International Journal of Future Generation Communication and Networking
Vol. 13, No. 3s, (2020), pp. 1191–1196

A Study on Noise Reduction Algorithms for speech signal in Hearing Aids

Mr. Hrishikesh B. Vanjari [#1], Dr. Mahesh T. Kolte[#2]

*[#]E&TC Engineering Department, Pimpri Chinchwad College of Engineering, Nigdi, Pune 411044,
Maharashtra, India*

Abstract

Today there is many people having hearing problems. The main complaint of people with hearing loss is the low ability to deduce speech in a noisy environment. The aim is to study various noise reduction techniques and compare them and arrive at a conclusion regarding which one of them is most suitable for enhancing speech signals. The hearing aid is a device that can acquire, process and feedback the acoustic signal in real time. In this case, several impedance matching algorithms, various filter bank techniques, signal processing algorithm and echo cancellation are discussed. The FFT algorithm is used to make this application, which is quite easy to implement and requires less computation. The different Noise reduction methods are – Spectral subtraction, Adaptive Filter based on LMS Algorithm, Weiner Filter, Chebyshev Filter and Kalman Filter. The purpose of noise reduction algorithms is to provide a quality and clarity voice signal in background noise, interference sources and reverberation effects. Most single channel algorithms operate in the time frequency domain, while multichannel algorithms rely on spatial filtering.

Keywords*- Hearing aid, Noise Reduction, SNR, Fast Fourier Transform, adaptive noise cancellation, digital signal processing, directional microphones, hearing aids, noise reduction, speech-in-noise*

INTRODUCTION

The intelligibility of human speech plays a vital role in day-to-day communication. The fundamental senses are hearing, sight, smell, taste and touch. These senses perceive information from the environment & hence human brain processes this information to make an accurate response. Complex communication between humans and their environment is favored by these senses. Sound acts as an information provider for these senses. The information must be noise free in order that we better understand the external environment. Noise is described as any unwanted information that hinders the power of the human to process valuable sensory information. Therefore, a sound without noise becomes essential for the right interaction of humans with their external world. The most focus is on voice signals that are providers of data in various communication systems. During signal transfer, the distortion caused by some unwanted signals causes the loss of useful data and information stored within the signals. [17],[18] The quality and intelligibility of the discourse are determined not only by the physical characteristics of the discourse itself, but also by the conditions of communication and the capacity for information, the ability to obtain context information, imitations and gestures. When intelligibility is discussed it is important to understand the difference between a real and recorded discourse. During a real conversation, a person can recognize the surrounding sounds and concentrate on the speech of another person, thus filtering the desired information from various audio environments. Therefore, the ability of a human to recognize and filter sounds significantly increases intelligibility and understanding of speech, even if communication takes place in a noisy environment, situation or condition. The hearing aid is a small electronic instrument that makes the sounds louder and makes speech easier. [2]

Noise can be defined as any form of unwanted energy that tends to interfere with the proper reception and reproduction of the desired signal. The voice signal is mainly divided into two parts, such as vowels and consonants. The low frequency characteristics are taken by the vowel sound, but the energy will be much higher while the consonant sound is high frequency and the energy share is of very low magnitude. Mainly the defect of the human ear results in a lower identification of high frequency in contrast to low frequency sound. In the presence of hearing aids, it is also difficult to hear speech. Speech intelligibility is reduced in the presence of noise. There are different types of noise depending on the noise source and frequency. Depending on the source of noise is: acoustic noise, thermal noise and trigger noise, electromagnetic noise, electrostatic noise, channel distortions, echo and fade, processing noise. Depending on the different frequency is: white noise, limited band

ISSN: 2233-7857 IJFGCN

noise, narrow band noise, colour noise, impulsive noise, and transient noise. There are many methods available for noise reduction in speech signal. This paper is to provide a brief overview and of few signal processing techniques.

