This document is downloaded from DR-NTU, Nanyang Technological University Library, Singapore.

Title	Inverse system design based on the volterra modeling of a parametric loudspeaker system
Author(s)	Ji, Wei; Gan, Woon-Seng
Citation	Ji, W., & Gan, W. S. (2012). Inverse system design based on the Volterra modeling of a parametric loudspeaker system. NONLINEAR ACOUSTICS State-of-the-Art and Perspectives, 1474, 383-386.
Date	2012
URL	http://hdl.handle.net/10220/10102
Rights	© 2012 American Institute of Physics. This paper was published in NONLINEAR ACOUSTICS State-of-the-Art and Perspectives and is made available as an electronic reprint (preprint) with permission of American Institute of Physics. The paper can be found at the following official DOI: [http://dx.doi.org/10.1063/1.4749374]. One print or electronic copy may be made for personal use only. Systematic or multiple reproduction, distribution to multiple locations via electronic or other means, duplication of any material in this paper for a fee or for commercial purposes, or modification of the content of the paper is prohibited and is subject to penalties under law.

AIP Conference Proceedings

Inverse system design based on the Volterra modeling of a parametric loudspeaker system

Wei Ji and Woon-Seng Gan

Citation: AIP Conf. Proc. **1474**, 383 (2012); doi: 10.1063/1.4749374
View online: http://dx.doi.org/10.1063/1.4749374
View Table of Contents: http://proceedings.aip.org/dbt/dbt.jsp?KEY=APCPCS&Volume=1474&Issue=1
Published by the American Institute of Physics.

Additional information on AIP Conf. Proc.

Journal Homepage: http://proceedings.aip.org/
Journal Information: http://proceedings.aip.org/about/about_the_proceedings
Top downloads: http://proceedings.aip.org/dbt/most_downloaded.jsp?KEY=APCPCS
Information for Authors: http://proceedings.aip.org/authors/information_for_authors

ADVERTISEMENT

Downloaded 15 May 2013 to 155.69.4.4. This article is copyrighted as indicated in the abstract. Reuse of AIP content is subject to the terms at: http://proceedings.aip.org/about/rights_permissions

Inverse System Design Based on the Volterra Modeling of a Parametric Loudspeaker System

Wei Ji and Woon-Seng Gan

Digital Signal Processing Laboratory, School of Electrical and Electronic Engineering, Nanyang Technological University, Singapore

Abstract. Parametric loudspeaker systems have been widely used for projecting high directional audible sound beams. However, the nonlinear interaction among primary waves also generates harmonics, which distort the desired signal and degrade the sound quality. In order to investigate this inherent nonlinear mechanism, a baseband distortion model is developed from nonlinear system identification using an adaptive Volterra filter. For the conventional double-sideband amplitude modulation (DSBAM) technique, it is found that the harmonic distortion is largely attributed to the second harmonic. The adaptation results derived from both simulation and measurement indicate that the first few coefficients of the second-order kernel are dominant. Based on the Volterra model, a *p*th-order inverse system is designed to compensate the harmonic distortions present in the demodulated signal. Simulation and measurement results demonstrate that the harmonic distortion can be greatly reduced to an acceptable level when the inverse system is introduced with a suitable recursive order.

Keywords: *p*th-order inverse, Volterra filter, Parametric loudspeaker system
PACS: 43.60.Mn

INTRODUCTION

Due to the ability of generating high directional sound beams, parametric loudspeaker systems have attracted increasing interests and studies since Bennett and Blackstock experimentally applied the acoustic parametric array effect to the audible beam generation in air in the 1970s [1]. The ultrasonic primary waves of finite amplitude undergo nonlinear interaction when propagating in air, which gives rise to the formation of an array of virtual sources along the propagation path. In the 1980s, Yoneyama et al. introduced the DSBAM method to their prototype of a parametric loudspeaker [2]. It was subsequently found that the reproduced audible sound suffered from high nonlinear distortion which was primarily attributed to the second harmonic present in the secondary waves.

