Proceedings of Symposium on Ultrasonic Electronics, Vol. 30 (2009) pp. 89-90
18-20 November, 2009

The effects of the non-ideal ultrasound transducer on the performance of a parametric loudspeaker

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1. Introduction

Parametric loudspeaker can project an audible sound to a specific area without disturbing the people in adjacent locations, which achieves much attention recently. Unlike ordinary loudspeaker, audio signal is coupled with a high frequency carrier signal by modulation techniques firstly. And then the modulated ultrasound is played back by a transducer at a sufficiently high sound pressure level. Finally, A highly directive audible sound is created due to the nonlinear effects in ultrasound propagation[1)].

Parametric loudspeaker was firstly fabricated by M. Yoneyama et al. in 1980s, where the double sideband amplitude modulation (DSB-AM) technique was employed to couple the low frequency audio signal with a high frequency carrier signal[2)]. As a result of using DSB-AM modulation technique, these early devices exhibited extreme distortion in the demodulated signal. Kamakura suggested two methods to eliminate the distortion[3)]. One is the square root modulation technique (SRAM). Based on this technique, Pompei firstly fabricated the low distortion parametric loudspeaker named audio spotlight in 1998[4)]. The other is single-sideband amplitude modulation (SSB-AM) technique[5)], which is adopted by ATC Company.

Much work has been done to analyze the performance of different modulation techniques based on the Berktay's solution[6)]. However, most of them assume that the ultrasound transducer is ideal. In this paper, the effects on the performance of parametric loudspeaker caused by non-ideal ultrasound transducer is discussed, which provides a guide to design distortion-reduction preprocessing technique for a given ultrasound transducer.

2. Analysis

The block diagram of parametric loudspeaker with DSB-AM modulation technique is shown in **Fig. 1.** The audio signal $g(t)$ is modulated by DSB-AM technique and then radiated into air from

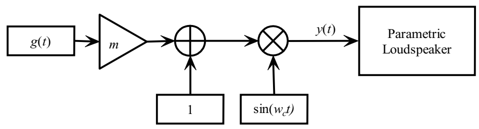

Fig. 1. The block diagram of parametric loudspeaker with DSB-AM modulation technique.

the ultrasound transducer. The input signal of ultrasound transducer is

$$y(t) = \mathrm{Im}\left\{[1 + mg(t)]e^{j\omega_c t}\right\}, \tag{1}$$

where m is modulate coefficient, ω_c is carrier frequency, and Im{} is the imaginary part of complex value. Assume

$$x(t) = 1 + mg(t)\ , \tag{2}$$

so equ. (1) can be rewritten as

$$y(t) = \mathrm{Im}\left\{\frac{1}{2\pi}\int_{-\infty}^{\infty} X(\omega)e^{j(\omega+\omega_c)t}d\omega\right\}, \tag{3}$$

where $X(\omega)$ is the Fourier transformer of $x(t)$.

Consider the ultrasound transducer response, the primary wave pressure $P_1(t)$ emitted by ultrasound transducer is

$$P_1(t) = \mathrm{Im}\left\{\frac{e^{j\omega_c t}}{2\pi}\int_{-\infty}^{\infty} X(\omega)H(\omega_c + \omega)e^{j\omega_c t}d\omega\right\} \tag{4}$$

where $H(\omega)$ is the frequency response of the ultrasound transducer. According to Berktay's solution, the frequency content of the selfdemodulated secondary wave pressure is

$$P_2(t) = \frac{k}{2\pi^2}\mathrm{Re}\left\{\int_{-\infty}^{\infty}\int_{\omega_1}^{\infty}(\omega_1 - \omega_2)^2 C(\omega_1, \omega_2)e^{j(\omega_1-\omega_2)t}d\omega_2 d\omega_1\right\} \tag{5}$$

$$C(\omega_1, \omega_2) = X(\omega_1)X^*(\omega_2)H(\omega_c + \omega_1)H^*(\omega_c + \omega_2) \tag{6}$$

where subscript "*" denotes complex conjugate and Re{} is the real part of complex value.

