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Speech Enhancement Based On Modified Least Mean Square Filter Using Xilinx System Generator

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ABSTRACT :

Speech enhancementis playing a major role in voice recognition application.so speech requires to be denoised and enhanced .in literature many filters exist to transform speech and make noise free but error estimation is fixed or static.as the amount of noise varies in speech based on acoustics a adaptive filter is necessary to be implemented .least mean square filter is a accurate filter to estimate error and filtering speech signal.to implement the LMS adaptive filter in integrated circuit level optimization in area and power parameters is essential so a novel method is proposed to implement LMS filter with parallel operations .PPG is the shared unit between error reduction and weight computation blocks.

Keywords : LMS filter, adaptive filter, speech denoising speech enhancement .

I.Introduction :

Fixed coefficient filters have fixed internal parameters and structure. To work properly, some filters may require matching of statistical characteristics of input signal with prior information along with that filter. These types of conditions may not meet all the time. Filter may require changing its coefficient according to input conditions or it may not have prior knowledge about incoming input. So filter must adjust its characteristics for unknown conditions with the help of some predefined algorithms, to make changes as per requirement any time. It may lead to increase in hardware as well as software cost design. Hence in such cases adaptive filters are useful.

Adaptive Filter:

Therefore once the higher than rationalization we are able to outline adaptive filter as "The purpose of the final adaptive system is to filter the input in order that it resembles (in some sense) the required signal input" OR "An adaptive FIR or IIR filter styles itself supported the characteristics of the input to the filter and an indication that represent the required behavior of the filter on its input." Adaptive filters area unit a category of filters that iteratively alter their parameters. The filter minimizes the error between some desired signal and a few reference signal. A results of the method may be a set of N faucet values that outline the character of the input being filtered.Now, as an example we would like to see the character of some channel, say an area during which an indication is being created and received to a electro-acoustic transducer.

To do this, it looks that we might ought to have a way of supply a famous signal and scrutiny it to a similar signal as received into the mic. The LMS filter ought to, give a collection of faucets outline the inverse of the space. once filtering out noise alone (which is another task altogether), the faucet weights may then be enforced because the corresponding filter, inversed and applied to the received signal therefore manufacturing (at least theoretically) the initial signal in its original type. – faucets area unit already inversed! once we feed.Associate in Nursing adaptive filter a coaching sequence, it permits it to adapt in order that the filtered signal is as shut as doable to the reference signal.

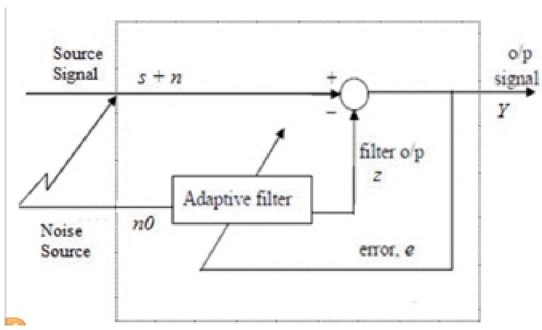

Fig. 1: Basic adaptive noise cancelled

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Adaptive noise canceller receives two inputs namely primary input and reference input. Primary input is combination of source signal and noise signal uncorrelated with each other. While the reference input is another noise signal , correlated in some extent with noise only shown in fig 1.

II.Related work :

Denoising or enhancement of speech signals plays a very important role in speech recognition and communication. Research has been conducted in force from 1970's and a vast number of techniques and algorithms have been proposed. The Kalman and extended Kalman filter based algorithms, technique using spectral subtraction and its modification are the most well known enhancement methods, see [1]– [3] and references therein. Recently, techniques using two or more microphones (mic array) have also been developed, see, e.g. [4], [5].

However, if the speech signal is corrupted by a harsh noise, all the above-mentioned techniques will lose its denoising power, leading to poor speech recognition rate and communication quality. On the other hand, in many real-life applications, speech recognition based handsfree electronic devices have to be operated in very noisy environments, and their performance degenerates considerably such that the lowest quality required cannot be ensured. That is one of the many reasons why more sophisticated and powerful techniques and systems are being pursued extensively for speech enhancement in the speech signal processing community.

III.Implementation:

LMS adaptive block diagram.

