Intelligibility Enhancement of Synthetic Speech: A Review

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Abstract: Current method of speech enhancement has been developed with adaptive filtering approach. The adaptive filter utilizes the least mean square algorithm for noise removal, but in LMS algorithm key parameter is step size. When step size is large speed and least mean square error is large and it is that computational cost increases to an undesirable level as the length of the impulse response increases. This paper provide a detail review on existing methodologies on enhancement on synthetic speech.

Keywords: *LMS, UNANR, Adaptive Filtering, Synthetic Speech*

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I. INTRODUCTION

In real speech signals time environment are corrupted by several forms of noise such as speaker Noise, background noise, car noise, and also they are subjected to distortion caused by communication channels; examples are room reverberation, low-quality microphones etc. In all such situations extraction of high resolution signals is a key task. In this aspect filtering come in to the picture, basically filtering techniques are broadly classified as non-adaptive and adaptive filtering techniques. In practical cases the statistical nature of all speech signals is non- stationary; as a result non-adaptive filtering may not be suitable. Speech enhancement improves the signal quality by suppression of noise and reduction of distortion. Speech enhancement has many applications; for example, mobile communication, robust speech recognition, low quality audio devices, and hearing aid [1].

Many approaches have been reported in the literature to address speech enhancement. In recent years, adaptive filtering has become one of the effective and popular approaches for the speech enhancement. Adaptive filters permit to detect time varying potentials and to track the dynamic variations of the signals.They modify their behavior according to the input signal. Therefore, they can detect shape variations in the ensemble and thus they can obtain a better signal estimation. Now, the least mean square based adaptive filters have been widely used for speech enhancement. However, a steady state convergence analysis for the least mean square algorithm with deterministic reference inputs showed that the steady-state weight vector is biased, and thus the adaptive estimate does not approach the wiener solution [2].

Digital signal processing (DSP) has been a major player in the current advancements such as noise filtering, system identification, and voice prediction[3]. Standard DSP techniques, however, are not enough to solve these problems quickly and obtain acceptable results. Adaptive filtering techniques must be implemented to promote accurate solutions and a timely convergence to that solution.

Speech is most primary human communication for that reason there exists a big trend to increase & improve telecommunication. However, the background noise is an important handicap. If it is joined with other distortion, it can seriously damage the service quality.

The acoustical environment is defined as a set of transformation that affects the speech signal, since the moment it leaves speakers mouth until it is in digital form. There are mainly two main sources of distortion are acoustic noise & channel distortion[4].

Additive noise is like a fan running in the background, a door slam, a conversion among others it is in our common daily life. It can be stationary or nonstationary. Stationary noisy is one made by a computer fan or air conditioning it has a spectral power density that does not change over time. Non stationary noise caused by door slam radio, TV voices has statistical properties that change over time. A signal captured with speaker close to the microphone has a little noise & reverberation. However if the microphone is far from the speakers mouth it can pick up a lot of noise and reverberation.

Channel distortion can be caused by the frequency response of a microphone, the presence of an electrical signal of the local loop of a telephone line etc. Reverberation caused by the reflection of acoustic waves of the walls and other objects can also dramatically alters the speech signal. The presence of background noise in speech significantly reduces the intelligence of speech noise reduction or speech enhancement algorithms.

These algorithms are used to suppress such background noise and improve the perceptual quality with intelligibility of speech. Removing various type of the noise and the inherent complexities of the speech noise reduction techniques usually have a trade-off between the amount of noise removal & speech distortion introduce due to processing of the speech signal. Several techniques have been proposed for this purpose in the area of speech enhancement, like Adaptive filtering method using LMS filter [4,5,6] and UNR [9] algorithms. The performance of these techniques depends on the quality & intelligibility of the processed speech signal, improvement in the speech signal to noise is the target of most technique.

Introduction To Speech Enhancement

The speech enhancement tells about the growth of communication system. Enhancement means improvement in the value or quality of something. When we applied it to speech this simply means the improvement in the intelligibility and quality of degraded speech signal by using signal processing tools. By speech enhancement, it refers not only to noise reduction but also to dereverberation and separation of independent signals. This is a very difficult problem due to two reasons. First, the nature and characteristics of the noise signal can change dramatically in time and between applications. It is measure can also be defined differently for each application. Two criteria are often used to measure performance: quality and intelligibility. It is very hard to satisfy both at time [2, 3]. Speech enhancement is an area of speech processing where the goal is to improve the intelligibility and pleasantness of a speech signal. The most common approach in speech enhancement is noise removal where estimation of noise characteristics can cancel noise components and retain only the clean speech signal. The basic problem with this approach is that if we remove those parts of signal that resemble noise. In other words, speech enhancement procedures, often inadvertently, also corrupt the speech signal when attempting to remove noise. Algorithms must therefore compromise between effectiveness of noise removal and level of distortion in the speech signal. The term filter is often used to describe a device in the form of a piece of physical hardware or software that is applied to a set of noisy data in order to extract information about a prescribed quantity of interest. The noise may arise from a variety of sources for example, the data may have been derived a useful signal component that has been corrupted by means of noisy sensors or may represent a useful signal component that has been corrupted by transmission through a communication channel.

