

Low Computational Complexity and High Computational Speed in Leading DCD ERLS Algorithm

Fatemeh Abdi¹, Parviz Amiri^{1,*}, and Mohammad Hossein Refan¹

¹Faculty of Electrical Engineering, Shahid Rajaei Teacher Training University, Tehran, Iran.

*Corresponding Author's Information: pamiri@sru.ac.ir

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ABSTRACT

Adaptive algorithm adjusts the system coefficients based on the measured data. This paper presents a dichotomous coordinate descent method to reduce the computational complexity and to improve the tracking ability based on the variable forgetting factor. Vedic mathematics is used to implement the multiplier and the divider operations in the VFF equations. The proposed method decreases the area and increases the computation speed. The linear exponentially weighted recursive least squares as the main algorithm is implemented in many applications such as the adaptive controller, the system identification, active noise cancellation techniques, and etc. The DCD method calculates the inverse matrix in the ERLS algorithm and decreases the resources used in the field-programmable gate array, also the designer can use the cheaper FPGA board to implement the adaptive algorithm because the method doesn't need lots of resources. The proposed method leads to implementing complex algorithms with simple structures and high technology. The proposed method is implemented with ISE software on the Spartan 6 Xilinx board. The proposed algorithm calculates the multiplication result with less than 15ns time and reduces the used FPGA resources to lower than 20% as compared with the classic RLS.

1. INTRODUCTION

A linear least-squares (LS) is one of the popular methods in digital signal processing (DSP) applications. The LS applications include adaptive antenna arrays [1], active noise cancellation [2], multiuser detection [3], system identification [4], [5], micro-physics effect, particle accelerator [6] and the controllers of power converter [7]. The system parameters and ambient conditions will be change during signal processing, and the adaptive algorithm adapts the system coefficients based on the changes which means a compensation on the system output. Adaptive algorithms have a wide range of applications at industries such as control, communications [8], radar and sonar signal processing [9], interference cancellation [10], and biomedical engineering [11]. The main specification of application is a filtering

process that leads to creating a matching between the input signal and the desired response. The filter coefficients are updated based on the measured data in the system and they are applied as an input signal to the adaptive algorithm. By this way, the difference between the filter output and the desired response will be minimized in an either statistical or deterministic sense.

Least mean squares (LMS) and recursive least squares (RLS) algorithms are common algorithms which adjust and measure the system parameters according to the changes which can happen in the system. Important features of the LMS algorithm are simplicity and robustness in digital signal processing.

The RLS filter as a powerful algorithm can be implemented in applications such as the adaptive filters, prediction algorithm, controller, and system

identification. Despite the convergence rate of the RLS algorithm is faster than the LMS algorithm, the computational complexity of the RLS algorithm is higher than the LMS algorithm because the RLS algorithm must calculate an inverse matrix [12], [13].

The new methods provide an optimal performance in solving the system equations. But some methods include complicated operations, so designers try to suggest different sub-optimal algorithms. There are methods such as the direct and iteration method which calculate the inverse matrix. These methods include complex operations such as multiplication, division, and square-rooting, which lead to high computational complexity and hence, the designer needs expensive hardware resources [14].

Using the direct methods, such as Gaussian elimination, Cholesky decomposition and QRD, one can obtain a high accuracy in the solutions. The main idea of the direct method is to reduce the system equations to an upper triangular or a lower triangular form. But these methods have complex operations such as multiplication, division, and square-rooting. These operations lead to high computational complexity and therefore, the system may need expensive hardware resources [15]. The iterative methods are suitable for a large or sparse system. These methods provide higher convergence rate and optimal performance and have some advantages compared with the direct methods. As they need less memory, they are faster and provide simpler solutions for complex structures [16].

There are two types of iterative methods: non-stationary iterative methods, such as the steepest descent and the CG and the stationary iterative methods, such as the Jacobi, Gauss-Seidel and coordinated descent (CD) algorithms. The stationary iterative methods have less computational complexity than non-stationary iterative methods [17].