In order to reveal the inherent physical nonlinear phenomenon taking place in parametric loudspeaker systems, we use the adaptive Volterra filter to develop a nonlinear model accounting for the baseband signal distortion. The Volterra filter is capable of modeling a large class of nonlinear systems with a straight forward structure composed of different orders of kernels [3]. In this paper, an inverse system is designed to compensate the nonlinear harmonic distortion in a parametric loudspeaker system based on the Volterra model, and its performance is compared with DSBAM and SRAM techniques.

The rest of this paper is organized as follows. Section 2 reviews the adaptive

NONLINEAR ACOUSTICS State-of-the-Art and Perspectives
AIP Conf. Proc. 1474, 383-386 (2012); doi: 10.1063/1.4749374
© 2012 American Institute of Physics 978-0-7354-1081-7/$30.00

Downloaded 15 May 2013 to 155.69.4.4. This article is copyrighted as indicated in the abstract. Reuse of AIP content is subject to the terms at: http://proceedings.aip.org/about/rights_permissions

modeling of a parametric loudspeaker system using the Volterra kernels. In Section 3, a *p*th-order inverse system is recursively implemented and evaluated theoretically and experimentally. Finally, conclusion is drawn in Section 4.

VOLTERRA MODELING OF A PARAMETRIC LOUDSPEAKER SYSTEM

A Volterra filter can render the input-output equation of a nonlinear system as a polynomial series. Since the nonlinear distortion in parametric loudspeaker systems mainly results from the second harmonic, a truncated Volterra filter up to the 2nd-order kernel is used as the system model and can be expressed as [4]

$$y(n)=\sum_{m_1=0}^{N_1-1}h_1(m_1)x(n-m_1)+\sum_{m_2=0}^{N_2-1}\sum_{m_1=0}^{N_2-1}h_2(m_1,m_2)x(n-m_1)x(n-m_2), \qquad (1)$$

where N_1 and N_2 are the memory lengths of the kernels $\mathbf{H}_1[\cdot]$ and $\mathbf{H}_2[\cdot]$, respectively. Using the adaptive NLMS algorithm [5] and a cascaded adaptive structure [6], the Volterra model of a parametric loudspeaker system is developed based on the KZK mathematical model [7]. The modeling results in terms of the 1st- and 2nd-order kernel coefficients are given in Figure 1.

It can be seen from Figure 1 that the 1st-order kernel approximates the impulse response of a distortionless transmission for linear output, while the 2nd-order kernel indicates the quadratic nonlinearity is nearly memoryless.

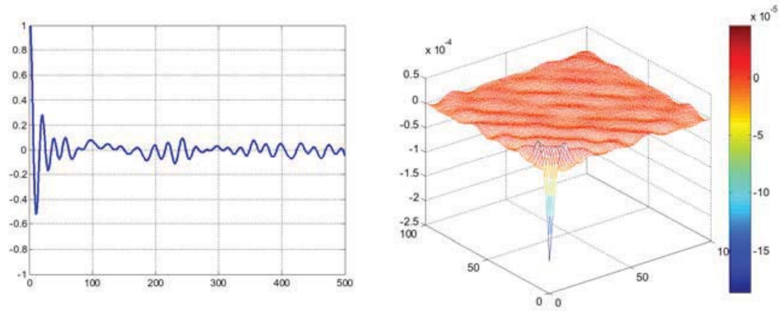

FIGURE 1. Coefficients of the 1st-order kernel (left), and the 2nd-order kernel (right).

DESIGN OF A *P*TH-ORDER INVERSE SYSTEM

Based on the Volterra modeling results, one way to compensate the distortion in a parametric loudspeaker system is to introduce a *p*th-order inverse system. Assume the parametric loudspeaker system can be represented by its Volterra model as $\mathbf{H}\{\cdot\}$. A preinverse system $\mathbf{G}_p\{\cdot\}$ can recover the input signal $x(n)$ with only residual higher-order products as

$$\mathbf{H}\left\{\mathbf{G}_p\left\{x(n)\right\}\right\}=x(n)+T_p\left\{x(n)\right\}, \qquad (2)$$

where $T_p\{\cdot\}$ represents a nonlinear system whose Volterra kernels are zero for order zero to p. In order to obtain the inverse system, Mathews et al. [8] proposed a

