International Journal of Science and Research (IJSR)
ISSN (Online): 2319-7064
Index Copernicus Value (2013): 6.14 | Impact Factor (2015): 6.391

A Time Efficient Approach to Enhance the Noise Cancellation in Speech Signal on FPGA Using Adaptive Filtering Algorithm

Pravin Bhadauria[1], Rakesh Kumar[2]

[1]Gurgaon Institute of Technology & Management, MDU, Rohtak

[2]Assistant Professor, ECE, Gurgaon Institute of Technology & Management, MDU, Rohtak

Abstract: ***Modern Field programmable door clusters (FPGAs) incorporate the assets expected to outline productive sifting structures. Versatile filter is a self-learning filter where in the estimation of filter taps are changed relying upon the sign measurements. The sign got at the yield of the filter is contrasted and the fancied flag and given to a calculation piece which computes the new estimation of the taps in each cycle. The study keeps giving a reproduction of a particular issue of clamor cancelation in discourse signal, utilizing two stages as a part of two distinct situations. The first uses a white Gaussian as the clamor signal and the second uses a hued noise signal. The simulation results by both, MATLAB and Xilinx were acquired and analyzed. The execution on Xilinx was finished by utilizing altered point number juggling. Subsequently numerous adjusting and truncation mistakes must be made.***

Keywords: Adaptive filter, FPGA, Matlab, noise signal, noise cancellation, Xilinx

1. Introduction

Adaptive filters are filters with the capacity of adjustment to an obscure domain. This group of filters has been generally connected in view of its flexibility (equipped for working in an obscure framework) and minimal effort (equipment expense of usage, contrasted and the non-versatile filters, acting in the same framework). The capacity of working in an obscure domain added to the ability of following time varieties of info measurements makes the versatile filter an effective gadget for sign handling and control applications [1]. To be sure, versatile filters can be utilized as a part of various applications and they have been effectively used throughout the years.

As it was before specified, the uses of versatile filters are various. Hence, applications are isolated in four fundamental classes: ID, opposite demonstrating, forecast and obstruction wiping out. Every one of the applications aforementioned, have a typical trademark: an info sign is gotten for the versatile filter and contrasted and a craved reaction, producing anerror. That error is then used to alter the flexible coefficients of the filter, by and large called weight, so as to minimize the mistake and, in some ideal sense, to make thaterror being upgraded, at times tending to zero, and in another tending to a desired signal [2].

2. Active Noise Cancelling

The active noise cancellation (ANC), additionally called versatile clamor dropping or dynamic clamor canceller has a place with the obstruction crossing out class. The point of this calculation, as the point of any versatile filter, is to minimize the clamor obstruction or, in an ideal circumstance, drop that irritation [3]. The methodology embraced in the ANC calculation, is to attempt to mirror the first flag s(n). In this paper, the last goal is to utilize an ANC calculation to drop discourse clamor impedance, however this calculation can be utilized to manage some other sort of tainted sign. A plan of the ANC can be seen in fig.1, depicted below.

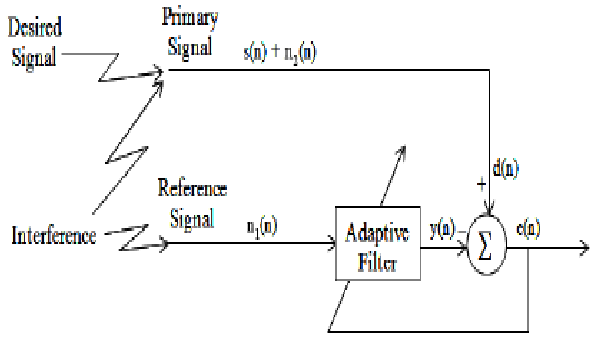


Figure 1: Active Noise Canceller

In the ACN, as clarified some time recently, the point is to minimize the noise interference1 that taints the first info signal. In the figure over, the fancied sign d(n) is created by an obscure sign, that we call s(n) tainted for an extra clamor n2(n), produced for the impedance. The versatile filter is then introduced in a spot that the main information is the impedance signal n1(n). The signs n1(n) and n2(n) are connected. The yield of the filter y(n) is contrasted and the fancied sign d(n), creating anerror e(n). That error, which is the framework yield, is utilized to change the variable weights of the versatile filter with a specific end goal to minimize the noise impedance. In an ideal circumstance, the yield of the framework e(n) is created by the sign s(n), free of the noise impedance n2(n) [5].