PROBLEM IN HEARING AID

Hearing is an important sense among the five senses: vision, taste, smell and touch. The human ear serves as a receptor. Hearing loss occurs most commonly due to damage to the ear, rather than the central auditory system. The audio frequency range that you are able to listen to is generally between 20Hz to 20 kHz. The human ear is only sensitive to hear the frequency range between 1 kHz to 4 kHz. Then, below 1 kHz, the ear does not respond and above 4 kHz, it can damage the hearing. Hearing loss is generally reserved for people who have a relative insensitivity to sound in the frequency range of speech. The main complaint of a person with hearing loss is the low ability to deduce speech in a noisy environment. In hearing aids, the sound is processed by the hearing aids and reaching the ear. It is usually made of three parts; Microphone, processor units, receiver module. Recently available digital headphones are not compatible with the environment.

NOISE REDUCTION

One of the most challenging hearing situations that a person with hearing loss encounters is to follow the speech in the presence of background noise. Noise reduction in a hearing aid attempts to recognize and separate audio from noisy environment, before reducing the noise level of noise so that it can be heard with less listening effort. Hearing aids use various algorithms to discriminate between speech and noise. Most focus on amplitude fluctuations (sound level) since speech tends to fluctuate more than noise. Several also focus on fluctuations in speech frequency as it includes a wider range of frequencies. Some algorithms also use probability scales to calculate how much noise and voice there is in a particular hearing aid channel, before providing fewer amplifications in the channels where there is likely to be more noise present than the voice.

Different manufacturers use not only different systems but also their own decision and management rules. While it depends on how much noise there is, other factors are also generally considered, such as general speech levels and noise. It is common to vary the amount of reduced gain depending on the frequency of a band. The most important frequencies for speech comprehension are not reduced as much as the less important frequency bands for speech comprehension. If we were to use this function in a broadband manner, it would cause difficulties. While this gain reduction reduces the amplification of competitive noises, it would also decrease the energy in speech signals. In other words, noise reduction would affect both speech and noise in exactly the same way. This is where the division of the amplification into channels or bands (according to the manufacturer). Modern headphones are multichannel devices with between 3 and 32 channels. The noise reduction function can work separately within each separate channel. Therefore, it is possible to reduce amplification only on the loudest channels, in the hope of avoiding channels that carry important speech information. Most of the signal processing system that interacts with the outside world contains some of the signal processing techniques or algorithms. The digital hearing aid system uses sophisticated algorithms and digital filters. They also contain several algorithms for the transformation of the time domain signal into the frequency domain. There is large number of human spoken languages in world; each languages has its own sound characteristics such as pitch, separation between vocal folds, tones etc. therefore the speech signals contains samples of different languages in both male and female voice. Noise reduction techniques have tremendous real time applications, but the implementation of these Noise reduction methods is challenging task.

The basic purpose of any digital hearing aid is to amplify the sound of a certain frequency band of interest that is below the patient's hearing threshold and compress the strong signal. There are some well-known techniques to achieve noise reduction. There are different filters used for de-noising speech signals. The filters that have been used are - Spectral subtraction, Adaptive Filter using LMS (least mean square) algorithm, Weiner Filter, Chebyshev Filter and Kalman Filter.

A. Spectral Subtraction

Spectral subtraction is one of the first and easy approaches to improve speech in a single channel. In this technique, the estimated noise spectrum is subtracted from the voice spectrum (clean voice and

ISSN: 2233-7857 IJFGCN

noise) and then the clean voice spectrum is estimated. Spectral subtraction affected by background noise, fan noise, musical noise and speech distortion. The spectral subtraction technique deals with the transformation domain approach, in which the signal becomes a frequency domain [7], [12]. The advantage of this transformation domain is that it is easy to differentiate between voice and noise components and, therefore, noise components can be easily removed.