Downloaded 15 May 2013 to 155.69.4.4. This article is copyrighted as indicated in the abstract. Reuse of AIP content is subject to the terms at: http://proceedings.aip.org/about/rights_permissions

recursive structure for the implementation of $\mathbf{G}_p\{\cdot\}$, which is given as

$$\mathbf{G}_p\left\{x(n)\right\} = G_1\left[x(n)\right] - G_1\left[\mathbf{H}'_p\left\{\mathbf{G}_{p-1}\left\{x(n)\right\}\right\}\right], \tag{3}$$

where $G_1[x(n)] = H_1^{-1}[x(n)]$, and $\mathbf{H}'_p\{\cdot\} = H_2[\cdot] + H_3[\cdot] + \cdots + H_p[\cdot]$. For the parametric loudspeaker system, we only consider up to the quadratic nonlinearity so that $\mathbf{H}'_p\{\cdot\} \approx H_2[\cdot]$. Figure 2 shows the block diagram for recursively synthesizing $\mathbf{G}_p\{\cdot\}$, with p identical cells connected in cascade.

In the chain of the recursive synthesis, the current $G_k[x(n)]$ is obtained based on the previous cell output $G_{k-1}[x(n)]$ by computing the distortion amount subtracted from

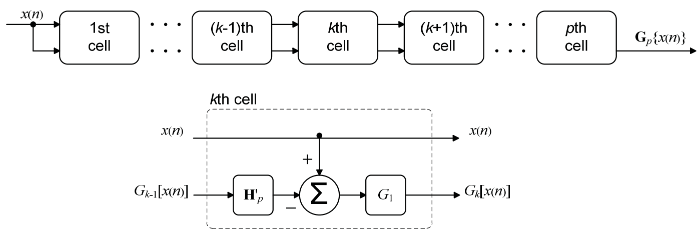

FIGURE 2. Recursive structure for the *p*th-order inverse (top), and implementation of the *k*th cell (bottom).

the original input $x(n)$, and then passing through G_1. The recursive computation will reach an acceptable level where more computation resource is consumed for only marginal distortion reduction.

Numerical simulations are conducted to evaluate the performance of the *p*th-order inverse system. The KZK equation is employed as the mathematical model, and a white-noise signal with frequency limited in the baseband ranging from 0 to 5 kHz is used as the input signal. The input signal is firstly passed through the *p*th-order inverse system, and then modulates a 40-kHz ultrasonic carrier using DSBAM scheme with modulation index $m = 1$ for reproducing the baseband signal with the maximum SPL [9]. Figure 3 shows the FFT spectra of the output signals of the parametric loudspeaker system without and with the *p*th-order inverse system for $p = 4$. It is seen that the second harmonic, whose frequency band extends from the baseband to 10 kHz, is significantly suppressed around 20 dB when the inverse system is introduced.

Next, experimental measurements are carried out in an anechoic chamber to compare the performances of the *p*th-order inverse system with the conventional DSBAM and SRAM techniques in terms of the SPL values and the THD levels for single tones from 1 kHz to 4 kHz as the input signals. Figure 4 shows the measurement results for p = 3, 4, and 5.

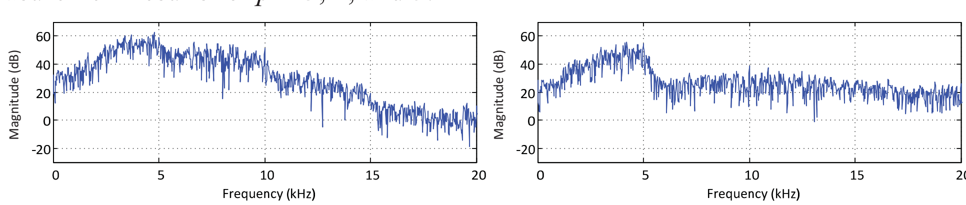

FIGURE 3. FFT spectra of the output signal without the *p*th-order inverse system (left), and with the *p*th-order inverse system for p = 4 (right).

