2013년 9월 전자공학회 논문지 제 50 권 제 9 호
Journal of The Institute of Electronics Engineers of Korea Vol. 50, NO. 9, September 2013http://dx.doi.org/10.5573/ieek.2013.50.9.172

논문 2013-50-9-20

Modified FxLMS Algorithm for Active Noise Control and Its Real-Time Implementation

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Abstract

This paper presents a modified filtered-x least mean square (FxLMS) algorithm to improve the stability of active noise control (ANC) system in realistic environment. A real-time ANC system employing modified FxLMS is designed and implemented on digital signal processor (DSP) board. The ANC system is evaluated for cancelling various tonal frequency noises in the range from 100 to 500 Hz and the performance is measured in terms of sound pressure level (SPL) attenuation. Experiment results show that a quiet zone with maximum 20 dB SPL attenuation can be generated around the location of error microphone.

Keywords : Real-time ANC, DSP board, FxLMS, quiet zone.

Ⅰ. INTRODUCTION

ANC cancels the primary noise by generating and combining an anti-noise (with equal amplitude but opposite phase) at the location of the error microphone, it efficiently attenuates low frequency noise with benefits in size and cost[1~3]. Over the past few decades a great progress has been made in ANC, yet the practical applications are limited, one important challenge comes from the impulsive noise in realistic environment[4]. This paper discusses a modified FxLMS based ANC system and its real-time implementation on DSP board. The famous FxLMS algorithm[3] based on minimization of variance of error signal has been widely used because of its robustness and simplicity. For the impulsive noise, the FxLMS algorithm may become unstable[4]. Hence, a modified FxLMS is applied in our system to overcome this problem.

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접수일자: 2013년8월9일, 수정완료일: 2013년9월2일

Our ANC system includes a ND-Tech TMS320C6713 DSP board with embedded ADC and DAC units, two microphones and a loudspeaker. The circuit part is self-designed and will be introduced in experimental setup section. The system is tested to cancel 100 to 500 Hz sine tones generated by function generator and primary loudspeaker. Experiment results prove that our system achieves a quiet zone around the location of error microphone.

The organization of the paper is as follows: Section Ⅱ presents the modified FxLMS algorithm. Section Ⅲ describes the experiments setup and implementation. Section Ⅳ illustrates the results of experiments. The conclusion is discussed in Section Ⅴ.

Ⅱ. MODIFIED FxLMS ALGORITHM

ANC cancels the primary noise by generating and combining an anti-noise. For generating this anti-noise, the reference signal $x(n)$ is picked up by microphone; and then, we generate digital anti-noise signal $y(n)$ by applying adaptive filter $w(z)$ to estimate the unknown primary transfer function $P(z)$; this digital signal is transformed to a real anti-noise $y'(n)$ in acoustic domain after passing through the secondary path $S(z)$ (Section Ⅲ.2.1). Using FxLMS algorithm, the adaptive filter is updated by minimizing the mean square error $e(n)$, as

$$w(n)=w(n)+\mu x'(n)e(n) \tag{1}$$

where μ is step size; $x'(n)$ is the filtered $x(n)$ signal by $S(z)$, but $S(z)$ is unknown and must be estimated by an additional filter $\hat{S}(z)$[3]. The estimation and significance of the secondary path transfer function will be discussed in Section Ⅲ.

The FxLMS algorithm may become unstable, especially in non-stationary impulsive noise environments[4~7]. This is a significant challenge for real-time ANC application. To solve this problem, Sun proved that the samples of the reference signal should be treated probabilistically[4]. According this, we proposed a modified FxLMS algorithm in impulse-like noise environments. The reference signal is treated probabilistically and update equation is modified as

$$w(n+1)=w(n)+\mu e(n)[\hat{s}(n)*[P_{x(n)}x(n)]] \tag{2}$$

where $\hat{S}(z)$ is the impulse response of estimated secondary path $\hat{S}(z)$, $*$ denotes linear convolution and $P_x(n)$ is the estimated probability of reference

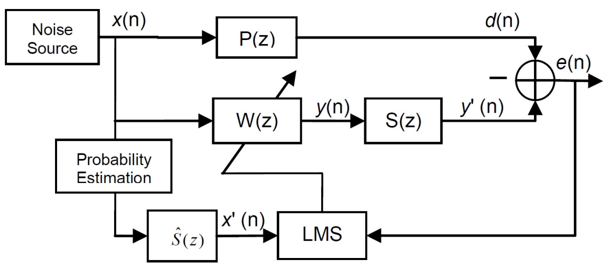

Fig. 1. Block diagram of ANC with modified FxLMS algorithm.

