한국음향학회지 제33권 제6호 pp. 375~382 (2014)
*The Journal of the Acoustical Society of Korea* Vol.33, No.6 (2014)
http://dx.doi.org/10.7776/ASK.2014.33.6.375

pISSN : 1225-4428
eISSN : 2287-3775

# A New Parallel Method for Narrowband Active Noise Control

# 협대역 능동 소음 제어를 위한 새로운 병렬 기법

Seong-Woo Kim, Young-Cheol Park,* Young-Soo Seo,** and Dae Hee Youn†

(김성우, 박영철,* 서영수,** 윤대희†)

School of Electrical and Electronic Engineering, Yonsei University
*Computer and Telecommunications Engineering Division, Yonsei University
**Agency for Defense Development

(Received July 13, 2014; revised August 25, 2014; accepted September 24, 2014)

**ABSTRACT:** In many practical active noise control applications, the primary noise contains multiple closely-spaced harmonics. A narrowband ANC system consists of adaptive filters excited by a composite reference signal, which is the set or sum of sinusoids. This paper analyzes and shows that the convergence speeds of the direct form, parallel form, and simplified parallel form narrowband ANC systems are affected by the fundamental frequency and frequency separation between two adjacent sinusoids in the reference signal. This paper also proposes the new simplified parallel form narrowband ANC system whose convergence speed is independent on the frequency of the reference signal. Computer simulations are conducted to verify the analysis presented in the paper and to compare the proposed narrowband ANC system with the conventional narrowband ANC system.

**Keywords:** Narrowband ANC system, FxGAL algorithm.
**PACS numbers:** 43.50.Ki

**초 록**: 많은 실제 능동 소음 제어 분야에서 주요 소음은 다수의 인접한 배음으로 구성된다. 또한 협대역 능동 소음 제어 시스템은 정현파 세트나 혹은 합으로 구성된 기준 신호에 의해 동작하는 적응 필터들로 구성된다. 본 논문은 직접 형, 병렬 형, 간소화한 병렬 형의 능동 소음 제어 시스템이 기본 주파수와 인접한 정현파의 주파수 간격에 영향을 받음을 보여주고 이를 분석한다. 또한 본 논문은 기준 주파수에 독립적인 새로운 간소화한 병렬 형 협대역 능동 소음 제어 시스템을 제안한다. 본 논문에서의 제시한 분석을 증명하고 새로운 협대역 능동 소음 제어 시스템을 기존 협대역 능동 소음 제어 시스템과 비교하기 위해 컴퓨터 시뮬레이션을 수행한다.

**핵심용어**: 협대역 능동 소음 제어 시스템, FxGAL 기법

## I. Introduction

Active noise control (ANC)[1-3] is based on the principle of superposition, where an unwanted primary noise is canceled by a secondary noise of equal amplitude and opposite phase. The primary noise produced by rotating machines, such as engines, is periodic and contains multiple harmonic-related narrowband components. In such applications, a nonacoustic sensor such as a tachometer or an accelerometer[3] is often used to synchronize an internally generated reference signal, and thus the feedback from the secondary source to the reference sensor can be prevented. The vast number of narrowband ANC algorithms have been proposed, which can be found from the previous publications[1-13] and references therein.

In practical narrowband ANC applications, e.g. electronic mufflers on automobiles, periodic noise usually contains multiple sinusoids at the fundamental and several dominant harmonic frequencies. Based on this observation, Glover[4] used a sum of cosine or sine waves as a reference signal of an adaptive filter with order much higher than two. However, the order of the adaptive filter required to achieve the same convergence speed increases as the frequency separation

†**Corresponding author:** Dae Hee Youn (dhyoun@yonsei.ac.kr)
2nd Engineering Building, Room b708, Department of Electrical and Electronic Engineering, Yonsei University, Seoul 120-749, Republic of Korea
(Tel: 82-2-2123-4534, Fax: 82-2-2123-4870)

between two adjacent sinusoids in the reference signal decreases or the fundamental frequency moves to $0$ or $\pi$.[5,6]

To overcome this problem, Kuo et al.[6] proposed a direct/parallel form method based on the filtered-x least mean square (FxLMS) algorithm. The idea is to separate a collection of many harmonically related sinusoids into mutually exclusive sets that individually have frequencies spaced out as far as possible. However, the direct/parallel form method doesn't completely eliminate the effect of the frequency separation and fundamental frequency on the convergence speed.[6]