The iterative methods provide higher convergence rate than the direct methods, also, they need less memory and simpler design than direct methods [18].

The dichotomous coordinate descent (DCD) algorithm is designed according to the CD iterative method. It requires no multiplication, division or square rooting operations, just uses additions and subtractions. Therefore, the hardware implementation of the DCD algorithm is optimal in real-time applications [16].

2. HARDWARE DESCRIPTION OF THE ERLS ALGORITHM

The RLS classic algorithm requires about $4N^2$ multiplications and $3N^2$ additions/subtractions. An ERLS algorithm can identify the system parameters in dynamic systems [1]. The ERLS algorithm is shown in Table 1, the numbers of the system coefficients are shown with the symbol N. Pm is the number of

multipliers, and Pa is the number of adders in the DCD algorithm which are employed to calculate the inverse matrix. The algorithm consists of an $N \times N$ symmetric positive matrix (R), and $N \times 1$ two vectors (\hat{h} and β_0). The R matrix is defined as the correlation matrix of the reference signal X, and the β_0 vector is the cross-correlation vector between the reference signal and the desired response [19]-[22].

An adaptive algorithm should estimate and identify an optimal vector ($\hat{h}(i)$) that leads to the error signal goes to zero and the zero value is the optimal value in the error signal [8] a regularization matrix is defined with Π symbol and a forgetting factor can be selected between zero and unity $0 < \lambda < 1$.

TABLE 1
ERLS ALGORITHM [19]

Step	Equation	×	+
	Initialization: $\hat{h}(0)=0, r(0)=0, R(0)=\Pi$		
	For $i=1, 2 \dots$		
1	$R(i) = \lambda R(i-1) + X(i)X^T(i)$	$N(N+1)/2$	$N(N+1)$
2	$y(i) = X^T(i)\hat{h}(i-1)$	N	$N-1$
3	$e(i) = d(i) - y(i)$	0	1
4	$\beta_0(i) = \lambda r(i-1) + e(i)X(i)$	N	2N
5	$R(i)\Delta h(i) = \beta_0(i) \rightarrow \Delta \hat{h}(i), r(i)$	Pm	Pa
6	$\hat{h}(i) = \hat{h}(i-1) + \Delta \hat{h}(i)$	0	N
Total	Multiplies = $(N^2+5N)/2+Pm$; Adds $\leq N^2+4N+Pa$		

The regularization matrix is a diagonal matrix $\Pi = \eta I$, a small positive number is assigned as the regularization parameter $\eta > 0$ and I is the $N \times N$ identity matrix [1], [8].

$$X(i)\hat{h}(i) = \beta(i) \quad (1)$$

The R matrix is an auto-correlation matrix with size $N \times N$, and the β vector is the cross-correlation vector with length N. The coefficient vector $h(i)$ can be calculated according to the normal Equation (1), and the algorithm identifies and estimates the upper triangle part in $R(i)$ and this method leads to reducing the computational complexity. The forgetting factor has effects on calculating the R and β elements.

The proposed method has been implemented to calculate the linear equations in the system based on

according to step 5 (Table 2). The β Updater updates the $r(i)$ residual vector. The “leading” component will be computed with the Max finder block according to step 1 in Table 2.

The C block computes the C value according to the third and fourth steps in Table 2. The Finite State Machine (FSM DCD) leads to pipeline-based running all of the blocks. The Max finder block and the r block will execute all steps synchronously [21]. The C block read the $r_n(i)$ values from the β RAM memory and the $R_{n,n}(i)$ value from the R RAM memory. The structure of the r Updater and the \hat{h} updater blocks are shown in Figs. 3 and 4 [24], [25].