signal. Assuming the noise beyond range $[-c, c]$ is impulsive noise, the probability is given as

$$P_{x(n)} = \begin{cases} 1, & -c \le x(n) \le c \\ \left(\dfrac{c}{x(n)}\right)^K, & otherwise \end{cases} \tag{3}$$

where $K (K=0, 2, 4, 6, \ldots)$ is the attenuation factor, c is the threshold of normal noise. Assuming that a sufficient number of samples follow Gaussian distribution $(x \sim N(\mu, \sigma^2))$, the parameters of this unknown distribution can be estimated using maximum-likelihood estimation as

$$\hat{\mu} = \frac{1}{n}\sum_{k=1}^{n} x_k, \quad \hat{\sigma}^2 = \frac{1}{n}\sum_{k=1}^{n}(x_k - \hat{\mu})^2 \tag{4}$$

where n is the number of samples and x_k is the magnitude of sample. We consider the samples beyond $3\hat{\sigma}$ as impulsive noise. This means, c equals $3\hat{\sigma}$ and the probability of impulsive noise is less than 0.2%.

Ⅲ. REAL-TIME ANC SYSTEM

3.1. SYSTEM SETUP

The ANC system setup is shown in Fig.2. The primary noise is generated using function generator and a 4 Ohm impedance loudspeaker. The frequency response of the primary loudspeaker ranges from 100 Hz to 20 kHz with a maximum power output of 30 W.

The reference microphone and error microphone is unidirectional condenser microphone with 2.2 KOhm impedance, dB sensitivity and 70 Hz to 20 kHz frequency response range; they are placed in location A and C respectively, as seen in figure. The cancelling loudspeaker has same specifications as the primary loudspeaker and is placed at location B.

Our system is implemented using TMS320C6713 DSP board. Throughout the experiments, the system is tested with 100 to 500 Hz sine tones; the sampling frequency F_S is set as 2000 Hz. The antialiasing

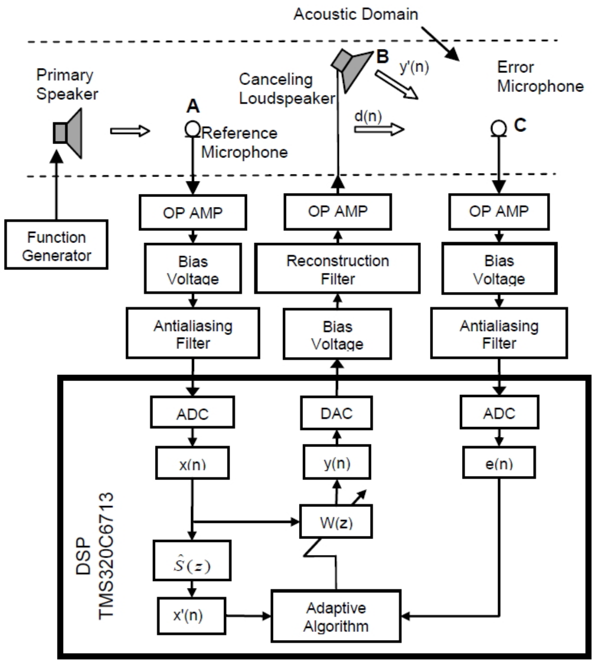

Fig. 2. ANC system setup.

filter and reconstruction filter is 4-order low-pass Butterworth filter. The passband frequency is 750 Hz and stopband frequency is 1000 Hz with -9 dB stopband attenuation. Our DSP board employs 12-bit 8-channel A/D convertor AD7888 and 12-bit 4-channel D/A convertor AD7564, both the analog input voltage of A/D convertor and analog output voltage of D/A convertor are 0 to 2.5 V. That is the reason we designed a bias voltage (+1.25 V) before the ADC signal input and a bias voltage (-1.25 V) after the DAC signal output. For the operational amplifier, a balance input microphone amplifier circuit with an adjustable gain is designed using high fidelity audio operational amplifier IC LME 49740.

Considering the superposition of primary noise and anti-noise at error microphone, the causality condition should be satisfied:

$$\delta_{AB} > \delta_S \tag{5}$$

where δ_{AB} is the acoustic delay, it equals the travel time of primary noise from A to B; δ_S is the system delay.

$$\delta_{AB} = \frac{D_{AB}}{c} \tag{6}$$

$$\delta_S \approx \frac{1}{F_S} + \frac{2}{120K} \tag{7}$$

where D_{AB} is the distance between A and B, c is sound speed constant in the air. We consider the system delay equals program computation delay $1/F_S$, ADC data receive delay $1/120K$ and DAC transmit delay $1/120K$.

