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# Proportionate Type LMS Adaptive Filters For Echo Cancellation Applications

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**ABSTRACT:** The best ideal approach to refine the weights as new is to utilize Least Mean Square (LMS) algorithms. LMS adjust the weights of a small amount in the direction that reduces the error occurrences. Critical path actualization of LMS adaptive filter is represented to accomplish a lower adaptation delay. To accomplish lower adaptation delay pipelining enactment is required. But conventional LMS adaptive filter does not permit when it outstrip the desired sample period. Proportionate-type normalized LMS filter is proposed in a way to adjust the weights of a small amount in the direction that reduces the error and to implement low-cost equalizer for ASIC fulfillments. Proposed adaptive filtering calculations facing more challenges because of their high computational complexity particularly for real-time applications like network echo cancelation. This idea traced the condition under which the fastest overall convergence will be achieved and designed in two different approaches which yield comparatively better results over area utilized and the power consumed. Design consideration focus on variable bit size for accuracy reinforce and simulation results were shown improvement over existing design.

**KEYWORDS:** LMS (Least Mean Square), FIR (Finite Impulse Response), IIR (Infinite Impulse Response), RLS (Recursive Least Square).

## I. INTRODUCTION

An adaptive filter is a computational device that iteratively models the relationship between the input and output signals of a filter. An adaptive filter self-adjusts the filter coefficients according to an adaptive algorithm. Figure 1 shows the diagram of a typical adaptive filter.

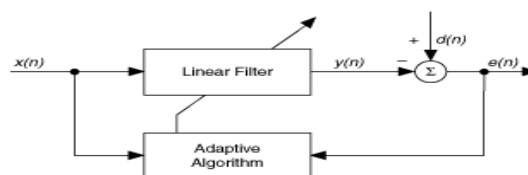


Fig. 1 TYPICAL ADAPTIVE FILTER

Where,

$x(n)$  is the input signal to a linear filter

$y(n)$  is the corresponding output signal

$d(n)$  is an additional input signal to the adaptive filter.

$e(n)$  is the error signal that denotes the difference between  $d(n)$  and  $y(n)$ .

The straight channel can be diverse channel types, for example, limited motivation reaction (FIR) or boundless drive reaction (IIR). A versatile calculation alters the coefficients of the straight channel iteratively to limit the intensity of  $e(n)$ .

The LMS calculation is a versatile calculation among others which changes the coefficients of FIR channels iteratively. Other versatile calculations incorporate the recursive least square (RLS) calculations.

*A. Convergence speed, Adoption rate, and Settling time*

Versatile channels upgrade the channel coefficients to limit the intensity of the blunder signal iteratively. The way toward limiting the intensity of the mistake signal is known as assembly. A quick assembly demonstrates that the versatile channel sets aside a short effort to figure the suitable channel coefficients that limit the intensity of the blunder signal. Settling time is the timeframe that versatile channels take to join. Littler settling time implies snappier union speed. Assembly speed is otherwise called selection rate.

*B. Steady State Error*

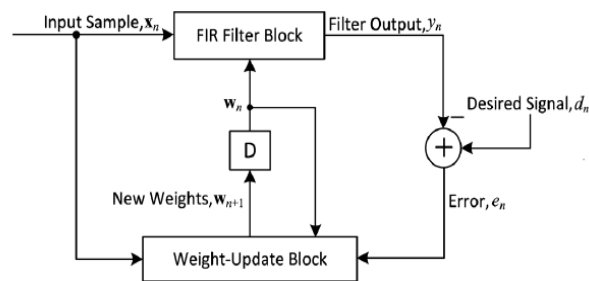
Steady state is the state when the versatile channel unites and the channel coefficients never again have noteworthy changes. Since signs may incorporate arbitrary commotion or in light of the fact that versatile channels are not ideal, the blunder signal yields of versatile channels are not really zero when the versatile channel meets. This blunder is known as the consistent state mistake.

