International Journal of Advanced Engineering and Nano Technology (IJAENT)
ISSN: 2347-6389, Volume-1, Issue-7, June 2014

Design of Effective Noise Cancellation Circuits in Acoustic

To-Po Wang, Yen-Chu Lee

Abstract—This paper presents an effective noise cancellation circuit in acoustic, which is designed, simulated, and measured. The feedforward earphone is with high performance of noise suppression. The main structure consists of a controller, a notch filter, a high-pass filter, a microphone, and a speaker. The noise detection microphones pick up sounds and divide them into human voice and low-frequency noise. After passing through the gain-controlled filters, the embedded filtered-X least mean-square (FxLMS) algorithm will generate anti-noise signals with 180° phase difference to ambient noise. Consequently, these anti-noise signals will be transformed from digital formats to analog format, and thereby transformed to acoustic signal by the speakers. The generated acoustic signal will cancel the ambient noise, leading to a quiet zone. According to the measured results, it is indicated that a 15-dB noise reduction is achieved.

Index Terms—Active noise cancellation, low-frequency noise, feed forward type, feedback type.

I. INTRODUCTION

Due to the requirements of high-quality acoustic, researches of active noise control (ANC) are rapidly increased. In [1], a set of acoustical signal processing algorithms were developed to control sound in three dimensions. Moreover, the presented technique in [1] can be used to quiet zones with arbitrary shapes. Compared to traditional ANC systems containing only an adaptive filter coupled to an adaptive algorithm, the created adaptive ANC system can be used to generate 3-D quiet zones in space. In order to minimize the noise, a variable-step-size filtered-X least mean-square (FxLMS) algorithm is presented in [2]. The algorithm with variable step sizes can effectively improve the convergence rate and reduce noise. In [3], a developed algorithm of a control filter with an un-symmetric and two-sided exponential decay envelop over its impulse response is presented. The algorithm is very suitable to a long tap-length filter, and it can speed up the convergence rate. In [4], an algorithm used to overcome instabilities in an adaptive feedback ANC is presented. The algorithm can detect changes in the secondary path because of an increased gain of lifted headphone. In [5], a time-varying gain for additive auxiliary noise is presented. Base on the method, it can achieve the fast convergence of a feedback path modeling (FBPM) filter and reduce steady-state power of the residual error in an error microphone.

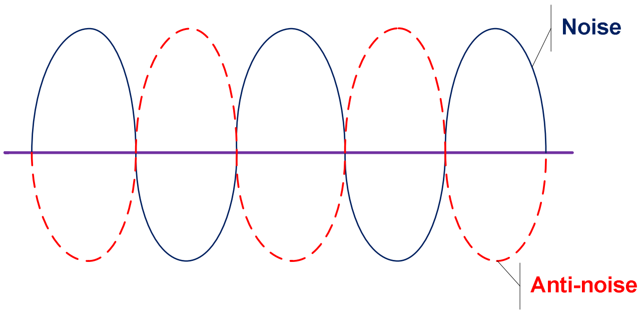

Fig. 1. Noise cancellation of ANC.

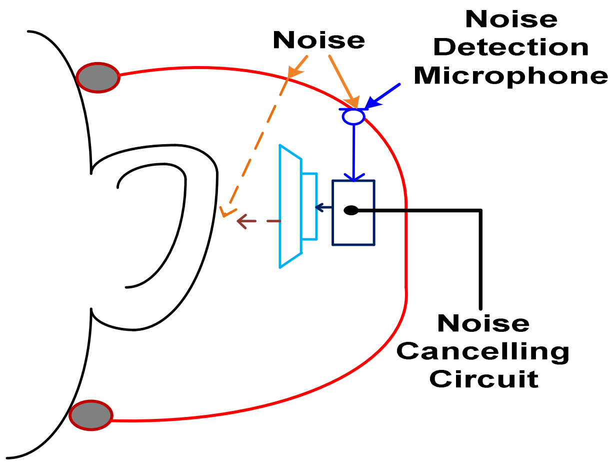

Fig. 2. Feedforward type of ANC.

