

# Implementation of blind adaptive filtering using VLSI technology

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## ABSTRACT

We are living in a very congested world where there are more and more vehicles. The noise and interference produced by these vehicles are enormous and their unwanted amplification of noise causes irritation. We aim to present a blind technique, enhancing speech, attenuating any kind of noise, by designing an adaptive filter depending on the nature of noise. The differentiation between speech and noise is made. This characteristic is used to derive a cost functional for speech enhancement. Adaptation is by changing two sets of weights. The simulation and synthesis of the above technique/algorithm is implemented using Verilogger pro and Leonardo spectrum.

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## Introduction

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 road, engine and wind noise. The echoes present in the near-end speaker's side-sources affect the car phone input. 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 research proposes

1. Study of speech coding and analysis [5], [7].
2. Adaptive signal processing [1], [3], [4], [8].
3. Noise and echo analysis and cancellation [2], [6].
4. To develop sound signal processing Algorithms to combat these imperfections.
5. The simulation is done using Verilogger pro [9], [10]
6. Synthesis is done using Leonardo spectrum
7. Back end tools are used to fuse the design into a chip and can be used for real time application.

### Approach to the problem

The primary requirement is a strategy by which we will be able to differentiate between noise and sound. This may be achieved by analyzing the properties of human speech. This idea has been put forth earlier [4]. To make the model much more effective, we have tried to implement it using VLSI technology. Strategy For Noise Identification

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

$$y[n] = s[n] + u[n] \quad (1)$$

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. [7].

### 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].

### 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].

### 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].

$$E_{ij} = E_{ni} + E_{sj} = E_{ni} + E_{sj} \quad (2)$$

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

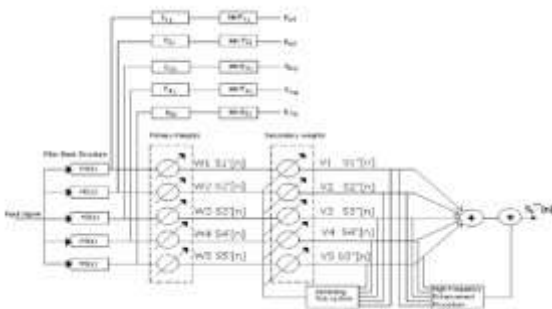
**Primary Weight Adjustment**

Consider a set of weights  $W_i$  (one for each pass band). The signals  $y_i$  [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]

$$W_i = \frac{1}{M} \sum_{j=1}^M (E_{ij} - E_{ni}) \quad , i = 1, \dots, I \quad (3)$$

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.



**Fig. 1. Schematic diagram of the Blind Adaptive Filter system**

**De-Noiseing 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. To arrive at the result we analyze the graphs:

From the Fig. 2.(a),(b), 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. Hence we may describe the functional based on the above characteristic.

We define,

$$E_{ij} = \sum_{k=0}^{L-1} (S_1^{(j-k)} - E_{ni})^2 \quad \begin{matrix} i = 1, 2, \dots, N \\ j = 1, 2, \dots, M \end{matrix} \quad (4)$$

Also,

$$E_{ij} = V_1^2 E_{ij} \quad (5)$$

Standard deviation of the  $j^{th}$  sub frame is defined as:

$$\sigma_j(s) = \sum_{i=1}^N V_1^2 E_{ij} \quad j = 1, 2, \dots, M \quad (6)$$

Hence RCSTD of the signal may be calculated from the

equation (7):

$$\Delta \sigma_j(s) = \left( \begin{matrix} \sigma_j(s) - \sigma_{j-1}(s) \\ \sigma_j(s) \end{matrix} \right) \quad (7)$$

This problem reduces to minimization of cost function  $J_c$  using simplex method to obtain an enhanced speech signal

$$s_i''[n].$$

Note: We must note that the area under the RCSTD curve greater in the case of the speech signal and comparatively smaller in the case of the noise signal. Putting it in a nutshell, we may describe the primary requirements of the functional as a tool to maximize the area under the RCSTD curve of the input signal.

We define the functional as,

$$J_{c1} = \sum_{j=2}^M \Delta \left( \sigma_j(s) \right) \quad (8)$$

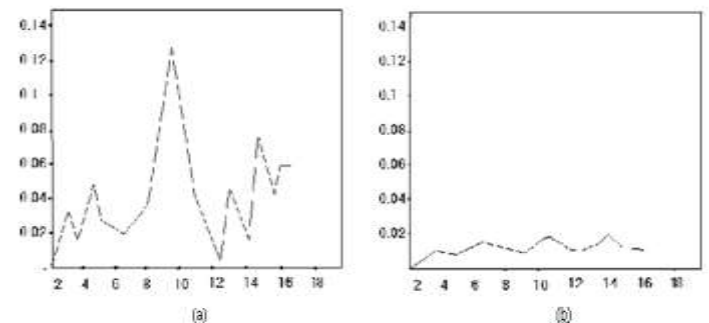
By maximizing the value of  $J_{c1}$  by altering  $v_i$  we may increase the area under the RCSTD curve there by enhancing the speech content. To prevent unwanted suppression of speech components we maximize the cost functional  $J_{c2}$ . This results in maximization of speech signal energy thereby solving the problem. The cost functional  $J_{c2}$  is defined by equation (9) as,

$$J_{c2} = \sum_{i=1}^N v_i \sum_{j=1}^M (E_{ij} - E_{ni}) \quad (9)$$

Finally we derive a cost function  $J_c$  given by equation (10)

$$J_c = c_1 J_{c1} + c_2 J_{c2} \quad (10)$$

Where  $c_1$  and  $c_2$  are real weight constants.



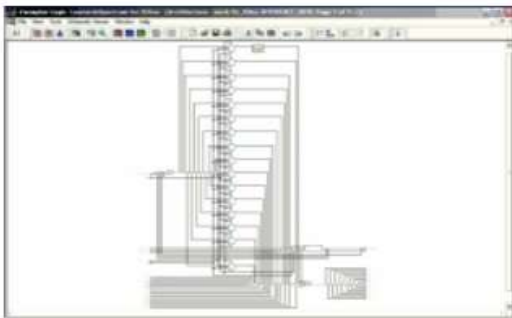
**Fig. 2. (a) Relative change of Standard deviation of the Noiseless signal s[n], and (b) relative change of standard deviation of the noisy signal y[n]**

**Pragmatic Implementation Using Verilog Hdl**

VERILOG HDL, an esteem software, used for practical circuit implementation, is especially a single chip design. The efficient and extremely powerful method described here could be very well established using VERILOG HDL, the resultant chip design could be practically implemented and can be eminently used for all practical applications. Blind Adaptive, the algorithm under speech, which is a paragon for design by itself, is handled in a professional way regarding practical chip design that involves simulation reports, chip design files and documents regarding implementation.[9],[10].

### Conclusion

Currently we are working on the simulation and synthesis part on Verilogger Pro and Leonardo Spectrum and the initial results have been satisfactorily obtained as below in Fig. 3. Crucial factors are the selection of threshold values, the time interval of the sub-frames etc. These can be determined only after experimentation. This paper mainly aims at proposing this methodology, which has potential applications ranging from Car telephones, cell phone hands free kits. Apart from these applications, it can be used for hearing aid applications [12]. Implementation is done using VLSI technology.



**Fig. 3. Synthesized Results**

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