ISSN: 2278-0181
Vol. 2 Issue 6, June - 2013

Improved Audio Filtering Using Extended High Pass Filters

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Abstract—Noise Filtering in audio signals has always been a challenge since the noise is spread across a large bandwidth and overlaps the spectral range of the audio signal being recovered. In audio systems all these can be kept below the audible level but ambient noise may not be avoided even if the audio system is designed according to the best practices. So, there have been several audio filtering techniques such as spectral subtraction, Dolby noise reduction, use of low-pass and high pass filters, FIR and IIR filtering, etc. Noise filtering improvements were assessed for both noise reduction and signal degradation effects by different signal to noise ratio computations.

Keywords: Audio Filtering, Low Pass Filters, High Pass Filters, FIR Filters, IIR Filters.

I. INTRODUCTION

With the development of communication technology, voice communication has become a major communication medium for people to transmit information more convenient. However, the widespread nature of noise makes the voice communication quality has declined. Therefore, to reduce the noise on the performance of voice communications, improve the quality of voice communications [1], voice denoising for technology has become a hot research topic. In this paper we will discuss various audio filtering techniques that will help in overcoming noise related issues.

Audio Filter: - Is a frequency dependent amplifier circuit, working in the audio frequency range,0Hz- beyond 20 KHz. Many types of filters exist for applications including graphic equalizers [2], synthesizers, sound effects, CD Players and Virtual Reality Systems.

The various types of filters can be defined according to the following classification:

- ✓ **Low pass (LP)** filters select low frequencies up to the cut-off frequency f_c and attenuate frequencies higher than f_c

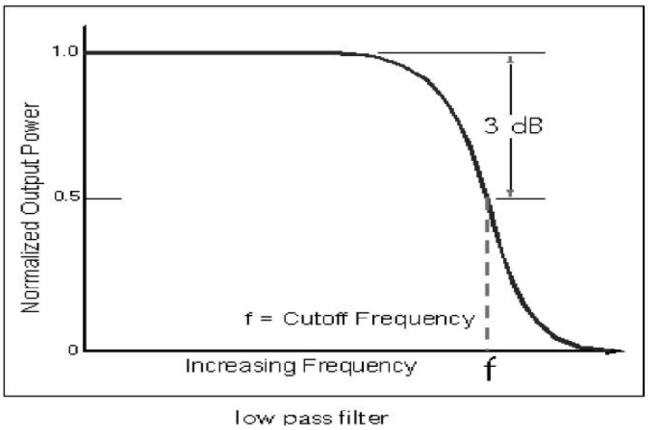
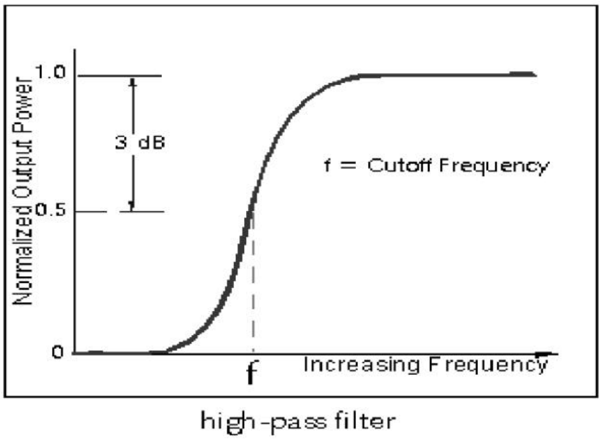


Figure 1: low pass filter

- ✓ **High pass (HP)** filters select frequencies higher than f_c and attenuate frequencies below f_c.

Figure 2: high pass filter



- ✓ **Band pass (BP)** filters select frequencies between a lower cut-off frequency f_{cl} and a higher cut-off frequency f_{ch}. Frequencies below f_{cl} and frequencies higher than f_{ch} are attenuated.