*C. Minimum Mean Square Error, Excess Mean Square error, and Misadjustment*

Winner filters are ideal channels that limit the mistake signal. At the point when versatile channels merge to victor channels, the blunder signals are symmetrical and uncorrelated to the info signals. The mean square of the victor channels' mistake signal characterizes the base mean square blunder of versatile channels. In any case, not every single versatile channel unite to champ channels. The channel coefficients of versatile channels may change around the champ channel coefficients and never equivalent victor channel coefficients precisely. The channel coefficients vacillation acquaints overabundance blunder with the mistake signal. Overabundance mean square blunder is the contrast between the mean-square mistake presented by versatile channels and the base mean square blunder delivered by relating victor channels. Mis change is the proportion of the overabundance mean square blunder to the base mean square mistake.

*D. STABILITY*

FIR filters are stable inherently (fig 2.). Be that as it may, versatile FIR channels are not constantly steady. Since versatile calculations modify channel coefficients iteratively, the channel coefficients can get unbounded. At the point when channel coefficients become limitless, the versatile channel is unsteady. The LMS adaptive filter is famous because of its low-multifaceted nature, yet in addition because of its solidness and palatable intermingling execution. Because of its few significant utilizations of current pertinence and expanding requirements on region, time, and force multifaceted nature, proficient usage of the LMS versatile channel is still very significant.



Structure of conventional LMS adaptive filter.

Fig .2 STRUCTURE OF CONVENTIONAL LMS ADAPTIVE FILTER

To execute the LMS calculation, one needs to refresh the channel loads during each examining period utilizing the assessed mistake, which rises to the contrast between the present channel yield and the ideal reaction. Since all loads are refreshed simultaneously in each cycle, direct-structure acknowledgment of the FIR channel is a characteristic possibility for execution.

In any case, the immediate structure LMS versatile channel is regularly accepted to have a long basic way because of an inward item calculation to acquire the channel yield. This is essentially founded on the presumption that a number-crunching activity begins simply after the total information operand words are accessible/produced.

## II. ARCHITECTURE DESIGN OF PROPOSED FIR FILTERS

### A. FIR Filter

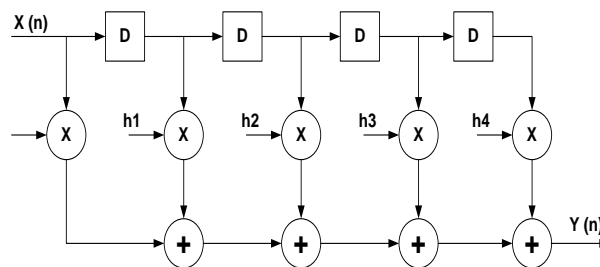


Fig.3 FIR Filter

In signal processing, a limited motivation reaction (FIR) channel (fig 3.) is a channel whose drive reaction (or reaction to any limited length input) is of limited term, since it settles to focus in limited time. This is as opposed to endless motivation reaction (IIR) channels, which may have inside criticism and may keep on reacting uncertainly (generally rotting).

The impulse response of a Nth-order discrete-time FIR filter lasts exactly  $N + 1$  samples (from first nonzero component through last nonzero component) before it at that point settles to zero. For a causal discrete-time FIR channel of request N, each estimation of the yield succession is a weighted entirety of the latest info esteems.

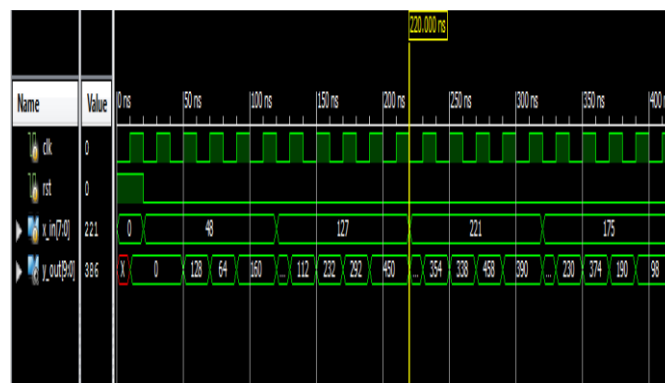


Fig.4 The output waveform of the FIR filter

The output window for FIR filter which was shows the output in the figure 4. It performs the hex conversion at desire time, area and delay.