In this work, a feedforward earphone with high noise suppression is presented in this paper. The main structure consists of a controller, a notch filter, a high-pass filter, a microphone, and a speaker.

II. DESIGN OF HIGH-NOISE-SUPPRESSION EARPHONE

A regular passive noise-canceling technique is suitable for high frequency range (i.e., above several kHz). However, it is not suitable in low frequency range for acoustic (i.e., below 1 kHz). Therefore, active noise control (ANC) technique is adopted in this work, as shown in Fig. 1. An anti-noise acoustic wave is generated from a controller. The anti-noise wave is with opposite phase compared to the noise (marked with blue color). This can effectively minimize noise. Fig. 2 shows a headphone with a feedforward type of ANC. A noise detection microphone is attached on the shell of the headphone. Once the noise is detected, the noise cancelling circuit will generate the anti-noise signal to minimize noises. To describe the feedforward type of ANC, the simplified model is depicted in Fig. 3. In the system, the feedforward delay (Dff) denotes the phase shift from the headset to the speaker. Moreover, the delay from the electrical impulses

Manuscript Received on June 2014.

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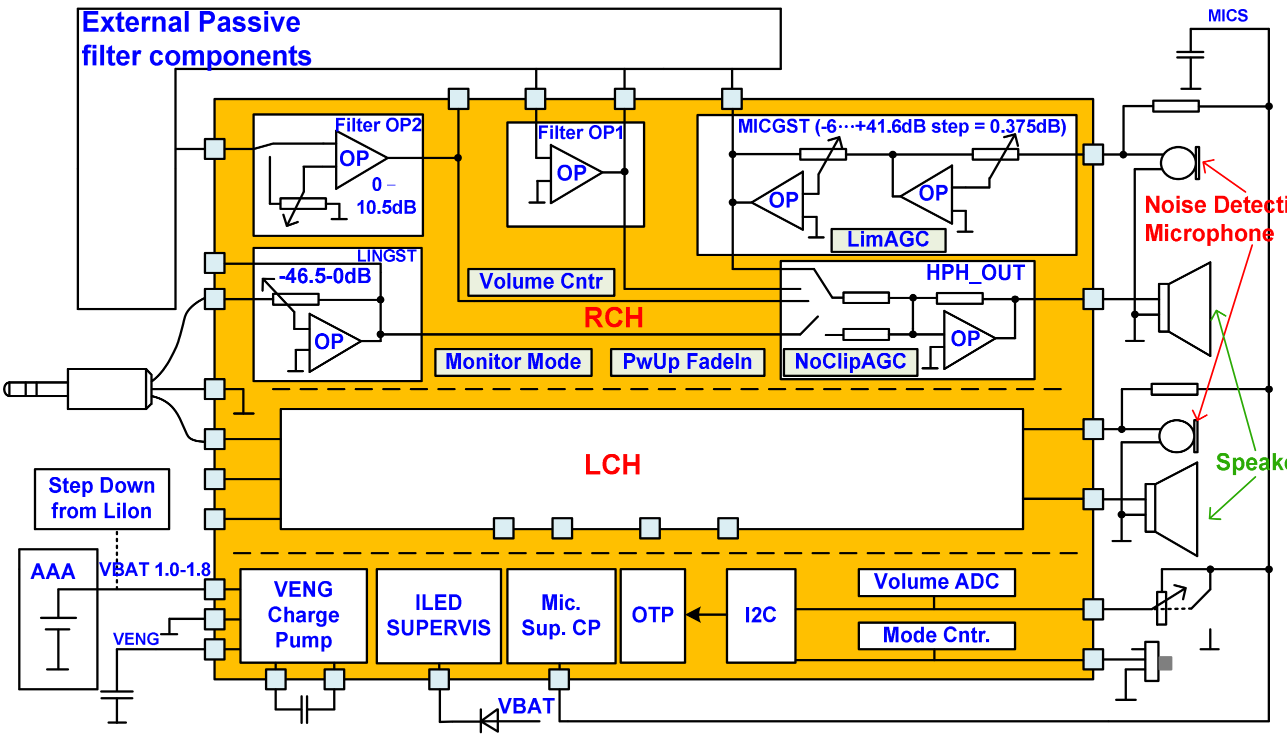

Fig. 4. Block diagram of the used driver integrated circuit (AS3410).

