International Journal of Engineering and Techniques - Volume 4 Issue 2, Mar-Apr 2018

RESEARCH ARTICLE OPEN ACCESS

Efficient Noise Reduction in Hearing AIDS Using Integrated Framework Approach

Dr. N. Shanmugapriya

Assistant Professor,
Department of Computer Applications (PG), Dr. SNS Rajalakshmi College of Arts & Science,Coimbatore.

Abstract:

Hearing defect is becoming more common and may cause a number of reasons. In our everyday life, we do not feel the danger of our ear skills, with the highest seriousness. Generally, two factors can be said to be the primary causes of hearing loss, i.e. the sound and duration of the sound of the sound. This inner ear or damaging exposure to external noise cells (NIHL) may be damaging exposure to the noise exposure rather than the noise-induced hearing loss (NIHL) especially for internal and external hair cells.Nowadays many deaf people have been affected. The main complaint of those with hearing loss is the ability to reduce the conversation in a noisy environment. Asking help is a device that receives real time, process and feedback sound signal. In this case, various barriers are applicable, various filter banking techniques, signal processing and echo cancellations. The purpose of this work is to reduce the deafness used by people of all kinds. To implement this using an integrated configuration approach, it is very easy and requires less computation.

Keywords -– **Noise Reduction, Hearing Aids, Integration, Framework, Approach**

1. NOISE – INTRODUCTION

Noise as no tragedy tends to hide the required signal. Noise can be created within a circle or taken from external or artificial sources. When noise is created in circuits, it is called internal noise. If noise is taken from an external source, it is called external noise. Interruption is a noise to hide the effective signal. This is usually caused by electricity sources, but may be triggered by other physical sources, such as mechanical frequency, acoustic concepts or electrochemical sources.

There are many types and sources of noise or distortions and they include:

- Electronic noise like heat noise and shot noise.
- Sounds like moving, vibrating or rotating engines, moving vehicles, keyboard clicks, wind and rain.
- Electromagnetic noise that can interfere with voice exchange and reception.

Depending on its frequency, spectrum or time characteristics, a noise process is further classified into several categories:

- **White noise:** purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.

- **Narrow Band 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.
- **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.

Any unwanted signal that intercepts communications or measurements of the signal carrying information is called noise. In almost all environments noise is in various forms or percentages. For example, in a digital cellular mobile phone system, there may be a variety of distorting sound quality, such as flattening background noise, electromagnetic radio frequency noise, co-channel radio interference, radio channel distortion, hang down. And signal processing noise.

Noise can cause various transmission errors and can affect a communication process, so noise processing is an important and integral part of modern signal processing.

On the basis of the range of frequency present, noise can be broadly categorized into Broadband Noise and Narrowband Noise:

- **Broadband: -** The energy is distributed equally across all frequency bands. E.g. low-frequency sound of a jet plane.
- **Narrowband: -** Most energy is concentrated around specific frequencies. E.g. caused by rotating machinery.

Noise pollution is a very big problem prevailing in the environment these days. The workers working in heavy noise industry gets affected by the high level of noise present in the industries.

2. HEARING AIDS AND NOISE

Plays an important role in the human intellectual relationship. It is both comfort and understanding. The quality and understanding of speech is limited to the characteristic characteristics of the speech, but also by contact stats and information skills, ability to receive information from the environment, compliance and gestures. When discussing intelligibility it is important to understand the difference between a real and recorded speech.

A small electronic tool to ask for help, it loudly loud and easy to talk and understand. It is designed to produce sound waves with a small microphone, shift the loudest sound, and send it to the ear to the smallest speaker, helping the patients to ask for losses, and then improve their listening. With the help of Microsoft today, AIDS is getting smaller and smaller and improving quality.

Digital trial AIDS has many advantages: They can get a higher signal-to-noise ratio, change profits, counteract the electromagnetic interference, removal of ideas, and are widespread worldwide. But in the real environment, a variety of noises are encountered, the performance of voice system under noise environment would drop dramatically or even completely fail, therefore the noise reduction performance of a voice system is critical to evaluate the quality of a hearing aid.

Improving the speech comprehension under the noise environment has been the bottleneck of enhancing the performance of hearing aids. At present, improving methods mainly include two categories: directional microphone and noise reduction algorithm [2]. The former is designed based on the differences of speech and noise in the space, and utilizes directional microphones or beam forming technology to enhance the speech signal characteristic in the specific direction.

Digital hearing aids are committed to minimize the negative impact of the noise, basically have the function of smart noise reduction. However the noise reduction effect of various types of hearing aids differ in thousands ways, which requires the establishment of a complete measurement system to evaluate the noise reduction performance of hearing aids, eventually help hearing-impaired persons to choose suitable hearing aids.

