Fast Evaluation of Preprocessing Methods of the Parametric Array Loudspeaker

Chuang Shi[1]* and Yoshinobu Kajikawa[1]

[1]Department of Electrical and Electronic Engineering, Kansai University, Osaka, Japan

*Corresponding email: r148005@kansai-u.ac.jp

Paper Number: P11000187

1. Abstract

This paper introduces a fast evaluation method for the parametric array loudspeaker (PAL). The present practice uses repeated time-consuming measurements to acquire total harmonic distortion (THD) and sound pressure level (SPL) curves for every preprocessing method. Instead, the proposed method requires only the frequency response of an ultrasonic emitter, since measuring one frequency response is relatively easier than measurements of many preprocessing methods.

***Index Terms*—Parametric array loudspeaker, double sideband modulation, square root modulation, modified amplitude modulation, ultrasonic emitter.**

2. Introduction

PAL is an efficient directional loudspeaker that is capable in many sound field control applications [1]. Its generation of a sharp audio beam is resultant from nonlinear acoustic effects in air. Difference frequency sound is generated in a bi-frequency ultrasonic beam and confined mainly within the ultrasonic beam [2]. This difference frequency sound is made audible in the PAL. Thus, the directivity of the PAL is similar to the ultrasonic beam, which is much narrower than the conventional sound device of the same size [3].

Figure 1 shows an experimental setup of the PAL [4]. The input of the PAL is read from an audio file and preprocessed before the conversion to an analog signal. The preprocessed input is amplified and drives an ultrasonic emitter. After that, a distorted waveform of the audio input is recovered in air. This process is also known as self-demodulation [5]. The bandwidth of the PAL is measured from 500 Hz to about 10 kHz. Such a bandwidth is often recognized as one of the major drawbacks of the PAL [6].

Past studies have primarily focused on developing preprocessing methods to suppress inherent nonlinear distortion of the PAL. Generally, the double sideband (DSB) modulation method was the first preprocessing method. It is widely applied nowadays, but the second harmonic distortion level is relatively high when the modulation index is large [7]. Hence, Kamakura *et al*. proposed the square root (SRT) modulation method to equalize the envelope of the DSB modulation method [8]. Due to the square root, the preprocessed input has an infinite bandwidth that can only be reproduced by an ideal ultrasonic emitter. The class of the modified amplitude modulation (MAM) methods were hence developed to cater to the bandwidth of a practical ultrasonic emitter [9].

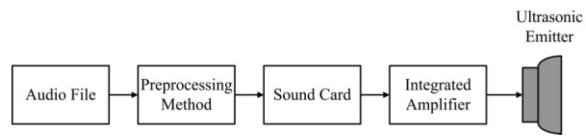

Figure 1 An experimental setup of the PAL (extracted from [4]).

In this paper, six of aforementioned preprocessing methods are comparatively evaluated by simulations and experiments. An objective score, as well as THD and SPL, is adopted to rate the overall performance of every preprocessing method [10]. The motivation of this paper is to figure out whether using numerical simulations based on prior knowledge of an ultrasonic emitter can achieve fast evaluation of the PAL.

3. Preprocessing Methods

Berktay's far-field solution provides that the self-demodulated pressure level of a modulated ultrasonic carrier is proportional to the second derivative of the squared envelope function, *i.e.*

$$p_d(z) \propto \frac{\partial^2}{\partial t^2} E^2\left(t - \frac{z}{c_0}\right), \tag{1}$$

where z is the distance from the ultrasonic emitter; c_0 is the sound speed; and $E(t)$ is the envelope function, usually assumed to have an unit amplitude [5].