One possible solution to the dependency on the fundamental frequency and frequency separation is to use adaptive $M$ filters connected in parallel, each of which is excited by a single-frequency sinusoid.[7] Using cosine and sine waves as reference signals, the parallel form method has a convergence speed independent on the fundamental frequency and frequency separation.[3] However, to implement the FxLMS algorithm in the parallel form, two estimated secondary-path filters are required for each reference channel. Normally, the secondary paths are modeled using finite impulse response (FIR)-type filters, which pose a computational complexity and result in a bottleneck in system implementation.[8]

To solve the complexity issue of the parallel form method, a simplified approach proposed in,[9] where simplified single-frequency ANCs are connected in parallel and the sine wave generator is replaced with a simple delay. In the simplified parallel form method, a single secondary-path filter is required for each reference channel, which is the half of the parallel form method.

However, in this paper, it is shown that the convergence speed of the simplified parallel form method in[9] becomes slower as the fundamental frequency moves to $0$ or $\pi$. Since the fundamental frequency in practical applications can be very low, the simplified parallel form method is likely to suffer from slow convergence speed.

In this paper, a new simplified parallel form method based on the filtered-x gradient adaptive lattice (FxGAL) algorithm is proposed. In the proposed algorithm the $m$ th reference signal vector is orthogonalized using four additional coefficients per channel. As a result, the eigenvalue spread of the $m$ th reference signal correlation matrix always becomes 1. Thus, the convergence speed of the proposed method is independent on the frequency of reference sinusoids and its computational complexity is similar to the simplified parallel form method.[3]

The rest of this paper is organized as follows. Section II starts with analyzing conventional narrowband ANC system. The proposed narrowband ANC system is presented in Section III. Section IV comprises the experimental results comparing the proposed narrowband ANC system with conventional narrowband ANC system. Finally, conclusions of this work are drawn in Section V.

## II. Conventional Narrowband ANC Systems

### 2.1 Narrowband ANC Model

Fig. 1 depicts the block diagram of the direct form narrowband ANC system based on the FxLMS algorithm. The primary noise $d(n)$ comprises $M$ dominant narrowband components at frequency $\omega_m$, $m=1,\ 2,\ldots,M$. The primary noise also contains a zero-mean additive white Gaussian noise $v_p(n)$ with variance $\sigma_{v_p}^2$. The transfer functions $P(z)$ and $S(z)$ represent the primary and secondary paths, respectively. $\hat{S}(z)$ is the secondary path estimate (or model). Synchronization signal triggers the sinewave generator that produces the reference signal, which is filtered by the adaptive FIR filter $W(z)$ to produce the anti-noise $y(n)$ to cancel the primary noise $d(n)$. The canceling signal $y(n)$ is generated as

$$y(n)=w^T(n)x(n), \tag{1}$$

where $T$ denotes transpose operation, $w(n)=[w_0(n)\ w_1(n)\ \ldots\ w_{L-1}(n)]^T$ is the weight vector of the adaptive filter $w(z)$, and $x(n)=[x(n)\ x(n-1)\ \ldots\ x(n-L+1)]^T$ is the reference signal vector. The adaptive filter length $L$

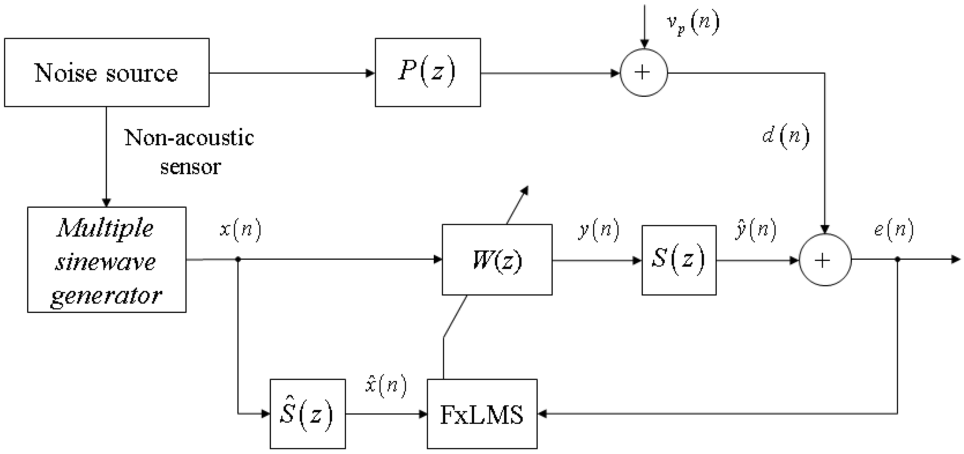



Fig. 1. Direct form narrowband ANC system.

