AN ENERGY EFFICIENT NOISE CANCELLATION CIRCUIT FOR EAR HEADPHONES

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Abstract**—***The adaptive filter algorithms are described as the least mean square method that makes successive corrections to the weight vector and leads to the minimum mean square error; the normalized Least Mean Square (LMS) algorithm, commonly used in applications due to its fast convergence and stability. Past strategies for mapping the least mean square finite impulse response filter onto parallel and pipelined architectures either introduce delays within the constant updates or have excessive hardware needs. LMS adaptive Finite Impulse Response (FIR) filter that produces identical output and error signals as would be created by the quality LMS adaptive filter design while not adaptation delays. Unlike existing architectures for delay less LMS adaptation, the new architecture's throughput is freelance of the filter length. The coefficient values are might to be stored in a shift register, which outputs are connected to multipliers and then to adders. Then registers are required for each and every weight coefficient. It has coded in Verilog, simulated using MODELSIM SE-64 10.1c, synthesized in Xilinx Spartan 3E trainer kit using Xilinx ISE 12.1. The Verilog code for the active noise cancellation filter uses B bits to represent the input, weight coefficients, and register. The lower section of the filter contains a MAC pipelined together with parallelism to increase the throughput, increase the speed with low power consumption*

Keywords—LMS filter, FIR filter, VLSI Design, Active Noise Cancellation

1. INTRODUCTION

Noise cancellation is a technique to reduce the unwanted noise from the outside of the surrounding things ex: plane, train, any sounds. To become with all headsets coming with some degrees .let it is talk about noise cancelling headphones which is usually meant active noise cancellation which electronically removes the surround.

Noise cancelling headphones are effectively rise sound by fit in the microphone in the headphone to listen to surround the noise then generated the waves that have to face which mean to opposite shape and feeds into ears. Generally, the two waves cancel out in the process calls destructive interference which sees the waves generates by the headphones. Effectively acting rise a sound riser results in hearing less of the outside world. The clever noise cancelling technique could make for happier reduce to the experience.

2.ACTIVE NOISE CANCELLATION OVERVIEW

A. Noise Cancellation works

All sounds are transmitted in waves described in terms of amplitude, wavelength and frequency. Fig.1 shows ANC removes unwanted sounds well preserving the original audio signals. If usually built directly into the headphones. A microphone picks up the ambient noise and transmit through the way of ANC there an identical 180 degree inverted waveform generated and send back to the headphones speakers. The result that the ambient noise wave and newly created waveform cancel each other. The red line has denoted the generated waveforms which is from the microphone. Another second green line has denoted which is external waves from any other out sources. Upcoming these cases have two types of controlling the noises. Maintaining the Integrity of the Specifications.

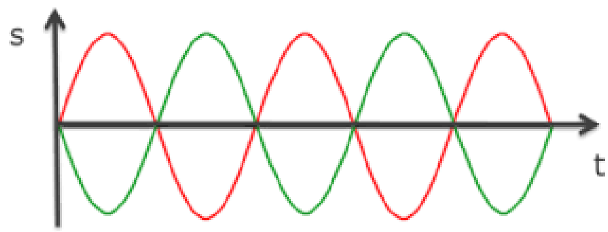

Fig.1: Active Noise Cancelling Technique

B. Active noise

- Required power
- Better at reducing lower frequencies

Need a power source to reducing the noise source and more effective to the lower frequency using more technologies implemented in the headphone

C. Passive noise:

- Not required power
- Better at reducing higher frequencies

It does not need a power supply to reducing the noise which is useful to reduce the higher frequency

3.NOISE CANCELLATION USING BUFFER METHOD

When the design factor of the detailed hardware architecture system using the values to optimise the circuit.to comparison in terms of the overall noise reduction and one by one different from the all computational complexity.so therefore, were using the adaptive filter design for the improvement and area efficient of all in-ear headphones

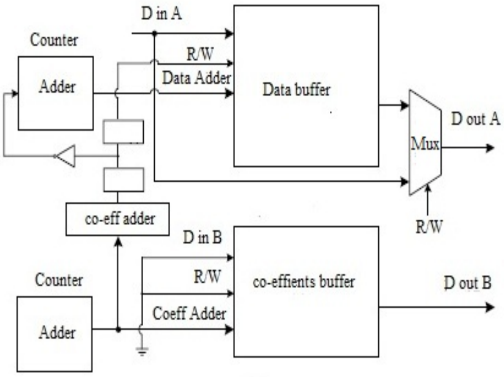

Fig.2: Hardware Architecture Buffer System

The above Fig.2 shows the structure of the ANC cancellation system .All the associated data can be stored in the buffer that has named as a data buffer.one sample values removed to the processing and another sample signals come and occupy the empty of the buffer space. This type of redundant processing where over the head for the entire system. We use the circular buffer means oldest value must be replaced by new one and rest values are the same position.