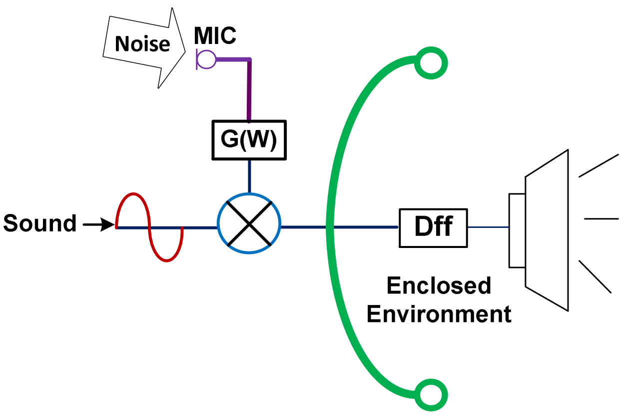

Fig. 3. Simplified model of feedforward type for ANC.

converted to the air pressure waves has to be included in the Dff. The transfer function G(w) is used to compensate the mechanical frequency of the headset and the delay from the microphone (MIC) and the ear. Fig. 4 shows the block diagram of the used driver integrated circuit (AS3410) [6], it includes two channels including left channel and right channel. In each channel, it consists of filters, gain controlled units, noise detection microphones, and speakers. The noise detection microphones pick up sounds and divide them into human voice and low-frequency noise. After passing through the gain-controlled filters, the embedded filtered-X least mean-square (FxLMS) algorithm will generate anti-noise signals with 180° phase difference to ambient noise. Consequently, these anti-noise signals will be transformed from digital formats to analog format, and thereby transformed to acoustic signal by the speakers. The generated acoustic signal will cancel the ambient noise, leading to a quiet zone. Fig. 5 depicts the software simulator of AS3410.Fig. 5(a) gives the design parameters of the functional blocks in this work. From the microphone to speaker, these are a notch filter and a high-pass filter to filter

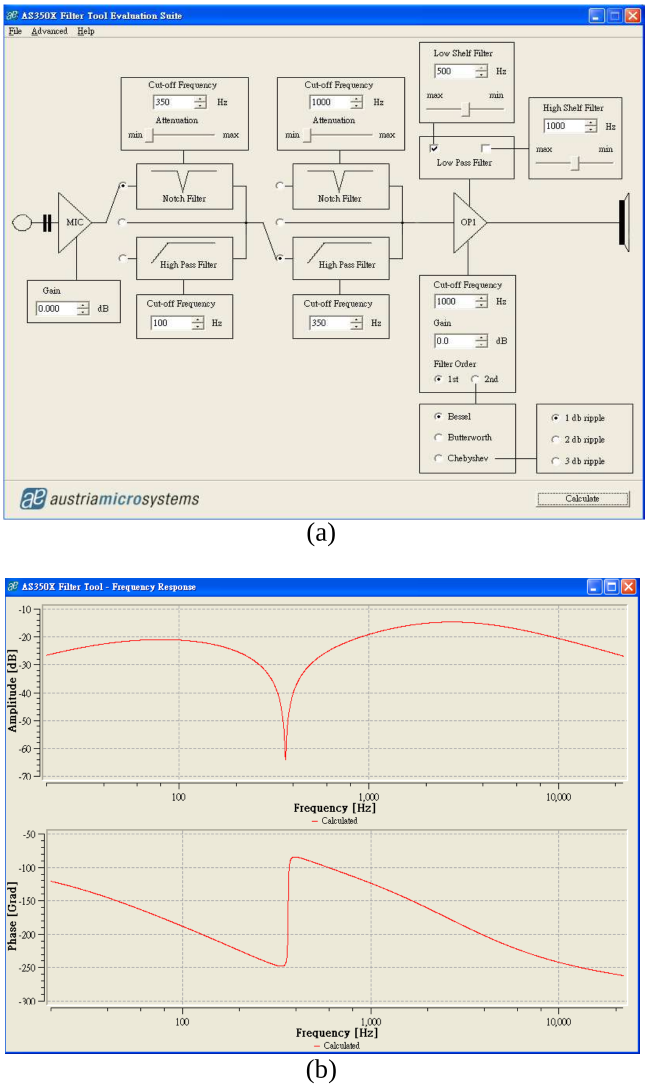

Fig. 5. Software simulator of ANC. (a) functional block, (b) results.

