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ADAPTIVE FILTER IN HEARING AID SYSTEM FOR IMPAIRED PEOPLE

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ABSTRACT

Most of we humans are gifted with our five senses intact. However an unfortunate few continue to suffer in their own domain. Science has given them gadgets, which may help them to lead a near normal life. Most of these gadgets have shortcomings. For instance, we may consider the hearing aid. A gadget used widely by people who are hard of hearing. Through it improves their hearing perception, unwanted amplification of noise could cause irritation and even damage under extreme conditions. There are noise cancellation algorithms, which currently exist. But they act only upon noises of kind that they are designed for. There is no algorithm currently existing, which will cancel any kind of noise. In this paper we aim at presenting an algorithm, which enhances speech by attenuating any kind of noise. To achieve this we design an adaptive filter, which would adapt itself depending on the nature of noise. we compared with LMS Algorithm and proposed blind technique, since we do not know the parameters of the noise signal. The primary differentiation that we make between speech and noise is that speech is highly non-stationary in a time interval of 250ms whereas noise stationary. This characteristic is used to derive a cost functional using which we may achieve speech enhancement. Adaptation takes place by changing two sets of weights. Apart from hearing aids this algorithm also finds application in mobile phones (Hands free kit). The above algorithms are simulated with the help of mat lab program and to implement using TMS320C6X DSP processor for real time application.

1. INTRODUCTION

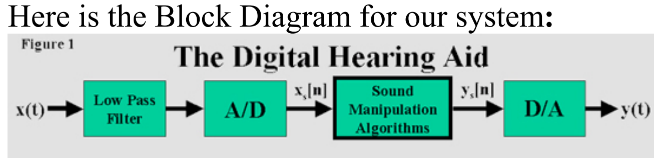
Traditional analog hearing aids are similar to a simple radio. They can be tuned and adjusted for volume, bass, and treble. But hearing is not just a technical loss of volume .Rather, hearing deficiency can increase sensitivity and reduce tolerance to certain sounds while diminishing sensitivity to others. For instance ,digital technology can tell the difference between the speech and background noise allowing one in while filtering out other .Approximately 10% of the world's population suffers from some type of hearing loss, yet only a small percentage of this statistic use a hearing aid. The stigma associated with the hearing aid, customer dissatisfaction with hearing aid performance, and the cost associated with a high performance solution are all causes of low market penetration. Through the use of digital signal processing, digital hearing aid now offers what the analog hearing cannot offer.

The primary problem faced during noise reduction pertaining to speech, is that no parameters are known about the characteristics of noise. Previous methods involve the usage of an anti-phase signal to cancel the primary source signal. This technique has been used successfully in many industrial applications to reduce noise levels. However such a technique is useless in the case of speech enhancement. Also usage of filters for noise reduction will also be useless owing to the uncertain nature of noise. Hence an adaptive approach is an apt solution to this problem [1],[2],[3]. Further a blind technique is well suited in this case wherein no prior assumptions are made regarding the properties of speech and noise [4],[5]. However the spectral properties of human speech and its phoneme structure are used to model the system. The existing technique used to enhance the speech quality is subject to both degradations due to surrounding environment All these tasks must be achieved with a single VLSI chip in order for the system to be both cost-effective, power efficient and widely accepted[6]. Hence the goal of this paper is to prove that this technique can better the existing one, using VLSI technology the same can be implemented and realized. This paper proposes

1. Adaptive signal processing [1], [3], [4], [8].
2. Noise and echo analysis and cancellation [2], [6].
3. The simulation is done using MATLAB [9], [10]
4. For real time application,TMS320C6X DSP processor is used.

2. BASIC ALGORITHM USEDOUR SYSTEM

Here is the Block Diagram for our system**:**



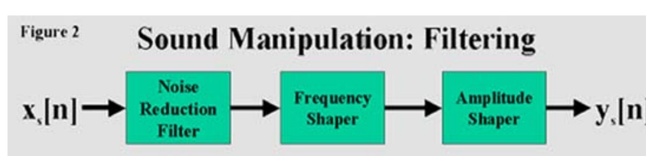


Fig. 1. Block Diagram for our system

In everyday situations, there are always external signals that may interfere with the sounds that the hearing aid user actually wants to hear. This ability to distinguish a single sound in a noisy environment is a major concern for the hearing impaired. For people with hearing loss, background noise degrades speech intelligibility more than for people with normal hearing, because there is less redundancy that allows them to recognize the speech signal. Often the problem lies not only in being able to hear the speech, but in understanding speech signals due to the effects of masking. To adjust for this loss, we developed noise reduction filter in MATLAB for our hearing aid.