## III. DESIGNING OF LMS ADAPTIVE FILTER

The best ideal approach to refine the weights as new is to utilize Least Mean Square (LMS) algorithms

- A. Two-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter and
- B. Zero-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter.

*A. Architecture for two-proportionate-delayed time-multiplexed lms adaptive-filter*

The proposed structure for a two – proportionate – postpone LMS versatile channel comprises of three pipeline stages, where the principal arrange finishes after the primary degree of the snake tree in the blunder calculation unit, and the remainder of the mistake calculation square contains the following pipeline organize. The weight-update square contains the third pipeline arrange. The two-adjustment defer structure includes extra registers over the one-adjustment postpone structure. The basic way of this structure is equivalent to either that of the weight-update unit or the subsequent pipeline organize. The time space LMS versatile channel calculation has been utilized in an assortment.

We find that in numerous well known applications like channel adjustment and direct estimation in remote correspondence, commotion scratch-off in discourse handling, and electrical cable impedance crossing out, evacuation of muscle antiques, and electro downgrade ancient rarities for ECG, the channel request could fluctuate from 5 to 100. Be that as it may, in certain applications like acoustic reverberation abrogation and seismic sign securing, the channel request necessity could be more than 1000. In this way, we examine here the effect of increment in channel request on basic way alongside the plan contemplations for execution of huge request channels for rapid applications.

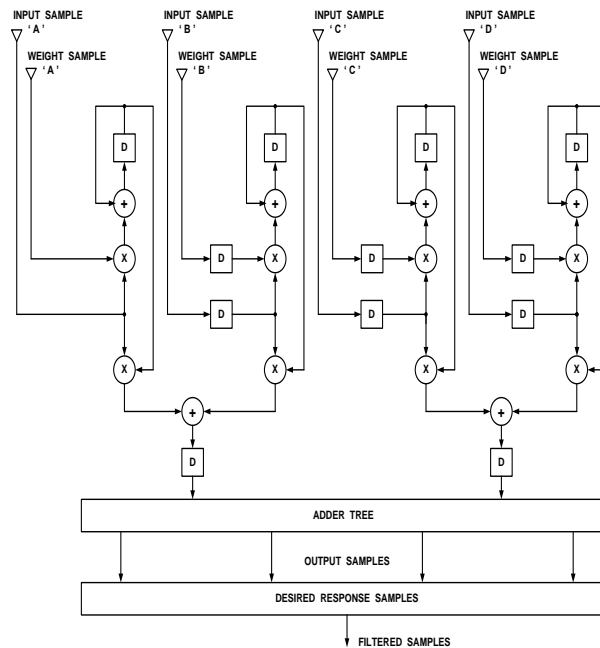


Fig.5 Architecture for Two-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter

*Pipeline Implementation*

For large-order filters, the basic way delay for 1-organize pipeline execution increments by when the channel request is multiplied. For 2-organize pipeline usage, could be bigger than , and could be the basic way deferral of the structure. Likewise increments by when the channel request is multiplied. In the point when 90-nm CMOS innovation is utilized, the basic way postponement could be about 5.97 ns and 3.66 ns for 1 and 2-organize pipeline usage, individually, In this way, so as to help input testing rates higher than 273 Msp, extra postponements could be consolidated at the last part of the snake tree utilizing just few registers. Note that if a pipeline arrange is presented just before the last degree of expansion in the snake tree, at that point just a single pipeline register is required.