Downloaded 15 May 2013 to 155.69.4.4. This article is copyrighted as indicated in the abstract. Reuse of AIP content is subject to the terms at: http://proceedings.aip.org/about/rights_permissions

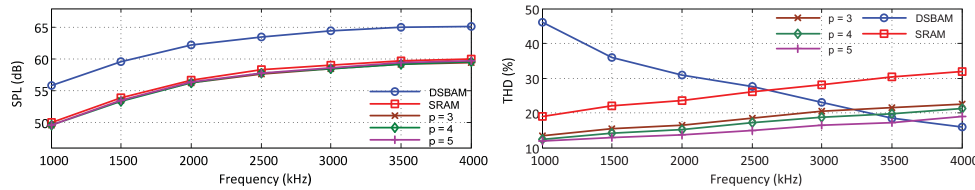

FIGURE 4. Measurement results of the SPL values (left) and THD levels (right) obtained from DSBAM, SRAM, and *p*th-order inverse system for $p = 3, 4, 5$.

The SPL values generated from the *p*th-order inverse system are almost identical to those from SRAM techniques, and the 6-dB discrepancies from the DSBAM curve can be compensated by multiplying the initial pressure with a factor of $\sqrt{2}$ according to the Berktay's solution [10]. From the THD curves, it can be observed that the *p*th-order inverse outperforms the SRAM and DSBAM for most frequency values when the actual emitter's response is taken into account, and higher order of p can yield lower THD levels. Therefore, the *p*th-order inverse technique can be employed as an alternative for reducing the harmonic distortion present in parametric loudspeaker systems. A suitable order can be selected as a trade-off between the acceptable THD amount and the computational load of available DSP processors.

CONCLUSION

In this paper, a *p*th-order inverse system is designed and implemented for compensating the harmonic distortion generated in a parametric loudspeaker system. Based on the Volterra model, the input signal is recursively predistorted in the inverse system before modulating the ultrasonic carrier. Simulation results showed that the second harmonic can be effectively suppressed, and experimental measurement results demonstrated that the *p*th-order inverse method can outperform the DSBAM and SRAM methods in terms of reducing the THD level.

REFERENCES

1. M. B. Bennett and D. T. Blackstock, "Parametric array in air," *J. Acoust. Soc. Am*. **57**(3), 562-568 (1975).
2. M. Yoneyama and J. Fujimoto, "The audio spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design," *J. Acoust. Soc. Am*. **73**(5), 1532-1536 (1983).
3. M. Schetzen, *The Volterra and Wiener theories of nonlinear systems*, New York: Wiley, 2006.
4. V. J. Mathews, "Adaptive polynomial filters," *IEEE Sig. Proc. Mag*. **8**, 10-26 (1991).
5. K. A. Lee, W. S. Gan, and S. M. Kuo, *Subband adaptive filtering: Theory and implementations*, New York: John Wiley & Sons, 2009.
6. W Ji, E. L. Tan, and W. S. Gan, "Baseband Distortion Modeling for a Parametric Loudspeaker System Using Volterra Kernels," in Proc. ASIPA Annual Summit and Conference 2011, Xi'an, China, 2011.
7. Y. S. Lee, "Numerical solution to the KZK equation for pulsed finite amplitude sound beams in thermoviscous fluids," Ph.D. Thesis, The University of Texas at Austin, 1993.
8. V. J. Mathews and G. L. Sicuranza, *Polynomial signal processing*, New York: John Wiley & Sons, 2000.
9. F. J. Pompei, "The use of airborne ultrasonics for generating audible sound beams," *J. Audio Eng. Soc*. **47**(9), 726-731 (1999).
10. H. O. Berktay, "Possible exploitation of non-linear acoustics in underwater transmitting applications," *J. Sound Vib*. **2**(4), 435-461 (1965).

Downloaded 15 May 2013 to 155.69.4.4. This article is copyrighted as indicated in the abstract. Reuse of AIP content is subject to the terms at: http://proceedings.aip.org/about/rights_permissions