3. Background Studies

Proposed approach is based on following studies:

In [5], M. Stella et. al. proposed a straightforward neural system called Adaline as versatile filter. Investigation depended on motor noise cancelation in autos. It demonstrates that SNR enhances in the wake of passing the clamor cancelation framework. In the main investigation with low motor noise change was 7.12 dB, and in the second explores different avenues regarding high motor clamor we accomplished a 8.46 dB change. Lower change in the third test can be clarified by wind and street noise in a driving auto.

In [6], M. Grimy et. al. presents a versatile noise cancelation calculation based fluffy and neural system. The significant point of preference of the proposed framework is its simplicity of usage and quick merging. The propose calculation is connected to clamor wiping out issue of long separation correspondence filter. The reenactment result demonstrated that the proposed model is adequacy.

In [7], N. akhter et. al. concentrates upon the examination of versatile noise canceller utilizing Recursive Least Square (RLS), Fast Transversal Recursive Least Square (FTRLS) and Gradient Adaptive Lattice (GAL) calculations. The execution examination of the calculations is done in light of joining conduct, union time, connection coefficients and sign to clamor proportion. Subsequent to contrasting all the reproduced results we watched that GAL plays out the best in clamor cancelation as far as Correlation Coefficient, SNR and Convergence Time. FTRLS, RLS, and GAL were never assessed and looked at on their execution in noise cancelation as far as the criteria considered here.

In [8], G. Goplani and A Jafari proposed another sort of nonlinear versatile filter the versatile neural fluffy filter (ANFF), based upon a neural system's learning capacity and fluffy if-then manage structure, is proposed in this paper. The ANFF is inalienably a food forward multilayered connectionist system which can learn without anyone else's input by preparing information or master information spoke to by fluffy if-then guidelines. At that point adjustment here incorporates the development of fluffy if-then standards (structure learning), and the tuning of the free parameters of participation capacities (parameter learning).

In this new ANFF, we additionally made the learning and fluffiness parameters versatile. In parameter learning stage, a back engendering like adjustment calculation is produced to minimize the yield error. There are no concealed hubs at first, and both the structure winning and parameter learning are performed simultaneously as the adjustment continues.

In [8], D. Nicolae and R. Rormulus presents a versatile noise canceller will be displayed and some valuable perceptions will be done over the sound signs. The versatile clamor canceller is exceptionally effective and valuable framework in numerous applications with sound video and so forth.

In [9], Y. Lau et. al. demonstrated a helpful execution study between the time-fluctuating LMS (TV-LMS) and other two primary versatile methodologies: The Least Mean Square (LMS) calculation and the Recursive Least Square (RLS) calculation. Their study unveiled the calculation execution time, the base Mean Square Error (MSE) and required filter request.

In [10], H. Shin et. al. utilizes averaging investigation to consider the mean-square execution of versatile filters, regarding solidness conditions as well as far as expressions for the mean-square error and the mean-square deviation of the filter, and additionally as far as the transient execution of the comparing incompletely found the middle value of frameworks.

In [11], S. A. Hadei and M. Loftizad exhibited a decent tradeoff between union properties and computational many-sided quality and demonstrated that the union property of quick relative projection (FAP) versatile separating calculation is better than that of normal LMS, NLMS, and RLS calculation.

In [12], R.K. Thenua and S.K. Agarwal examines the execution of LMS and NLMS versatile calculations when actualized on Texas Instruments (TI) TMS320C6713 DSP equipment and tried for two sorts of signs; sinusoidal tone sign and ECG signal. The got results from DSP unit are broke down with the assistance of Digital Storage Oscilloscope (DSO) and indicate extensive change in SNR level of a separated sign.

In [13], Bernard Widrow et.al built up a model for clamor cancelation with the assistance of versatile filter and utilized for assortment of down to earth applications like the crossing out of different types of intermittent obstruction in electrocardiography, the dropping of occasional impedance in discourse signals, and the scratching off of wide band impedance in the side-flaps of a receiving wire exhibit.