should be at least twice the total number of sinusoids to deal with both the in-phase and the quadrature components, i.e., $L \geq 2M$.

The error signal is expressed as

$$e(n) = d(n) + s(n)*y(n), \tag{2}$$

where $s(n)$ is the secondary path filter.

Assuming that the estimated secondary path model $\hat{S}(z)$ is an FIR filter with length $\hat{L}_s$:

$$\hat{S}(z) = \sum_{l=0}^{\hat{L}_s-1} \hat{s}_l z^{-l}. \tag{3}$$

The filtered reference signals are computed as[3]

$$\hat{x}(n) = \sum_{l=0}^{\hat{L}_s-1} \hat{s}_l x(n-l), \tag{4}$$

where $\hat{s}_l$ are the coefficients of $\hat{S}(z)$.

The weight vector is updated using the FxLMS algorithm[3]

$$w(n+1) = w(n) - \mu\hat{x}(n)e(n), \tag{5}$$

where $\mu$ is the step-size parameter.

## 2.2 Convergence speed comparison

For a stationary input and sufficiently small step-size parameter $\mu$, the convergence time $\tau_{mse}$ is dependent on the eigenvalue spread $\rho$ of the autocorrelation matrix as[14]

$$\tau_{mse} \leq \frac{\lambda_{\max}}{\lambda_{\min}} T_s = \rho T_s (\text{seconds}), \tag{6}$$

where $T_s$ is the sampling period. Small eigenvalue spread can be obtained by reorganizing the filtered reference signal vector. Thereby, a number of different narrowband ANC systems have been developed.[1,3,5,7]

For the simplification of analysis, the reference sinusoids are expressed as[6]

$$\hat{x}_m(n) = \hat{A}_m \cos\{[\omega_0 + (m-1)\Delta\omega]n + \hat{\phi}_m\}, \tag{7}$$

where $\omega_0$ is the fundamental frequency, $\Delta\omega$ is the frequency separation, and $\hat{\phi}_m$ is the phase of the estimated secondary path. The amplitude of the estimated secondary path $\hat{A}_m$, $m = 1, 2, ..., M$ are assumed to be unity.

In the direct form method,[6] a close-form expression of eigenvalues is difficult to obtain when the total number of sinusoids is greater than two. Previously in,[6] bounds for the extreme eigenvalues were used to analyze the eigenvalue spread of the direct form. The lower bound for eigenvalue spread can be expressed as

$$\frac{L|\sin[\omega_0+(l_1-1)\Delta\omega]|+|\sin\{L[\omega_0+(l_1-1)\Delta\omega]\}|}{L|\sin[\omega_0+(l_1-1)\Delta\omega]|-|\sin\{L[\omega_0+(l_1-1)\Delta\omega]\}|}\le\rho, \quad (8)$$

where $l_1 = \arg\max \sin\{L[\omega_0+(l-1)\Delta\omega]\}/\sin[\omega_0+(l-1)\Delta\omega]$. According to the analysis, the eigenvalue spread of the direct form method is significantly affected by the fundamental frequency and frequency separation.[6]

However, in the parallel form method, using cosine and sine wave as the $m$ th channel reference signal, the eigenvalue spread of the parallel form method is 1.[3]

In the simplified parallel form, only cosine or sine waves is used as the reference signal, so that the eigenvalue can be expressed as[3]

$$\rho=\frac{1+\cos\,\omega_0}{4-\cos\,\omega_0}. \quad (9)$$

Fig. 2 shows the eigenvalue spread versus different fundamental frequencies for $M=4$, $L=8$. The frequency separation is $\Delta\omega=0.1\pi$. Fig. 2 clearly shows that the fundamental frequency affects the eigenvalue spreads of the direct and simplified parallel form methods, but does not affect that of the parallel form method.[3,6] Especially, this figure shows eigenvalue spreads of the direct and simplified parallel form methods significantly increase as the fundamental frequency decreases, which is problematic because the primary noise in practical situation can have very low fundamental frequencies.