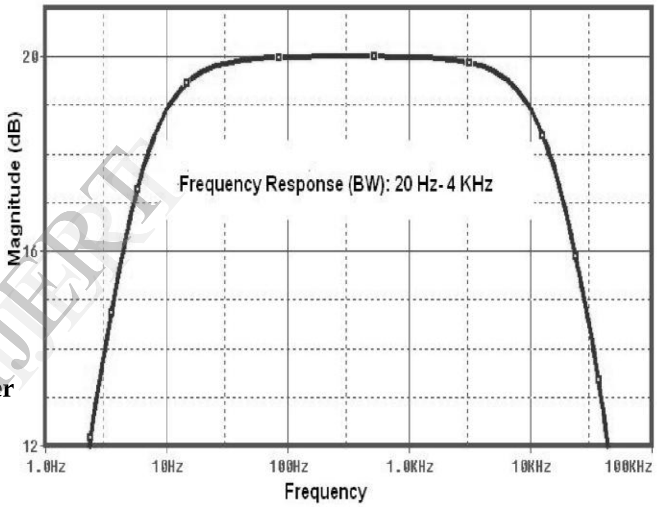
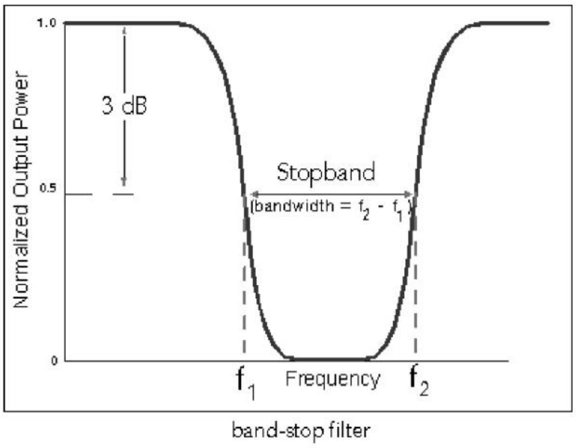


Figure 3: band pass filter

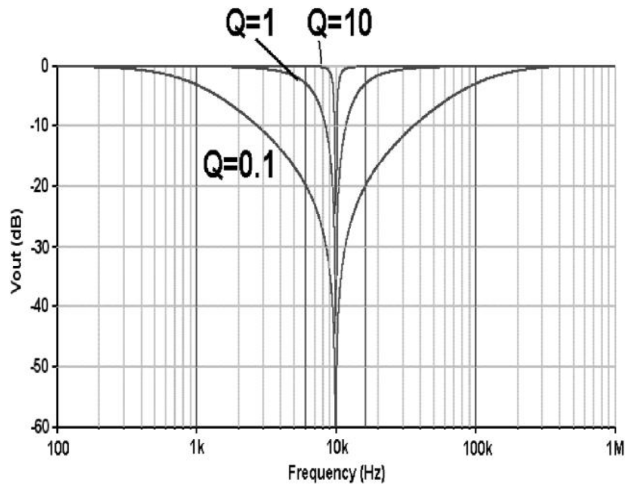
- ✓ **Band reject (BR)** filters attenuate frequencies between a lower cut-off frequency higher cut-off frequency fch. Frequencies below fcl and frequencies higher than fch are attenuated.

Figure 4: band reject filter



✓ **Notch** filters attenuate frequencies in a narrow bandwidth around the cut-off frequency f_c.

Figure 5: notch filter



✓ **Time-domain filters**

1) Finite Impulse Response (FIR) filters- A finite impulse response (FIR) filter is one in which the output signal goes to zero in some finite amount of time.

2) Infinite Impulse Response (IIR) filters- An infinite impulse response (IIR) filter may run for an arbitrarily long period of time and never have the output go to zero.

The difference between the two types of filters is that an IIR filter is recursive; i.e., an IIR filter will take its previous output and use it as input for the next output, whereas an FIR filter will take as input the current sample in addition to some previous sample. In other words, an IIR filter has internal feedback while an FIR filter does not.