4. Proposed Work

Proposed work involves following steps:

a) Adaptive filter implementation

Adaptable filter is characterized by four viewpoints:

1) The signs being prepared by the filter.
2) The structure that characterizes how the yield sign of the filter is processed from its information signal.
3) The parameters inside this structure can be iteratively changed to modify the filter's information yield relationship.
4) The versatile calculation that portrays how the parameters are balanced starting with one time moment then onto the next.

By picking a specific versatile filter structure, one determines the number and sort of parameters that can be balanced. The Filter can be a FIR or an IIR filter. The quantity of taps of the filter chooses the precision of preparing. Progressively the quantity of taps more will be the exactness. In any case, the deferral created likewise increments with the expansion in number of taps. The Weight Adaptation piece is the square of calculation used to change the weights of the filter. Distinctive calculations like LMS, RLMS, and NLMS and so forth can be utilized to change the filter weights [13].

b) LMS proposed algorithm

Least mean squares (LMS) calculations are a class of versatile filter used to impersonate a wanted filter by finding the filter coefficients that identify with creating the minimum mean squares of the mistake signal (distinction between the craved and the real flag). It is a stochastic inclination plummet strategy in that the filter is just adjusted in view of the error at the present time. In straightforward terms this calculation predicts the following tap esteem by adding some remedy to the present tap esteem. The redress is such that the new filter tap esteem merges to Wopt i.e. the MSE ought to lessen. The slope of MSE measures the closeness of tap quality to Wopt [14].

In the event that angle is certain then Wk >Wopt and something should be subtracted from Wk.

In the event that angle is negative then Wk <Wopt and something should be added to Ck so that Wk+1 approaches Copt.

$Wk(n+1)=Wk(n)-\mu\nabla J$
Where
∇ is gradient of MSE J
μ is step size
Wk(n) is tap value of kth tap and nth iteration

The progression size can be variable or steady. In LMS calculation, it is a steady positive number whose worth reaches from $0 < \mu < 2/Ymax$ Where Ymax is greatest eigen estimation of R. In the event that μ surpasses the cutoff then direction of Wk gets to be unstable.

$$Wk(n+1) = Wk(n) + \mu E[e(n)*x(n)]$$

This is the condition of steepest dive calculation. Presently LMS calculation assesses the angle as

$$Wk(n+1)= Wk(n) + \mu*en *xn$$

By this algorithm the calculation of next tap esteem gets to be less demanding furthermore equipment and space required is greatly decreased. The redesign for kth esteem requires just 1 augmentation and 1 expansion. Henceforth, for a filter of request P+1, P+1 multipliers and adders are required. One viper is required for discovering e(n) and one multiplier for μ*e(n). At long last P adders and P+1 multipliers are important to discover yield y(n). Subsequently aggregate of 2P+3 multipliers and 2P+2 adders are required.

5. Results

Results of our proposed technology will be like following below figures:

Results using Xilinx software:

Run the Xilinxsoftware and initialize the project.
Xilinx ISE 9.li improvement environment was utilized for usage of above VLSI outline. The configuration has been exchanged to VHDL code and it's the reenactment finished with the Xilinx ISE test system furthermore executed on Spartan-3 FPGA XC3s400pq208-5 board. The circuit used to make the sinusoidal signs is appeared in fig.2. It gives six results, where every result is a straight grade moving some spot around 0 and 360 degrees.