Fig. 3 shows the eigenvalue spread versus different frequency separations for $M=4$, $L=8$. The fundamental frequency is $\omega_0=0.2\pi$. Fig. 3 clearly shows that the frequency separation affects the eigenvalue spreads of the direct form method, but does not affect those of the parallel and simplified parallel form methods.[3,6]

Figs. 2 and 3 show that the parallel form method is independent on the frequency of the reference signal. However, the parallel form method based on the FxLMS algorithm requires $2M\hat{L}_s$ multiplications for secondary

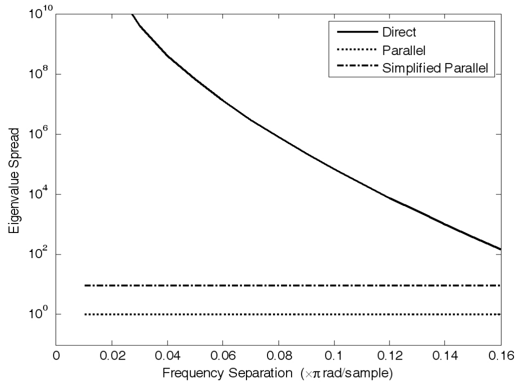



Fig. 2. Effect of frequency separation $\Delta\omega$ on eigenvalue spread ($M$ = 4, $L$ = 8, $\omega_0=0.2\pi$).

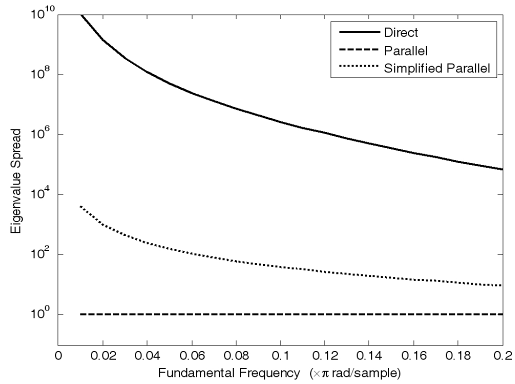



Fig. 3. Effect of fundamental frequency $\omega_0$ on eigenvalue spread ($M$ = 4, $L$ = 8, $\Delta\omega_0=0.1\pi$).

path filtering. In practical situations, the length of secondary path model should be long, so that the secondary path filtering becomes burden. In the simplified parallel form method, the sine wave generator is replaced with simple time delay, which saves the $M\hat{L}$ multiplications.[3] However, its eigenvalue spread depends on the fundamental frequency.

## III. New Narrowband ANC System

### 3.1 Parallel form FxGAL Algorithm

In this paper, we propose a new narrowband ANC system. In an effort to reduce the eigenvalue spread ratio and thus to improve the convergence speed, the filtered-x gradient adaptive lattice (FxGAL) algorithm[15,16] is applied to each two-weight adaptive filter of the simplified parallel form method.

Consider a second-order lattice predictor that transforms the $m$ th channel filtered reference signal into the orthogonal filtered backward prediction error. This orthogonalization is carried out in the lattice through formulas[15,16]:

$$\hat{f}_{m,1}(n) = \hat{f}_{m,0}(n) - k_m \hat{b}_{m,0}(n-1), \tag{10}$$

$$\hat{b}_{m,1}(n) = \hat{b}_{m,0}(n-1) - k_m \hat{f}_{m,0}(n), \tag{11}$$

where $k_m$ is the th channel reflection coefficient and $f_{m,0}(n) = b_{m,0}(n) = \hat{x}_m(n)$. The $m$ th channel filtered backward prediction errors are orthogonal to each other as

$$E\{\hat{b}_{m,0}(n)\hat{b}_{m,1}(n)\} = 0. \tag{12}$$

Assuming the filtered reference signal $\hat{x}_m(n)$ is the pseudorandom noise signal,[17] the $m$ th channel reflection coefficient can be expressed as

$$k_m \frac{E[\hat{x}_m(n)\hat{x}_m(n-1)]}{E[\hat{x}_m(n)\hat{x}_m(n)]} = \cos\omega_m. \tag{13}$$