Figure 6: FIR filter

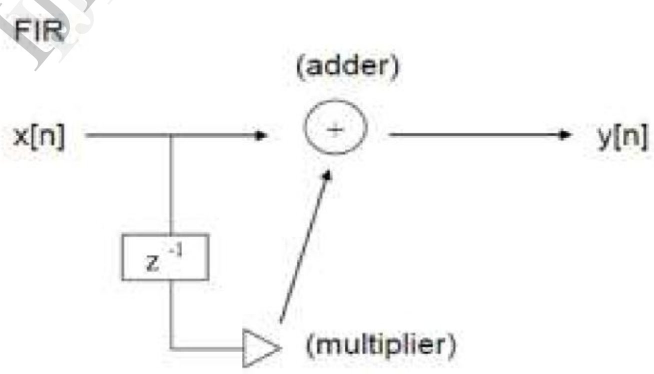
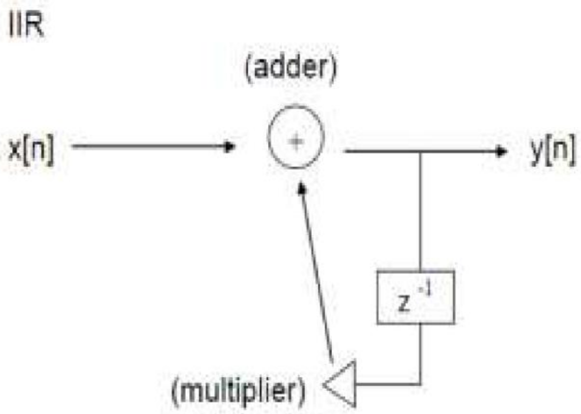


Figure 7: IIR filter



II. RELATED WORK

O'hern, William A. (2012.)[1] in "System and method for filtering audio content." U.S. Patent No. 8,156,518 has discussed that-A system and method for filtering audio content are provided. An audio data filtering system may include an input interface to receive audio data. The system may also include an analysis module to compare a textual representation of the audio data to data identifying prohibited audio content. The system may also include a packet filter to prevent transmission of portions of the audio data[3] matching prohibited audio content to remote user device. The system may further include an output interface to transmit the portions of the audio data not identified as prohibited audio content to the remote user device.

SongYu, Lihue (2010)[2] International Conference on Computer, Mechatronics, Control and Electronic Engineering (CMCE))in "The filter in the DAC of audio processing"-This paper designs a kind of filters used in audio processing of the DAC. Through the study of FIR filter parameters, this paper completed the over-sampling of the audio filter, so that over-sampling rate achieves 128 times. It also use the analysis tools in MATLAB fdatool to design filter[4].From the simulation waveform of it ,the noise of signal ratio is achieved more than 100 dB, and it can better suppress pass-band noise.

Ibrahim Almajai and Ben Milner (IEEE transactions on audio, speech, and language processing, vol. 19, no. 6, august 2011)[5] "Visually Derived Wiener Filters for Speech Enhancement"-The aim of this work is to examine whether visual speech information can be used to enhance audio speech that has been contaminated by noise. First, an analysis of audio and visual speech a feature is made, which identifies the pair with highest audio-visual[6] correlation. The study also reveals that higher audio-visual correlation exists within individual phoneme sounds rather than globally across all speech. This correlation is exploited in the proposal of a visually derived Wiener filter that obtains clean speech and noise power spectrum statistics from visual speech features. Clean speech statistics are estimated from visual features using a maximum *a posteriori* framework that is integrated within the states of a network of hidden Markov models to provide phoneme localization. Noise statistics are obtained through a novel audio-visual voice activity detector which utilizes visual speech features to make robust speech/no speech classifications. The effectiveness of the visually derived Wiener filter is evaluated subjectively and objectively and is compared with three different audio-only enhancement methods over a range of Signal-to-noise ratios.

Zhiyao Duan (IEEE journal of selected topics in signal processing, vol. 5, no. 6, October 2011)[7]"Sound prism: An Online System for Score-Informed Source Separation of Music Audio"-Sound prism, as proposed in this paper, is a computer system that separates single-channel polyphonic music audio played by harmonic sources into source signals in an online fashion. It uses a musical score to guide the separation process[7]. To the best of our knowledge; this is the first online system that addresses score-informed music source separation that can be made into a real-time system. The proposed system consists of two parts:
1) A score follower that associates a score position to each time frame of the audio performance.
2) A source separator which reconstructs the source signals for each time frame, informed by the score.