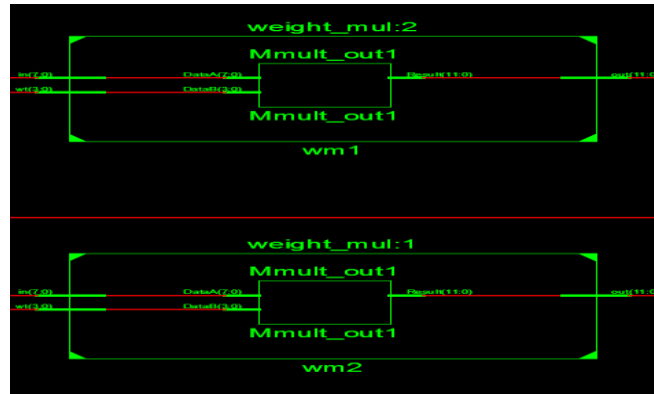


Fig.6 Pipeline stage

If we introduce the pipeline stage the pipeline organize at step up from the last viper in the snake tree, at that point we need extra registers. The deferral of the viper square anyway doesn't build quick with the channel request since the snake tree is possibly expanded one level when the channel length is multiplied, and presents just a single additional postponement of in the basic way. The basic way could be diminished just gradually on the off chance that we pipeline the versatile channel after each expansion, which will include huge register unpredictability.

Along these lines, pipelining of the increase square or snake tree after each expansion is definitely not a best choice to execute versatile channels for high-testing rate or for huge channel orders.

*B.Architecture for Zero-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter*

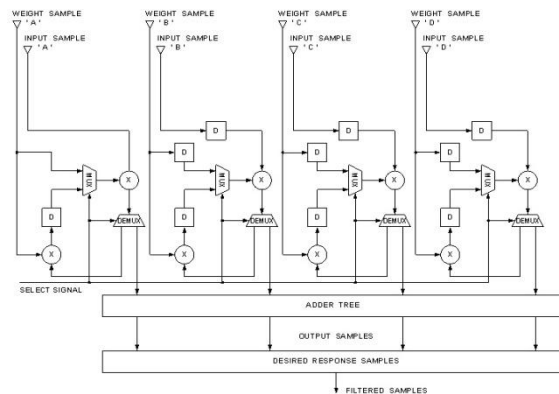


Fig.7 Architecture for Zero-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter

Error computation and weight update square are the two calculation obstructs in Direct-Form Zero Adaptation LMS channel appeared in Figure 7. In both the squares, zone expending parts are multipliers, weight enrolls and tapped postpone line. For zero adjustment, Error calculation square and weight update square calculation are acted in a similar clock period. As per the proposed direct structure (Non-pipelined), blunder calculation and weight update can't happen all the while and henceforth augmentation of both these stages could be multiplexed by a similar arrangement of multipliers and same registers could be utilized by methods for performing mistake calculation in first half cycle and weight update in the subsequent half cycle.

#### IV. EXPERIMENTAL RESULTS

Table I

Comparative analysis of conventional LMS filter, 2-Proportionate LMS filter and 0-Proportionate LMS filter.

| Parameters                               | Conventional LMS filter | 2-Proportionate LMS filter | 0-Proportionate LMS filter |
|--|-------------------------|----------------------------|----------------------------|
| Number of slice registers                | 745                     | 344                        | 278                        |
| Number of slice LUT                      | 824                     | 570                        | 547                        |
| Number of Bonded IOB's                   | 321                     | 182                        | 135                        |
| Minimum input arrival time before clock  | 13.253 ns               | 11.694 ns                  | 10.420 ns                  |
| Maximum output required time after clock | 4.91 ns                 | 3.634 ns                   | 3.634 ns                   |
| Maximum combinational path delay         | 17.647 ns               | 14.882 ns                  | 13.608 ns                  |

Table I represents the parameters of conventional type LMS filters. We evaluate the minimum arrival time, maximum combinational path delay of both filters. Table I is clear that 0-proportionate LMS filter shows a good performance over the 2-proportionate LMS filter in terms of combinational path delay, arrival time and number of slice. These comparison will be helpful for designers for choosing better parameters while design on the LMS filter.

