International Journal on Applications in Information and Communication Engineering
Volume 1: Issue 5: May 2015, pp 26-28. www.aetsjournal.com ISSN (Online) : 2394-6237

Vuvuzela based Noise Reduction Method for Hearing Aid Applications

Sandhya P, Vanitha Lakshmi M

Abstract**—** Speech has always been one of the most important carriers of information for people and it becomes a challenge to maintain its high quality. In many application of noise cancellation, the changes in signal characteristics could be quite fast. Speech signal is often degraded by additive background noises like babble noise, train noise, restaurant noise, car noise, white noise etc. In such noisy environment listening task is very difficult at the end user. This paper describes a new approach for noise cancellation in speech signal using the algorithm named Vuvuzela Denoising algorithm for attenuating noise in speech signals. The simulation results demonstrate the good performance of the algorithm in attenuating the noisy signals.

Index Terms-Hearing Aid, Noise Reduction, and Vuvuzela Method.

I. INTRODUCTION

Communication via speech is one of the essential functions of human beings. Humans possess varied ways to retrieve information from the outside world or to communicate with each other, and the three most important sources of information are speech, images and written text. For many purposes, speech stands out as the most efficient and convenient one. Speech not only conveys linguistic contents, but also communicates other useful information like the mood of the speaker. Language communication through speech is closely intertwined with the evolution of human civilization. Noise presented in speech signal Recorded under the real conditions can impair the quality of the signal, reduce intelligibility, and Increase listener fatigue. Since in practice many of of noise is presented in recording speech the problem noise reduction is essential in the the world of Telecommunications and has gained much attention in recent years.

Noise reduction algorithms in general, attempt to improve the performance of communication systems when their input or output signals are corrupted by noise. The main objective of speech is to improve the speech quality or intelligibility.

A hearing aid is a battery-powered, electronic device that makes listening easier for people with a hearing loss. A hearing aid consists of a microphone, an amplifier and a receiver. The microphone picks up sounds in your acoustic environment and turns them into electronic signals. The amplifier selectively amplifies the acoustic electronic signals. The receiver is a very small speaker at changes the electric signals back to sounds and delivers the sound to the ear.

The Hearing Aids are used by the people who have:

- Hearing loss with the birth
- Hearing loss with the old age
- Hearing loss with the accident
- Not all hearing loss need Hearing aid, need doctor's correct examination for determination.

The considerations of the hearing aid are given as follows: they are; degree of disability, appearance, budget, lifestyle, hearing condition etc.

II. NOISE REDUCTION SCHEMES

Noise reduction has always been a non-trivial problem for engineers. The total removal of background noise is practically impossible and the distortion of the speech content is inevitable. There are several methods used to reduce the noise. Traditional noise reduction algorithms include spectral subtraction, Wiener filtering, time varying speech model-based or state-based methods, and microphone array based techniques. The considerations in using hearing aid are:

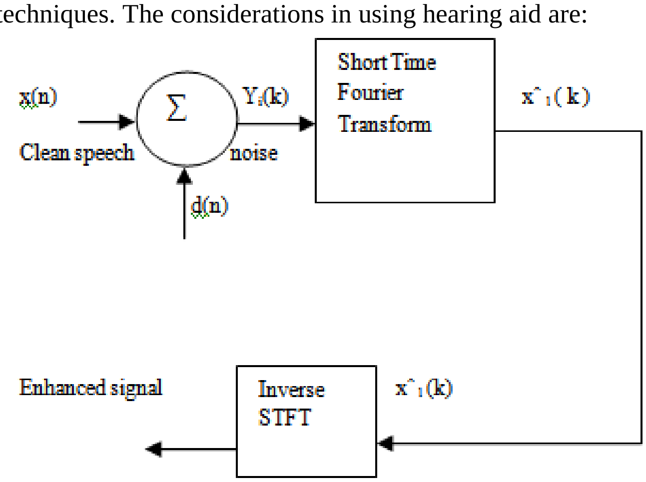

Figure 1: Block Diagram of the System

In this paper, we are going to discuss about the noise reduction in hearing aid using Vuvuzela Denoising algorithm.

