A Preprocessing Method to Increase High Frequency Response of A Parametric Loudspeaker

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Abstract— Unlike a conventional loudspeaker, a parametric loudspeaker whose principle is derived from nonlinear acoustic effects in air can create a controllable sound beam with a compact emitter. Therefore, the parametric loudspeaker is advantageous in deployments where personal audio zones are in demand. However, it is a dilemma that the sound generation principle of parametric loudspeaker leads to a narrow bandwidth of itself, which hinders its popularization in applications requiring high fidelity. In this paper, we propose a tentative treatment of harmonic distortions incurred in parametric loudspeakers, whereby the asymmetrical amplitude modulation (AAM) method is used to extend the high frequency response by doubling the higher cut-off frequency.

I. INTRODUCTION

The fundamental theory of a parametric loudspeaker is based on the parametric array effect in air [1, 2]. When two primary signals at close frequencies are transmitted from an ultrasonic emitter, the sum and difference frequency waves as well as other harmonics are produced due to the nonlinear interaction between the primary signals. Ultrasonic waves decay rapidly due to the absorption in air, but the difference frequency wave being lower in frequency is less attenuated. Because a virtual end-fire array is formed by the vibrating air molecules in front of the ultrasonic emitter, sharp radiation patterns are observed at low frequencies.

A parametric loudspeaker usually consists of three parts, as shown in Fig. 1. They are the processor, the amplifier, and the ultrasonic emitter. The processor, which can be digital, analog, or hybrid, realizes preprocessing methods to reduce harmonic distortions to an acceptable sound quality level [3]. Class-D amplifiers operating in the ultrasonic range are often used to drive the ultrasonic emitters. The latter is normally made of a polyvinylidene fluoride film [4] or an array of piezoceramic transducers [5].

In the past, several acoustic equations have been developed to mathematically describe the nonlinear process on or off the axis of a parametric loudspeaker [1-2, 6-8]. Therein, Berktay's far-field solution is one of the most widely used [6]. It is an on-axis one-dimensional model, given by

$$p_{d}(r,0) = \frac{p_{0}^{2}}{r} SK \frac{\partial^{2}}{\partial t^{2}} \Big[E^{2}(\tau) \Big], \qquad (1)$$

where p_d is the pressure level of the demodulated difference frequency wave; p_0 is the initial pressure level of the primary waves; r is the distance between the observation point and the



Figure 1 Block diagram of the parametric loudspeaker.



Figure 2 Frequency response of a commercial parametric loudspeaker measured in a recording room.

surface of the ultrasonic emitter; *K* is a scalar relating to a few acoustic parameters of air; *S* is the cross-sectional area of the primary beams; $E(\tau)$ is the envelope function varying slowly as compared to the carrier wave; and τ is the retarded time.

Two well-known limitations of the parametric loudspeaker are readily observed from (1): firstly, the square operator on $E(\tau)$ results in unavoidable harmonic distortions; secondly, the second derivative performs as a high pass filter, leading to the poor low frequency response of a parametric loudspeaker [9]. Preprocessing methods can be applied to overcome these two limitations. In the past two decades, preprocessing methods have been proposed to suppress harmonic distortions for the parametric loudspeaker [10-16, 18-20]. Only until recently, a psychoacoustical preprocessing technique was validated to be able to generate virtual bass effect at one octave lower than the cut-off frequency of a parametric loudspeaker [21]. A commercial parametric loudspeaker unit was measured in a sound-proof recording room, and its frequency responses at discrete distances were plotted in Fig. 2. The lower and upper bounds of the usable frequency range were observed at 500 kHz and 14 kHz, respectively, where the sound pressure level dropped to -25 dB of the maximum level. This inspired us to look for a solution to extend the higher cut-off frequency, which has not gained enough attention in past studies.