3.LEAST MEAN SQUARE ALGORITHM :

The objective is to change (adapt) the coefficients of an FIR filter, W, to match as closely as possible the response of an unknown system, H. The unknown system and the adapting filter process the same input signal $x[n]$ and have outputs $d[n]$ (also referred to as the desired signal) and $y[n]$.The adaptive filter, W, is adapted using the least mean-square algorithm, which is the most widely used adaptive filtering algorithm.

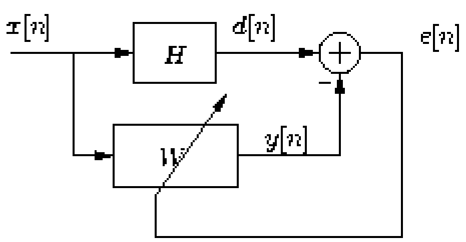


Fig.2.LMS Algorithm

First the error signal, $e[n]$, is computed as

$$e[n] = d[n] - y[n] \text{-------------(1)},$$

this measures the difference between the output of the adaptive filter and the output of the unknown system. On the basis of this measure, the adaptive filter will change its coefficients in an attempt to reduce the error.

LMS coefficient update,

$$h_{n+1}[i] = h_n[i] + \mu\, e\, x[n-i] \text{ ---------(2)}$$

The step-size μ directly affects how quickly the adaptive filter will converge toward the unknown system. If μ is very small, then the coefficients change only a small amount at each update, and the filter converges slowly. With a larger step-size, more gradient information is included in each update, and the filter converges more quickly; however, when the step-size is too large, the coefficients may change too quickly and the filter will diverge. (It is possible in some cases to determine analytically the largest value of μ ensuring convergence.)

4. FREQUENCY SHAPER:

· Applies gain > 1 for hard-to-hear frequencies
· Modifies gain for other specified ranges

The frequency shaper is designed to correct for loss of hearing at certain frequencies. The filter applies a gain greater than one to the frequencies that the user has difficulty hearing. As one of its parameters, the filter takes in a vector of frequencies, determined by an audiologist, that define the user's hearing characteristics. For each range, the frequency shaper applies a certain gain based on the user's specific hearing loss. Thus, our frequency shaper is completely configurable to any user.

5.AMPLITUDE SHAPER:

Once the signal has been passed through the Noise Reduction Filter and the Frequency Shaper, it will be passed through our Amplitude Shaper. The dynamic range of hearing is measured in terms of sound pressure, in decibels. A normal hearing range extends from approximately 0 dB to 120 dB, where 0 dB is the Threshold of Hearing and 120 dB is the Threshold of Pain. Discomfort usually begins to occur around a saturation level of about 90 dB of sound.

Classification of hearing loss	Hearing level
Normal Hearing	-10dB - 26dB
Mild hearing loss	27dB - 40dB
Moderate hearing loss	40dB - 70dB
Severe hearing loss	70dB - 90dB
Profound hearing loss	> 90dB

Table 1.Different degree of hearing loss

Hearing loss compresses the range of hearing, raising the Threshold of Hearing and typically lowering the Threshold of Pain. For example, a person with moderate hearing loss would have a Threshold of Hearing around 40 - 70 dB and a Threshold of Pain around 100 dB.

How does the Amplitude Shaper Work?

We assume that the Frequency Shaper raises the frequencies that the user has difficulty hearing to sound pressure levels within his dynamic range of hearing. Therefore, all that our Amplitude Shaper has to do is check, bit by bit, that output power does not exceed a given saturation level, Psat. Since noise is concentrated in the low power levels as well, the filter also removes a significant amount of noise. Output power is equal to zero for levels below Psat.

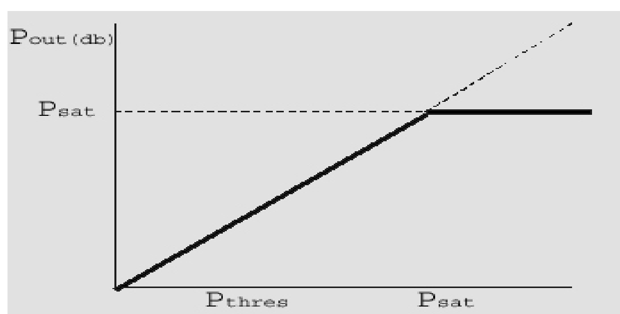


Fig 3. Amplitude Shaper Response

6.PROPOSED BLIND TECHNIQUE:

A.Strategy For Noise Identification

We express the incoming signal as y[n] which is a combination of speech s[n] and noise u[n].

$$y[n] = s[n] + u[n] \qquad (3)$$

No prior knowledge about the parameters of s[n] and u[n] are known. we described some of the important features of the human speech signal s[n]. However, the characteristic that plays prominence is its non-stationary nature while considering a time frame of over 250ms. In this same time frame noise is predominantly stationary in nature. For noise, the autocorrelation structure and the power spectrum density remains constant over long time intervals. This basic property is used to differentiate between noise and speech

B. Solution Requirements

1. Considering the fact that speech content is a mixture of various frequency components, we must analyze the incoming signal over the entire speech signal bandwidth. This may be achieved by a filter bank structure.
2. Having split the input into pass bands we must analyze each of these pass bands individually to analyze which of these pass bands have noise. This may be achieved by calculating a noise estimate for each of the pass bands.
3. Knowing which pass bands are noisy we require a methodology to adapt the system to attenuate those pass bands which are noisy and amplify the noise free pass bands [4].

C. Noise Energy Estimation

Before applying any enhancement procedure we need to establish which pass bands contain noise. The problem faced here is that noise energy cannot be estimated separately. Only y[n] (speech + noise) is available with us. To determine noise we use the "indirect method" wherein we calculate the autocorrelation of the input sequence and then find the discrete Fourier transform of the autocorrelation sequence. Theoretically it is not possible to extract the noise separately at this stage, hence the energy calculated in these gaps will represent the noise energy and this will be a minimum value. [7].

D. Approximated Procedure

After filtering we have five separate sequences as shown in Fig.1. We have separated each of these pass band sequences into M sub-frames (represented by index k).

We must note that the value of M is critical (i.e.) it must satisfy the primary condition that within the sub frame interval speech must be non-stationary and noise must be stationary. Each of the M sub frames contains L samples each (represented by index). For the simulation we assumed a value of 20ms as the time interval of each sub frame [4].

$$Eij = Enij + Esij = Eni = Esij \tag{4}$$

Using (2) we may determine the noise energy estimate of the input signal y[n].

E. Primary Weight Adjustment

Consider a set of weights W_i (one for each pass band). The signals yi [n] is multiplied with each of these weights. The primary function of these weights is to attenuate the noise. The values of W_i are calculated such that the band with the largest average speech signal energy is passed without attenuation. [11]

$$W_i = \begin{cases} 1 & , i \notin I \\ \frac{1}{q} \sum_{j=1}^{M} (E_{ij} - E_{ni}) & , i \in I \end{cases} \tag{5}$$

q is a quantifying value which restricts the value of W_i within 0 and 1.

Each sequence $y_i[n]$ is multiplied with the corresponding weight W_i to give $S_i[n]$ which is of much better quality and contains lesser noise compared to the input sequence.

F. De-Noising Subsystem

This subsystem plays a prominent role in speech enhancement. The speech characteristics discussed earlier are used in this subsystem. Primarily, we need to describe a functional, which would separate noise from speech. we may infer that the relative change in standard deviation (RCSTD) of speech signal is much greater than a noise signal[4]. Also we may note that speech signal is composed of many peaks.

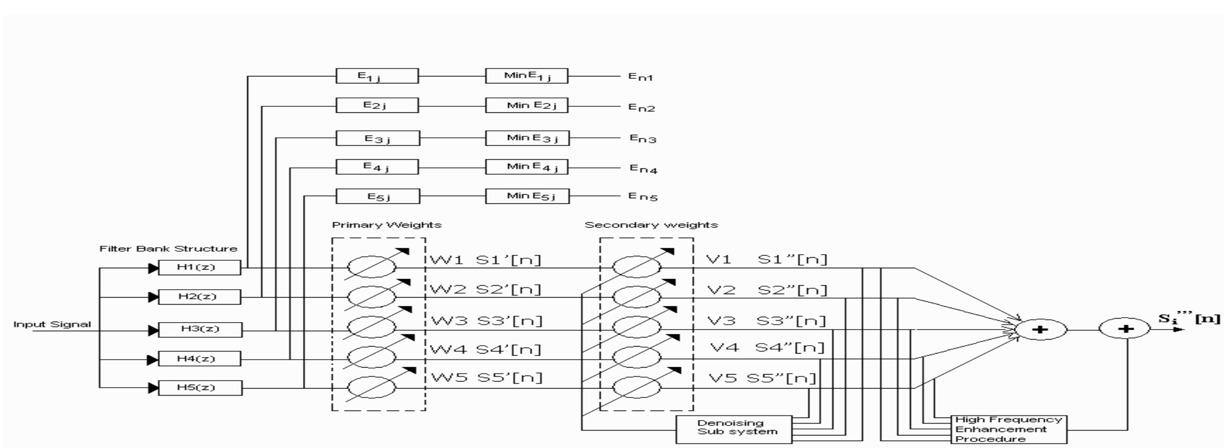


Fig. 4. Schematic diagram of the Blind Adaptive Filter system.

