality margin of the system is large enough. A Filtered-x Multichannel Wiener Filter is presented and applied to integrate noise reduction and active noise control. The cascaded scheme and the integrated scheme are compared experimentally with a Multi channel Wiener Filter in a classic noise reduction framework without active noise control, where the integrated scheme is found to provide the best performance [1].

***Eric Martin [2012*]-** have introduces an adaptive audio block thresholding algorithm. The denoising parameters are computed according to the time-frequency regularity of the audio signal using the SURE (Stein Unbiased Risk Estimate) theorem. The author studied unlike the diagonal estimators, the adaptive audio block thresholding algorithm based on a non-diagonal estimator is very much elective with white noise. However there are some defects. The sounds which are like a white Gaussian noise will be deleted. For instance, it’s impossible to hear cymbals from a drum kit after a denoising [2].

B. JaiShankar and K. Duraiswamy [2012]- have introduced 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 denoising techniques that remove noises from a digital signal. Many denoising technique have been proposed for the removal of noises from the digital audio Signals. But the effectiveness of those techniques is less. In this paper, an audio denoising technique based on wavelet transformation is proposed [3].

C Mohan Rao, Dr. B Stephen Charles, Dr. M N Giri Prasad [2013]- have presents 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. When the environment is varying, the rate is increased while it would be decreased when the environment is stable and the computation complexity of adaptive filter can be significantly reduced. In the simulation, additive white Gaussian noise is added to the randomly generated information signal and efficiently reduced this noise with minimum or no error by using evolutionary computation with Least Mean Square (LMS) algorithms. Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal [4].

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

ISSN: 2278 – 7798

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.

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, different filters are used to denoised the audio signals and enhance speech and audio signal quality.

VI. OBJECTIVES OF WORK:

- The objective of a noise reduction system is heavily dependent on the specific context and application. In some scenarios, for example, we want to increase the intelligibility or improve the overall speech perception quality.
- Study of noise cancellation Techniques.
- To Implement the Exiting Various Techniques studied as in literature review such as Butter worth filter.
- Study and Analyze the Results Being Obtained such as frequency, time signal and phase angle.
- Noise reduction technology is aimed at reducing unwanted ambient sound, and is implemented through different methods.

VII. METHODOLOGY:

This dissertation is removes noise from the audio signal. It is based upon GUI (graphical user interface) in MATLAB. It is an effort to further grasp the fundamentals of MATLAB and validate it as a powerful application tool. There are basically different files. Each of them consists of m-file and figure file. These are the programmable files containing the information about the filter and figure files are the way to analyze the given audio and enter the various filter related data. In this work we will firstly upload the sound in the format .wav in the given window**.** Listen the sound which will appear to be noisy .In the GUI we will take the filter button and when click on the filter sound button than a new window will open named filter sound. then choose the desired filter for denoising and enter its various parameters namely type , order, pass-band frequency, stop-band frequency, pass-band ripple and stop band ripple. Then click on ok than a new window filtered sound will open. This window shows filtered sound along with the graphs and various details about the filters like Transfer function, Step impulse and Frequency response.

VIII. REFERENCES:

[1] Roshahliza M. Ramli, Ali O. Abid Noor, and Salina Abdul Samad ***"A Review of Adaptive Line Enhancers for Noise Cancellation"*** Australian Journal of Basic and Applied Sciences, 6(6): 337-352, 2012 ISSN 1991-8178.

[2] Eric Martin**," *Audio denoising algorithm with block thresholding*"** Published in Image Processing On Line on YYYY {MM {DD.ISSN 2105-1232.

[3] B. JaiShankar1 and K. Duraiswamy **"audio denoising using wavelet transform"** International Journal of Advances in Engineering & Technology, Jan 2012. ISSN: 2231-1963

[4] C Mohan Rao1, Dr. B Stephen Charles ***"A Variation of LMS Algorithm for Noise Cancellation"*** International Journal of Advanced Research in Computer and Communication Engineering Vol. 2, Issue 7, July 2013 ISSN (Print) : 2319-5940 .

[5] B. Jai Shankar, K.Duraiswamy **"signal denoiser using wavelets and block matching process "**Asian Journal of Computer Science and Information Technology2: 1 (2012) 1 – 3. **ISSN 2249 - 5126** .

[6] J. Jebastine , Dr. B. Sheela Rani **"design and implementation of noise free Audio speech signal using fast block least Mean square algorithm"** Signal & Image Processing : An International Journal (SIPIJ) Vol.3, No.3, June 2012.

[7] Guoshen Yu, Stéphane Mallat **"Audio Denoising by Time-Frequency Block Thresholding "**IEEE TRANSACTIONS ON SIGNAL PROCESSING, VOL. 56, NO. 5, MAY 2008.

[8] Raghavendra Sharma, Vuppuluri Prem Pyara**" A Robust Denoising Algorithm for Sounds of Musical Instruments Using Wavelet Packet Transform "***Circuits and Systems*, 2013, 4, 459-465 Published Online November 2013.

[9] Matheel E. Abdulmunim, Rabab F. Abass **"Novel Video Denoising Using 3-D Transformation Techniques "**International Journal of Engineering and Advanced Technology (IJEAT) ISSN: 2249 – 8958, Volume-2, Issue-5, June 2013.

[10] K.P. OBULESU1 P. UDAY kumar **"implementation of time frequency block thresholding algorithm in audio noise reduction "**ISSN: 2278 – 7798 International Journal of Science,

ISSN: 2278 – 7798

Engineering and Technology Research (IJSETR) Volume 2, Issue 7, July 2013 .

[11] S. N. Sampat1*, Dr. C. H. Vithalani**" Customized Neighborhood Threshold Speech Denoising Using Wavelet Transform Based On Filter Bank Method "**international journal of darshan institute on engineering research & emerging technologies vol. 2, no. 1, 2013 .

[12] Rajeev Aggarwal**" Noise Reduction of Speech Signal using Wavelet Transform with Modified Universal Threshold "**International Journal of Computer Applications (0975 – 8887) Volume 20– No.5, April 2011.

[13] Kai Siedenburg1, Monika D¨orfler **"Audio Denoising by Generalized Time-Frequency Thresholding "**aes 45th international conference, helsinki, finland, 2012 march 1–4.

ISSN: 2278 – 7798