Fig. 4 calculates the new value of the r vector based on step 5 in Table 2. This figure needs two addresses to read the $R_{:,n}$ value and the previous value of the r vector. Those addresses will be read through RAM reader block. After reading the vectors, the previous value of r vector should be as an input signal to the shifter block. This vector is multiplied in $2^{\Delta m}$, and the shifter block will generate the result without the multiplier operation. The next block will add or subtract new vectors according to the sign of (rn) value.

Calculator block C gets the index of $\arg \max_{p=1 \dots N} \{|r_p|\}$ from max finder block and computes the C value. The \hat{h}_n value is read from \hat{h} RAM memory based on the \hat{h} update block which computes the new value and writes it in the \hat{h} RAM. The residual vector r read from β RAM, is shifted, added and subtracted each element of R parameters. These steps are pipelined to achieve an effective time in updating.

TABLE 2
REAL-VALUED LEADING DCD ALGORITHM FOR SERIAL FPGA
IMPLEMENTATION [20]

State	Operation
0	Initialization: $\hat{h} = 0, r = \beta, m = Mb, k = 0, \Delta m = 0$
1	$n = \arg \max_{p=1, \dots, N} \{ r_p \}$
2	If $m = 0$, algorithm stops Else, $m = m-1, \alpha = 2^m, \Delta m = \Delta m+1$
3	$c = R_{n,n} - r_n 2^{\Delta m+1}$
4	If $c < 0$, go to state 5 Else, go to state 2
5	$\hat{h}_n = \hat{h}_n + \text{sign}(rn)\alpha$ $r = r \times 2^{\Delta m} - \text{sign}(rn) R_{:,n}$ $n = \arg \max_{p=1, \dots, N} \{ r_p \}$ $\Delta m = 0, k = k+1$ If $k = N_u$, algorithm stops; else, go to state 3

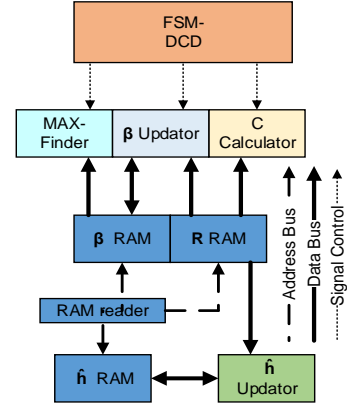


Figure 2: The DCD algorithm architecture.

4. IMPROVED VARIABLE FORGETTING FACTOR USING THE VEDIC MATHEMATICS

The proposed method is based on an improved Variable forgetting factor and DCD-Exponentially Weighted RLS algorithm (IVFF-Leading DCD-ERLS) and it is an optimal method in system identification and the controllers. The forgetting factor λ has an important role in the features of the LS algorithms such as convergence rate, tracking ability, and stability. If the λ value is near to one, the algorithm has proper stability and fast convergence rate while has low tracking ability. The improved IVFF-DCD ERLS algorithms have been developed to find the desired performance and a good tradeoff between parameters.

The variable forgetting factor is calculated according to the below equation:

$$\lambda_n = \min \left\{ \frac{\sigma_{qn} \sigma_{vn}}{\varepsilon + |\sigma_{en} - \sigma_{vn}|}, \lambda_{\max} \right\} \quad (12)$$

The λ_n value will be replaced by the λ value in Table 1, the ERLS algorithm will be executed based on the variable forgetting factor. The new structure will lead to speeding the tracking ability. Where $E\{e_n^2\} = \sigma_{en}^2$ is the power of the prior error, the power of the system noise is defined with σ_{vn}^2 , which is the PRBS signal that is the system noise in this application:

$$q_n = X_n^H R_{n-1}^{-1} X_n \text{ and } E\{q_n^2\} = \sigma_{eqn}^2$$

The $\varepsilon > 0$ value is a very small constant to prevent the denominator to become zero and λ_{\max} number is smaller than one. $E\{q_n^2\}$ denotes the statistical expectation operator [26]. The multiplier and divider are main operations in digital signal processing. Their parameters such as area and delay have important role in the design. Equation (12) needs the multiplier and the divider operations to calculate the optimal forgetting factor. For this reason, the Vedic multiplier (32-bits) is designed using the 16 bits Vedic multiplier and high-speed carry-save adder. The carry-save

adder is used to improve the delay and operation frequency [27].