Hence we may describe the functional based on the above characteristic. We define

$$E'_{ij} = \sum_{k=0}^{L-1} (S'_1[(j-1)L+n])^2 \qquad \begin{array}{l} i = 1, 2, \ldots, N \\ j = 1, 2, \ldots, M \end{array} \tag{6}$$

Also,

$$E''_{ij} = V_1^2 E'_{ij} \tag{7}$$

Standard deviation of the j th sub frame is defined as:

$$\sigma_j(s'') = \sqrt{\sum V_1^2 E'_{ij}} \qquad j = 1, 2, \ldots, M \tag{8}$$

Hence RCSTD of the signal may be calculated from the equation (7):

$$\left|\Delta\sigma_j(s'')\right| = \left|\frac{\sigma_j(s'') - \sigma_{j-1}(s'')}{\sigma_{j-1}(s'')}\right| \tag{9}$$

Where c_1 and c_2 are real weight constants.

we may describe the primary requirements of the functional as a tool to maximize the area under the RCSTD curve of the input signal. By maximizing the value of J_{c1} by altering v_i we may increase the area under the RCSTD curve there by enhancing the speech content. To prevent unwanted suppression of speech components we maximize the cost functional J_{c2}. This results in maximization of speech signal energy thereby solving the problem we derive a cost function Jc given by equation (10)

$$J_c = c_1 J_{c1}^{-1} + c_2 J_{c2}^{-1} \tag{10}$$

Thus problem reduces to minimization of cost function Jc using simplex method to obtain an enhanced speech signal $s_i''[n]$.

7. MATLAB RESULT:

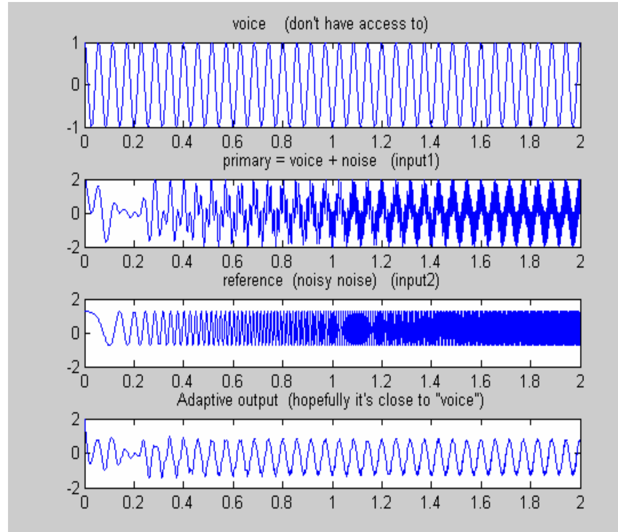


Fig 5.LMS OUTPUT (μ =.005)

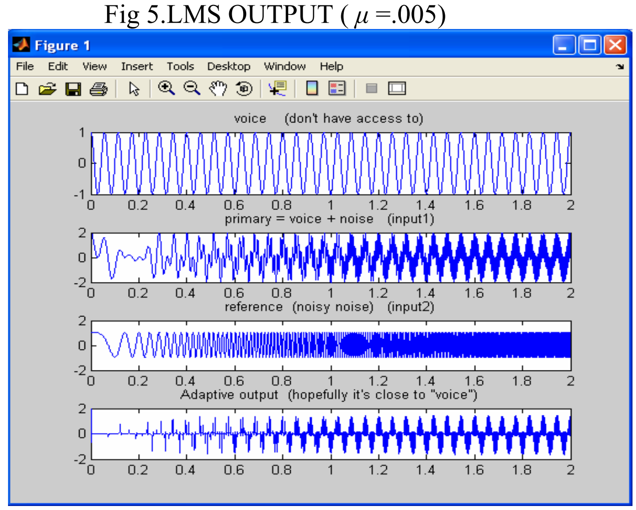


Fig 6.LMS OUTPUT (μ =0.9)

