

# VLSI Circuit Design for Noise Cancellation in Ear Headphones

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**ABSTRACT:** Noise cancellation ear headphones (NCEHs) supplement the acoustic isolation characteristic of headphones with active noise reduction. By their nature, headphones block out some degree of external noise because the ear cups absorb it, but NCEHs go a step further and diminish the noise that manages to get through. This report presents some of the methods used in implementing noise cancellation headphone. The principles of passive attenuation after which active attenuation which includes feed forward and feedback methods are described. Active noise control (ANC) is an efficient technique to deal with low frequency noise that is difficult to be abated by noise barrier or sound absorbing material. Many successful ANC systems have adopted the feed forward filtered-x least mean squares (FxLMS) algorithm to reduce machinery noise. The noise canceling headset is another well known example, where the feedback control structure is favorable due to the small size. However, the feedback control structure is incapable of reducing broad band noise. Therefore, this paper investigates the feasibility of implementing the feed forward FxLMS algorithm in the noise canceling headset.

**KEYWORDS:** Active noise cancellation, feedforward filtered-x least mean square (FxLMS), in-ear headphone, VLSI design

## I. INTRODUCTION

The goal of active noise cancellation (ANC) is to reduce the amplitude of the sound pressure level of the noise incident on the receiver or ear by "actively" introducing a secondary, out-of-phase acoustic field, "anti noise". The resulting destructive interference pattern reduces the unwanted sound. Over the past two decades, significant advances in control theory and the development of flexible, programmable, high-speed digital signal processing computers have made it possible to model and implement more complex active noise control systems. ANC is based on either feed forward control or feedback control. In feed forward control, a reference input coherent with the noise is sensed before it propagates past the secondary source. In feedback control, the active noise controller attempts to cancel the noise without the benefit of an "upstream" reference input. Structures for feed forward ANC are classified into (1) broadband adaptive feed forward control with a control field reference sensor, (2) narrowband adaptive feed forward control with a reference sensor that is not influenced by control field. Feed forward ANC is generally more robust than feedback ANC particularly when the feed forward system has a reference input isolated from the secondary anti noise source. When dealing with broad band noise, the feed forward control structure has been widely applied. The feed forward control structure requires a reference sensor to be placed near the noise source to provide time-advanced information of the noise wave to be canceled. The distance from the reference sensor to the secondary source generating the anti-noise wave is usually made long enough in order that real-time control is feasible. However, when the distance is short, a fast controller is essential to ensure the causality of the feed forward control structure.

Due to the compact size of the noise canceling headset, it is not feasible to put a reference sensor far from the secondary source. The current practice in the noise canceling headset is to apply the feedback control structure that is only efficient to reduce narrow band noise. As summarized in Table I, the general purpose digital signal processor (DSP) and microcontroller are adequate to carry out the feedback control structure.

## II. THEORY

### Proposed System

The proposed feed forward filtered-x least mean square ANC circuit design provides the features of using lower operating frequency and consuming much less power that facilitate better performance than the conventional ANC headphones.

To verify the effectiveness of the proposed design, a series of physical measurements is executed in an anechoic chamber. Measurement results show that the proposed high-performance/low-power circuit design can reduce disturbing noise of various frequency bands very well, and outperforms the existing works.

Item	Specifications
Process	TSMC 90nm CMOS technology
Chip size	1.053×1.058 mm <sup>2</sup>
Power supply (standard)	1.0 V for core circuit
Power consumption of the core only	0.52 mW at 20 MHz, 0.59 V
Total power consumption (including I/O power at 3.3 V)	6.59 mW at 20 MHz, 0.59 V
Required operating frequency	20 MHz
Maximum operating frequency	104 MHz
Gates	84.2 k gates
Throughput	149.25 k samples/s at 20 MHz

Fig: proposed system specification

### Filtered- X LMS algorithm

The use of adaptive filter is complicated by the fact that electrical reference signal must be obtained from the acoustic pressure using a microphone. Also, an electrical error signal must be obtained from the residual acoustic noise using as error microphone. Finally, the canceling sound must be produced from the electrical output signal using a 6 loud speaker. Therefore, a number of other transfer functions must be included. The summing junction represents acoustic superposition in the space from the canceling loudspeaker to the error microphone, where the primary noise is combined with the output of the adaptive filter. Therefore it is necessary to compensate for the secondary path transfer function  $S(z)$  from  $y(n)$  to  $e(n)$ , which includes the D/A converter, reconstruction filter, power amplifier error microphone etc.

Fig. 1 illustrates the block diagram of the FxLMS algorithm composed by a N-tap adaptive filter, a secondary path model  $\hat{S}(z)$ , and a LMS coefficient update mechanism, where  $z^{-1}$  denotes the unit delay block;  $w_i$  denotes the  $i$ -th coefficient of the adaptive filter;  $x(n)$  and  $y(n)$  are the input and output of the adaptive filter, respectively; and  $e(n)$  is the error signal. The critical path in Fig. 1 is recognized from the leftmost multiplier to the rightmost adder.

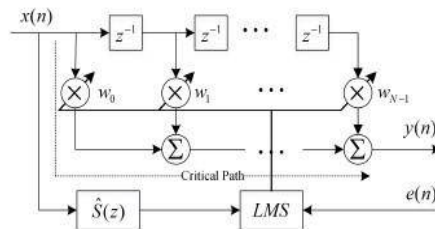


Fig. 1. Block diagram of the FxLMS algorithm.