B. Wiener Filter

Among the many techniques developed for noise cancellation, one of the most fundamental is the Weiner filter. This filter reduces noise by making a comparison between the received signal and the estimate of the speech signal we want. The Weiner filter is the most fundamental technique used for canceling noise in voice signals. This method causes speech degradation that reduces the values of SNR and PSNR. Therefore, this method cannot be applicable for real-time applications. However, there are some measures that determine the performance of the filter by relating the amount of speech distortion with noise reduction factors. [12] The error is also relatively high compared to other signals. Therefore, it is not recommended to use this filter for noise cancellation in voice signals.

In signal processing, the Wiener filter is another approach to reduce noise in voice signals. This filter method is based on the reduction of the mean square error (MSE) between the original signal and the estimated signal [12]. Using this optimal linear technique, SNR a is always greater than or equal to SNR a priori, which means that this technique is useful for cleaning noisy voice signals [6]. The improved signal is obtained with less noise. The Wiener filter achieves a satisfactory noise reduction for some applications; it also introduces distortions in the voice signal that may be perceptibly unacceptable when the SNR is very low. However, this filter assumes that the input is stationary. For its implementation, it is eminent that the user knows the spectral properties (power functions, etc.) of both the noise and the original audio. [19] The process works by minimizing the mean square error between the new signal and the original signal. It is widely known that, although the Weiner filter reduces noise, it has the cost of speech distortion. Therefore, it is only recommended to use this filter when noise cancellation-voice distortion is acceptable.

In this method, the Weiner filter is using the TSNR (Two Step Noise Reduction) method. The signal observed is the combination of clean audio and noise. This speech is divided into small squares. These frames overlap with successive frames. Each of them is smoothed with a Hamming window. Then, using the Fast Fourier Transform, the signal is converted from the time domain to the frequency domain. The phase and magnitude are extracted from the FFT thus obtained. The required modification to its value is calculated using the TSNR method. Once a new phase and magnitude are obtained, its IFFT (Fast Fourier Inverse Fast Transform) is calculated and the frames are superimposed and added to produce the voice signal without noise. [20]

C. Least mean squares

The adaptive filter is like any other linear filter, except that its transfer function is variable. It can be changed by adjusting some parameters. An error signal is generated when comparing the input signal and the training signal and this is used to adjust the parameters. Therefore, adaptive filters are regulated according to the incoming signal. Adaptive filters have an algorithm to optimize parameter values. The LMS algorithm (mean square minimum) employs the technique of stochastic gradient descent. This means that the filter coefficients are only affected by the current error value. As the name implies, the filter coefficients are directly related to the mean minimum square of the difference between the signal we want and the original signal. The primary principle behind this approach is to continue updating the filter weights until they reach their optimum value. Initially, very small values are assigned to the weights and are updated according to the MSE gradient (mean square error). The filter is adjusted to minimize the output power and the output noise power is also minimized. Therefore, by minimizing the total output power, the filter maximizes the signal to noise output ratio.

When selecting the size of the step in the LMS algorithm, care should be taken because if the size of the step is large, the dependence of the weights on the gradient increases, so any minor change in the gradient would lead to the oscillation of the weights . However, if the step size is too small, the weights will take a long time to reach the ideal weight values. The adaptive filter used by the LMS algorithm cancels the noise of a loud voice signal. The main advantage of this method includes its ability to adapt to noise, its tendency to provide output signals with low noise and to be able to do so

ISSN: 2233-7857 IJFGCN

without distorting the signal. Being an adaptive filter, it can eliminate unexpected noise even when it has non-stationary properties. The method has the advantage of not requiring a priori knowledge of the properties of the noise signal, and obtains an improvement in speech quality by reducing granular quantification noise. In the relationship between noise reduction and speech distortion with the single-channel Wiener filter, it is formally studied. NLMS Algorithm (Normalized Average Minimum Square) and the RLS Algorithm (Minimum Recursive Minimum Square) to cancel the noise signal in the hearing aids. Background noise is negatively affecting the speech intelligibility of people with hearing loss. [3]. This method is used to cancel the internal noise or error signal in the digital headphones caused by the acoustic coupling between the microphone and the speaker. The main idea of this method is to replace the receiver's input signal with a synthesized signal, which sounds perceptually similar to the original signal. These methods only reduce the internal noise of the hearing aids.