Let x(n) and y(n) denote speech and uncorrelated additive noise signal, respectively, where *n* is a discrete-time index. The observed signal y(n)=x(n)+d(n), given by y(n)=x(n)+d(n), is transformed into the time-frequency domain by applying the short-time Fourier transform (STFT). Specifically

$$Y_l(k) = \sum_{n=0}^{N-1} y(n+lM)h(n)e^{\frac{-j2\pi}{N}nk} \tag{1}$$

Sandhya P, PG Scholar, Department of ECE, S.A Engineering College, Poonamallee-Avadi Main Road, Veeraraghavapuram, Thiruverkadu post, Chennai 600 077,India (Email : sandhyaparanthaman@gmail.com)

Vanitha Lakshmi M, Assistant Professor , Department of ECE, S.A Engineering College, Poonamallee-Avadi Main Road, Veeraraghavapuram, Thiruverkadu post, Chennai 600 077,India (Email : vanithahitesh08@yahoo.co.in)

where K is the frequency-bin index (K=0,1,.,N-1),L is the time frame index (*l=0,1,…)*is an analysis window of size N (e.g., Hamming window), and M is the framing step (number of samples separating two successive frames). Given an estimate for the STFT of the clean speech, an estimate for the clean speech signal is obtained by applying the inverse STFT

$$\hat{x}(k)=\sum_{\ell}\sum_{k=0}^{N-1}\hat{X}_{\ell}(k)\tilde{h}(n-\ell M)e^{\frac{j2\pi}{N}nk(n-\ell M)} \quad (2)$$

Where h (n) is a synthesis window that is bi-orthogonal to the analysis window(n) [19], and the inverse STFT is efficiently implemented by using the weighted overlap-add [10].Let it denotes a set of spectral measurements, and let be a given distortion measure between and. Our objective is to find an estimator, which minimizes the conditional expected value of the distortion measure, given the set of spectral noisy measurement

$$\hat{X}_{\ell}(k)=\arg\min_{\hat{X}} \mathrm{E}\left\{d\left[X_{\ell}(k).\hat{X}\right]/y_0^{\ell}(k)\right\} \quad (3)$$

we consider a causal estimation of (in which case as well as a non causal estimation while the spectral components are not assumed statistically independent. Then an estimate of the clean speech

Short-time spectral magnitude can be obtained as the gain function takes on a value Depending on the *a posteriori* SNR defined by.

$$SNR_{post}(n,k)=\gamma_{n,k}=\frac{\gamma_{n,k}}{\sigma_d^2(n,k)} \quad (4)$$

Where

$$\sigma_d^2(n,k)=\mathrm{E}\left\{D_{n,k}\right|^2\right\} \quad (5)$$

On the other hand, *a priori* SNR is given by,

$$SNR_{prior}(n,k)=\xi_{n,k}=\frac{\mathrm{E}\left\{X_{n,k}\right|^2\right\}}{\sigma_d^2(n,k)} \quad (6)$$

III. VUVUZELA METHOD

The vuvuzelas have the potential to cause hearing loss. vuvuzelas can have a negative effect when a listener's eardrums are exposed to the instrument's high-intensity sound.

The Vuvuzela is like a straightened trumpet and is played by blowing a raspberry into the mouthpiece. The player's lips open and close about 235 times second, sending puffs of air down the tube, which excite resonance of the air in the conical bore. A single Vuvuzela played by a decent trumpeter is reminiscent of a hunting horn – but the sound is less pleasing when played by the average football fan, as the note is imperfect and fluctuates in frequency. It sounds more like an trumpeting. This happens because the player does not keep the airflow and motion of the lips consistent.

A flared instrument has louder higher-frequency harmonics than a cylindrical one. The flared instrument is perceived as louder because the higher harmonics are at frequencies where our hearing is most sensitive. This is partly why the conical saxophone sounds louder than the cylindrical clarinet. Since it produces 116 decibels at 1 metre. A whole crowd produces even higher levels, and measurements at a training match have shown temporary hearing loss among spectators. Experiments on other noise sources show that louder sounds are more annoying. Our hearing is an early-warning system: we listen out for sudden changes in the sounds around us which might indicate threats, and ignore benign, persistent noise.

