International Journal of Advanced Biotechnology and Research ISSN 0976-2612,
Vol 2, Issue 1, 2011, pp 183-189 http://www.bipublication.com

A PRACTICAL APPROACH: ADAPTIVE FILTERING FOR SENSORINEURAL HEARING IMPAIRMENT

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ABSTRACT

Hearing impairment is the number one chronic disability affecting many people in the world. Background noise is particularly damaging to speech intelligibility for people with hearing loss especially for sensorineural loss patients. Several investigations on speech intelligibility have demonstrated that sensorineural loss patients need 5-15 dB higher SNR than the normal hearing subjects. This paper describes a practical approach using adaptive filtering for sensorineural impairment to improve the SNR for sensorineural loss patients. The computer simulations results show superior convergence characteristics of the adaptive algorithm by improving the SNR at least 8dB for input SNR's less than and equal to 0 dB and better time and frequency characteristics.

Keywords
Hearing Impairment, Adaptive filter, Sensorineural loss, SNR improvement.

1. Introduction

Hearing impairment is the preamble chronic disability, affecting people in the world. Many people have great difficulty in understanding speech with background noise. This is especially true for a large number of elderly peoples and sensorineural impaired persons.

Hearing loss or deafness can be broadly classified into 2 types. Conductive loss: This type of hearing disability can be measured by audiograms and is considered as a mild disability. So the intelligibility of the signal can be easily resorted by amplification.

Sensorineural loss: This is a broad class of hearing impairments its origin is in the cochlea or auditory nervous system. sensorineural loss disorders are difficulty to remedy. This type of defects may be due to congenital or hereditary factors, disease, tumors, old age, long-term exposure to industrial noise, acoustic trauma or the action of toxic agents etc. The sensorineural loss patient's experiences difficulty in making fine distinction between speech sounds, particularly those having a predominance of high frequency Energy [5], [6]. He may hear the speaker's voice easily, but be unable to distinguish. For example between the words 'fat' and 'sat' [7], [28]. Two features of sensorineural impairment particularly detrimental to the perception of speech are high tone loss and compression of the dynamic range of the ear. A high tone loss is analogous to low pass filtering. Amplification of the high tones may improve intelligibility, but in these circumstances dynamic range of the ear is a handicap [9], [4]. Because, the dynamic range of the impaired ear may not be sufficient to accommodate the range of intensities in speech signals. So, the stronger components of speech are perceived at a level, which is uncomfortably loud, while the weaker components are not heard at all [2], [4], [5].

Several investigations on speech intelligibility have demonstrated that subjects with sensorineural loss patients need 5 to 15db higher

SNR than the normal hearing subjects. While most of the defects in transmission chain up to cochlea can now-a-days be successfully rehabilitated by means of surgery. The great majority of the remaining inoperable cases are sensorineural hearing impaired patients [5], [3]. Digital technology has made an important contribution in the field of audio logy. Digital signal processing methods offer great potential for designing a hearing aid but, today's Digital Hearing Aid are not up to the expectation for sensorineural loss patients. Hearing-impaired patients applying for hearing aid reveal that more than 50% are due to sensorineural loss. So for only Adaptive filtering methods are suggested in the literature for the minimization of noise from the speech signal for sensorineural loss patients [7].

2. Adaptive Filtering Method

The least mean square algorithm was first introduced by Widrow and Hoff in 1959 is simple, robust and is one of the most widely used algorithms for adaptive filtering. LMS algorithm is generally the best choice for many different applications [1], [14]. This method can be effectively applied to reduce the noise i.e. to improve the SNR for sensorineural loss patients [6], [7], [10].

2.1 Adaptive noise canceller with two microphones

In the adaptive noise cancellation process, the desired speech signal $d(n)$ is to be calculated from a noise-corrupted speech signal $x(n) = d(n) + v_1(n)$. In this method, reference signal, $v_2(n)$ is correlated with $v_1(n)$. This reference signal may be used to estimate the noise $v_1(n)$, and the estimate can be then be subtracted from $x(n)$ to get the estimate of $d(n)$.