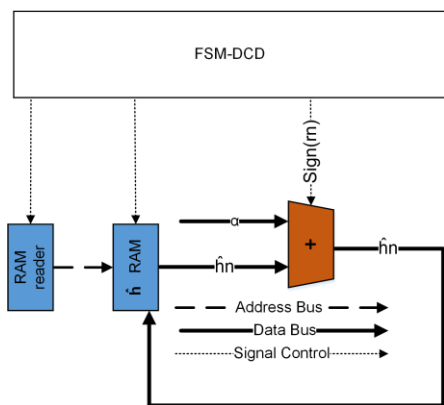


Figure 3: The \hat{h} updater block.

The Vedic multiplier leads to fast performance in the current processors. The proposed multiplier will start to calculate the result with small size from the input number (multiplicand size of (2×2)). Large bits $(N \times N)$ break into smaller bits $(N/2 = n)$. In this method, the input bits will be divided to $n/2$ and this method will be repeated until the number bits reach to 2×2 . The 24 bit or 30 bit Vedic multiplier using Urdhva-Tiryakabhyam method is implemented to multiply the mantissa part in the float point using the IEEE754 standard. The Vedic multiplier improves some parameters such as memory and area in comparing with conventional multiplier [28], [29]. When designers need more accuracy, they can increase the bits in mantissa or fractional numbers. Fig.5 shows the 32 bit Vedic multiplier.

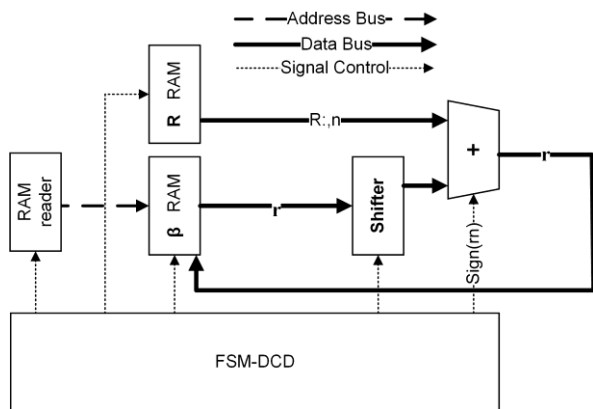


Figure 4: The r updater block.

The structure below shows how the Vedic multiplier operates:

- A2 A1 A0
- B2 B1 B0
- $M0 = A0B0$
- $K0M1 = A1B0 + A0B1$
- $K2K3M2 = A2B0 + A1B1 + A0B2 + K0$
- $K4K5M3 = A1B2 + A2B1 + K2 + K3$
- $M5M4 = A2B2$

This method needs logical gates and Vedic mathematics to calculate the M0-M5 intermediate signals. A2A1A0 and B2B1B0 are inputs and M0-M5 is the multiplication results. The Vedic multiplier with 3-bits is explained to design 16 bits Vedic multiplier in the figure below [25]. The proposed algorithm will identify and calculate the system coefficients and forgetting factor based on 1 and 4 steps in Table 1 and the variable forgetting factor traces the system changes better with the Vedic structure [30-32].