Ponce, H. (Electrical Engineering, Computing Science and Automatic Control (CCE), 2012 9th International Conference on 26-28 Sept. 2012)[9] "A novel adaptive filtering for audio signals using artificial hydrocarbon networks"-The present paper introduces a novel adaptive filtering for audio signals corrupted with white noise. The audio filter is based on artificial hydrocarbon networks (AHNs), a novel approach inspired on organic chemistry for stability, well-forming and easy-spanning topologies. In that sense, an AHN-molecular structure adapts its parameters in a specific window to cancel signal noise from original audio sequence using the notion of filtering given by AHN-topologies. Results of the present approach show that artificial hydrocarbon networks can achieve audio filtering for noisy signals[9]. Then, a comparison between AHN-filtering and FIR filtering is presented. In addition, the AHN-filtering is implemented on DSP NI-Speedy 33 hardware.

III. PROBLEM DEFINITION

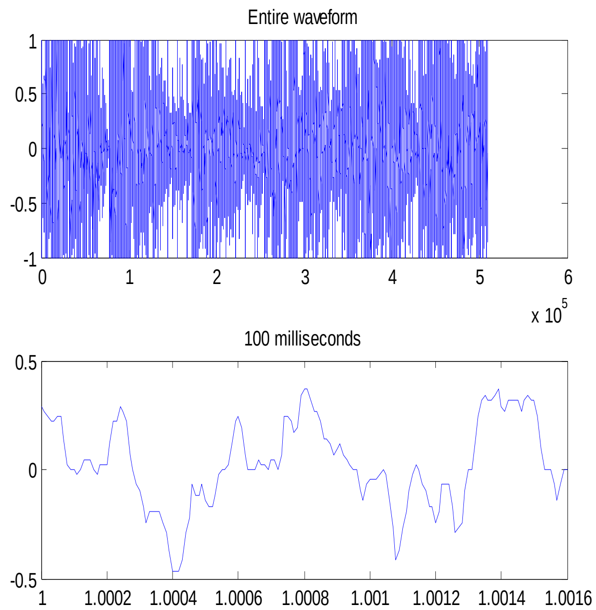
In original, FFR filter and high pass filter has the functionality to get the lower and upper limit of audio frequency input by the user manually. In this research, an effort will be done to make FFR filter and high pass filter adaptive in nature by automatically selecting the lower as well as upper limit of audio frequency.

IV. RESULT & ANALYSIS

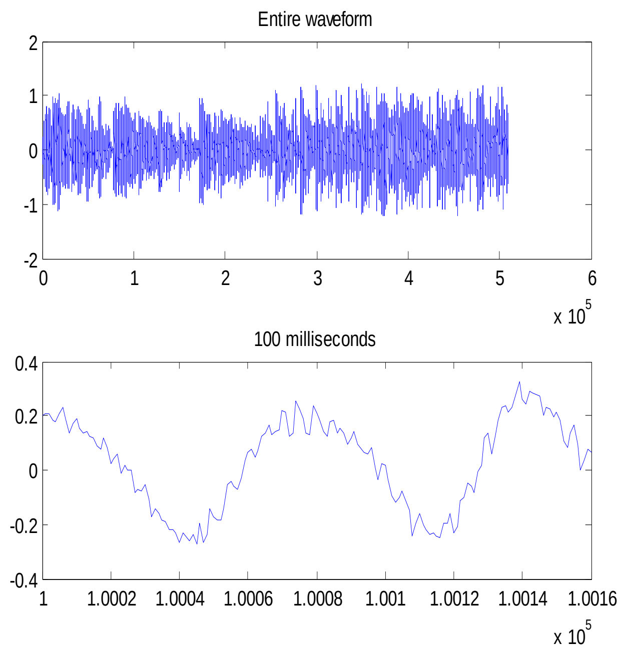
To improve the accuracy of the audio filtering algorithms, High pass filter and FIR filter is extended in this research work. The main objective is to improve the efficiency of available algorithms so that they provide good results in some critical applications. Various tests is performed using developed and existing algorithms to verify that proposed algorithms provide better results. Suitable simulation is performed in MATLAB.