$$\hat{d}(n) = x(n) - \hat{v}_1(n) \quad \ldots..1$$

If $d(n)$, and $v_1(n)$ are jointly wide-sense stationary processes, and if the autocorrelation $r_{v_2}(k)$ and the cross-correlation $r_{v_1 v_2}(k)$ are known, then filter can be designed to find the minimum means square estimate of $v_1(n)$. But, the desired speech signal $d(n)$ and the noise $v_1(n)$ are non-stationary signals and their autocorrelations are unknown [14], [1]. Therefore, as an alternative to the Wiener filter, adaptive noise canceller is considered as shown in Fig.1. If the reference signal $v_2(n)$ is uncorrelated with $d(n)$, then the minimization of mean square error $E\{|e(n)^2|\}$ is equivalent to minimizing

$$E\{|v_1(n) - \hat{v}_1(n)|^2\} \ldots\ldots.2$$

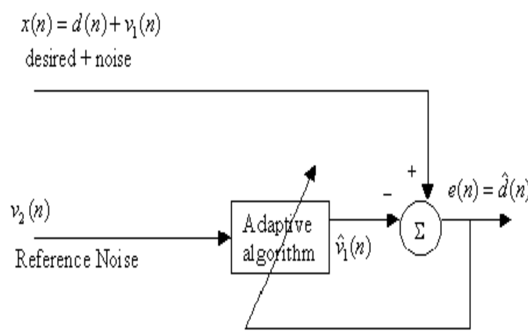


Figure 1. Adaptive noise canceller

In other words, the output of the adaptive filter is the minimum mean square estimate of the noise $v_1(n)$. If there is no information about desired speech signal $d(n)$ in the reference signal $v_2(n)$, then the adaptive filter can minimize noise, by estimating $v_1(n)$. Since the output of the adaptive filter is the minimum mean square estimate of $v_1(n)$, then it follows that $e(n)$ is the minimum mean square estimate of the desired speech signal $d(n)$. It is an efficient noise canceller and is implemented in digital hearing aid for reducing noise effects especially for SNHL persons [10], [13].

2.2 Single Microphone Adaptive Algorithm

Under ideal conditions, at least one microphone (omni directional) must be placed at the noise source. If the ideal adaptive noise canceller is used with two microphones, one microphone picking up more speech than noise and the other microphone picking up more noise than speech, the noise will not be cancelled completely, but the level of the noise will be reduced [30], [29], and [7]. An improved speech-to-noise ratio will result, with improved intelligibility and can be effectively used for noise reduction in digital hearing aid for SNHL persons. Although these adaptive noise cancellation methods are extremely effective, they are restrictive in their requirements on the availability and placement of multiple microphones. In these situations, adaptive filters can be employed with single microphone. Since, it is possible to derive a reference signal by simply delaying the input noisy speech signal $x(n) = d(n) + v_1(n)$ as shown in Fig. 2.

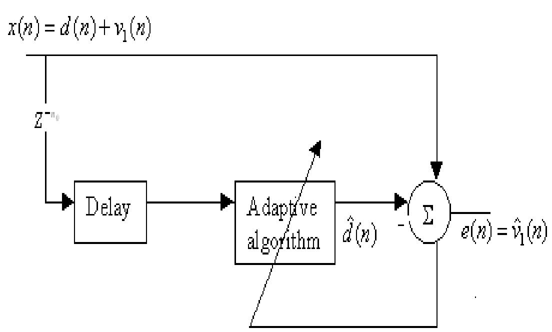


Figure 2. Adaptive noise canceller with single microphone

The desired speech signal $d(n)$ is a narrowband signal and that $v_1(n)$ is a broadband noise with

$$E\{v_1(n)v_1(n-k)\} = 0; \qquad |k| > k_0 . \quad \text{….3}$$

If $d(n)$ and $v_1(n)$ are uncorrelated, then

$$E\{v_1(n)x(n-k)\} = E\{v_1(n)d(n-k)\} + E\{v_1(n)v_1(n-k)\} = 0; \qquad |k| > k_0 \quad \text{….4}$$

Therefore, if $n_0 > k_0$ then the delayed process $x(n-n_0)$ will be uncorrelated with the noise $v_1(n)$, and correlated with $d(n)$. Thus, $x(n-n_0)$ may be used as a reference signal to estimate $d(n)$ as shown in Fig.2. In contrast to the adaptive noise canceller in Fig. 1, the adaptive filter in Fig. 2 produces an estimate of the broadband process, $d(n)$ and the error $e(n)$ corresponds to an estimate of the noise $v_1(n)$. This adaptive structure can be effectively used for noise cancellation, but best suited for stationary signals [14]. In the case of non-stationary signals like speech signals, the delay would normally be a multiple of the pitch period [1], [8], and [13]. If, the delay is chosen properly, then this structure can be effectively used for noise cancellation in digital hearing aids. In this work, the delay is approximately chosen, whose length is very much lesser than the length of the signal.

3. Results and evaluation

In this work normalized LMS and single source NLMS has been implemented for the enhancement of the speech signal in digital hearing aid for SNHL persons. The performance of both algorithms has been measured using output SNR, eigenvalue ratio, time plots and intelligibility tests.

3.1 Performance evaluation by using output SNR, eigenvalue ratio and time plots

The algorithm is evaluated for corrupted speech signals with different types of noise like cafeteria, low frequency and babble noise with different SNR. The various parameters like β and filter order were changed and the performance of the algorithm was evaluated. Results show that, both parameters SNR and eigenvalue ratio are strongly depending on β and filter order.