Combination of the equation above yields:

$$D_{AB} > 0.18 \tag{8}$$

The distance between the reference microphone and the cancelling loudspeaker should be larger than 0.18 meter.

3.2. SYSTEM IMPLEMENTATION

3.2.1. OFFLINE MODELING AND SECONDARY PATH ESTIMATION.

The secondary path transfer function $S(z)$ includes the D/A converter, reconstruction filter, amplifier, loudspeaker, acoustic path from the

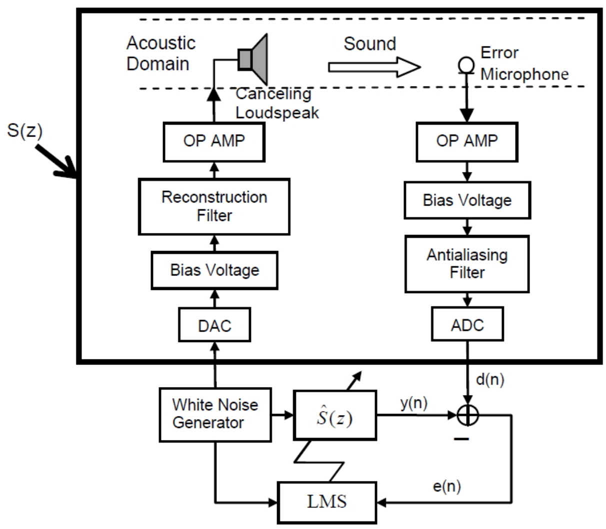

Fig. 3. Block diagram of the offline secondary estimation.

canceling loudspeaker to the error microphone, error microphone, antialiasing filter, A/D converter and error microphone. Usually, the secondary path is invariable; it can be estimated during the initial training stage, the offline modeling hardware setup is shown in Fig.3. White noise is used to train the system.

The length of adaptive filter $\hat{S}(z)$ is 128. SNR for offline modeling is defined to shown the performance as:

$$SNR = -10\log_{10}\frac{\sum e^2(n)}{\sum d^2(n)} \tag{9}$$

We compared the average SNR achieved in offline modeling with different D_{BC} in Table 3. The distance between cancelling speaker and error microphone should be selected neither too close nor too far; 0.1 meter is an appropriate choice.

Table 3. Average SNR in offline modeling for different DBC.

D_{BC}	Average SNR
0.05 m	23.5 dB
0.10 m	31.6 dB
0.15 m	27.2 dB
0.20 m	23.3 dB
0.30 m	19.6 dB
0.50 m	16.7 dB

3.2.2. ONLINE MODELING.

The length of adaptive filter $W(z)$ in online modeling is 256. The real-time ANC system is tested to cancel several sine tones generated by function generator in the frequency range 100 to 500 Hz.

Ⅳ. RESULTS

A SPL meter is used to measure SPL close to the error microphone. SPL attenuation with ANC system is shown in Fig.5. The reference microphone and error microphone is place at (−50, 0) and (0, 0)

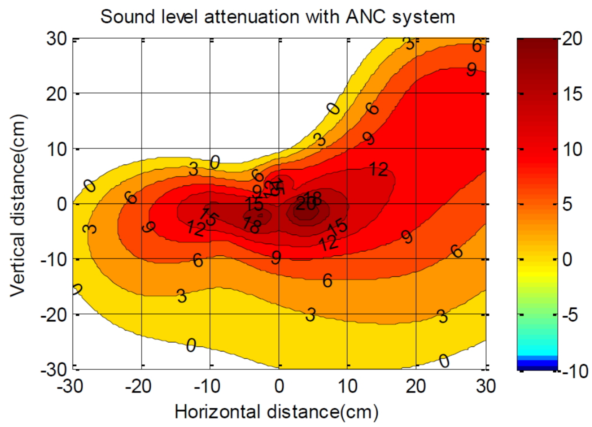

Fig. 5. SPL attenuation with ANC.

respectively. The front face of canceling speaker is place from (−5, 10) to (5, 15). The figure proves a quiet zone with maximum 20 dB SPL attenuation is generated around the location of error microphone.

It must be noted that our system uses one error microphone and one cancelling loudspeaker, the area of quiet zone is limited. To increase the area of quiet zone, multi-channel ANC with multi-error microphones and multi-speakers will be applied next to obtain desired quiet zone.

Ⅴ. CONCLUSION

A real-time ANC system using DSP is implemented, the performance is evaluated by cancelling sine tonal noise. A quiet zone with maximum 20 dB SPL attenuation is generated around error microphone. It is noted that the area of quiet zone is limited, this can be overcome by applying multi-channel ANC.

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