The drawback of the DSB modulation method has been well understood [11, 12]. The primary pressure level of the DSB modulation method is written as

$$P_{DSB}(t) = \left[\frac{1}{1+m} + \frac{m}{1+m} x(t)\right] \cos\omega_c t, \tag{2}$$

where m is the modulation index; $x(t)$ is the audio input; and ω_c is the carrier frequency. Substituting (2) into (1) yields the self-demodulated pressure level of the DSB modulation method, which is given by

$$p_{DSB}(t) \propto \frac{\partial^2}{\partial t^2}\left[\frac{2m}{(1+m)^2}x(t)+\frac{m^2}{(1+m)^2}x^2(t)\right]. \quad (3)$$

The second term in square brackets contributes to the second harmonic distortion.

The SRT modulation method has been proposed to equalize the squared envelope function in Berktay's far-field solution [8]. Therefore, the primary pressure level of the SRT modulation method is derived as

$$P_{SRT}(t)=\left[\sqrt{\frac{1}{1+m}+\frac{m}{1+m}x(t)}\right]\cos\omega_c t. \quad (4)$$

Similarly, the self-demodulation pressure level of the SRT modulation method is given by

$$p_{SRT}(t) \propto \frac{\partial^2}{\partial t^2}\left[\frac{1}{1+m}+\frac{m}{1+m}x(t)\right]. \quad (5)$$

The SRT modulation method can eliminate inherent nonlinear distortion of the PAL in theory.

The class of MAM methods have been developed by introducing a quadrature term. Firstly, the original MAM method is derived as

$$P_{MAM}(t)=P_{DSB}(t)+\frac{m}{1+m}\hat{x}(t)\sin\omega_c t, \quad (6)$$

where the quadrature term is given by

$$\hat{x}(t)=\sqrt{1-x^2(t)}. \quad (7)$$

The original MAM method has the same difficulty as the SRT modulation method, due to the square root operation in the quadarture term.

Hence, the class of MAM methods are proposed to take the truncated Taylor expansion of (7), where the truncation order is determined by the bandwidth of an ultrasonic emitter. The quadrature term is rewritten as

$$\hat{x}(t) \approx \hat{x}_q(t)=\sum_{i=0}^{q}\frac{(2i)!}{(1-2i)(i!)^2 4^i}x^{2i}(t), \quad (8)$$

where q defines the truncation order and the order of the MAM method.

4. Experiment

The experimental setup of the PAL (see Figure 1) was adopted in the experiment. The carrier frequency was set at 40 kHz. The audio input was a sine sweep ranging from 1 kHz to 8 kHz with an interval of 500 Hz. Six preprocessing methods were implemented. They were the DSB, SRT modulation methods and MAM0 to MAM3 methods. For every preprocessing method, the modulation index varied from 0.1 to 1.0.

The experiment was carried out in a sound proof room (2.9 m $\times$ 3.1 m $\times$ 2.1 m). The microphone (B&K Type 4191L) was placed 3 meters away from the ultrasonic emitter (Mitsubishi Electronic Engineering Company). Figure 2 shows the measured frequency response of the ultrasonic emitter. Moreover, an finite impulse response (FIR) filter was designed to fit this measured frequency response.

THD, SPL and an objective score, were adopted as performance measures. The THD level describes the level of harmonic distortion in a nonlinear system, which is defined as

$$THD(f)=\sqrt{\frac{A^2(2f)+A^2(3f)+A^2(4f)+\cdots}{A^2(f)}}, \quad (9)$$

where $A(nf)$ denotes the amplitude of the fundamental frequency at f or the harmonic frequency at nf.

The SPL in this paper describes the amplitude of the fundamental frequency, which is expressed as

$$SPL(f)=20\log_{10}\frac{A(f)}{20\mu Pa}. \quad (10)$$

When the THD level and SPL are measured at a wide band of fundamental frequencies, the averaged THD level and SPL are given by their arithmetic means.