5. RESULTS

Here, the results are presented using the computer simulation in the ISE software. FSM DCD block generates the control signals and applies them to other blocks. This block has some input signals such as Done-C and stop-xXT. When Done-C is equal to 1, it means that the new value of C signal is calculated and the algorithm will be transferred to a new step. The En-Write signals will enable the RAM memories to write the new value according to the addresses which RAM-reader block generates them. When the algorithm calculates the xXT result, the stop-xXT signal goes to unity. The output signals are defined to enable the write-signal in RAMs and some blocks such as C calculator and MAX finder [33]. Fig. 6 shows the obtained result in the FSM DCD block. Fig. 7 shows the Max Finder result. Four numbers are defined in the residual vector and this block will execute step 1 in Table 2. Table 3 shows the used FPGA resources for MAX finder block. The Max finder will find the maximum value and its index between the values in the residual vector r .

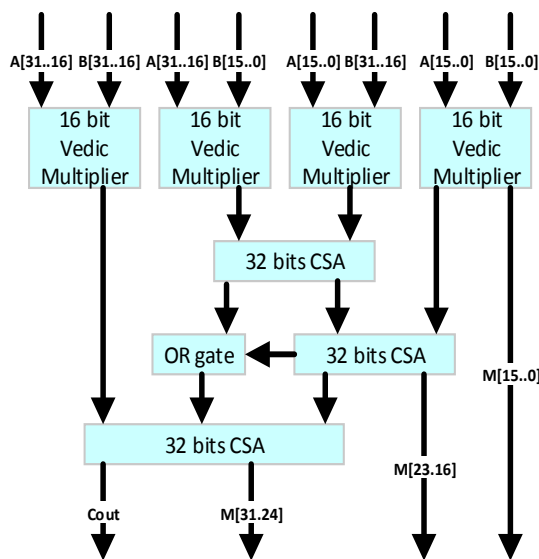


Figure 5: The 32 bits Vedic multiplier.

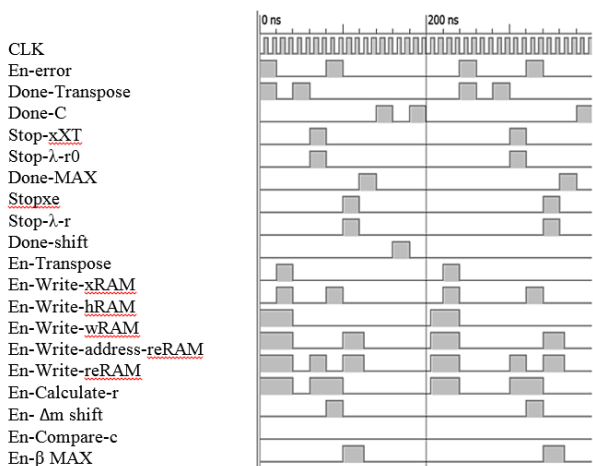


Figure 6: The FSM DCD result.

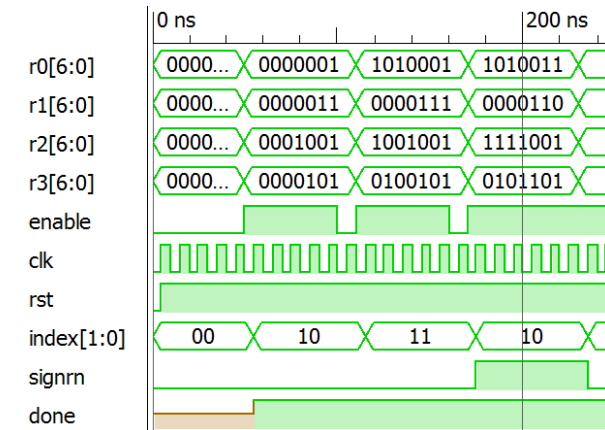


Figure 7: The result of Max finder block.