Hence, the maximum sampling rate is calculated by

$$F_{max} = \frac{1}{t_{multiplier} + (N - 1)t_{adder}} \quad (1)$$

where  $t_{multiplier}$  and  $t_{adder}$  are lapsed time of a multiplier and an adder, respectively. When the tap length increases, the maximum sampling rate decreases. When  $N$  registers are inserted, the critical path in the adaptive filter can be shortened to one multiplier and one adder. The LMS algorithm implemented in such a way is known as the delay LMS algorithm when  $N$ -unit delay is used. It is a trivial problem to extend the delay LMS algorithm to include a secondary path model and become the delay FxLMS algorithm. The critical path of the delay FxLMS algorithm is twice that of the delay LMS algorithm, because the secondary path model introduces one multiplier and one adder. However, the delay LMS and delay FxLMS algorithms have small step size bounds that lead to slow convergence.

The compensation can be carried by placing an inverse filter in series with the loudspeaker or placing as identical filter in the reference signal path to the weight update of LMS algorithm. The placement of the secondary path transfer function following the digital filter  $W(z)$  controlled by the LMS algorithm.

The placement of the secondary path transfer function following the digital filter  $W(z)$  controlled by the LMS algorithm is shown in figure 2. The residual error is expressed as

$$\begin{aligned} e(n) &= d(n) - y'(n) \\ &= d(n) - s(n) * y(n) \\ &= d(n) - s(n) * [w T(n) x(n)] \end{aligned}$$

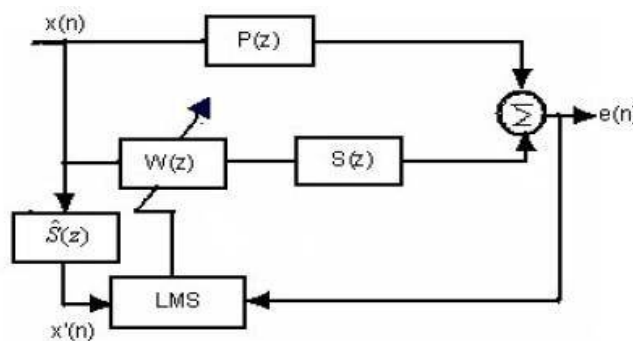


Fig 2: Block diagram of ANC system using the FxLMS algorithm

where  $s(n)$  is the impulse response of the secondary path  $S(z)$  at time  $n$ , \* denotes linear convolution,

$$w(n) = [w_0(n) \ w_1(n) \ \dots \ w_{L-1}(n)]^T$$

$$x(n) = [x(n) \ x(n-1) \ \dots \ x(n-L+1)]^T$$

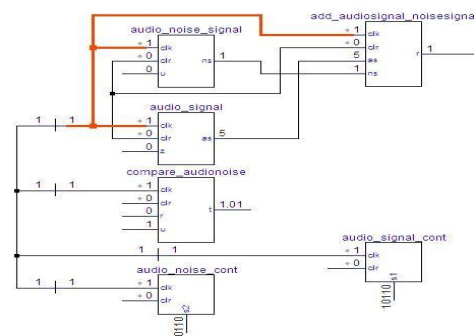
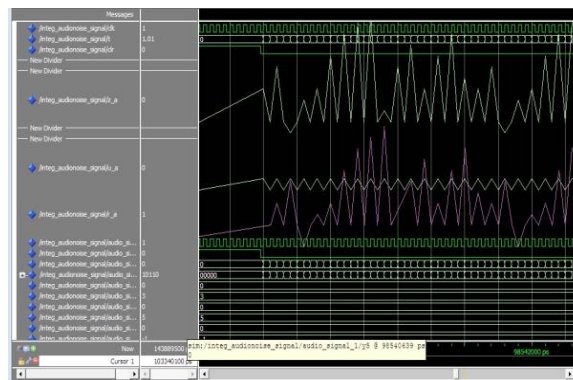
is the signal vector at time  $n$  and  $L$  is the order of the filter  $W(z)$  The objective of the adaptive filter is to minimize the instantaneous squared error,

$$\hat{\xi}(n) = e^2(n)$$

The most widely used method to achieve this is the stochastic gradient or LMS algorithm, which updates the coefficient in the negative gradient direction with step size  $\mu$

$$w(n+1) = w(n) - \frac{\mu}{2} \nabla \hat{\xi}(n)$$

**Results and discussions**



### Practical applications

Noise canceling headphones are particularly useful for workers, operating or working near heavy machinery and engines. The noise is selectively eliminated thus enabling the reception of the desired sounds, such as speech and warning signals. Cabin noise in small aircraft is a combination of noise from a variety of sources, the major ones being engine, wind, and propeller. So plane pilots routinely wear noise attenuating headsets to get rid of the unwanted noise. Such headsets usually employ passive noise attenuation in the form of an annular cushion carried on the rim of each ear cup. But NCH is proved to be worthy in attenuating noise to a further level which makes the pilot as well as passengers to hear warnings and instructions effectively.

### III.CONCLUSION

Active noise reducing headphone has been shown to provide a promising protection from noise using a combination of passive, analog and digital techniques. Future development in DSP could even bring better solutions for more demanding noise environments. The active headset application demonstrated how control and signal processing can be implemented successfully in a single application

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