A Review on Various Audio signals Filtering Procedures for Hearing aids

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Abstract: **Long distance communicated audio signal recordings may contain unwanted noise that can impair source recognition, speech recognition, and other audio signal method requirements. In this paper several custom analysis/synthesis algorithms are presented based on a time-varying spectral representation of noisy signal. Enhancement method adapts to instantaneous signal behaviour and alters noisy signal so that enhanced output signal is higher in quality than method noisy input signal. A compound of Ramez exchange method filter in order to frequency filtering and discrete wavelet Transform base noise filtering is likely to be design in near future in order to retain features that are attributable to desired signal, such as human speech, while detach features that are more likely to be due to noise in long distance contamination.**

Keywords: RX (Ramez Exchange), DWT (Discrete Wavelet Transform), SNR (Signal to Noise Ratio), MSE (Mean Square Error), BER (Bit Error Rate)

I-INTRODUCTION

When a recorded audio signal contains unwanted additive noise it is desirable somehow to enhance perceived signal-to-noise ratio before playback [1, 2, 3, 4]. Assuming noisy signal of interest is a digital data file, enhancement can be performed off-line (not in real time) with a digital copy of original without risk of damage to evidence itself. Off-line method also allows multiple passes through data, use of iterative algorithms, and opportunity in order to subjective evaluation of results.

A truly useful enhancement procedure needs to be *single-ended*, meaning that it must operate with no information available at receiver other than noise-degraded signal itself. Thus, it is necessary to devise an enhancement method that can adapt to immediate signal behaviour and alter noisy signal in both time domain and frequency domain so that a listener or forensic examiner will rate enhanced signal as both higher in quality and more useful than method noisy signal.

Fundamental issue in order to single-ended noise reduction is that problem is ill-posed. A desired signal, $s(t)$, is corrupted with unwanted additive noise, $n(t)$, resulting in observed signal $x(t) = s(t) + n(t)$. With only signal $x(t)$ available there is one equation with two unknowns, and therefore it is generally impossible to determine $s(t)$ from $x(t)$ unless some other information is available. Thus, in order to enhance desired signal $s(t)$ while attenuating unwanted $n(t)$, some means is needed to determine which features of composite signal $x(t)$ are attributable to noise and which are due to desired signal. Once undesirable features are somehow identified, some means is needed to reduce or detach those components from composite signal. Finally, some adaptive control scheme is needed to adjust detection and removal procedures to compensate in order to expected time varying behaviour of signal and noise.

An issue not considered in this paper is how to handle signal gaps, or "dropouts," which can occur if signal is lost momentarily due to mechanical or electrical interruptions in recording or playback systems. Dropouts can pose a serious problem in order to forensic audio signal enhancement. Signal enhancement method must detect signal dropout and take some suitable action, either muting playback momentarily or preferably synthesizing an evaluate of missing material [5].

II-LITERATURE SURVEY

Alexander Schasse et al [1] represent filter-bank system implemented in hearing aids has to fulfil various constraints such as low latency and high stop-band attenuation, usually at cost of low frequency resolution. In context of frequency-domain noise-reduction algorithms, insufficient frequency resolution may lead to annoying residual noise artifacts since spectral harmonics of speech cannot properly is resolved. Especially in case of female speech signals, noise between spectral harmonics causes a distinct roughness of method signals. Therefore, this work proposes a two-stage filter-bank system, such that frequency resolution can be improved in order to purpose of noise reduction, while original first-stage hearing-aid filter-bank system can still be used in order to compression and amplification. They propose procedures to implement second filter-bank stage with little additional algorithmic delay. Furthermore, computational complexity is an important design criterion. This finally leads to an application of second filter-bank stage to lower frequency bands only, resulting in ability to resolve harmonics of speech.

James T. G. [2] et al presents a low complex design of a non-uniformly spaced digital finite impulse response (FIR) filter bank in order to digital hearing aid application. Frequency response masking (FRM) technique is used in order to design of 8 non-uniformly spaced sub-band filters, with a single half-band filter as a prototype filter. With FRM technique and half-band filter, a drastic reduction in number of multipliers and adders in linear phase FIR filter can be achieved. Further complexity-effective design can be achieved by producing masking filter from prototype filter. FRM technique is achieved by cascading various combinations of prototype filter and its interpolated filters to develop sub bands. Simulation results shows that, proposed filter bank gives 120 dB attenuation with 13 multipliers only.

Problem Statement: A lot of effort has been devoted to design of uniform and non-uniform filter banks in order to hearing aid applications.

But in most of cases, it never deals with large change of hearing losses at mid frequencies. [1] is good in filtering at mid frequency but sharp variation of hearing loss occurs at low and high frequency range, [2] further need to improve noise reduction performance by adapting number of first-stage frequency channels to signal in terms of a voiced/unvoiced sound detection.

III-CONCLUSION

We can conclude that available methods for are good and significant enough but there is still problems like Frequency resolution at receiver end can be improvised with combination of different filter or multistage filter. A new combination of Remez exchange and DWT filter is planned to be develop in near future. And it is expecting to achieve high frequency resolution and less MSE for the audio signal received.

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