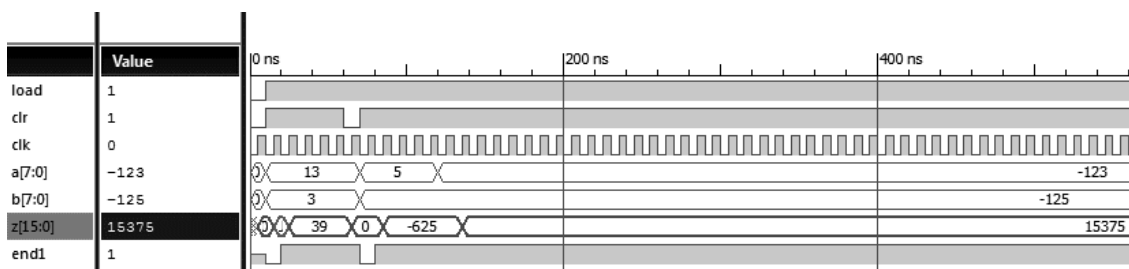


Figure 8: The result of Vedic multiplier.

TABLE 3
FPGA RESOURCES FOR MAX FINDER BLOCK

Resource	Utilization
Number of Slice Registers	1%
Number of Slice LUTs	1%
Number of BUFG/BUFGMUXs	6%
Number of OLOGIC2/OSERDES2s	5%

TABLE 4
USED FPGA RESOURCES AT Spartan 6

Algorithm	Utilization	
	IVFF-Leading DCD-ERLS	Classic RLS
Slice Register	15%	38%
Slice LUTs	54%	62%
Fully used LUT-FF pairs	18%	20%
BUFG/BUFGCT RLS	18%	38%
DSP48A1S	37%	52%

The index of the maximum value is used in the next step to calculate the coefficient and the residual vectors. The Max finder should find the maximum value without any attention to sign bit. In Fig. 7, the index is the location of the biggest value and the signrn is the sign of the biggest value.

Fig. 8 shows the result of the Vedic multiplier. A and b input values are two signed numbers, z number is the result of multiplier and end1 signal shows the time when the output is valid and stable. Clr signal is a clear signal and low active. When Clr signal goes to zero, the algorithm deletes the previous value. The multiplier ensures the output value at a high speed. The used resources in FPGA are shown in Table 4. The proposed algorithm has fast tracing ability and reduces the used resources in FPGA. Fully used LUT-FF (Look up table-Flip Flop) pairs mean that how well your design uses the slice components. We have also used the adjacent flip-flop within that slice for every LUT in the design. Normally the designs have some logics that only use the LUT pair. The proposed method (Improved IVFF leading -DCD-ERLS) includes more operations because it has faster convergence rate and good tracing ability [34].

The LUTs are organized in Slices which mean that those elements share connections in order to utilize fast carry chain. LUT is the truth table and this truth table effectively defines how your combinatorial logic behaves. DSP48A1S is the digital signal processing

unit in FPGA and includes the adder, multiplier, register, logic, and ALU. The proposed method needs FPGA with less resources, such as DSP48A1S, slice LUT and etc.

6. CONCLUSION

The leading DCD-ERLS algorithm is improved based on variable forgetting factor with Vedic mathematics to calculate the multiplier and divider operations in the relevant equations. This structure leads to good tracking ability, less delay and area, and faster convergence rate. The IVFF-Leading DCD ERLS algorithm calculates the inverse matrix according to the leading DCD algorithm and it does not need multiplier and divider operations. This structure just uses simple operations and reduces the used FPGA resources such as register less than 15% and the DSP48A1S more than 37%. The Vedic multiplier is done with a lower time about 15 ns. The proposed method reduces the cost of the final product because the producer can implement the proposed method with cheaper FPGA.

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BIOGRAPHIES



Systems, Power Amplifier, Voltage Converter and regulator.



research interest is in RF and power electronic circuits, with focus on high efficient and high linear power circuit design. He is currently with the Faculty of Electrical Engineering at Shahid Rajaei Teacher Training University (SRTTU), Tehran, Iran.



Engineering, Shahid Rajaei Teacher Training University (SRTTU), Tehran, Iran. He is the author of about 50 scientific publications on journals and international conferences. His research interests include GPS, DCS, and Automation System.

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