This paper is organized as follows. Section 2 overviews the past literatures of preprocessing methods to reduce harmonic distortions incurred in the parametric loudspeaker. In Section 3, the asymmetrical amplitude modulation (AAM) method is proposed to make use of the second harmonics, which are by-produced from the parametric array effect, to extend the high frequency response of a parametric loudspeaker. Preliminary results are obtained through measurement to validate the AAM method. Finally, Section 4 concludes the main contributions of this paper.

II. OVERVIEW OF PREPROCESSING METHODS FOR THE PARAMETRIC LOUDSPEAKER

Inventors of the "audio spotlight" [1] has originally noticed that the frequency response of a parametric loudspeaker had a slope of 12 dB per octave, and the second harmonic distortion ratio was proportional to the modulation index used in the double sideband modulation (DSBAM, see Fig. 3(a)). Since then, a range of preprocessing methods have been thoroughly studied for the parametric loudspeaker. In 1984, Kamakura et al. [10] noted that the square operator contributed most of the harmonic distortions. Therefore, a square root operation was suggested to pre-distort the envelope function in the DSBAM. Such a preprocessing method is now known as the square root amplitude modulation (SRAM, see Fig. 3(b))that substantially suppress the second harmonics. Later, a variant of the SRAM was again proposed by Kamakura et al. [11] to reduce the average power consumption of a parametric loudspeaker to one third as compared to the one using the DSBAM.

A double integral operator was introduced into the SRAM as an equalizer to further reduce harmonic distortions [12, 13]. In Pompei's measurement [13], the double integral operator reduced the total harmonic distortion (THD) of the SRAM to below 5%. However, the necessity of such a double integral operator is debatable. It is recommended for the SRAM only when ultrasonic emitters have flat frequency responses across a wide frequency range.

One drawback of the SRAM is that even though the square root operator results in an infinite bandwidth of the modulated signal, ultrasonic emitters can only achieve sufficiently high radiation power within a narrow frequency range [4-5]. Thus, the single sideband amplitude modulation (SSBAM, see Fig. 3(c)) was proposed for parametric loudspeakers [4, 14]. The sign of multiplier on the Hilbert transformed branch indicates two types of SSBAM, known as the lower and the upper sideband amplitude modulation (LSBAM and USBAM) [15]. In theory, the THD value of the SSBAM is zero when the modulating input is a single tone. However, in measurements, the THD values are higher than the theoretical value, which



(d) Modified Amplitude Modulation (MAM) [19]

Square Root

Figure 3 Block diagrams of four preprocessing methods: (a) the DSBAM; (b) the SRAM; (c) the SSBAM; and (d) the MAM.

could be due to the approximations made in the derivation of the Berktay's far-field solution [16] or system errors in the implementation of a parametric loudspeaker [17, 18]. Sakai *et al.* [16] summarized that the SSBAM was preferred when the reproduced sound level was relatively low, while the DSBAM outperformed the SSBAM when the reproduced sound level was large. Similar to the SRAM, there was also an energyefficient variant of the SSBAM, in which the amplitude of the carrier signal was adjusted according to the envelope of the audio signal. By using this method, the power consumption of the parametric loudspeaker and the ultrasonic exposure to the users could be significantly reduced [16].

The SSBAM is essentially a quadrature modulation method. Referring to the quadrature modulation, Liew [19] proposed a class of modified amplitude modulation (MAM) preprocessing methods, which were adaptable to bandwidths of a wide range of ultrasonic emitters. The core MAM contains a pre-distorted term modulated with the orthogonal carrier. When different orders of Taylor expansion of the pre-distorted term substitute itself, the core MAM is extended to a class of methods. The block diagram of the MAM is shown in Fig. 3(d). It possesses the advantages of both the SSBAM and the SRAM, therefore achieves a similar THD performance as compared to the SSBAM (which theoretically is zero) and retains a low level of the inter-modulation distortion (IMD) as compared to other preprocessing methods. Another method proposed to improve on the SSBAM was the recursive SSBAM (RSSBAM), which was conceptualized by Croft *et al.* [20]. As a recursive algorithm, its performance depended on the total iteration times and the model equation used in each iteration. The trade-off between high fidelity and computational complexity is an unavoidable issue. So far, the nonlinear model used in most analyses of the RSSBAM is as simple as a square operator [3, 20].