In this study a .wav file with noise is used and randomly filtered using MATLAB filters so as to remove unwanted noise and to convert it into Noise-free signal and suitable graphs are plotted to know the distinction between the two.

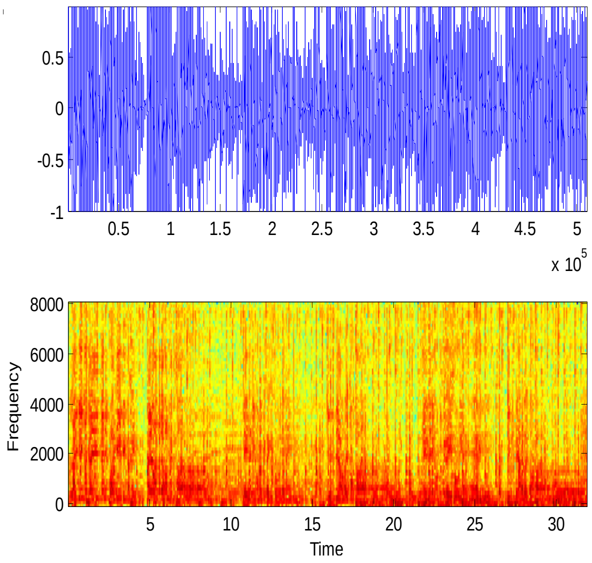
For Noisy Signal



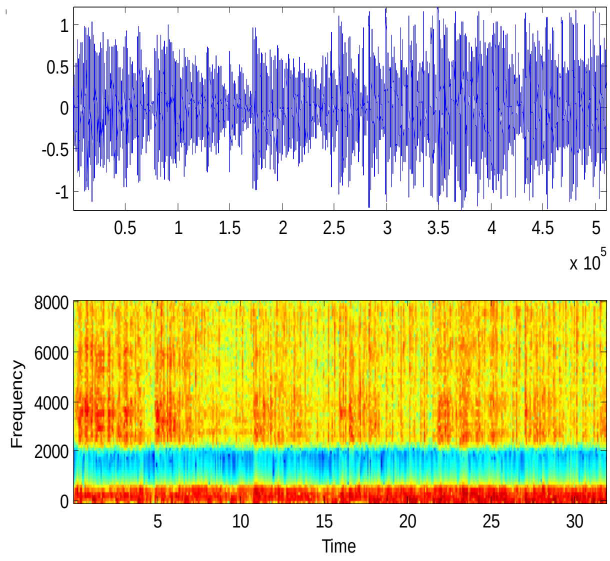
For Noiseless Signal



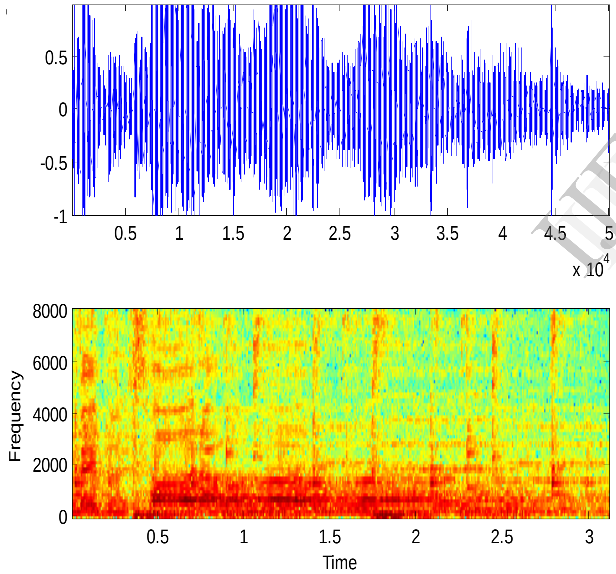
For Noisy Signal



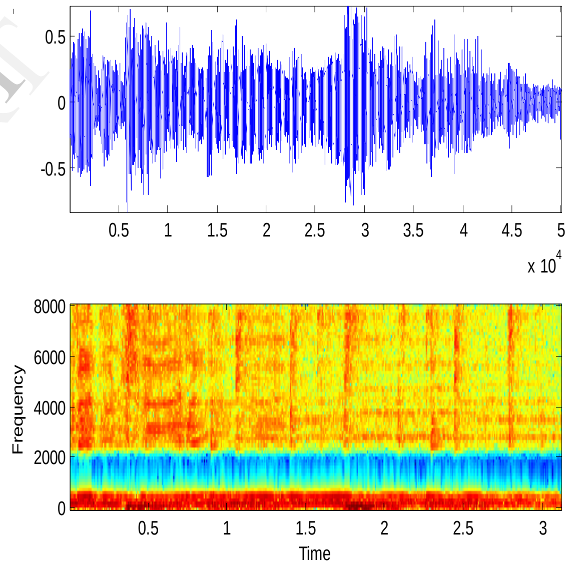
For Noiseless Signal



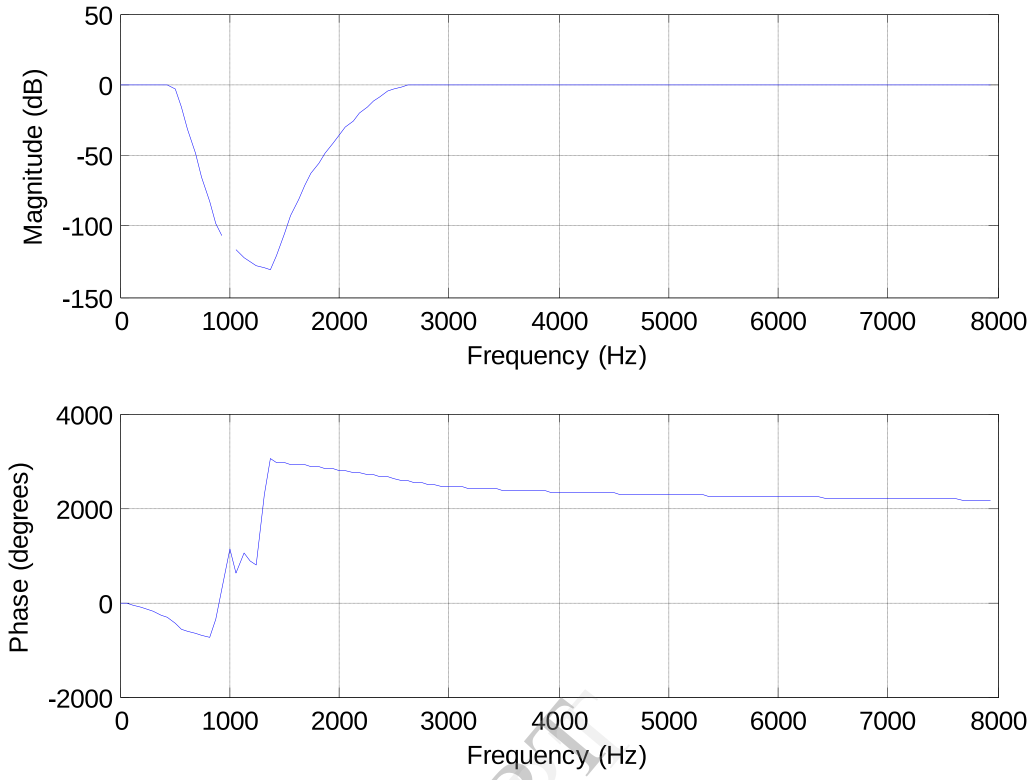
For Noisy Signal



For Noiseless Signal



Noisy v/s Noiseless



CONCLUSION

Audio Filtering is one of the most commonly used effective tools for sound recording and production. Nevertheless, its successful application is heavily dependent on the specialized skills of the operator. In this paper we have described basic filters for time-domain audio processing. These algorithms perform the filtering operations by the computation of difference equations. The coefficients for the difference equations are given for several filter functions such as low pass, high pass, band pass, shelving and peak filters. Simple design formulas for various equalizers lead to efficient implementations for time-varying filter applications.

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