D. Chebyshev *Filter*

Chebyshev filters are very attractive, since they are capable of allowing the passage of a particular frequency band while blocking all others. Another advantage of these filters is that the time required to process the signal is much shorter. This is because the transition of the Chebyshev filter between its pass band and the stop band is extremely fast. [21] [22] A Chebyshev type 2 low pass digital filter has been designed to eliminate noise from the voice signal. In this, it is difficult to eliminate the noise of the voice signal without affecting its quality. Therefore, the filter output signal has a large mean square error. Due to a medium high square error value, the SNR and PSNR values are low, but the calculation time is very small compared to other filters..

E. Kalman Filter

The Kalman filter is a mathematical method to estimate the state of a phenomenon by minimizing the mean square error. One can predict past, current and subsequent values without knowing of the system model. The main idea of the Kalman filter is to predict future original values from current and past values in a repetitive manner. This method is generally used when there is uncertainty in the information. When the required data cannot be recovered directly, we use the Kalman filter. In some situations, there is no availability of the required data, therefore, the available data that is close to the actual data is used as input in the Kalman filter. Estimate how a signal will behave in the future. It is the ideal filter for non-stationary signals. [23] Here, the Kalman filter has been used to filter the random noise of a voice signal. The process used in this algorithm, which is recurring, occurs in two steps. The first step is "the prediction" in which the filter generates estimates of the current state variables, including the uncertainties with which they are associated. The second step is called "estimation." Depending on the result of the next measurement, the estimates are updated with the weighted average. Higher weights are assigned to estimates with greater certainty. [24] The Kalman filter provides satisfactory results by filtering the noise of the original speech. The average quadratic error algorithm is used to find the parallelism between the original voice signal and the Kalman filter output. The main advantage of this filter is that it is fast and has fast processing.

Conclusions

In this paper speech noise reduction techniques are reviewed. These noise reduction techniques are spectral subtraction and Wiener filter. These filter algorithms help improve SNR. In spectral subtraction, a noisy spectrum is subtracted from the noisy speech spectrum to obtain the estimate of the clean speech spectrum. In addition, the noise spectrum estimate is updated in cases where there is no useful voice signal and there is only noise. In the analysis of the spectral subtraction algorithm, the SNR is improved with the linear increase of the input SNR, but not as much as the user requires. It means that noise is eliminated in fewer quantities. Therefore, this noise reduction technique is not useful. Another noise reduction technique is Wiener filtering, where an SNR is a multiplicative gain function. According to the Wiener filter analysis, the SNR is improved as the power level of the input SNR increases. This increase in SNR leads to noise reduction. Therefore, the Wiener filter offers the best performance in terms of SNR compared to the spectral subtraction algorithm.

Acknowledgment

ISSN: 2233-7857 IJFGCN

REFERENCES

[1] 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.

[2] Er. Mannu Singla and Er. Harpal singh "Review Paper on Frequency Based Audio Noise Reduction Using Different Filters" International Journal of Science Engineering and Technology Research, Volume 4, Issue 5, May 2015

[3] P. Rajesh, K. Umamaheshwari, P. Vinodh Babu and S. Srinivasa Rao "Application of Adaptive Filter in Digital Hearing Aids for Cancellation of Noise" 978-1-4 799- 8081-9/15/$3l.00 © 2015 IEEE ICCSP Conference.

[4] M. Poornima and E. Rajinikanth "Multi Stage Filter Bank System for Improved Noise Reduction in Hearing Aids" 2016 International Conference on Circuit, Power and Computing Technologies [ICCPCT].

[5] Arun Sebastian and James T. G. "Digital Filter Bank for Hearing Aid Application Using FRM Technique" 978-1- 4799-1823-2/15/$31.00 ©2015 IEEE

[6] 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.