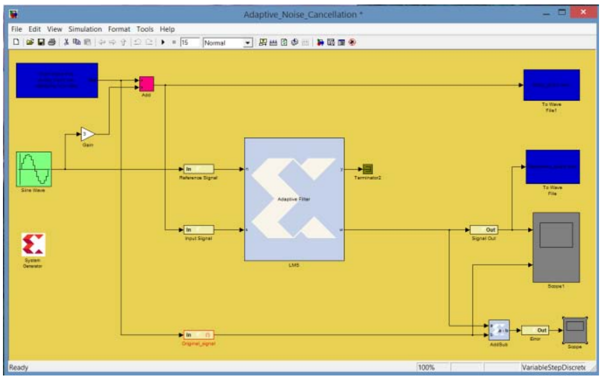


Figure 2: Proposed simulation system on Xilinx

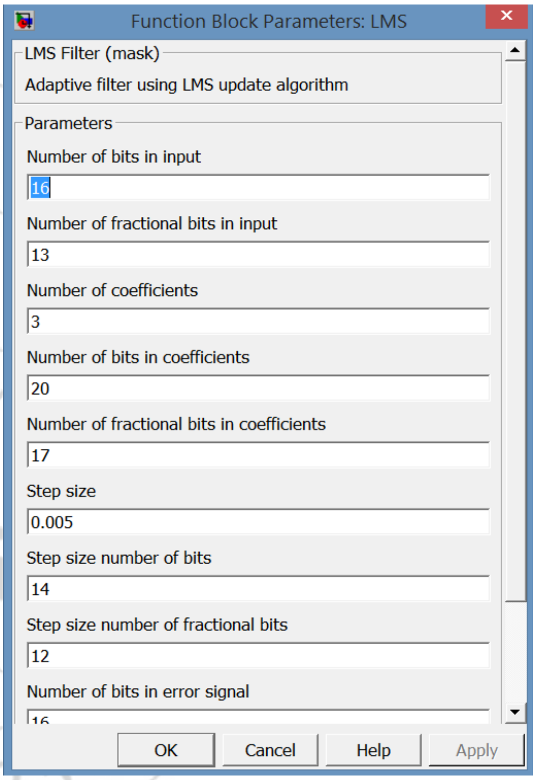
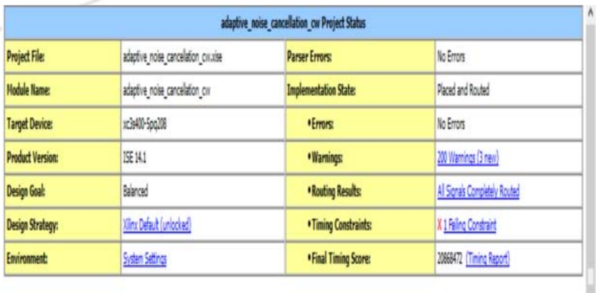


Figure 3: LMS block parameters



adaptive_noise_cancellation_cw Project Status			
Project File:	adaptive_noise_cancelation_cw.xise	Parser Errors:	No Errors
Module Name:	adaptive_noise_cancelation_cw	Implementation State:	Placed and Routed
Target Device:	xc3s400-5pq208	•Errors:	No Errors
Product Version:	ISE 14.1	•Warnings:	200 Warnings (3 new)
Design Goal:	Balanced	•Routing Results:	All Signals Completely Routed
Design Strategy:	Xilinx Default (unlocked)	•Timing Constraints:	X 1 Failing Constraint
Environment:	System Settings	•Final Timing Score:	20868472 (Timing Report)

Figure 4: Adaptive noise cancelation results

International Journal of Science and Research (IJSR)
ISSN (Online): 2319-7064
Index Copernicus Value (2013): 6.14 | Impact Factor (2015): 6.391

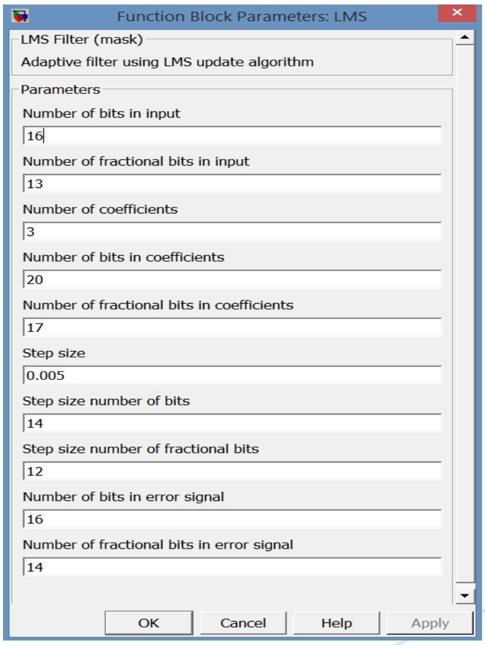


Figure 5: Function block parameters for LMS mask filtering

Device Utilization Summary				
Logic Utilization	**Used**	**Available**	**Utilization**	**Note(s)**
Number of Slice Flip Flops	92	7,168	1%	
Number of 4 input LUTs	396	7,168	5%	
Number of occupied Slices	233	3,584	6%	
Number of Slices containing only related logic	233	233	100%	
Number of Slices containing unrelated logic	0	233	0%	
Total Number of 4 input LUTs	411	7,168	5%	
Number used as logic	396			
Number used as a route-thru	15			
Number of bonded IOBs	83	141	58%	
Number of MULT18X18s	10	16	62%	
Number of BUFGMUXs	1	8	12%	
Average Fanout of Non-Clock Nets	2.05			