Hardware architecture not only for high computing as the same time used for better flexibility and produce the antinoise signal take over adaptively.

4.DESCRIPTION OF ACTIVE NOISE CANCELLATION IN PIPELINED SYSTEM

Past strategies for mapping the Least Mean Square (LMS) Finite Impulse Response (FIR) filter design parallel and pipelined. The architectures may either introduce delays within the constant updates or have excessive hardware needs.

we have a tendency to describe a hardware-efficient pipelined design for the LMS adaptive FIR filter that produces identical output and error signals as would be created by the quality LMS adaptive filter design while not adaptation delays. Unlike existing architectures for delayless LMS adaptation, the new architecture's throughput is freelance of the filter length.

The Fig.3 Designed Register Level Transistor (RTL) has shown the figure. The input values such as name x(n) are stored in shift register then which outputs are connected to increase values with multipliers and adding the total data to the adder. Weight coefficient needed the shift register to shifting it is values one position to another one.

The number of equal models which has divided by N modules called as a TAP 0, TAP 1, TAP N-1.where the entire whole module consists the TAP procedure and same partial elements such as multiplier, adder and shift register. The values are stored in an output register which is also needed to the process of the system. The propagation would increase the ripple and adder.

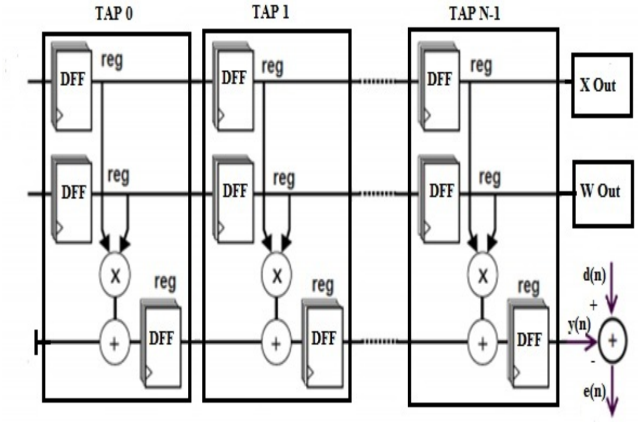

Fig.3: Proposed Design System

The adaptive filter has coded in Verilog, simulated using MODELSIM 6.5, synthesised in Xilinx Spartan 3E trainer kit using Xilinx ISE 13.1. The Verilog code for the active noise cancellation filter uses B bits to represent the input, weight coefficients, and register. During whereas 2 signals transmitted by using 2B bits. The section of the filter contains a MAC (Multiply Accumulate) pipelined together with parallelism to increase the throughput, increase the speed with low power consumption.

A.Synthesised Output with LUT and Flip-flop's count

To all the elements of the terms and the efficient elements how it is used on the problematic of this function and the design variables.

The following Table.1 which shows the number of lookup tables and the counting number of the flip-flop's where is used by the programming

TABLE I.SYNTHESIED OUTPUT WITH LUT AND FLIP-FLOP'S COUNTS

Logic Utilization	Used	Available	Utilization
Number of Slices	299	4656	6%
Number of Slices of Slice Flip Flops	288	9312	3%
Number of 4 input LUTs	550	9312	5%
Number of bonded 18IOs	67	158	42%
Number of MULT 18x 18IOs	1	24	4%

B.Output waveform in MODELSIM

To develop the signals with the ratio of the 1000 to 5000 ns in the design were using in the modelsim with the help of already designed function then get the output waveform in the normal way.If supposed to substitute the signal in the analogue form then only get the waveforms in the sine waveforms.Fig.4 shows the analog form of the waveforms.