II. Literature Survey

B. Windrow, J. R. Glover, J. M. McCool [1]explained concept of speech enhancement in a theoretical approach, using different speech enhancement algorithms. The speech enhancement methods aimed at suppressing the background noise are based on one way or the other on the estimation of the background noise. If the noise is evolving more slowly than the speech, it is easy to estimate the noise during the pauses in the speech. If the noise is varying rapidly then estimation is more difficult. This paper addresses the problem of reduction of additive background noise in speech. Here speech enhancement is implemented using three methods. They are speech enhancement using spectral subtraction, speech enhancement using Weiner filter and Kalman filter respectively.

Sayed A. Hadei, M. lotfizad [2]studied in many application of noise cancellation, the changes in signal characteristics could be quite fast. This requires the utilization of adaptive algorithms, which converge rapidly. Least Mean Squares (LMS) and Normalized Least Mean Squares (NLMS) adaptive filters have been used in a wide range of signal processing application because of its simplicity in computation and implementation. The Recursive Least Squares (RLS) algorithm has established itself as the ultimate adaptive filtering algorithm in the sense that it is the adaptive filter exhibiting the best convergence behavior.

RadhikaChinaboina, D.S.Ramkiran[3]studied the adaptive filtering constitutes one of the core technologies in digital signal processing and finds numerous application areas in science as well as in industry. Adaptive filtering techniques are used in a wide range of applications, including echo cancellation, adaptive equalization, adaptive noise cancellation, and adaptive beam forcing. Acoustic echo cancellation is a common occurrence in today‟s telecommunication systems. The signal interference caused by acoustic echo is distracting to users and causes a reduction in the quality of the communication. This paper focuses on the use of LMS and NLMS algorithms to reduce this unwanted echo, thus increasing communication quality.

Sambur M. I., Nutley N. J. [6]described that a novel constrained-stability least-mean-squares algorithm for filtering speech sounds is proposed in the adaptive noise cancellation problem. It is based on the minimization of the squared Euclidean norm of the weight vector change under a stability constraint over the a posteriori estimation errors. To this purpose, the Lagrangian methodology has been used in order to propose a nonlinear adaptation in terms of the product of differential input and error.

A.S.N.Murthy and D.Elizabeth Rani [7] described the removal of noise from speech signals has applications ranging from cellular communications to front ends for speech recognition system. Optimal estimate of adaptive filtering using least mean square algorithm implemented for the observed noisy speech. The algorithm yields better results in noise reduction with significantly less distortions and artificial noise.

.Longbiao Wang, NorihideKitaoka [8] studied a blind dereverberation method based on spectral subtraction using a multi-channel least mean squares (MCLMS) algorithm for distant-talking speech recognition. In a distant-talking environment, the channel impulse response is longer than the short-term spectral analysis window. By treating the late reverberation as additive noise, a noise reduction technique based on spectral subtraction was proposed to estimate the power spectrum of the clean speech using power spectra of the distorted speech and the unknown responses.

Yunfeng Wu, Rangaraj M. Rangayyan,Yachao Zhou and Sin-ChunNg [9] presentednovel unbiased and normalized adaptive noise reduction (UNANR) system to suppress random noise in electrocardiographic (ECG) signals. The system contains procedures for the removal of baseline wander with a two-stage moving-average filter, comb filtering ofpower-line interference with an infinite impulse response (IIR) comb filter, an additive white noise generator to test the system‟s performancein terms of signal-to-noise ratio (SNR), and the UNANR model that is used to estimate the noise which is subtracted from the contaminatedECG signals. The UNANR model does not contain a bias unit, and the coefficients are adaptively updated by using the steepest-descentalgorithm. The corresponding adaptation process is designed to minimize the instantaneous error between the estimated signal power and thedesired noise-free signal power.

Suleyman S. Kozat, Andrew C. Singer, AlperTungaErdogan [10] was said that consider model combination methods for adaptive filtering that perform unbiased estimation. In this widely studied framework, two adaptive filters are run in parallel, each producing unbiased estimates of an underlying linear model. The outputs of these two filters are combined using another adaptive algorithm to yield the final output of the system. The author requires that the final algorithm produce an unbiased estimate of the underlying model. Then later specialize this framework where he combine one filter using the least-mean squares (LMS) update and the other filter using the least-mean fourth (LMF) update to decrease cross correlation in between the outputs and improve the overall performance. The author studied the steady-state performance of previously introduced methods as well as novel combination algorithms for stationary and non-stationary data. These algorithms use stochastic gradient updates instead of the variable transformations used in previous approaches. He explicitly provides steady-state analysis for both stationary and non-stationary environments.