The objective score previously proposed quantizes the overall performance of a preprocessing method [10]. It is a known trade off that a lower modulation index results in less THD but also a lower SPL. The objective score is thus proposed as the weighted sum of the averaged THD level and SPL, *i.e.*

$$Score=\log_2\frac{\overline{THD(f)}}{1\%}+\frac{100-\overline{SPL(f)}}{3}. \quad (11)$$

The THD levels of the six preprocessing methods obtained in the experiment are plotted in Figure 3. The THD levels of the DSB modulation and MAM0 to MAM3 methods increase almost linearly with the modulation index. Figure 4 presents the measured

SPLs. It is observed that a lower modulation index leads to a lower SPL. Moreover, the SRT modulation method using the modulation index of 0.6 provides the lowest objective score as shown in Figure 5.

5. Numerical Simulation

The numerical simulation was further carried out. The simulation flow was based on a similar structure to the experimental setup. The audio input was read from the same sine sweep file used in the experiment. Preprocessing methods were carried out in MATLAB. The ultrasonic emitter were simulated as an FIR filter, of which the frequency response was shown in Figure 2. Berktay's far-field solution was adopted to be the nonlinear acoustic model, which would be replaced by Volterra filter identification in the future [13].

The THD levels, SPLs, as well as objective scores obtained in the numerical simulation are plotted in Figures. 6, 7, 8. In Figure 6, the simulated THD levels are slightly higher than the THD levels in the experiment. In Figure 7, the SPLs of the DSB and SRT modulation methods are about 1 dB lower than the SPLs in the experiment. In Figure 8, due to the aforementioned mismatches, the objective scores are compressed into a smaller range as compared to the experiment results. However, the effectiveness of the numerical simulation has been validated, since the SRT modulation method using the modulation index of 0.6 is recognized as the best preprocessing method in agreement with the experiment results.

6. Conclusions

Six preprocessing methods of the PAL, including the DSB, SRT modulation methods and the class of MAM methods with the order of 0 to 3, have been comparatively evaluated. THD, SPL and a previously proposed objective score are selected as performance measures. In both the experiment and simulation, the SRT modulation method using the modulation index of 0.6 demonstrates the best overall performance. The numerical simulation based on prior knowledge of an ultrasonic emitter achieves good predictions to the experiment results. Hence, a fast evaluation method of the PAL is proposed that measures the frequency response of the ultrasonic emitter only and carries out numerical simulations at a trivial cost of efforts.

Acknowledgements

This work is supported by MEXT-Supported Program for the Strategic Research Foundation at Private Universities, 2013-2017.

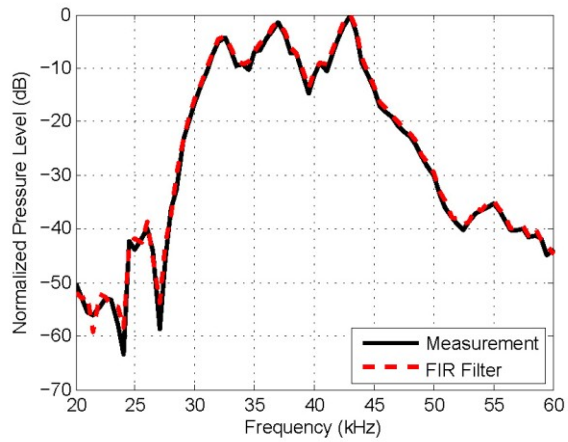

Figure 2 Frequency response of the ultrasonic emitter.

References

[1] C. Shi, Y. Kajikawa, and W. S. Gan, "An overview of directivity control methods of the parametric array loudspeaker," *APSIPA Trans. Signal Inform. Process.*, vol. 3, no. E20, pp. 1-12, 2014.

[2] P. J. Westervelt, "Parametric acoustic array," *J. Acoust. Soc. Am.*, vol. 35, no. 4, pp. 535-537, 1963.

[3] C. Shi and Y. Kajikawa, "A convolution model for computing the far-field directivity of a parametric loudspeaker array," *J. Acoust. Soc. Am.*, vol. 137, no. 2, pp. 777-784, 2015.