The input signal is a speech sentence in English and is recorded with sampling frequency 22050 Hz in different noisy conditions to evaluate the performance of the algorithm. Performance of the algorithm was studied, for different values of β and filter order. From the studies, we have noticed that for $\beta = 0.45$ and filter order = 32 normalized LMS gives better SNR and intelligibility improvement.

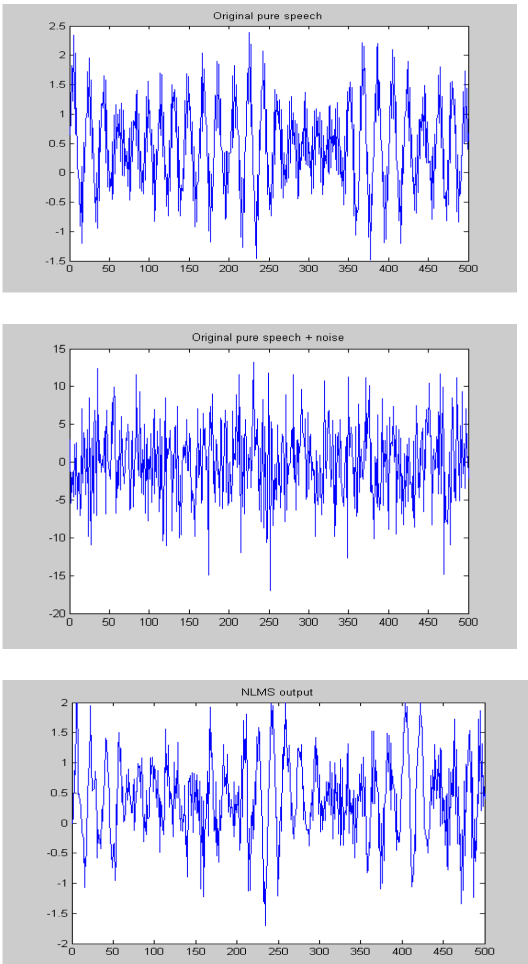
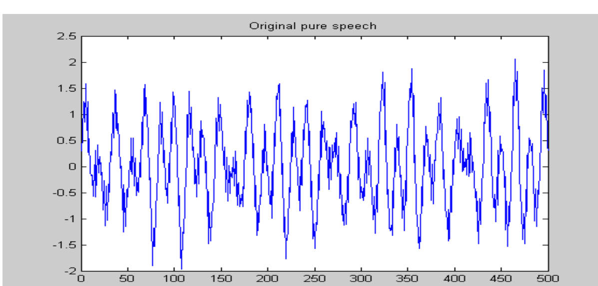
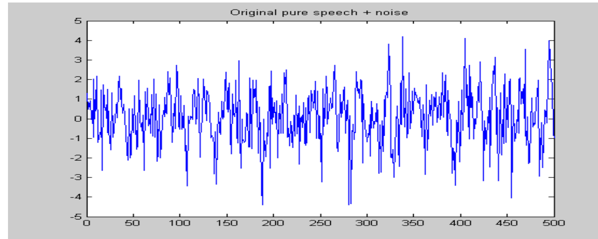


Figure 3 Original, contaminated and filtered signals of NLMS noise canceller.





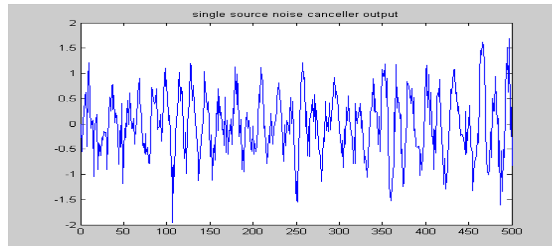


Figure 4 Original, contaminated and filtered signals of single source noise canceller.

	Input SNR in dB	Output SNR in dB	Eigenvalue ratio
NLMS	-5	7.33	1069.43
	0	9.32	1060.32
	+5	11.15	1003.06
Single Source NLMS	-5	6.2	1129.67
	0	8.51	1132.35
	+5	11.15	1102.53

Table 1 Eigenvalue ratio and SNR of the output signal for direct LMS, NLMS and Single Source LMS

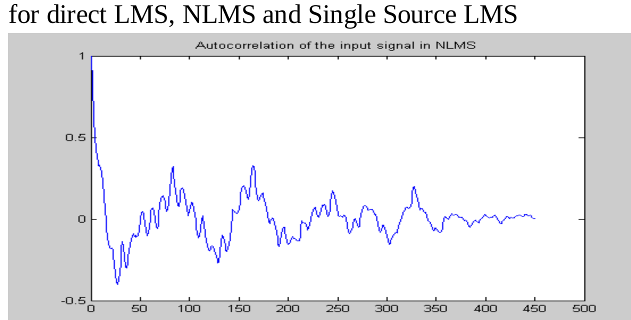


Figure 5 Autocorrelation of the input signal

For different input SNR, the output SNR and eigenvalue ratios are calculated as shown in Table 1. The eigenvalue ratio is calculated to find out how well the algorithm converges to the optimum Wiener solution. The main disadvantage of both the algorithms is their high eigenvalue ratio. Because, filters having high eigenvalue ratio requires longer time to converge and vice versa.