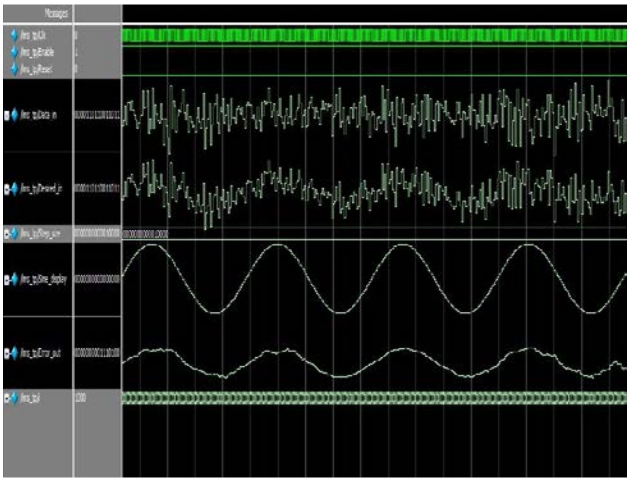
Fig.4: Simulated Output in MODELSIM

The upcoming signal values which are developed by the Xilinx navigator and implemented the function whether uses the implementation or uses the simulation process. The maximised values where is used the several inputs and a desired output which is called as a data out. These all values are stored in the data buffer that helps to generate the prescribed clock signal like as the timing diagrams are following by the below figure can be mentioned values. How could be generated by the procedure shown the below function.the folowing timing diagram Fig.5.

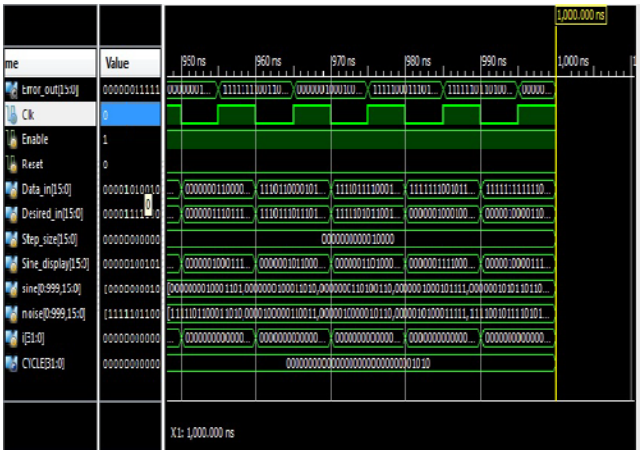
Fig.5:Timingdiagram for proposed buffers

B. Applications:

This technique can be used in the home applications and vehicles. Which is also helpful for reducing traffic noise cancellation. Many numbers of smart and great individual applications are invites to reduce or restrict the active noise cancellation techniques. It has requires, when sources need to cancel the sounds like as any loud speaker arrives at near area.

5. MODEL-BASED DESIGN OF AN ANC FILTER IN MATLAB

The ANC circuit sent it is model-based design used in the MATLAB coding technique. Where the digital signal processing the sine wave and the values are transmitted to the adder in front of placed to the digital filter design .Fig 6 shows the extra random source also connected to the digital filter design system which is use to Analyse both signals and reduce the noise and generated the mute or reduced original signal.

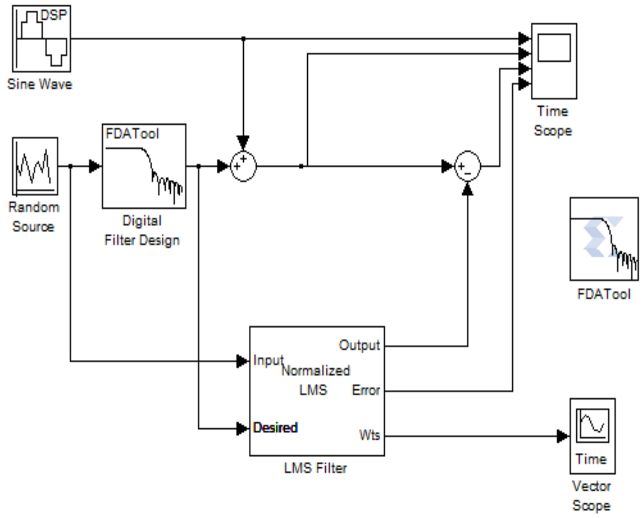

Fig.6: ANC in MATLAB Workout

LMS adaptive filter which is use to receive the signals from the random source and digital system design. Where is FDA tool optimise both the source emitted waves?LMS filter output,error,vectorscopes are to be connected with time scope normalized LMS filter uses this design.

So these are the total function which is use in the MATLAB process design developed by the ANC procedure.If the process required use to develop otherwise do not need to use this method for further development procedure.The performance of the noise cancellation circuits fully designed by the MATLAB procedure tools.However it has works depend upon the first noise came into the headsets microphone portal then the generated waves to be perfomed as well as in the opposite to reduce the overall noise and gives a combined or neutral waves to the listner.Designed circuitry will help in the future to be restrict all the noise which will be arise from any other environment.

