

# Parametric Array in Air: Distortion Reduction by Preprocessing

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**Abstract:** Although several examples of using parametric generation from an ultrasonic carrier as an audio loudspeaker have been discussed in the literature, practical constraints have received little attention. We discuss an effect of transducer bandwidth that must be overcome in order for the “audio spotlight” to become a practical, high-fidelity source for reproduction of music.

## INTRODUCTION

In a parametric array, highly-directional low-frequency sound is generated by the self-demodulation of an intense, amplitude-modulated high-frequency sound beam as a result of nonlinear propagation effects. The term “audio spotlight” was introduced by Yoneyama et al. (1) for a parametric array in air used to generate directional audio-frequency sound with an ultrasonic primary beam.

Berktay’s far-field solution (2) predicts a demodulated secondary waveform along the axis of the beam that is proportional to the second time derivative of the square of the modulation envelope. The secondary wave is therefore generated with high levels of harmonic distortion, even for moderate modulation indices (3). Integrating the modulation signal twice and taking the square root removes this distortion; however, the resulting reduction in distortion is limited by the bandwidth of the ultrasonic transducer (4). In the present work, we consider the effect of finite transducer bandwidth on this predistortion scheme.

## SIMULATIONS

The simulations in this paper are based on Bertkay’s solution, and are therefore subject to its underlying assumptions: The array length  $\ell$ , which determines the half-power beam width of the parametric array [ $4/\sqrt{k_s \ell}$  radians, where  $k_s$  is the audio-frequency wavenumber], is assumed to be less than the Rayleigh distance for the transducer radiating at the carrier frequency  $f_c$  [see Fig. 1, where the effective array length is defined as the reciprocal of the absorption coefficient for air (5) at  $f_c$ ]; the modulation signal varies slowly in comparison with the carrier wave; and shock formation does not occur. A more accurate analysis requires numerical solutions that account for the combined effects of nonlinearity, diffraction, and absorption in the sound beam, but the main effect of transducer bandwidth is adequately revealed by the present use of the Bertkay solution.

The various steps in the simulation are depicted in Fig. 2, where  $x(t)$  is the audio input signal that is approximated by  $y(t)$ , the self-demodulated signal. The two integrals in the first block, giving  $x_1 = \iint x dt^2$ , compensate for the two time derivatives in the Bertkay solution, shown in the last block. Gain and offset are introduced in the second block [ $x_2 = 1 + mx_1$ , where  $m$  is the modulation index ( $0 \leq m \leq 1$ ), and  $x_1$  is normalized to unity amplitude], followed by the square root operation  $x_3 = \sqrt{x_2}$  to compensate for the quadratic nonlinearity in the Bertkay solution. The low-pass filter simulates the finite bandwidth  $\Delta f$  of the transducer; its output is denoted  $E(t)$ . Amplitude modulation by  $E(t)$  yields  $E(t) \sin(2\pi f_c t)$ . If  $\Delta f = \infty$  (infinite bandwidth) then  $E(t) = x_3(t)$ , and from the Bertkay solution [ $y(t) \propto d^2 E^2/dt^2$ ] we thus obtain the desired result,  $y(t) \propto x(t)$ .

The square root operation introduces infinite harmonics of the input signal  $x$ , and in principle a source with infinite bandwidth is required to obtain  $y(t) \propto x(t)$ , because  $E(t) \neq x_3(t)$  for  $\Delta f \neq \infty$ . To model the reality of finite transducer bandwidth, the transducer response is assumed to be flat in the passband defined by  $f_c \pm \frac{1}{2}\Delta f$ , outside of which it falls off at a rate of 24 dB per octave. For simplicity, our simulations are for an input signal having a single frequency  $f_s$ , such that  $x(t) = \sin(2\pi f_s t)$ .

In Fig. 3 the relative bandwidth  $\Delta f/f_c$  is varied from 5% to 50% and the effect on generation efficiency (dB in arbitrary units) is considered as a function of the modulation index  $m$  for  $f_c/f_s = 40$ . The amplitude of the output  $y$  at frequency  $f_s$  is very nearly proportional to  $m$  for all values of  $\Delta f/f_c$  shown. For given values of  $m$ , the reduction in conversion efficiency with decreasing bandwidth reflects the fact that the transducer radiates less energy in the passband.