Fig. 3 shows the time plots of pure signal, corrupted signal with –5dB SNR, and the NLMS filtered signal. Fig 4 shows the time plots of pure signal, corrupted signal and filtered signal for single source noise canceller. Fig. 5 shows the autocorrelation of the input corrupted signal verses number of samples. The NLMS with two source gives better results compared to single source NLMS.

3.2 Intelligibility Test

In order to measure the performance of clinical intelligibility tests of the algorithms, listening tests were carried out. The tests were conducted on both hearing impaired and normal hearing persons. The experiment was carried out in a room whose size was about 4 m by 5 m. The room was carpeted but no attempt was made to improve the room acoustics otherwise. The main speaker and the noise source were placed 2.5 feet away from the microphones. For speech intelligibility test, we processed 10 sentences with different noise. These tests were performed on 15 subjects, 5 with normal hearing (Group 1), 5 with a mild to moderate SNHL (Group 2) and 5 with moderate to severe SNHL loss (Group 3).

In the experimental evaluation, the target source was a male speaker reading sentences and interference consisted of 3 different types of noise (1) cocktail party noise (2) five speaker babble (3 male and 2 female) (3) low frequency noise. The noise level is varied to get different SNR. The subjects were listened the original, the noisy and the filtered signals. The percentage of correct responses was recorded. The results are displayed in Tables 2, 3 and 4 for –5dB input SNR. The results indicate that a considerable improvement is obtained, particularly for moderate to severe SNHL subjects.

The result of recognition test of NLMS and single source noise canceller signals are displayed in Table 4. It is seen that after adaptive processing the intelligibility improvement is achieved in all the cases. NLMS filter showed an average intelligibility improvement of 1 % with normal subjects, 4 % with mild to moderate SNHL subjects and 6 % with moderate to severe SNHL subjects as compared to single source noise canceller with cocktail party noise.

Group1	Group 2	Group 3
96 %	78 %	63 %

Table 2 Average intelligibility score for the noiseless signal

Types of noise	Cocktail party noise	Babble noise	Low frequency noise
Group1	73 %	78 %	83 %
Group 2	31 %	34 %	38 %
Group 3	15 %	13 %	16 %

Table 3 Average intelligibility score for the signal plus noise

Types of noise		Cocktail party noise	Babble noise	Low frequency noise
Group1	NLMS	94 %	93 %	94.5 %
	NLMS with single source	93 %	92 %	93 %
Group2	NLMS	75 %	73 %	74.5 %
	NLMS with single source	71 %	70.5 %	72 %
Group3	NLMS	63 %	61 %	64 %
	NLMS with single source	57 %	53 %	58 %

Table 4 Intelligibility improvements for three groups of subjects.

3.3 Conclusion

Adaptive algorithms can be successfully used for noise reduction in digital hearing aid for SNHL persons. Off-line tests in different conditions show improvements in SNR up to 10.06 dB with NLMS-algorithm and up to 8.6 dB with single source NLMS for zero dB input SNR. In a more realistic environment, they may have the variations of ± 3 dB. But, the results show that, both the algorithms have high eigenvalue ratio. Hence, they need more time to converge into the optimal solution to the identity matrix. In that case, the inputs are perfectly uncorrelated and have equal power [1], [14]. In LMS filter, convergence rate is highly dependent on the conditioning of the autocorrelation matrix of its inputs. The mean square error of an adaptive filter trained with LMS, decreases over time as a sum of exponentials. The time constants of mean square error are inversely proportional to the eigenvalue of the autocorrelation matrix of the filter inputs. Therefore, small eigenvalue create slow convergence modes in the mean square error function. Large eigenvalue, on the other hand, put a limit on the maximum learning rate that can be chosen without encountering stability problems. Best convergence properties are obtained when all the eigenvalue are equal, that is, when the input autocorrelation matrix is proportional

Convergence performance of the standard LMS algorithm can be improved by using frequency domain filtering, by exploiting the orthogonal property of DFT and related orthogonal transforms. Hence transform domain adaptive algorithms can be implemented with less computational complexity, good decorrelation efficiency and good SNR and intelligibility improvement for SNHL persons [15]-[27].

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