2.1. Canal Aids

Canal aids are designed to fit entirely in the ear-canal and can be of two types: completely-in-the-canal (CIC) or in-the-canal (ITC). CIC hearing aids are seated completely in the ear canal and are therefore almost invisible. CIC are however difficult to adjust or remove. ITC hearing aids are slightly larger than CIC and they protrude a bit from the ear canal. They are therefore slightly easier to manipulate. Canal aids are limited in gain and signal processing power due to their small size. They are therefore more appropriate for mild to moderate hearing losses.

Figure 1: - Completely in the Canal (CIC)

Figure 2: - In the Canal (ITC)

2.2. In-the-air Hearing Aids

In-the-ear (ITE) hearing aids are designed to fit entirely in the outer-ear. ITE hearing aids are bigger than CIC and ITC hearing aids and therefore leave more room for a larger receiver and integrated circuits and batteries.

Figure 3: - In-the-Ear

2.3. Behind-the-ear hearing aids

In a behind-the-ear (BTE) hearing aid the electronics are in a small case hooked behind the ear. The receiver is in the BTE case and the sound is conducted to the ear canal through a plastic tube which terminates by an earmold. The earmold is custom-made to fit the ear canal of the user and prevent ambient sound to reach directly the eardrum. The earmold also prevents the sound from the receiver to reach the BTE microphones, hence reducing the feedback effects and allowing for high amplifications.

Figure 4: - Behind-the-Ear

2.4. Open fitting BTE Aids

Open fitting BTE hearing aids are a kind of BTE hearing aids which has been used more commonly during the past years. The main difference with a classic BTE hearing aid is that the earmold is removed (and sometimes replaced by a small piece of silicon). The absence of the earmold reduces the occlusion effect, the risk of infections in the ear canal and it improves the physical comfort [5] but it also leaves nothing to prevent the ambient sound to reach directly the eardrum. The amplification in open fitting hearing aids is also limited due to the higher risk of feedback caused by the absence of the earmold.

The knowledge on the noise signals is usually less extensive than the knowledge on the speech signal. Indeed, a large variety of noise signals can be present in common listening scenarios. The noise can be broadband or band-limited, intermittent or persistent, stationary or non-stationary. In the case of concurrent speakers, the noise signal can even have the same characteristics as the desired speech signal.

Finally, the noise can be diffuse or localized. Furthermore, when the noise is localized, the position of the noise sources is usually unknown, whereas the speech source is most commonly assumed to be facing the listener.

3. PROBLEM OBJECTIVE

Hearing is one of the five senses along with vision, taste, smell and touch. The ear serves as a receiver of incoming sounds. Hearing loss most commonly occur because of damages of the ear, rather than the central auditory system. The audio frequency range which is capable to hear is generally between- 20Hz to 20kHz.The human ear is only sensible to hear the frequency range between 1kHz to 4kHz. So below 1 kHz, ear will not respond and above the 4 kHz, it may damage the hearing capability.

Hearing loss is usually reserved for people who have relative insensitivity to sound in the speech frequency range. The main complaint of person with hearing loss is low ability to deduce speech in a noisy environment. In hearing aids the sound is processed by hearing aid and reaching to the ear. Normally it is made of three parts; Microphone, Processor units, receiver module. Recently available digital hearing aids are not compatible with environment.

4. PROPOSED METHOD FOR NOISE REDUCTION

The aim of these noise reduction procedures is to obtain significant noise reduction performance even for signals whose desired signal and noise components are located in the same frequency range.

However, a high audio quality noise-reduced signal cannot be easily obtained with this method. The main reason is the nonoptimal estimation of power spectral densities which are required in (1). Here, especially the estimation of the noise power spectral density poses problems since the noise signal alone is not available.

In order to nevertheless obtain reliable estimates, well known methods can be utilized. These are

- Estimating the noise power spectral density in pauses of the desired signal which requires an algorithm to detect these pauses,
- Estimating the noise power spectral density with the minimum statistics method or its modifications.

Both methods, however, exhibit a major disadvantage: they only provide long-term smoothed noise power estimates. However, for power spectral density estimation of the noisy signal, which can easily be obtained by smoothing the subband input signal power, short-term smoothing has to be applied in order that the Wiener-filter gains can follow shortterm fluctuations of the desired signal.

Calculating the Wiener-filter gain with differently smoothed power spectral density estimates causes the well known musical tones phenomenon [12].

To avoid this unpleasant noise, a large number of procedures have been investigated of which the most widely used are

- Overestimating the noise power spectral density estimates,
- Lower-limiting the Wiener-filter values to a minimum, the so-called *spectral floor*.

With the overestimation of the noise power spectral density, short-time fluctuations of the noise no more provoke a random “opening” of the Wiener-filter coefficients—the cause of musical tones.