Using the FxGAL algorithm,[13,14] the update equation for the th channel regression coefficient $w_m(n)$ can be expressed as

$$w_m(n+1) = w_m(n) - \mu \Lambda_m^{-1}(n)\hat{b}_m(n)e(n), \tag{14}$$

$$e(n) = d(n) + s(n) * \left[\sum_{m=1}^{M} w_m^T(n) b_m(n)\right], \tag{15}$$

where $w_m(n) = [w_{m,0}(n)\ w_{m,1}(n)]^T$ is the th channel regression coefficients vector, $b_m(n) = [b_{m,0}(n)\ b_{m,1}(n)]^T$ and $\hat{b}_m(n) = [\hat{b}_{m,0}(n)\ \hat{b}_{m,1}(n)]^T$ are the $m$ th channel backward prediction errors vector and $m$ th channel filtered backward prediction errors vector, respectively. The matrix $\Lambda_m(n)$ is a diagonal matrix with diagonal elements given by the power of the $m$ th filtered backward prediction error $\sigma_{\hat{b}^2_{m,1}}(n), l-0,1$. The power of $m$ th filtered backward prediction error can be recursively estimated using the single-pole low-pass filter as

$$\sigma_{\hat{b}^2_{m,1}}(n) = \lambda \sigma_{\hat{b}^2_{m,1}}(n-1) + (1-\lambda)\hat{b}^2_{m,1}(n), \tag{16}$$

where $\lambda$ is the smoothing factor.

Finally, the proposed parallel FxGAL-based narrowband

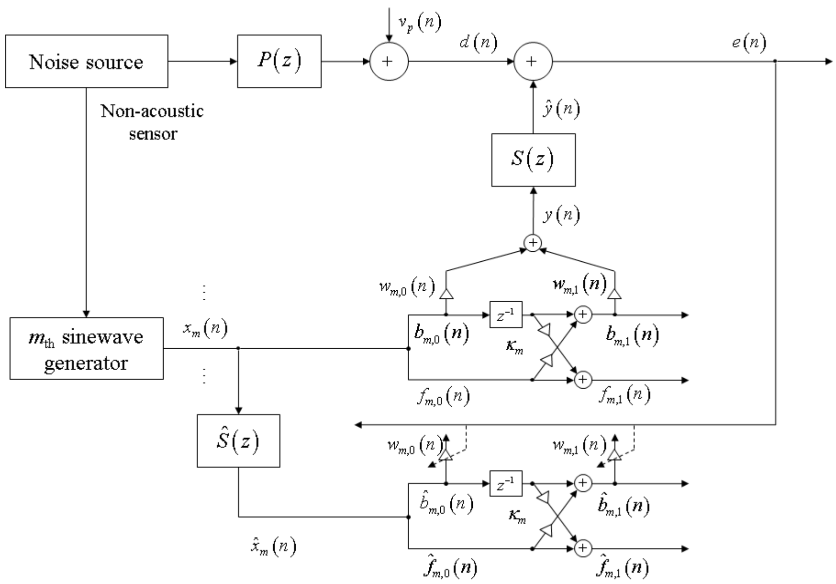


Fig. 4. Proposed parallel FxGAL-based narrowband ANC system.

ANC system is shown in Fig. 4.

### 3.2 Convergence Analysis

For ease of analysis, the cosine wave $x_m(x)$ is treated as a pseudorandom noise signal which leads to simple derivations and elegant equations without sacrificing the accuracy of the analysis.[17] We also assume that $\hat{S}(z) = S(z)$ or the secondary path has been very closely modeled. An optimum weight vector $w_{m,0}$ can be obtained by minimizing the mean square error (MSE)[18]

$$\mathbf{w}_{m,o} = \arg\min_{\mathbf{w}_m} E\left[e^2(n)\right]. \tag{17}$$

We first rewrite the update equation in (13) using the eight error vector, defined as $\varepsilon_m(n) = w_m(n) - w_{m,0}$[18]:

$$\begin{aligned}\varepsilon_m(n+1) = {} & [I - \mu \Lambda_m^{-1}(n)\hat{b}_m(n)\hat{b}_m^T(n)]\varepsilon_m(n) \\ & - \beta_m(n) - \mu b_m(n) v_p(n)\end{aligned}, \tag{18}$$