III. PROPOSED PREPROCESSING METHOD TO INCREASE HIGH FREQUENCY RESPONSE

As introduced in Section 2, the second harmonics generated in the DSBAM are totally unwanted. However, in this paper, we propose to make use of selected second harmonics to extend the high frequency response of a parametric loudspeaker, and the second harmonics remaining unwanted can be suppressed by using the SSBAM.

A. Retaining the Second Harmonics Using the DSBAM

The envelope function of the DSBAM is given by

$$E_{DSBAM1}(\tau) = 1 + m \cos \omega_d \tau, \qquad (2)$$

where *m* is the modulation index (see Fig. 3(a)), and ω_d is the frequency of the single tone input. Thus, the self-demodulated wave can be computed by substituting (2) in (1), i.e.

$$\omega_{DSBAM1} \propto 2m\omega_d^2 \cos \omega_d \tau + 2m^2 \omega_d^2 \cos 2\omega_d \tau.$$
 (3)

As noted from (3), the modulation index can be set around 0.7 to suppress the second harmonic at $2\omega_d$ but maintain sufficient output level of the reproduced wave at ω_d [1, 9].

However, when two sine tones at ω_1 and ω_2 are fed into the ultrasonic emitter, the inter-modulation takes place due to the inherent nonlinearity of the parametric loudspeaker. Thus, the envelope function of the DSBAM is changed to

$$E_{DSBAM2}(\tau) = 1 + m_1 \cos \omega_1 \tau + m_2 \cos \omega_2 \tau, \qquad (4)$$

where m_1 and m_2 are modulation indices of the two sine tones at ω_1 and ω_2 . After we substitute (4) into (1), the expression of the self-demodulated wave is obtained as

$$p_{DSBAM2} \propto 2m_1 \omega_1^2 \cos \omega_1 \tau + 2m_1^2 \omega_1^2 \cos 2\omega_1 \tau + 2m_2 \omega_2^2 \cos \omega_2 \tau + 2m_2^2 \omega_2^2 \cos 2\omega_2 \tau + m_1 m_2 (\omega_1 - \omega_2)^2 \cos(\omega_1 - \omega_2) \tau + m_1 m_2 (\omega_1 + \omega_2)^2 \cos(\omega_1 + \omega_2) \tau.$$
(5)

In (5), there are two additional harmonics generated besides the two fundamental tones and their second harmonics. These two harmonics are resulted from inter-modulation effects, and their amplitudes are proportional to the square of frequencies. Hence, for being higher in frequency, the sum frequency at ω_1 + ω_2 is dominant over the difference frequency.

Measurements are carried out using four cases of inputs (see Fig. 4), which are (1) $\omega_d = 7$ kHz using the DSBAM with m = 0.5; (2) $\omega_d = 14$ kHz using the DSBAM with m = 0.5; (3) $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the DSBAM with $m_1 = m_2 = 0.5$; (4) $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the DSBAM with $m_1 = m_2 = 0.5$; (4) $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the DSBAM with $m_1 = 0.3$ and $m_2 = 0.7$, respectively. Spectrums of measured self-demodulated waves are plotted in Fig. 5. A comparison between case 1 and case 2 demonstrates a drop of 12.6 dB in





Figure 4 Generated inputs of a parametric loudspeaker using the DSBAM.



Figure 5 Spectrums of the measured self-demodulated waves of the DSBAM.

the sound pressure level of the self-demodulated wave due to the limited bandwidth of the ultrasonic emitter. But in case 3, the self-demodulated wave at 14 kHz is 1 dB higher than that in case 2, as a result of the second harmonic generated by the 7 kHz tone. Furthermore, case 4 validates the controllability of the relative amplitude between the self-demodulated 7 kHz and 14 kHz tones. The sound level difference is reduced from 12.6 dB to 5.3 dB.