However, this overestimation reduces the audio quality of the desired signal since especially low-power signal components are more strongly attenuated or vanish due to the overestimation. Limiting the noise reduction to the spectral floor reduces this problem but, unfortunately, also reduces the overall noise reduction performance. Nevertheless, this reduced noise reduction performance is generally preferred against strong audio quality distortion. More sophisticated methods utilize, that is, speech characteristics or masking properties of the ear, to limit the Wiener attenuation and thus reduce the signal distortion without compromising the noise reduction effect too much.

5. PARAMETER & MEASUREMENT

An important question we have to answer is that how to measure the speech quality after noise reduction. Normally, it is the best way to use human testers for this purpose (subjective evaluation), however, this method is costly and not every researcher can afford it.

5.1. Signal-to-Noise Ratio

A common way for measuring signal quality is to measure Signal to Noise Ratio (SNR). It is defined as the ratio of total energy of speech signal to total energy of noise signal,

The SNR using for speech quality evaluation is not calculated on the whole signal, but only on the parts with speech activity. Therefore, in order to calculate the SNR for speech signal, we must employ a Voice Activity Detector (VAD) to remove all the parts without speech activity.

However, the SNR has a weakness that it doesn’t take into account the distribution of noise and speech over time. The SNR in one section of signal can be very low, but the overall SNR can still be high, because the total noise energy is small compare to the total energy of speech.

5.2. Segment SNR

The segmental SNR is one of the most popular methods for measure speech quality. It is calculated by dividing the signal into many segments and calculating the SNR of each segment.

The size of segment is chosen according to the characteristics of speech signal (around 20 ms to 40 ms). This type of SNR is more suitable for speech quality assessment than the global SNR. In general, SNR measures have the benefit of low computational cost, but because it is calculated on time domain signals, some pre-processing must be done to align the tested signals.

6. CONCLUSION

Hearing loss can also be age-related which makes the research on hearing aids very important. For hearing aid users background noise and acoustic feedback imposes a major problem in terms of speech understanding and listening comfort. If these problems are not resolved some hearing aid users may even choose not to use their hearing aids.

The development of hearing aids covers a wide range of different signal processing components. They are mainly motivated by audio logical questions. This paper focuses on integrated framework method dealing with the compensation of the recruitment phenomenon, the improvement of speech intelligibility, and the enhancement of comfort while using the hearing aid in everyday life.

7. REFERENCES

[1] R. Bentler, C. Palmer, and H. G. Mueller. Evaluation of a second-order directional microphone hearing aid: I. speech perception outcomes. *Journal of the American Academy of Audiology*, 17(3):179–189, 2006.

[2] A. Beck and Y. C. Eldar. Doubly Constrained Robust Capon Beamformer With Ellipsoidal Uncertainty Sets. *IEEE Transactions on Signal Processing*, 55(2):753–758, February 2007.

[3] Y. Hu and P. Loizou, "Subjective evaluation and comparison of speech enhancement algorithms," *Speech Communication,* vol. 49, pp. 588-601, 2007.

[4] P. C. L. Yi Hu, "Evaluation of Objective Quality Measures for Speech Enhancement," *IEEE Transactions on Audio, Speech, and Language Processing,* vol. 16, no. 1, pp. 229-238, 2008.

[5] Li Zhang, Xiaomei Chen, Bo Zhong, Longbiao He, Huan Xu and Ping Yang "Objective Evaluation System for Noise Reduction Performance of Hearing Aids"2015 IEEE International Conference on Mechatronics and Automation August 2-5 , Beijing, China.

[6] Hui SHEN and Linghua ZHANG "A New Variable Step-Size Algorithm on Acoustic Feedback Suppression for Digital Hearing Aids" 978-1-4799-3903-9/14/$31.00© 2014 IEEE.

[7] Abdulmunim Matheel E., Abass Rabab F. "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 .

[8] Anju , Mamta Katiyar " Design of Butterworth and Chebyshev1 Lowpass Filter for Equalized Group Delay" International Journal of Advanced Research in Computer Science and Software Engineering, Volume 2, Issue 5, ISSN: 2277 128X, May 2012.

[9] Niedzwiecki M. and Ciolek M., "Elimination of impulsive disturbances from archive audio signals using bidirectional processing," Audio, Speech, and Language Processing, IEEE Transactions on, vol. 21, no. 5,pp. 1046–1059, 2013.

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

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

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

[12] M. A. Abd El-Fattah, "*Speech enhancement using an adaptive Wiener filtering approach*", Progress InElectromagnetic Research, Vol. 4,pp.167-184, 2008.

[13] Centre for Law-Enforcement Audio Research, "*Speech enhancement tutorial*", University College London, England, accessed 2010.

[14] T. Petković, "*Optimal Filtering*", internal material, course: Stochastic process systems, FER, University of Zagreb, 2006.