where $\beta_m = \mu \Lambda_m^{-1}(n)\hat{b}_m(n)\sum_{l \neq m}\hat{b}_l^T(n)\varepsilon_l(n)$ and $I$ is the identity matrix. Using (11) and taking the expected value of (17), we obtain

$$E[\varepsilon_m(n+1)] = [I - \mu]\varepsilon_m(n). \tag{19}$$

Significance of (18) is that the convergence speed of the parallel FxGAL algorithm is independent of the eigenvalue spread, and it only depends on the step-size parameter $\mu$. Hence, the proposed parallel FxGAL-based narrowband ANC system has a convergence speed independent on the reference frequency and/or frequency separation. But the proposed FxGAL-based narrowband ANC system requires only one secondary path filtering in each reference channel.

In summary, the proposed FxGAL-based narrowband ANC system has similar convergence speed to that of the parallel form method, but the proposed FxGAL-based narrowband ANC system has similar computational complexity to that of simplified parallel form method.

## IV. Simulation Results

Computer simulations were conducted to evaluate the performance of the proposed FxGAL-based narrowband ANC system. The sampling rate was 2 kHz. The primary path $P(z)$ shown in Fig. 5 was used. To analyze the effect

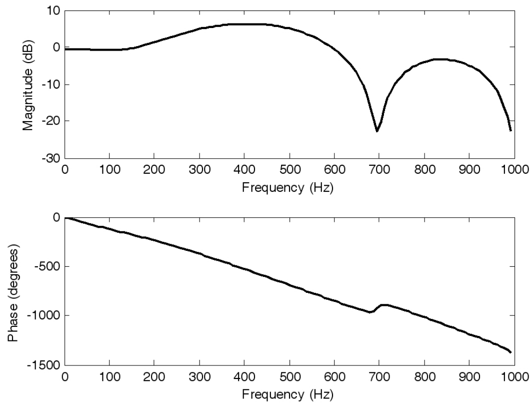
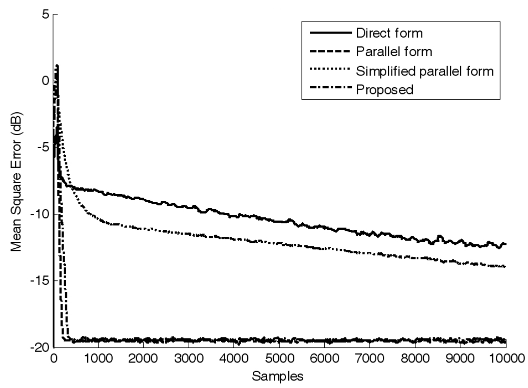
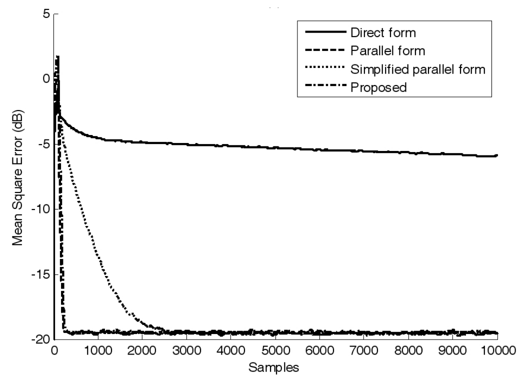



Fig. 5. Magnitude (top) and phase (bottom) responses of primary path $P(z)$.




(a)




(b)

Fig. 6. MSE of narrowband ANC systems for (a) $\omega_0 = 20$ Hz and (b) $\omega_0 = 100$ Hz, and $\Delta\omega_0 = 100$ Hz.