A.MATLAB Worked Output

The MATLAB turns to simulation and implement results Fig.7 describes when design with the MATLAB coding use to show the noise cancellation techniques.

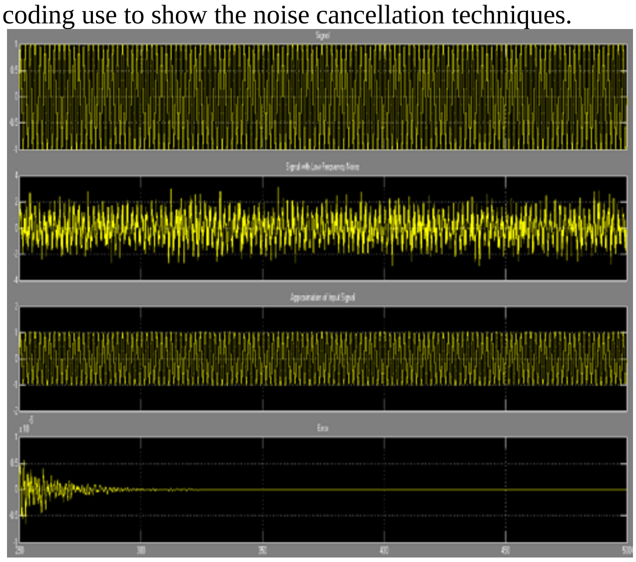
Fig.7: Waveform Analysis with MATLAB coding

6. GRAPHICAL REPRESENTATION OF RESULT ANALYSIS

The values are approximately calculated and it provides by the waveforms are generated in the Xilinx tool used. The following figures Fig.9&10 shows the representation. To generate the waveforms which help of the designed utilities that maximized results of the active noise cancellation .In order to eliminate noise sources input to the microphone signals are comes from the any sources to restrict the all noises to generate a noise source from the microphone which is place by in the headsets. The variable ranges of frequency responses and that's analysis are to be shows by the following bar chats.

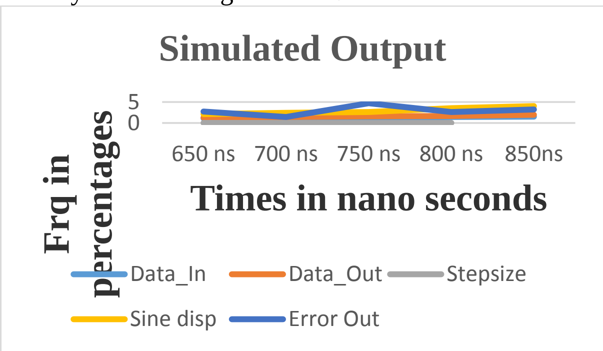

Fig.8: Simulated Analog output for Input Signals

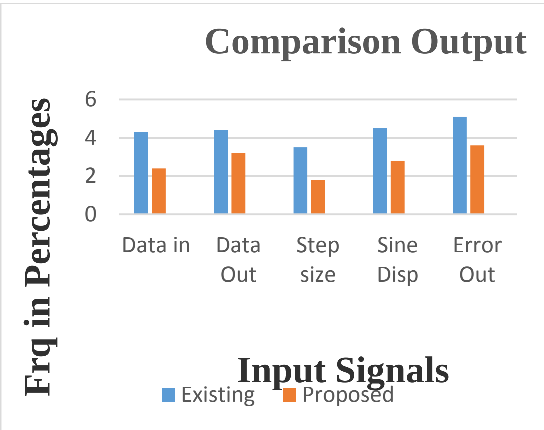

Fig.9: Comparisons of Existing & Proposed methods using bar chat

7. CONCLUSION

In this paper using the circuit which is called active noise cancellation designed and executed successfully. The result shown in the MODELSIM and verified an analogue waveform .to reduce the noise in the desired function have to hold that the proposed design systems. The design has coded in the Verilog and to be implemented in the XILINX 12.1 and MODELSIM-64-ss10.1c.it has more speed and less power consumption

8. FUTURE WORK

The experimental results show that the proposed high-performance/low-power ANC design circuit can reduce the unwanted and disturbing noises from outside are and the outputs perform well then the frequency bands also good. Proposed method achieves a good performance. Time delays should be controlled by the Pipelined architecture when the input gives to the system D Flip-Flops. The optimised output will be implementing in the earmuffs.

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BIOGRAPHY

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