Mamta M. Mahajan, S.S. Godbole [11] said that the concept of adaptive noise cancelling, an alternative method of estimating signals corrupted by additive noise or interference. The method uses a „primary‟ input containing the corrupted signal and a „reference‟ input containing noise correlated in some unknown way with the primary noise. A desired signal corrupted by additive noise can often be recovered by an adaptive noise canceller using the least mean squares (LMS) algorithm. Computer simulations with uncorrelated Gaussian noise and signals conforms the results of the analysis and demonstrate the effectiveness of the least mean squares (LMS) algorithms. The author said that this Adaptive Noise Canceller is then useful for enhancing the S/N ratio of data collected from sensors working in noisy environment, or dealing with potentially weak signals.

S. C. Douglas[12] studied that an adaptive filter is a digital filter that can adjust its coefficients to give the best match to a given desired signal. When an adaptive filter operates in a changeable environment the filter coefficients can adapt in response to changes in the applied input signals. Adaptive filters depend on recursive algorithms to update their coefficients and train them to near the optimum solution. An everyday example of adaptive filters is in the telephone system where, impedance mismatches causing echoes of a signal are a significant source of annoyance to the users of the system. The job of the adaptive filter here is to estimate the characteristics of the echo path, generating the echo and compensate for it. To do this the echo path is viewed as an unknown system with some impulse response and the adaptive filter must mimic this response.

Literature Survey Table

Author	Title	Methodology	Result
Yunfeng Wu, Rangaraj M. Rangayyan, Yachao Zhou and Sin-Chun Ng. (2009Elsevier)	Filtering electrocardiographic signals using an unbiased and normalized adaptive noise reduction system	Unbiased Normalized Adaptive Noise reduction algorithm and combination IIR filtering.	Effective of Noise, SNR elimination improvement
Sayed.AHadai, M.Lotfizad April 2010 (International journal IJCEE)	A family of Adaptive filter Algorithms in Noise cancellation for Speech Enhancement	Fast Affine Euclidean Direction Search algorithm.	Attenuation of signals
SuleymanS.Kozat, AndrewC.Singer, Alpha TungaErdogen and Ali H. Sayed Aug-10 (IEEE Transaction)	Unbiased model combinations for adaptive filtering	Combination methods for adaptive filtering-combination of parallel Adaptive filter.	A steady state mean square error
A.S.N Murthy, D.Elizabeth Rani (May 2011 International Journal)	Adaptive LMS filtering Approach for Speech Enhancement	Least Mean Square Algorithm(LMS).	Speech Signal Distortion
K.Prameela, M.AjayKumar, Mohammad Ziaur-Rahman, DrB.V.Rama ,MohanaRao Oct-11 (International Journal IJCSCN)	Non Stationary Noise removal from speech signals using variable step strategy	Robust variable stepsize LMS(RVSSLMS) Modified robust variable step size LMS(MRVSSLMS).	Effective than conventional LMS in terms of SNR improvement and convergence rate
Roshalizam.Ramli , Ali o.Abid Noor, Salina Abdul Samad 2012 (International Journal AJBAS)	A Review of Adaptive Line Enhancer for Noise Cancellation	Adaptive line Enhancer method based on noise cancellation System.	Fast convergence speed, low computational complexity
AnujaUntawale And KishoriDegaonkar 2015 (ICPC)	Survey on Noise Cancellation Techniques of Speech Signal by Adaptive Filtering	DCT LMS and Adaptive Filtering.	Effective elimination of Random Noise than other conventional algorithms

Table 2.1:- Literature Survey Table

All above methods are observed by different researcher. From above literature survey it is summarized that adaptive filtering algorithms like LMS, NLMS and RLS algorithms which constitute the adjusting mechanism for the filter coefficients are in fact closely related to classical optimization techniques although ,in latter, all calculations are carried out in an off-line manner. Moreover, an adaptive filter, due to its real-time self-adjusting characteristics, is sometimes expected to track the optimum behavior of a slowly varying environment. There are many drawbacks such as Step size, order of filter, convergence rate. When convergence rate is high that time SNR improvement is low but when we concentrated on SNR improvement then we used variable step size for improvement of SNR. In Variable Step convergence rate is main issue so that from given literature survey we conclude that with constant step size and order of filter LMS algorithm gives better performance.