Total harmonic distortion (THD) is plotted in Fig. 4 as a function of  $\Delta f/f_c$  and  $m$  for  $f_c/f_s = 40$ . Note that  $\text{THD} \rightarrow 0$  for any  $m$  as  $\Delta f \rightarrow \infty$ , and that THD increases with  $m$  and decreases with  $\Delta f$ . For the nominal values  $\Delta f/f_c = 10\%$  and  $m = 0.5$ , we find  $\text{THD} \simeq 5\%$ . Although this level of distortion may be acceptable for reproduction of speech, it may be excessive for high-fidelity reproduction of music.

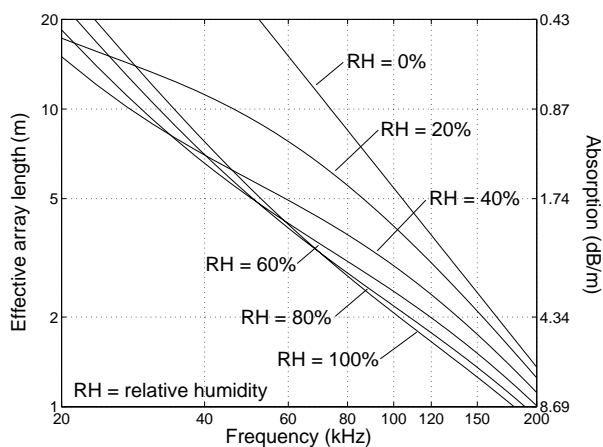


FIGURE 1. Effective array lengths in air.

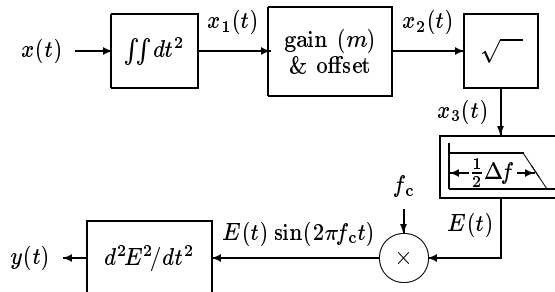


FIGURE 2. Block diagram of preprocessing scheme.

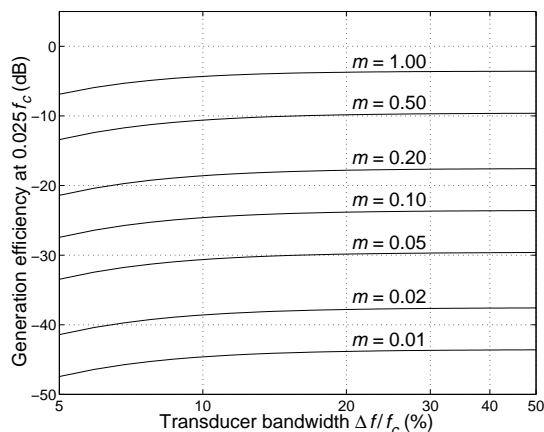


FIGURE 3. Generation efficiency.

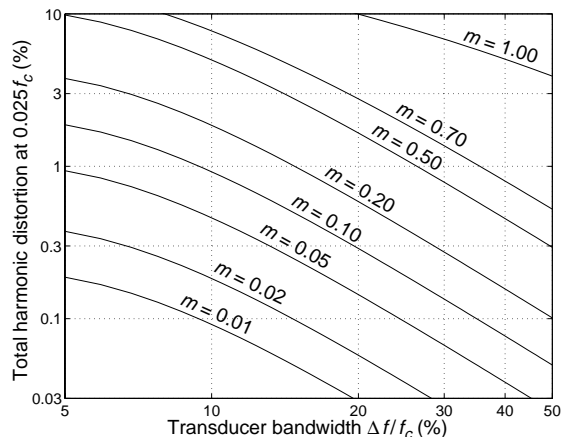


FIGURE 4. Total harmonic distortion.

### ACKNOWLEDGMENTS

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### REFERENCES

1. Yoneyama, M., Fugimoto, J., Kawamo, Y., and Sasabe, S., "The audio spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design," *J. Acoust. Soc. Am.* **73**, 1532–1536 (1983).
2. Berkta, H. O., "Possible exploitation of non-linear acoustics in underwater transmitting applications," *J. Sound Vib.* **2**, 435–461 (1965).
3. Blackstock, D. T., "Audio application of the parametric array," *J. Acoust. Soc. Am.* **102**, 3106(A) (1997).
4. Davy, B. A., "Acoustic self-demodulation of predistorted pulsed carriers," Master's Thesis, The University of Texas at Austin, May 1972.
5. Bass, H. E., Sutherland, L. C., Zuckerwar, A. J., Blackstock, D. T., and Hester, D. M., "Atmospheric absorption of sound: Further developments," *J. Acoust. Soc. Am.* **97**, 680–683 (1997).