IV. VUVUZELA DENOISING ALGORITHM

The Vuvuzela Denoising algorithm is given by the following flowchart. The vuvuzela Denoising algorithm is done by loading the .wav file and then estimating the noise is carried out by taking the short time fourier transform, the windowing function used here is hamming window, and then noise is estimated and inverse short time fourier transform is performed, and then the signal is retained by using overlap and add method

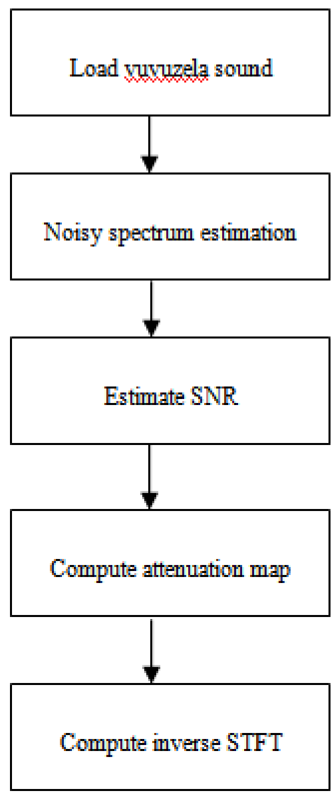

Figure 2: Flowchart of the Vuvuzela Denoising algorithm.

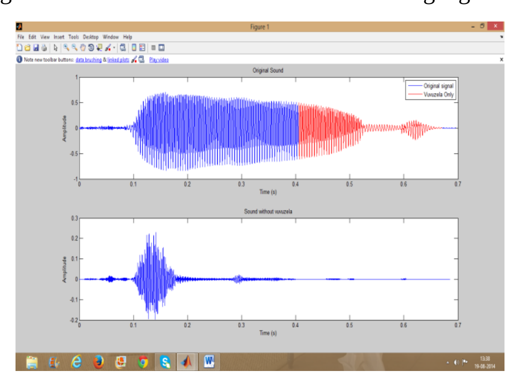

Figure 3: Original sound and sound without vuvuzela

The output of this algorithm is given by the following figures which are as follows: The figure 3 describes the original input signal which is given when the Vuvuzela sound is introduced. This figure consists of speech and also the noise signal which is differentiated by the different colours and also the other graph describes the sound when it comes without the Vuvuzela sound that is our corresponding noise.

The figure 4 describes the spectrogram which corresponds to both speech and the noise. Here the signal is accompanied along with the noise. This noise is reduced by using the Vuvuzela denoising method and by using this we can able to reduce the noise.

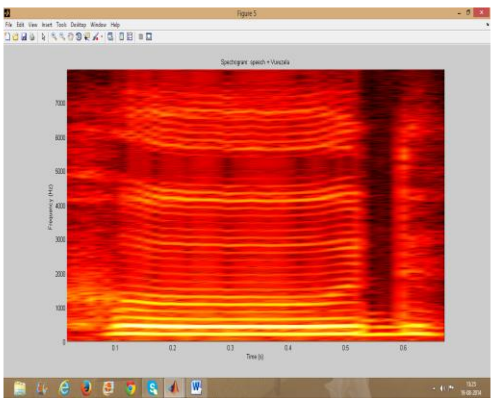

Figure 4: Spectrogram which has speech signal

The figure 5 shows the signal without noise and this is achieved by following the steps of the algorithm which is given in the figure 2. Here the noise is fully degraded and the noiseless signal is achieved.

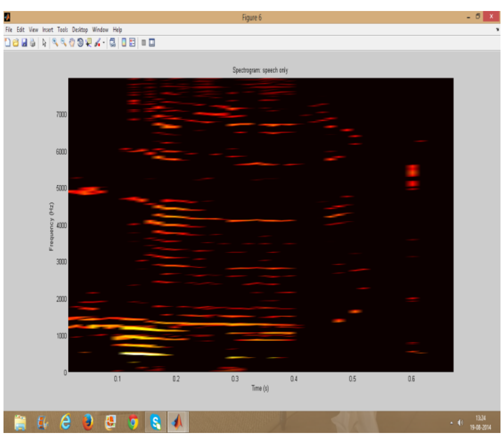

Figure 5: spectrogram speech only

V. RESULT

In this paper Vuvuzela Denoising method based is introduced. For effective noise reduction, this algorithm takes in account to perceptual aspects of human ear. It can be seen from the experimental results that it effectively reduces background noise in comparison with commonly used other types of algorithms. This method results in greater improvement of noise reduction and considerably improvement of perceptual speech quality in compassion to conventional method.