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out the unwanted noise. The center frequency of the notch filter is at 350 Hz, and the cut-off frequency of the high-pass filter is at 350 Hz. The simulated results are depicted in Fig. 5(b). From this figure, it is observed that the phase change is about 180° at 350 Hz. This can result in minimized noise around this frequency range.

III. MEASUREMENT RESULTS

Fig. 6 shows the measured results of the left and right channels. The red lines present the ANC off, and the blue lines are ANC on. From Fig. 6, it is indicated that a 15-dB noise reduction is achieved at 150 Hz in left channel due to ANC.

IV. CONCLUSION

The paper presents the design of active noise-canceling headphones having a wide band noise cancellation. The noise detection microphones pick up sounds and divide them into human voice and low-frequency noise. After passing through the gain-controlled filters, the embedded filtered-X least mean-square (FxLMS) algorithm will generate anti-noise signals with 180° phase difference to ambient noise. According to the measured results, it is indicated that a 15-dB noise reduction is achieved.

V. ACKNOWLEDGMENT

This work is supported in part by the National Science Council of Taiwan, R.O.C., under Contract NSC 103-2623-E-027-001-NU. The authors would like to thank the National Chip Implementation Center (CIC), Hsinchu, Taiwan, and National Nano Device Laboratories (NDL), Hsinchu, Taiwan, for technical supports.

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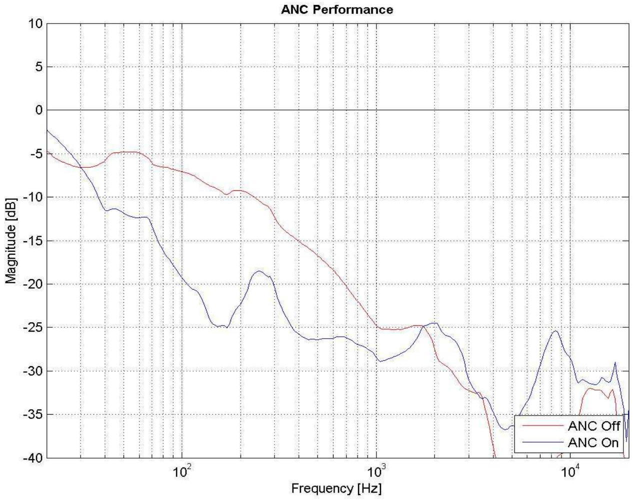
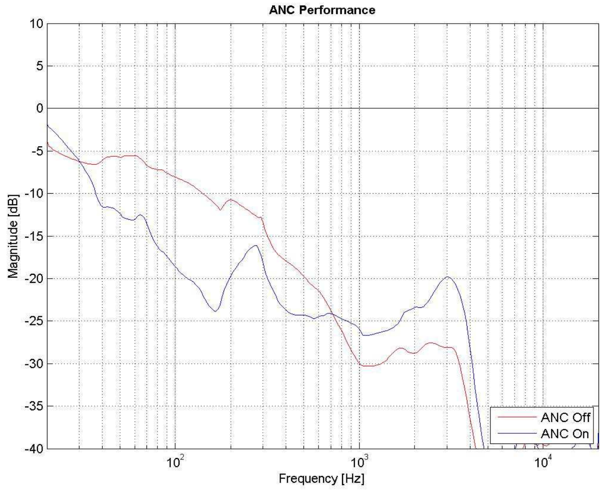


(a)



(b)

Fig. 6. Measured results (a) Left channel, (b) Right channel.

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