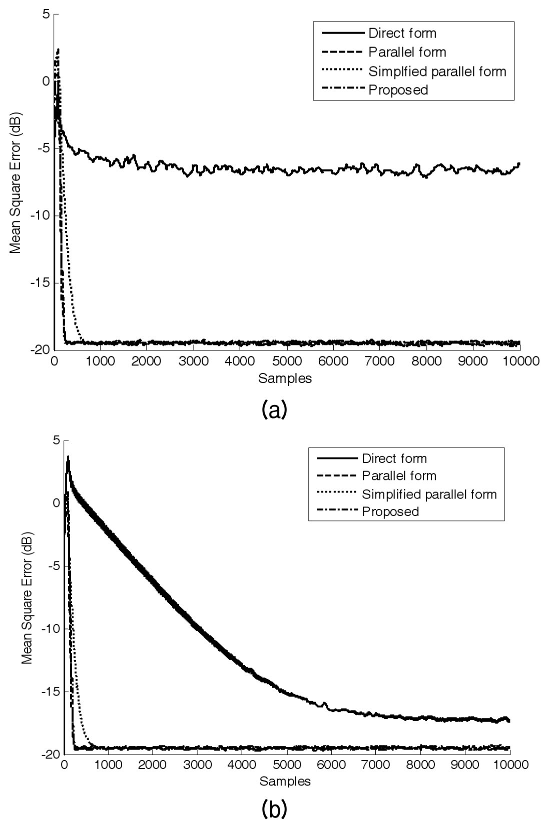



Fig. 7. MSE of narrowband ANC systems for $\omega_0 = 200$ Hz, and (a) $\Delta\omega_0 = 60$ Hz and (b) $\Delta\omega_0 = 150$ Hz.

of the fundamental frequency and frequency separation, the secondary path was assumed to be $S(z) = 1$. The number of sinusoids was $M = 4$, and the length of the adaptive filter was $L = 8$. A white Gaussian noise $v_P(n)$ was added at 20 dB SNR. The results shown below were ensemble averaged over 100 trials.

Fig. 6 compares the convergence behavior of narrowband ANC systems for low-fundamental frequency. The fundamental frequency was $\omega_0 = 20$ Hz [6(a)] and 100 Hz [6(b)], and frequency separation was $\Delta\omega = 100$ Hz. Step-size parameters were experimentally selected to equalize the steady-state MSEs of each narrowband ANC system: $\mu = 0.05$ for the direct form, parallel form, and simplified parallel form methods, and $\mu = 0.03$ for proposed mthod. Results show that the convergence speeds of the direct form and simplified parallel form methods vary according to the fundamental frequency. However, the parallel form and the proposed methods show robustly similar convergence speed for all test cases.

Fig. 7 compares the convergence behavior of narrowband ANC systems according to the frequency separation. The fundamental frequency was $\omega_0 = 20$ Hz and frequency separations of $\Delta\omega = 60$ Hz and 50 Hz were tested. Step-size parameters were experimentally selected to equalize the steady-state MSEs of each narrowband ANC system: $\mu = 0.05$ for the direct, parallel, and simplified parallel form methods, and $\mu = 0.03$ for the proposed method. It can be seen that the parallel form, simplified parallel form, and proposed methods have the same convergence speed regardless of the frequency separation.

## V. Conclusions

In this paper, a new simplified parallel form narrowband ANC system based on the FxGAL algorithm has been proposed. Like the parallel form narrowband ANC system, the proposed narrowband ANC system has a convergence speed that is independent of both the fundamental frequency and frequency separation, but it requires significantly lower computational complexity than the parallel form narrowband ANC system. However, if the nonacoustic reference sensor is not available, then the proposed narrowband ANC system cannot be utilized.

## Acknowledgement

This work was supported by Agency for Defense Development of Korea (UD120021JD).

## References

1. J. R. Glover, J. M. McCool, J. Kaunitz, C. S. Williams, R. H. Hearn, J. R. Zeidler, Jr.Eugene Dong, and R.C.Goodlin, "Adaptive noise canceling: principles and applications," Proc. IEEE **63**, 1692-1716 (1975).
2. P. A. Nelson and S. J. Elliott, *Active Control of sound,* 3rd ed. (Academic, New York, 1995), pp. 34-40.
3. S. M. Kuo and D. R. Morgan, *Active Noise Control Systems-Algorithms and DSP Implementations* (Wiley, New York,