[4] C. Shi and Y. Kajikawa, "A comparative study of preprocessing methods in the parametric loudspeaker," *Proc. 2014 APSIPA ASC*, Siem Reap, Cambodia, 2014.

[5] H. O. Berktay, "Possible exploitation of nonlinear acoustics in underwater transmitting applications," *J. Sound Vib.*, vol. 2, no. 4, pp. 435-461, 1965.

[6] C. Shi, H. Mu, and W. S. Gan, "A psychoacoustical preprocessing technique for virtual bass enhancement of the parametric loudspeaker," *Proc. 38th ICASSP*, Vancouver, Canada, 2013.

[7] M. Yoneyama, J. Fujimoto, Y. Kawamo, and S. Sasabe, "The audio spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design," *J. Acoust. Soc. Am.*, vol. 73, no. 5, pp. 1013-1020, 1983.

[8] T. Kamakura, M. Yoneyama, and K. Ikegaya, "Developments of parametric loudspeaker for practical use," *Proc. 10th Int. Symp. Nonlinear Acoust.*, Kobe, Japan, 1984.

[9] Y. H. Liew, "Signal processing techniques for sound eproduction in parametric arrays," Master Thesis, Nanyang Technological University, Singapore, 2002.

[10] C. Shi and Y. Kajikawa, "Objective evaluation of preprocessing methods of the parametric array loudspeaker," *Proc. 2015 Autumn Meeting Acoust. Soc. Jpn.*, Aizuwakamatsu, Japan, 2015.

[11] D. Ikefuji, M. Nakayama, T. Nishiura, and Y. Yamashita, "Weighted double sideband modulation toward high quality audible sound on parametric loudspeaker," *Proc. 38th ICASSP*, Vancouver, Canada, 2013.

[12] P. Ji, E. L. Tan, W. S. Gan, and J. Yang, "A comparative analysis of preprocessing methods for the parametric loudspeaker based on the Khokhlov-Zabolotskaya-Kuznetsov equation for speech reproduction," *IEEE Trans. Audio Speech Lang. Process.*, vol. 19, no. 4, pp. 937-946, 2011.

[13] C. Shi and Y. Kajikawa, "Identification of the parametric array loudspeaker with a Volterra filter using the sparse NLMS algorithm," *Proc. 40th ICASSP*, Brisbane, Australia, 2015.

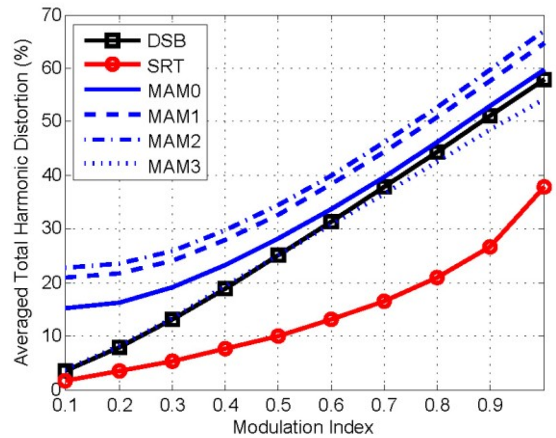

Figure 3 THD levels of preprocessing methods obtained in the experiment (extracted from [10]).

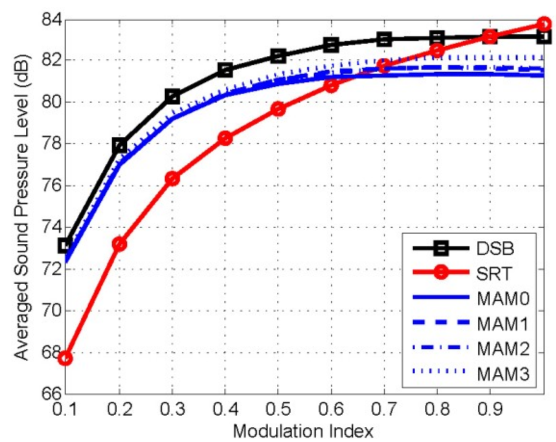

Figure 4 SPLs of preprocessing methods obtained in the experiment (extracted from [10]).