III. Conclusion

InSpeech enhancement the LMS filter algorithm and UNANR filter algorithm is introduced as the main adaptive algorithm in the time domain and its operation has been examined. There are mainly three parameters PSNR, RMSE and Time are considered for performance analysis .The comparison of these different parameters considered for filtering speech signal using LMS filter and UNANR Filter is done. This comparison shows that the signal to noise improvement in the input signal after UNANR filtering is much higher than that LMS filter algorithm. The above paper provides a detailed review on speech enhancement methodologies with adaptive filtering as well.

REFERENCES

[1] B. Widrow, J. R. Glover, J. M. McCool, J. Kaunitz, C. S. Williams, R. H. Hearn, J. R. Zeidler, Eugene Dong and R. C. Goodlin: "Adaptive Noise Cancelling: Principles and Applications," Proceedings of the IEEE, vol. 63, no. 12, Dec. 1975.

[2] Sayed A. Hadei, M. lotfizad; "A Family of Adaptive Filter Algorithms in Noise Cancellation for Speech Enhancement," IJCEE, Vol.2, No.2, April 2010.

[3] RadhikaChinaboina, D.S. Ramkiran, Habibulla Khan, K.PhaniSrinivas, Adaptive noise canceling for speech signals," This paper appears in: Acoustics, Speech and Signal Processing, IEEE Transactions on Issue Date: Oct 1978.

[4] N. J. Bershad, J. C. M. Bermudez, and J. Tourneret, "An affine combination of two LMS adaptive filters: Transient mean-square analysis, IEEE Trans. Signal Process., vol. 56, no. 5, pp. 1853–1864, 2008.

[5] Md. Zia Ur Rahman, K. Murali Krishna, G.V.S. Karthik, M. John Joseph and M. Ajay Kumar, " Non Stationary Noise Cancellation in Speech Signals using an Efficient Variable step size higher order filter", International Journal of Research and Reviews in Computer Science, UK, Vol. 2, No. 1, 2011.

[6] Sambur M. I., Nutley N. J.; "LMS Adaptive filtering for enhancing the quality of noisy speech," IEEE International Conference on ICASSP '78.

[7] A.S.N Murthy and D.Elizabeth Rani ,"Adaptive LMS Filtering Approach for Speech Enhancement" ,International journal of Communication Engineering and Applications,Vol 2, May 2011

[8] Longbiao Wang, NorihideKitaoka, and Seiichi Nakagawa, "Distant-Talking Speech Recognition Based on Spectral Subtraction by Multi-Channel LMS Algorithm," IEICE TRANS. INF. & SYST, Vol. E94-D, No.3 March 2011.

[9] Yunfeng Wu, Rangaraj M. Rangayyan, Yachao Zhou, Sin-Chun Ng, "Filtering electrocardiographic signals using an unbiased and normalized adaptive noise reduction system," Elsevier Medical Engineering & Physics 31 (2009) 17–26.

[10] SuleymanS.Kozat,AndrewC.Singer.AlphaTungaErdogen, and Ali H.Sayed,""Unbiased Model Combinations for Adaptive Filtering", IEEE Transactions, Vol 58,No 8,August 2010

[11] Mamta M. Mahajan, S. S. Godbole, "A New Approach for Speech Enhancement Based on Eigen value Spectral Subtraction," Signal Processing: An International Journal, vol. 3, Issue. 4, pp. 34-41.

[12] S. C. Douglas, "Adaptive Filters Employing Partial Updates," IEEE Trans.
Circuits SYS.II, vol.44, pp. 209-216, Mar 1997.

[13] Simon Haykin, "Adaptive Filter Theory," 4th ed., Prentice-Hall, Inc., 1991.

[14] Simon Haykin: "Adaptive Filter Theory," Third Edition, Prentice- Hall, Inc., Upper Saddle River, NJ, 1996.

[15] [16]Roshahliza M. Ramli, Ali O. Abid Noor, and Salina Abdul Samad, "Noisy speech enhancement using discrete cosine transform," Speech Communication, vol. 24, pp.249-257, 1998.

[16] Anuja N Untwale, KishoriS.Degonkar, "Survey on Noise Cancellation techniques of speech signal by Adaptive Filtering," IEEE International Conference Pervasive Computing, 2015.

[17] V. R. Vijaykumar, P. T. Vanathi, P. Kanagasabapathy, "Speech enhancement using a Kalman-based normalized LMS algorithm" This paper appears in:Telecommunications, 2008. International Symposium on Issue Date 27-28 Aug.2008.

[18] H. Kaur, "Applications and Simulation of Adaptive Filter in Noise Cancellation," *June 2011*.

[19] M. H. Hayes, "Statistical Digital Signal Processing and Modeling," *Wiley-India June 2010.*

[20] A. T. Erdogan, S. S. Kozat, and A. C. Singer, "Comparison of convex combination and affine combination of adaptive filters," in Proceeding of ICASSP , Taipei, Taiwan, 2009.