[7] Er. Mannu Singla and Er. Harpal singh "Review Paper on Frequency Based Audio Noise Reduction Using Different Filters" International Journal of Science Engineering and Technology Research, Volume 4, Issue 5, May 2015

[8] P. Rajesh, K. Umamaheshwari, P. Vinodh Babu and S. Srinivasa Rao "Application of Adaptive Filter in Digital Hearing Aids for Cancellation of Noise" 978-1-4 799-8081-9/15/$3l.00 © 2015 IEEE ICCSP Conference.

[9] M. Poornima and E. Rajinikanth "Multi Stage Filter Bank System for Improved Noise Reduction in Hearing Aids" 2016 International Conference on Circuit, Power and Computing Technologies [ICCPCT].

[10] Arun Sebastian and James T. G. "Digital Filter Bank for Hearing Aid Application Using FRM Technique" 978-1-4799-1823-2/15/$31.00 ©2015 IEEE

[11] 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.

[12] DAN CHAZAN, Y. MEDAN and Hearing Aids "Noise Cancellation for Hearing Aids " IEEE TRANSACTION ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, VOL. 36. NO. 11, NOVEMBER 1988.

[13] H. Puder, "Hearing Aids: An Overview of the State-of-the-Art, Challenges, and Future. Trends of an Interesting Audio Signal Processing Application," In Proceedings of the 6th International Symposium on Image and Signal Processing and Analysis, 2009.

[14] V. Hamacher, J. Chalupper, J. Eggers, E. Fischer, U. Kornagel, H. Puder, and U. Rass, "Signal Processing in High-End Hearing Aids: State of the Art, Challenges, and Future Trends," EURASIP Journal on Applied Signal Processing 2005:18, pp. 2915–2929.

[15] W. Zheng, M. Liu, "Hearing Environment Recognition in Hearing Aids, In 12th International Conference on Fuzzy Systems and Knowledge Discovery (FSKD)", pp. 1556-1560, 2015.

[16] V. Littmann, J. Beilin, M. Froehlich, E. Branda, P.J. Schäfer, "Clinical Studies Show Advanced Hearing Aid Technology Reduces Listening Effort", The Hearing Review, April 2016

[17] Rodriguez, J. E. F. F. R. E. Y. J., Lim, J., & Singer, E. (1987, April). Adaptive noise reduction in aircraft communication systems. In ICASSP'87. IEEE International Conference on Acoustics, Speech, and Signal Processing (Vol. 12, pp. 169-172). IEEE.

[18] Kumari, M., Talukdar, N., & Ali, I. (2016, November). A new gender detection algorithm considering the non-stationarity of speech signal. In 2016 2nd International Conference on Communication Control and Intelligent Systems (CCIS) (pp. 141-146). IEEE.

[19] Jingfang, W. (2011, August). Noisy speech in real time iterative Wiener filter. In 2011 International Conference on Mechatronic Science, Electric Engineering and Computer (MEC) (pp. 2102-2105). IEEE.

ISSN: 2233-7857 IJFGCN

[20] Plapous, C., Marro, C., & Scalart, P. (2006). Improved signal-to-noise ratio estimation for speech enhancement. IEEE Transactions on Audio, Speech, and Language Processing, 14(6), 2098-2108.
[21] Kim, K., Chung, Y., Park, C., Son, Y., & Yoon, J. (2007, December). Speech quality enhancement based on sinusoidal model using chebyshev filter. In Future generation communication and networking (fgcn 2007) (Vol. 1, pp. 323-327). IEEE.
[22] Shahid, M. B., Abbasi, M. A., & Muazzam, M. (2015, December). Detection of noise in high pass IIR digital filters. In 2015 International Conference on Humanoid, Nanotechnology, Information Technology, Communication and Control, Environment and Management (HNICEM) (pp. 1-5). IEEE.

ISSN: 2233-7857 IJFGCN