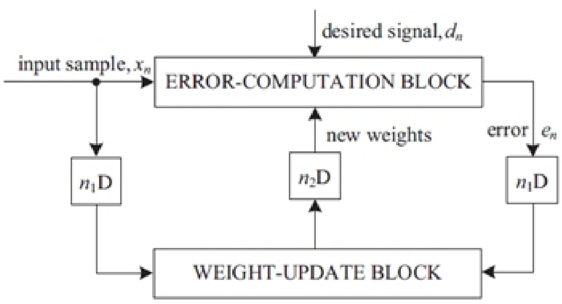

Fig 2 block diagram of implementation

Error computational block is the noise free output generator .the wieights or coefficients for the noise removal are calculated by the weight update block .nd are the delay blocks where input sample and error sample has the same delay.

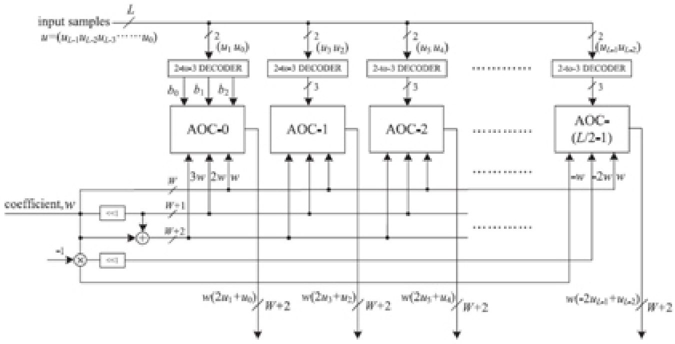

Fig 3 PPG cell

In the above block decoder is to decide the coefficient to be poped out with respect to the input given .the decoder output is tabulated in the below table. And - or cell is to to carry forward the selected cofficient to the error computational block. The coefficient is multiplied with multiples of 2 and 3 .as the input of 2 bit is considered a maximum of 3 multiplied coefficient is to be given to the and-or cell

U0	U1	B0	B1	B2
0	0	0	0	0
0	1	0	0	1
1	0	0	1	0
1	1	1	0	0

Table : 2 by 3 decoder resultant

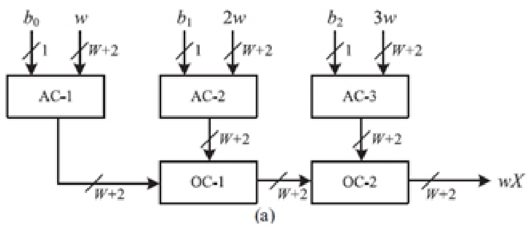

Fig 4And-or cell architecture design

As the 2 by 3 decoder truth table it is clear that any one of the output bit may be 1 .so considering different scenarios the truth table of the and-or cell design is tabulated.

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B0	B1	B2	W	2w	3w	Ac-1	Ac-2	Ac-3	Oc-1	Oc-2
0	0	0	0	0	0	0	0	0	0	0
0	0	1	1	2	3	0	0	3	0	3
0	1	0	3	6	9	0	6	0	6	6
1	0	0	4	8	12	4	0	0	4	4

Table: and-or truth table

Error computation block:

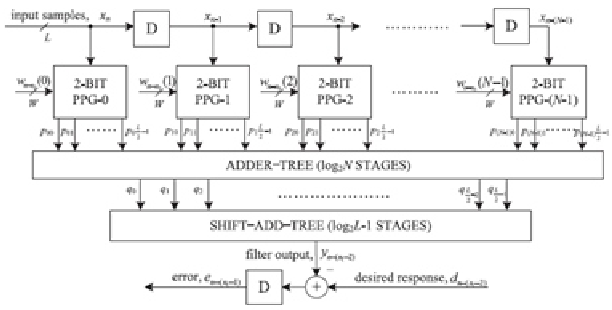

Fig 5 error computation architecture

As the equation describes the input begin multiplied with the weight co-efficient to make the implementation parallel and pipe lined the input bits are made into blocks of 2 bits being input to each PPG .the partial products generated are summed up using the adders called adder tree and shift adder tree .finally error is computed by subtracting the resultant output with the desired signal.