Performance Summary			
Final Timing Score:	20868472 (Setup: 20868472, Hold: 0, Component Switching Limit: 0)	**Pinout Data:**	Pinout Report

Figure 6: Performance summary using Xilinx for proposed work

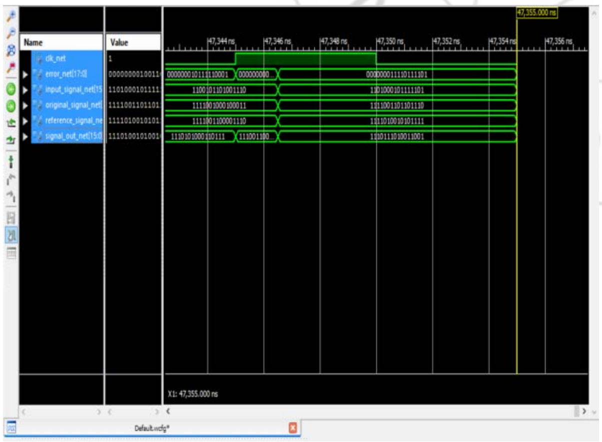


Figure 7: VHDL simulation system for LMS filter

Results using Matlab
Run the Matlab software and initialize the project.

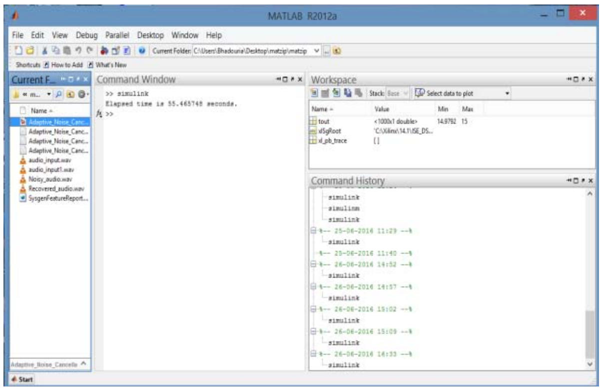


Figure 8: Proposed simulation Adaptive filter for noise cancellation

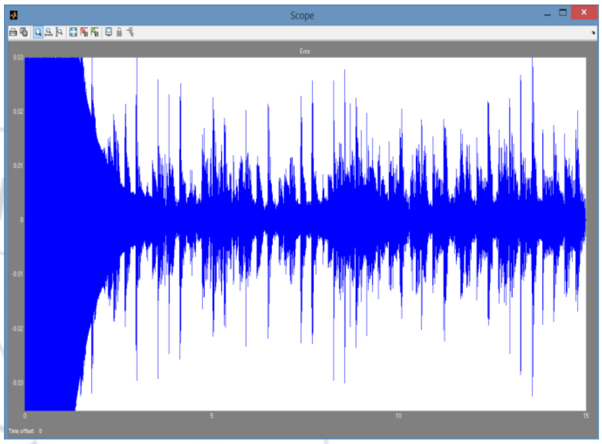


Figure 9: results of speech signals using proposed approach

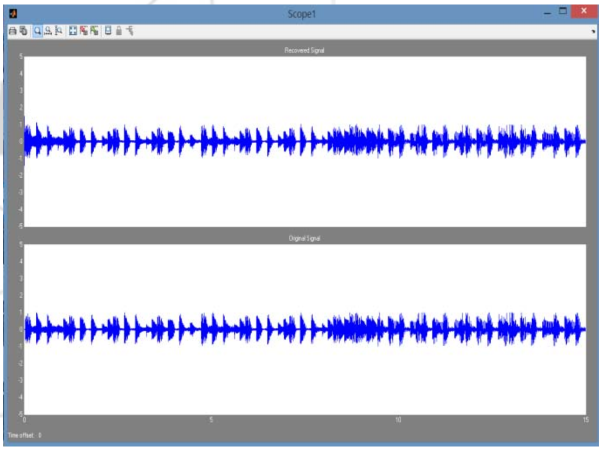


Figure 10: Results of original speech signals and processes i.e. noise free speech signals using proposed approach