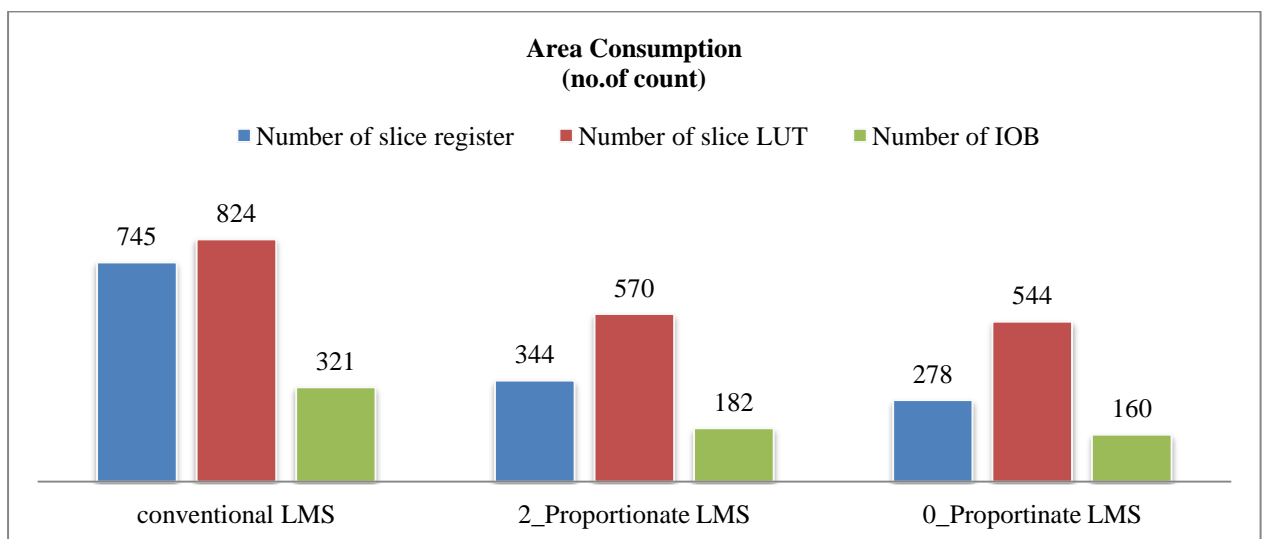


Fig. 7 Area consumption of conventional LMS, 2-proportionate LMS and 0-proportionate LMS filter

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Figure 7 shows the area consumption of conventional, 2-proportionate and 0-proportionate LMS filters. These results considered some parameters like number of slice register, number of slice LUT and number of slice IOB. From the Figure 7 observed that area for the 0-proportionate LMS filter is 37 % reduced when compared to conventional LMS filter and area for the 0-proportionate LMS filter is 46% reduced when compared to 2-proportionate LMS filter. The proposed 0-proportionate filtering technique is more precise for denoising of EEG signals.

| Device Utilization Summary (estimated values) |      |           |             |
|---|------|-----------|-------------|
| Logic Utilization                             | Used | Available | Utilization |
| Number of Slice Registers                     | 344  | 4800      | 7%          |
| Number of Slice LUTs                          | 570  | 2400      | 23%         |
| Number of fully used LUT-FF pairs             | 230  | 684       | 33%         |
| Number of bonded IOBs                         | 182  | 102       | 178%        |
| Number of BUFG/BUFGCTRLs                      | 1    | 16        | 6%          |
| Number of DSP48A1s                            | 4    | 8         | 50%         |

Fig.8 The Device utilization report of the proposed two proportionate type LMS adaptive filter

| Device Utilization Summary (estimated values) |      |           |             |
|---|------|-----------|-------------|
| Logic Utilization                             | Used | Available | Utilization |
| Number of Slice Registers                     | 278  | 4800      | 5%          |
| Number of Slice LUTs                          | 547  | 2400      | 22%         |
| Number of fully used LUT-FF pairs             | 222  | 603       | 36%         |
| Number of bonded IOBs                         | 135  | 102       | 132%        |
| Number of BUFG/BUFGCTRLs                      | 1    | 16        | 6%          |

Fig.9 The Device utilization report of the proposed two proportionate type LMS adaptive filter