1996), pp. 1-12.
4. J. R. Glover, "Adaptive noise canceling applied to sinusoidal interferences," IEEE Trans. Acoust. Speech, Signal Process. **25**, 484-491 (1977).
5. Y. Yuan, N. S. Kapsokavathis, K. Chen, and S. M. Kuo, "Active Noise Control System," U. S. patent 5359662, (1994).
6. S. M. Kuo and A. B. Puvvala, "Effects of frequency separation in periodic active noise control systems," IEEE Trans. Audio, Speech, Lang. Process. **14**, 1857-1866 (2006).
7. E. Ziegler Jr, "Selective active cancellation system for repetitive phenomena," U.S. Patent, 4,878,188, (1989).
8. Y. Xiao, "A new efficient narrowband active noise control system and its performance analysis," IEEE Trans. Audio, Speech, Lang. Process. **19**, 1865-1874 (2011).
9. S. M. Kuo, S. Zhu, and M. Wang, "Development of optimum adaptive notch filter for fixed-point implementation in active noise control," in Proc. 1992 Int. Conf. Industrial Electronics, 1376-1378 (1992).
10. F. Yang, A. Gupta, and S. M. Kuo, "Parallel multi-frequency narrowband active noise control systems," in Proc. 2009 IEEE Int. Conf. Acoust., Speech, Signal Process. (ICASSP), 235-256 (2009).
11. C. Y. Cheng and S. M. Kuo, "Complete parallel narrowband active noise control systems," IEEE Trans. Audio, Speech, Lang. Process. **21**, 1979-1986 (2013).
12. S.-H Oh, H.-S. Kim, and Y.-J. Park, "Active control of road-booming-noise with constraint multiple filtered-x LMS algorithm" (in Korean), J. Acoust. Soc. Kr. **19**, 3-7 (2000).
13. O.-S. Kwon, "Robust active noise controller with hybrid adaptive nonlinear compensator" (in Korean), J. Acoust. Soc. Kr. **28**, 16-22 (2009).
14. S. Haykin, *Adaptive Filter Theory*, 4th ed. (Prentice-Hall, New Jersey, 2002), pp. 270-271.
15. Y. C. Park and S. D. Sommerfeldt, "A fast adaptive noise control algorithm based on the lattice structure," Applied Acoustics, Elsevier Science Limited **40**, (1996).
16. S.-W. Kim, Y.-C. Park, and D. H. Youn, "A variable step-size filtered-x gradient adaptive lattice algorithm for active noise control," in Proc. 2012 IEEE Int. Conf. Acoust., Speech, Signal Process. (ICASSP), 189-192 (2012).
17. Y. Xiao, A. Ikuta, L. Ma, and K. Khorosani, "Stochastic analysis of the FXLMS-based narrowband active noise control system," IEEE Trans. Audio, Speech, Lang. Process. **16**, 1000-1014 (2008).
18. S. J. Elliott, I. M. Stothers, and P. A. Nelson, "A multiple error LMS algorithm and its applications to the active noise control of sound and vibration," IEEE Trans. Acoust., Speech, Signal Process. **35**, 1423-1434 (1987).

## Profile


### ‣ Seong-Woo Kim (김성우)

He received the B.S. degree in school of electrical and electronic engineering from Yonsei University, Seoul, Korea, in 2008. He is currently undertaking a combined master-doctoral program in school of electrical and electronic engineering from Yonsei University, Seoul, Korea. His research interests include the area of signal processing, adaptive signal processing, and array signal processing.

### ‣ Young-Cheol Park (박영철)

He received the B.S., M.S., and Ph.D. degrees electronics engineering from Yonsei University, Seoul, Korea, in 1986, 1988, and 1993, respectively. He is currently a Professor of computer and telecommunications engineering at Yonsei University. His research interests include the area of signal processing, high-quality audio coding, and 3-D audio signal processing.

### ‣ Young-Soo Seo (서영수)

He received the B.S., M.S., and Ph.D. degrees in school of mechanical engineering from Pusan National University, Pusan, Korea, in 1999, 2001, and 2005, respectively. He is currently a researcher at Agency for Defense Development, gyeongsangnam-do, Korea.

### ‣ Dae Hee Youn (윤대희)

He received the B.S. degree in electronics engineering from Yonsei University, Seoul, Korea, in 1977, and the M.S. and Ph.D. degrees in electrical and electronic engineering from Kansas State University, Manhattan, KS, in 1979 and 1982, respectively. He is currently a Professor at school of electrical and electronic engineering, Yonsei University, Seoul, Korea. His research interests include the area of signal processing, adaptive signal processing, and 3-D audio signal processing.