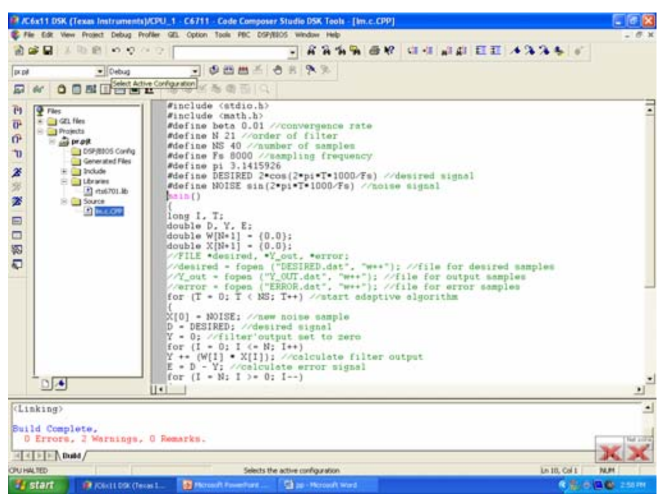


Fig .7. LMS C CODE OUTPUT(C6711 DSP processor)

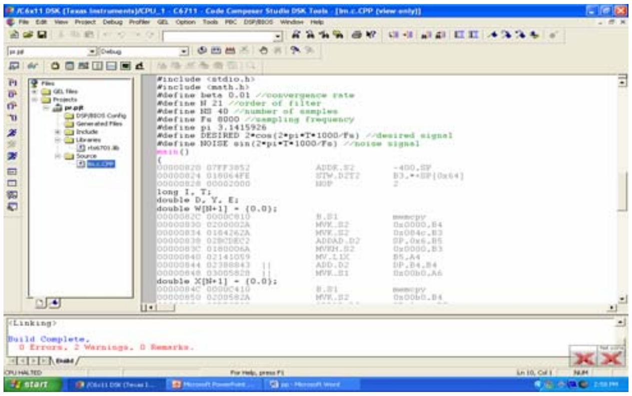


Fig .8. LMS MIXED SOURCE CODE OUTPUT
(C6711 DSP processor)

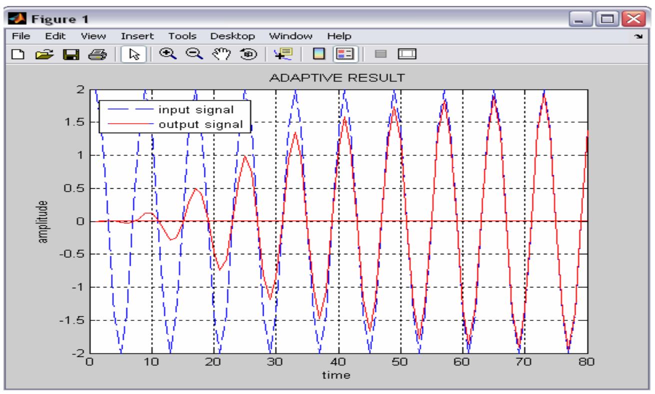


Fig .9. LMS (μ =0.005) OUTPUT(C6711 DSP processor)

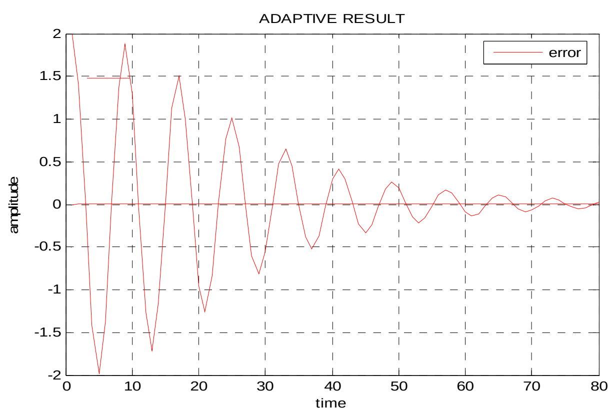


Fig .10. LMS (μ =0.005) ERROR SIGNAL
(C6711 DSP processor)

8. CONCLUSION

Currently we are working on the simulation using MATLAB and the initial results have been satisfactorily obtained.This paper mainly aims at proposing this methodology for noise cancellation, which has potential applications for impaired people as hearing aid. Apart from these applications, it can be used for Car telephones, cell phone hands free kits [2].Real time implementation is done using TMS320C6X DSP processor.

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