B. Suppressing Unwanted Harmonics Using the SSBAM

The modulation function of the SSBAM is given by

$$F_{SSBAM1}(\tau) = (1 + m\cos\omega_d \tau)\cos\omega_c \tau$$
(6)

$$\pm m \sin \omega_d \tau \cos \omega_c \tau$$
,

where ω_c is the frequency of the carrier wave (see Fig. 3(c)). Using the auxiliary angle formula [18], the envelope function of the SSBAM is obtained by

$$E_{SSBAM1}(\tau) = \sqrt{1 + m^2 + 2m\cos\omega_d \tau}.$$
 (7)

By substituting
$$(7)$$
 in (1) , we can obtain

$$p_{SSBAM1} \propto 2m\omega_d^2 \cos \omega_d \tau.$$
 (8)



Figure 6 Block diagrams of the proposed AAM.

It is readily observed in (8) that SSBAM ideally leads to zero THD level.

Similar to the IMD analysis of the DSBAM, when two sine tones at ω_1 and ω_2 are inputted into a parametric loudspeaker, the modulation function of the SSBAM is given by

$$F_{SSBAM2}(\tau) = (1 + m_1 \cos \omega_1 \tau + m_2 \cos \omega_2 \tau) \cos \omega_c \tau$$

$$\pm (m_1 \sin \omega_1 \tau + m_2 \sin \omega_2 \tau) \sin \omega_c \tau.$$
(9)

Hence, the envelope function of the SSBAM is given by

$$E_{SSBAM2}(\tau) = \sqrt{\begin{array}{l} 1 + m_1^2 + m_2^2 \\ +2m_1 \cos \omega_1 \tau + 2m_2 \cos \omega_2 \tau \\ +2m_1 m_2 \cos \omega_1 \tau \cos \omega_2 \tau \\ +2m_1 m_2 \sin \omega_1 \tau \sin \omega_2 \tau \end{array}}, \quad (10)$$

where m_1 and m_2 are modulation indices of the two sine tones at ω_1 and ω_2 . Thus, the self-demodulated wave is obtained by substituting (10) in (1), i.e.

$$p_{SSBAM2} \propto 2m_1 \omega_1^2 \cos \omega_1 \tau + 2m_2 \omega_2^2 \cos \omega_2 \tau + 2m_1 m_2 (\omega_1 - \omega_2)^2 \cos (\omega_1 - \omega_2) \tau.$$
(11)

It is observed in (11) that the SSBAM only generates one IMD component, as compared to the DSBAM that generates two IMD components. Hence, the IMD level of the SSBAM is expected to be less than that of the DSBAM.

С. Asymmetrical Amplitude Modulation and Its Validation

By combining the DSBAM with the SSBAM, we propose an Asymmetrical Amplitude Modulation (AAM) aiming to achieve individual control of the amplitudes of different fundamental frequencies and their corresponding second harmonics.

In the proposed AAM, the fundamental frequencies leading to desired harmonics are preprocessed using the DSBAM, and the SSBAM is applied to the other frequency components. For instance, two sine tones at ω_1 and ω_2 are reproduced from the parametric loudspeaker. The harmonic at $2\omega_1$ is assumed to be useful in the extension of the frequency response, while the harmonic $2\omega_2$ is assumed to be unwanted. As shown in Fig. 6, a spectrum detector splits the modulating input signal into two paths, which are preprocessed to retain and reduce harmonics. Hence, the modulation function of the AAM is given by

Waveforms of Four Cases of Inputs



Figure 7 Generated inputs of a parametric loudspeaker using the AAM.