Weight update block:

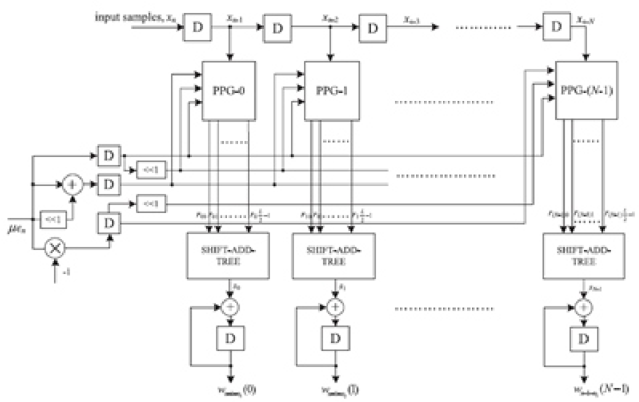

Fig 6 weight update block

Weight update block is used for the generation of coefficients by multiplying the input with the step size parameter according to the equation .the partial products generated are added with the shift adder tree .the previous weight are added to the present weights for maintaining the periodicity

IV.Results:

System generator schematic

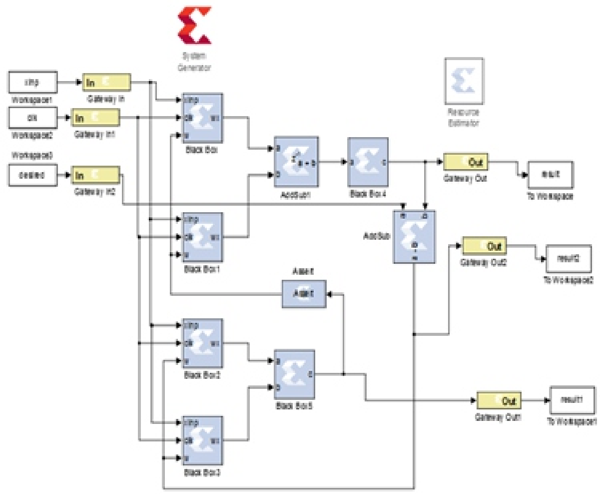

Fig 7 system generator model least mean square error

Waveform result:

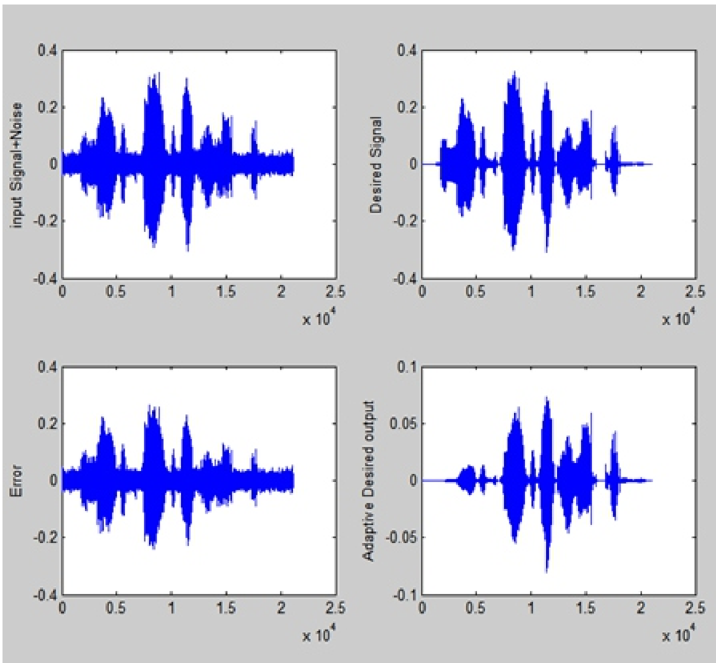

Fig 8 denoised speech signal

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V.Conclusion and future scope :

Area–delay-power efficient low adaptation delay architecture for fixed-point implementation of LMS adaptive filter. We used a novel PPG for efficient implementation of general multiplications and inner-product computation by common sub expression sharing.

Besides, we have proposed an efficient addition scheme for inner-product computation to reduce the adaptation delay significantly in order to achieve faster convergence performance and to reduce the critical path to support high input-sampling rates. Aside from this, we proposed a strategy for optimized balanced pipelining across the time-consuming blocks of the structure to reduce the adaptation delay and power consumption, as well.

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