VI. FUTURE ENHANCEMENT

This paper is further enhanced by introducing some more algorithms and comparing it. And those results are compared and the best method is undergone for the procedure.

REFERENCES

[1] J. S. Lim and A. V. Oppenheim, "Enhancement and bandwidth compressionof noisy speech," *Proc. IEEE*, vol. 67, pp. 1586–1604, Dec. 1979.

[2] S. F. Boll, "Suppression of acoustic noise in speech using spectral subtraction,"*IEEE Trans. Acoust., Speech, Signal Process.*, vol. 27, pp.113–120, Feb. 1979.

[3] M. Berouti, R. Scwartz, and J. Makhoul, "Enhancement of speech corruptedby acoustic noise," in *Proc. ICASSP*,Washington, DC, 1979, pp.208–211.

[4] P. Scalart and J. Vieira-Filho, "Speech enhancement based on a priorisignal to noise estimation," in *Proc. ICASSP*, 1996, pp. 629–632.

[5] B. L. Sim, Y. C. Tong, J. S. Chang, and C. T. Tan, "A parametric formulationof the generalized spectral subtraction method," *IEEE Trans.Speech Audio Processing*, vol. 6, pp. 328–337, July 1998.

[6] H. Gustafsson, S. E. Nordholm, and I. Claesson, "Spectral subtractionusing reduced delay convolution and adpativeaveraging," *IEEE Trans.Speech Audio Processing*, vol. 9, pp. 799–807, Nov. 2001.

[7] Y. Ephraim and D. Malah, "Speech enhancement using a minimummean square error short-time spectral amplitude estimator," *IEEE Trans.Acoust., Speech, Signal Processing*, vol. ASSP-32, pp. 1109–1121,Dec. 1984.

[8] O. Cappe, "Estimation of the musical noise phenomena with theEphraim and Malah noise suppressor," *IEEE Trans. Speech AudioProcessing*, vol. 2, pp. 345–349, Apr. 1994.

[9] C. Beaugeant and P. Scalart, "Noise reduction using perceptual spectralchange," in *Proc. EUROSPEECH Conf.*, vol. 6, 1999, pp. 2543–2546.

[10] I. Y. Soon and S. N. Koh, "Low distortion speech enhancement," *Proc.Inst. Elect. Eng., Vis. Image Signal Process.*, vol. 147, pp. 247–253,2000.

[11] N.Virag, "Single channel speech enhancement system based on maskingproperties of the human auditory system," *IEEE Trans. Speech AudioProcessing*, vol. 7, pp. 126–137, Mar. 1999.

[12] J. H. L. Hansen and B. L. Pellom, "An effective evaluation protocolfor speech enhancement algorithms," in *Proc. ICSLP*, vol. 7, Sydney,Australia, 1998, pp. 2819–2822.

[13] J. Sohn, N. S. Kim, and W. Sung, "A statistical model-based voice activitydetector," *IEEE Signal Processing Lett.*, vol. 6, no. 1, pp. 1–3, Jan.1999.

[14] Y. D. Cho and A. Kondoz, "Analysis and improvement of a statistical model-based voice activity detector," *IEEE Signal Processing Lett.*, vol. 8, no. 10, pp. 276–278, Oct. 2001.

[15] S. GazorandW. Zhang, "A soft voice activity detector based on a Laplacian- Gaussian model," *IEEE Trans. Speech Audio Processing*, vol. 11, no. 5, pp. 498–505, Sep. 2003.

[16] M. Matassoni, M. Omologo, A. Santarelli, and P. Svaizer, "On thejoint use of noise reduction and MLLR adaptation for in-car hands-freespeech recognition," in *Proc. 27th IEEE Int. Conf. Acoust. SpeechSignal Process., ICASSP-02*, Orlando, FL, May 13–15, 2002, pp.I-289–I-292.

[17] I. Cohen, S. Gannot, and B. Berdugo, "An integrated real-time beamformingand postfiltering system for nonstationary noise environments,"*Special Issue of EURASIP JASP on Signal Processing for Acoustic Communication System*, vol. 2003, no. II, pp. 1064–1073, Oct. 2003.

[18] A. Gilloire, P. Scalart, C. Lamblin, C. Mokbel, and S. Proust, "Innovative speech processing for mobile terminals: An annotated bibliography,"*Signal Process.*, vol. 80, no. 7, pp. 1149–1166, July 2000.