Figure 8 Spectrums of the measured self-demodulated waves of the AAM.

$$F_{AAM}(\tau) = (1 + m_1 \cos \omega_1 \tau + m_2 \cos \omega_2 \tau) \cos \omega_c \tau$$

$$\pm m_2 \sin \omega_2 \tau \sin \omega_c \tau.$$
(12)

Thus, the envelope function of the AAM is computed as

$$E_{AAM}(\tau) = \sqrt{\frac{1 + m_2^2 + 2m_1 \cos \omega_1 \tau + m_1^2 \cos^2 \omega_1 \tau}{+ 2m_2 \cos \omega_2 \tau + 2m_1 m_2 \cos \omega_1 \tau \cos \omega_2 \tau}}.$$
 (13)

By substituting (13) in (1), we can obtain

$$p_{AAM} \propto 2m_1\omega_1^2 \cos \omega_1 \tau + 2m_1^2\omega_1^2 \cos 2\omega_1 \tau + 2m_2\omega_2^2 \cos \omega_2 \tau + m_1m_2(\omega_1 - \omega_2)^2 \cos(\omega_1 - \omega_2)\tau + m_1m_2(\omega_1 + \omega_2)^2 \cos(\omega_1 + \omega_2)\tau.$$
(14)

It is observed in (14) that the inter-modulation in the AAM is identical to that in the DSBAM, thus the IMD level of the AAM is same as the DSBAM and higher than the SSBAM in theory.

Measurements are carried out to verify the proposed AAM. Four cases of audio tones are reproduced from the parametric loudspeaker (see Fig. 7), which are (1) $\omega_d = 4$ kHz using the SSBAM with m = 0.5, $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the

DSBAM with $m_1 = m_2 = 0.5$; (2) $\omega_d = 4$ kHz using the SSBAM with m = 0.5, $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the DSBAM with $m_1 = 0.3$ and $m_2 = 0.7$, respectively; (3) $\omega_d = 6$ kHz using the SSBAM with m = 0.5, $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the DSBAM with $m_1 = m_2 = 0.5$; (4) $\omega_d = 6$ kHz using the SSBAM with m = 0.5, $\omega_1 = 7$ kHz and $\omega_2 = 14$ kHz using the DSBAM with $m_1 = 0.3$ and $m_2 = 0.7$, respectively. Spectrums of the self-demodulated waves obtained in this group of measurements are plotted in Fig. 8. A comparison between case 1 and case 2 shows that by using the AAM, we are able to control the amplitudes of the harmonics of different fundamental frequencies. In case 1 and case 2, all the harmonics of 4 kHz tone are relatively low, as a result of using the SSBAM. But the desired second harmonic of the 7 kHz tone can achieve a relatively high sound pressure level. The most significant inter-modulation component occurs at 11 kHz, being the sum frequency of the two input tones at 4 kHz and 7 kHz. However, their difference frequency at 3 kHz is not found in the measurements. Similar trends are observed in case 3 and case 4. The second harmonic at 14 kHz is able to be generated at a relatively high level. The inter-modulation component is found at 13 kHz, which is the sum frequency of the two input tones at 6 kHz and 7 kHz. The proposed AAM has demonstrated its capability in improving the high frequency response of a parametric loudspeaker.

IV. CONCLUSIONS

In this paper, a proposed preprocessing method, the AAM, was presented to physically improve the ability of generating high frequency audio contents from a parametric loudspeaker despite the narrow bandwidth of ultrasonic emitter used inside. This proposed AAM combines the DSBAM and the SSBAM, and adopts a spectrum detector to split the input signal into two portions. The first portion is preprocessed using the DSBAM to retain its second harmonics, which are used for compensating the high frequency response of the parametric loudspeaker. The second portion is preprocessed by using the SSBAM to suppress remaining unwanted second harmonics. Experimental results demonstrated the feasibility of individual amplitude control of second harmonics resulted from different fundamental frequencies by using the proposed method. This work will be developed to work with music or other wideband contents in a real-time implementation. Listening tests will be conducted for evaluation the proposed method in the future.

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