6. Conclusions

The study demonstrates that the RLS calculation is more powerful than the LMS calculation, having a littler mistake and a speedier meeting for the instance of the white Gaussian clamor obstruction. For the shaded noise obstruction issue, the RLS has introduced a section from the past favorable circumstances, an intense steadiness, being fit for keeping its cancelation quality even with non-white variety in the clamor source. It is the inverse to the LMS calculation, which has demonstrated its wastefulness in such environment, having enormous varieties in the clamor cancelation mistake when the hued noise introduced a solid sign. Those error varieties are sufficiently enormous to be listened in the mistake yield signal.

The RLS calculation has a greater unpredictability and computational expense, yet relying upon the quality required for the ANC gadget, it is the best answer for be embraced.

References

[1] C. Mosquera, l.A. Gomez "Adaptive Filters for Active Noise Control" , Sixth international congress on sound and vibration Copenhagen, Denmark
[2] Colin H. Hansen "Understanding Active Noise Cancellation" lOS Press -2002
[3] A. B. Diggikar and S. S. Ardhapukar "*Design and implementation of Adaptive filtering algorithm for Noise Cancellation in speech signal on FPGA*" International Conference on Computing, Electronics and Electrical Technologies (ICCEET), 2012
[4] Simon Haykin "*Communication Systems"* Prentice Hall Information System Sciences Series, 4rd edition.
[5] Stella M., Begusic D. and Russo M., ‚Adaptive Noise Cancellation Based on Neural Network', IEEE 2006
[6] Miry M. H., Miry A. H. and Khleaf H. K, "Adaptive Noise Cancellation for speech Employing Fuzzy and Neural Network", 1st International Conference on Energy, Power and Control (EPC-IQ), College of Engineering, University of Basrah, Basrah, Iraq, November 30 - December 2, 2010.
[7] Ferdouse L., Akhter N., Nipa T. H and Jaigirdar F. T., ‚Simulation and Performance Analysis of Adaptive Filtering Algorithms in Noise Cancellation', IJCSI International Journal of Computer Science Issues, Vol. 8, Issue 1, January 2011.
[8] Golpayegani G. N. and Jafari A. H., ‚Improved Adaptive Neural Fuzzy Filter and Its Application In Noise Cancellation', 3rd International Conference on Bioinformatics and Biomedical Engineering , 2009.
[9] Nicolae D. and Romulus R., , Noise cancelling in audio signal with adaptive filter', ACTA Electrotehnica, volume 45 , November 6, 2006
[10] Lau Y. S., Hossain Z. M., and Harris R., "Performance of Adaptive Filtering Algorithms", Proceedings of the Australian Telecommunications, Networks and Applications Conference (ATNAC), Melbourne, 2003.
[11] Shin H. C., Sayed A. H., and Song W. J., "Mean Square Performance of Adaptive Filters using Averaging Theory", IEEE Signal Processing Letters, Vol. 6, pp. 106-108, May 1999.
[12] Hadei S. A. and Loftizad M., "A Family of Adaptive Filter Algorithms in Noise Cancellation for Speech Enhancement", International Journal of Computer and Electrical Engineering, Vol. 2, No. 2, pp. 307-315, April 2010.
[13] Thenua R. K. and Agrawal S. K., "Hardware Implementation of Adaptive Algorithms for Noise Cancellation", International Journal of Information and Electronics Engineering, Vol. 2, No. 2, March 2012.
[14] Widrow B., Glover J. R., J. Mccool M., Kaunitz J., Williams C. S., Hean R.H., Zeidler J. R., Dong E. Jr, and Goodlin R. C., "Adaptive Noise Cancelling: Principles and Applications," in Proc. of the IEEE, vol. 63 , no. 12 , pp. 1692 – 1716, 1975.