According to equ. (5), the amplitude of demodulated signal pressure $P_s(\omega_0, t)$, harmonic distortion signal $P_h(\omega_0, t)$ and intermodulation distortion signals $P_m(\omega_1 \pm \omega_2, t)$ are obtained, which are

$$P_s(\omega_0,t) = \frac{km\omega_0^2 A(\omega_c)}{2}\times \sqrt{A^2(\omega_c+\omega_0) + A^2(\omega_c-\omega_0) + 2A(\omega_c+\omega_0)A(\omega_c-\omega_0)\cos[\theta(\omega_0)]}\ , \tag{7}$$

$$P_h(2\omega_0, t) = km^2\omega_0^2 A(\omega_c + \omega_0)A(\omega_c - \omega_0)\ , \tag{8}$$

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$$P_m(\omega_1-\omega_2,t)=\frac{km^2(\omega_1-\omega_2)^2}{2}\times\sqrt{B^2(\omega_1,\omega_2)+B^2(-\omega_1,-\omega_2)+2B(\omega_1,\omega_2)B(-\omega_1,-\omega_2)\cos[\theta(\omega_1,\omega_2)]}\,, \quad (9)$$

and

$$P_m(\omega_1+\omega_2,t)=\frac{km^2(\omega_1+\omega_2)^2}{2}\times\sqrt{B^2(\omega_1,-\omega_2)+B^2(-\omega_1,\omega_2)+2B(\omega_1,-\omega_2)B(-\omega_1,\omega_2)\cos[\theta(\omega_1,-\omega_2)]} \quad (10)$$

respectively, where $A(\omega)$ is the magnitude response of ultrasound transducer, $\varphi(\omega)$ is the phase response,

$$\theta(\omega_0)=2\varphi(\omega_c)-\varphi(\omega_c+\omega_0)-\varphi(\omega_c-\omega_0), \quad (11)$$

$$B(\omega_1,\omega_2)=A(\omega_c+\omega_1)A(\omega_c+\omega_2), \quad (12)$$

and

$$\theta(\omega_1,\omega_2)=\varphi(\omega_c+\omega_1)+\varphi(\omega_c-\omega_1)-\varphi(\omega_c+\omega_2)-\varphi(\omega_c-\omega_2). \quad (13)$$

For SSB modulation technique, the amplitude of demodulated signal pressure, harmonic distortion signal and intermodulation distortion signals can derived by in a similarly way, which are

$$P_s(\omega_0,t)=km\omega_0^2A(\omega_c)A(\omega_c+\omega_0), \quad (14)$$

$$P_h(2\omega_0,t)=0, \quad (15)$$

$$P_m(\omega_1-\omega_2,t)=km^2(\omega_1-\omega_2)^2A(\omega_c+\omega_1)A(\omega_c+\omega_2), \quad (16)$$

and

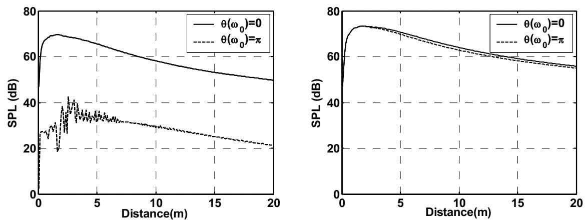
$$P_m(\omega_1+\omega_2,t)=0. \quad (17)$$

respectively.

3. Simultion

In order to test the effectiveness of the above theory analysis, a set of simulations were done using the KZK equation. In the simulation, the radius of the piston source is 0.1 m, the primary wave pressure is 120 dB, the carrier frequency is 40 kHz, and the temperature and the relative humidity is set at 28ºC and 60%, respectively. The input signal is a sine signal with frequency 1kHz. The output signals produced by two different transducers are compared. The magnitude responses of the two ultrasound transducer are the same while the phase responses are different.

The phase response of the first ultrasound transducer at 39 kHz, 40 kHz, and 41 kHz are -90 degrees, 0 degree and 90 degrees, which satisfies $\theta(\omega_0)=0$ and the phase responses of the secondary transducer are 90 degrees, 0 degree and 90 degrees, which satisfies $\theta(\omega_0)=\pi$. According to equ. (7), for DSB-AM modulation technique, the demodulated signal reaches its maximum value for $\theta(\omega_0)=0$ and minimum vale for $\theta(\omega_0)=\pi$. As shown in **Fig. 2(a)**, the demodulated signal produced by the first ultrasound transducer (dotted line) is much less than the one produced by the secondary ultrasound transducer (dashed line). While for the SSB-AM modulation technique, the demodulated signal is



(a) (b)

Fig. 2. The demodulated signal produced by different ultrasound transducer using (a) DSB-AM modulation, (b) SSB-AM modulation.

almost not affected by the phase response of transducer as shown in equ. (14) and **Fig.2(b)**. Simultions were also done for harmonic distortion signal and intermodulation distortion signals. The results are not shown because of the limited length of the paper. All the simutions validate the analytical results.

4. Conclusion

In this paper, the effects on the performance of parametric loudspeaker caused by non-ideal transducer are analyzed. For DSB-AM modulation technique, the amplitude of demodulation signal and intermodulation signal are effected by both phase response and magnitude response of ultrasound transducer. The harmonic signal for DSB-AM modulation technique is only effected by the phase response. While for SSB-AM, the amplitude of demodulation signal and intermodulation signal are only affected by the magnitude response of transducer. The harmonic signal is still zero.

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