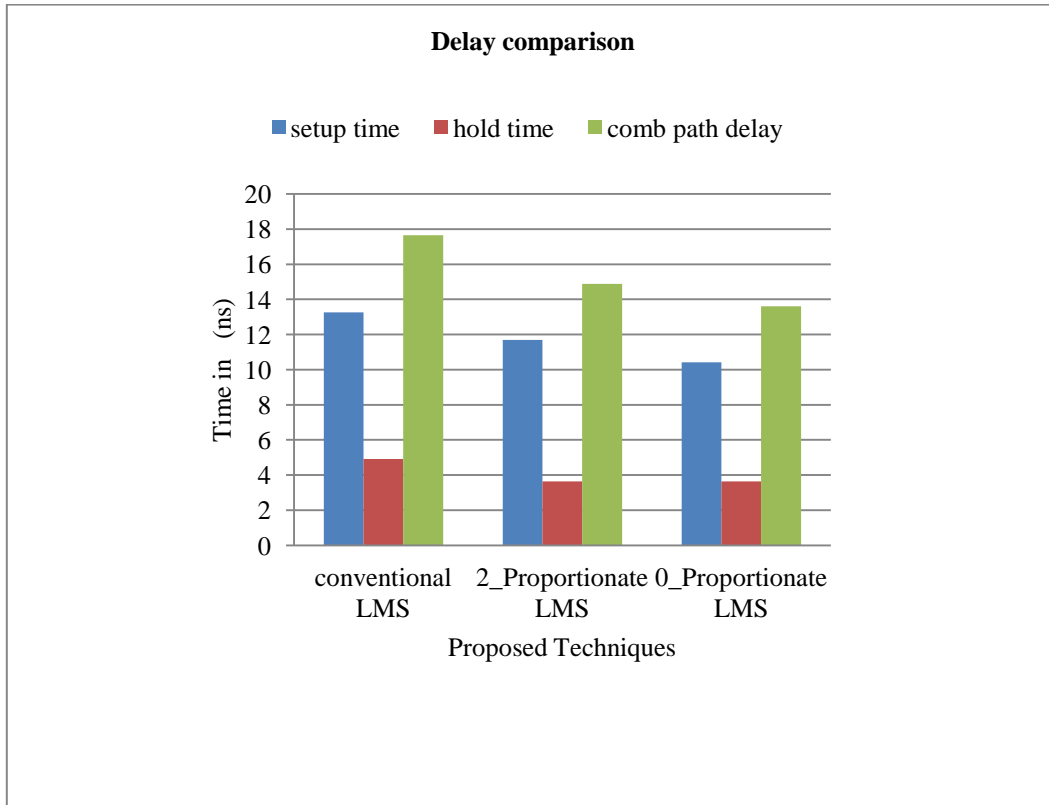


Fig.10 Delay comparison

Timing Summary:

Speed Grade: -3

```
Minimum period: 3.450ns (Maximum Frequency: 289.859MHz)
Minimum input arrival time before clock: 11.694ns
Maximum output required time after clock: 3.634ns
Maximum combinational path delay: 14.882ns
```

Fig.11 Timing report of the proposed two proportionate type LMS adaptive filter

Timing Summary:

Speed Grade: -3

```
Minimum period: 3.899ns (Maximum Frequency: 256.479MHz)
Minimum input arrival time before clock: 10.420ns
Maximum output required time after clock: 3.634ns
Maximum combinational path delay: 13.608ns
```

Fig.12 Timing report of the proposed zero proportionate type LMS adaptive filter

Figure 10 shows delay comparison of LMS filters. Setup time, hold time and combinational path delay was calculated for the proposed algorithms. 0-proportionate LMS filter has very less time and less delay when compared to other algorithms.



## V. CONCLUSION

We designed proportionate type LMS filters to reduce error occurrences in ASIC implementations and also to achieve lower adaptation delay. The Two-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter has been designed and the design analysis has been done. Zero-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filter has to be designed. A performance analysis has been done on the fir using Zero and two-Proportionate-Delayed Time-Multiplexed LMS Adaptive-Filters. Zero PDTMLMS Adaptive filter states comparatively better results over area, delay. Design consideration focus on variable bit size for accuracy and simulation results were shown improvement over existing system. This is the first attempt to implement the proportionate-type algorithms in hardware and this result will motivate other researchers to explore more efficient hardware solutions to further improve the sparse adaptive filter architectures.

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