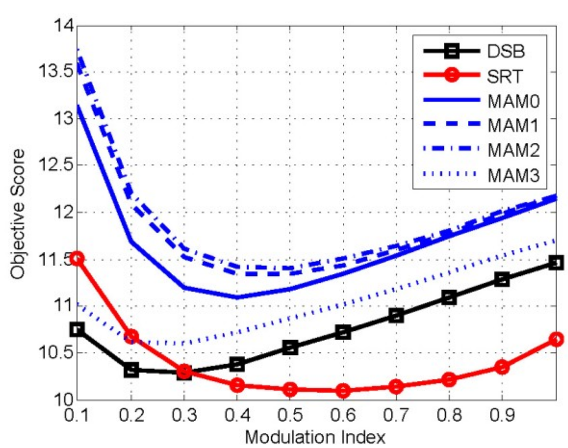

Figure 5 Objective scores of preprocessing methods obtained in the experiment (extracted from [10]).

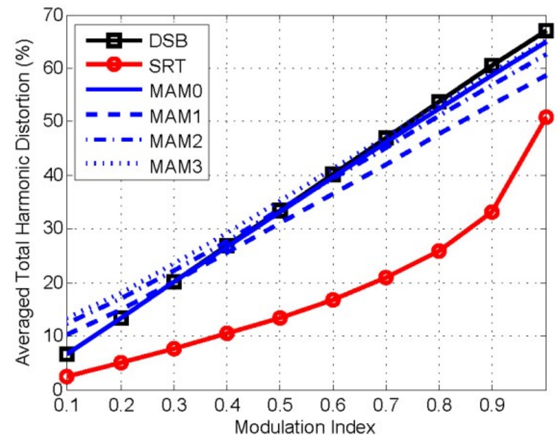

Figure 6 THD levels of preprocessing methods obtained in the simulation.

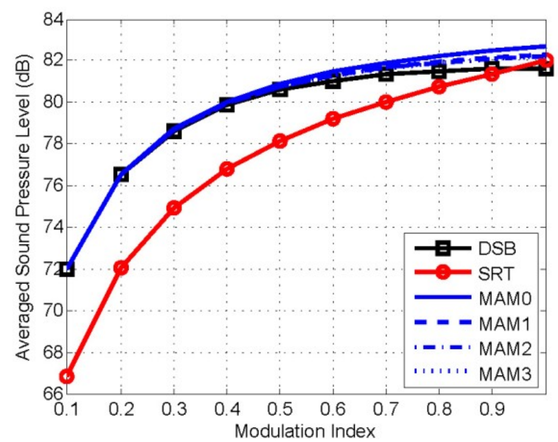

Figure 7 SPLs of preprocessing methods obtained in the numerical simulation.

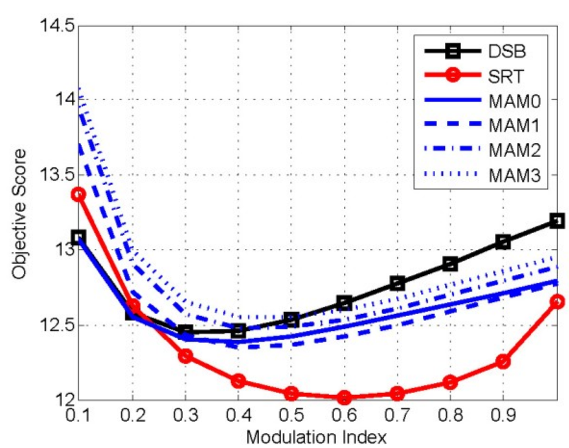

Figure 8 Objective scores of preprocessing methods obtained in the numerical simulation.