Directional Loudspeaker Using a Parametric Array

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The theory for sound reproduction of parametric arrays is based on nonlinear acoustics. Due to the nonlinearity of the air, a finite amplitude ultrasound interacts with itself and generates audible secondary waves in the sound beam. A special feature of this loudspeaker is its sharper directivity compared to conventional loudspeakers of the same aperture size.

This paper describes the basis of the theory used for parametric arrays, and presents the influence of the main parameters, e.g., carrier frequency. It also describes some signal pre-processing needed to obtain the desired audible sound. A PVDF (polyvinylidenefluoride) film transducer is also studied in order to produce a prototype to confirm the theory.

Keywords: Parametric array, directional loudspeaker, PVDF.

1 Headline of a section

The nonlinearities of air cause high frequency wave components (primary waves) to interact. This interaction produces new frequencies (secondary waves), which are a combination of the sums and the differences of their frequency components. This process is similar to AM demodulation; therefore the term self-demodulation is often used. Loudspeakers based on parametric arrays use this phenomenon to generate audible sound with a narrow directivity. This directivity allows distant targeting of specific listeners.

The basis of nonlinear acoustics is presented, in order to understand the parametric array and to show the influence of the primary wave characteristics. Self-demodulation generates some known distortions in the audible sound. Therefore pre-processing must be applied to the signal before it is emitted. Two possible methods of signal processing are presented and their efficiency is estimated by simulation. Finally, as we are going to produce a prototype, we choose a PVDF film based transducer and study its response according to different characteristics, e.g., the size of the transducer.

2 Nonlinear acoustics

The parametric array was first analyzed by Westervelt [1]. He suggests that the sound pressure of secondary waves (p_s) produced by the nonlinear interaction is

$$\Delta p_s - \frac{1}{c_0^2}\frac{\partial^2 p_s}{\partial t^2} = -\frac{\beta}{\rho_0 c_0^4}\frac{\partial^2 p_p^2}{\partial t^2}, \tag{1}$$

where p_p is the sound pressure of primary waves, β is the nonlinear coefficient, ρ_0 is air density, and c_0 is sound velocity. The square of p_p in the source term (the right side of Eq. (1)) allows self-demodulation in the area where primary waves exist. Secondary waves are generated along this area, in the so-called parametric array. This is limited by the dissipation of the primary waves, which increases with frequency, and by the shock formation distance l_p, where strong attenuation occurs. l_p decreases when the frequency or the amplitude of the primary waves increases.

Berktay [2] studied the self-demodulated wave in the far field and gives the sound pressure (p_s) on the axis of propagation,

$$p_s(z,t) = \frac{p_{0p}^2 S\beta}{16\pi\rho_0 c_0^4 z\alpha}\frac{\partial^2 f^2(\Omega t)}{\partial t^2}, \tag{2}$$

where f is the envelope function of the amplitude modulated primary waves, p_{0p} is the primary wave emission amplitude, S the sound beam cross-section, α the attenuation coefficient of the primary wave, and Ω the frequency of the self-demodulated wave. This expression allows us not only to compute the level of the audible sound but also to predict the distortion contribution.

The self-demodulated wave characteristics depend on many parameters of the primary waves. First, the directivity of the self-demodulated wave increases when the parametric array length and the transducer surface increase. The parametric array length is related to the primary wave frequency (dissipation of the wave and shock formation distance) and to the sound pressure level of the primary waves (shock formation distance). Next, the level of the self-demodulated wave depends on the sound pressure level of the primary waves, the transducer surface and the amplitude of the envelope function, which is related to the modulation rate (Eq. (2)).

Another important characteristic of the self-demodulated wave is the distortion rate. This can be reduced by decreasing the modulation rate: if the envelope function is $f = 1 + m\cos(\Omega t)$ then according to relation (2), the demodulated wave is proportional to $m\cos(\Omega t)$ and the distortion is proportional to $m^2\cos(2\Omega t)$. So with a small modulation rate m we have very small distortion but also a small demodulated wave. As we will see in the next section, the distortion may also be reduced by pre-processing the signal.

3 Signal pre-processing

Signal pre-processing has three aims: amplitude modulation, distortion reduction, and transducer response compensation. In this paper, we consider the transducer to have an ideal response, so we are only interested in amplitude modulation and distortion reduction.

If classical amplitude modulation is used, the envelope function is given by $f = 1 + ms(t)$, where $s(t)$ is the audible signal to be transmitted. The Berktay equation (Eq. (2)) shows that the demodulation wave is proportional to the second derivative of f^2, therefore to obtain $s(t)$ as a self-demodulation wave in the far field we have to use the modulation function

$$s_1(t) = \sqrt{1 + \iint s(t)\,\mathrm{d}t^2}\,. \tag{3}$$

This processing is the ideal. However the square root of a signal has an infinite spectrum, while a transducer has a limited bandwidth. In practical terms, in order to have a self-demodulated wave with a satisfactory distortion rate, the transducer must have a bandwidth that is at least four times larger than the highest frequency of $s(t)$. This constraint can be difficult to fulfil when complex signals, e.g., music, have to be transmitted.

Another solution is to use a single side band amplitude modulation (SSB). In this case the self-demodulated signal has less distortion than when classical amplitude modulation is used without other processing. To decrease the residual distortions, it is possible to simulate the self-demodulation, extract the distortions and correct the signal, as shown in Fig. 1. The advantage of this correction is that it does not increase the necessary bandwidth and can be used in iterative processing.

These two processing methods give good results, but both have constraints: if the transducer has sufficient bandwidth the first method can be used, but if the transducer bandwidth is narrow and the calculation time is not a problem, then the second method is better.

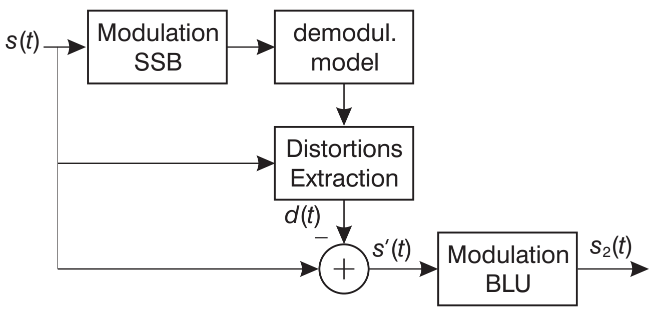

Fig. 1: Correction of distortions

4 PVDF transducer

The transducer characteristics, e.g., bandwidth or efficiency, are very important in the loudspeaker. With a view to producing a prototype, we have started to study a PVDF film based transducer.

These transducers are made up of one PVDF membrane placed on a support with one cavity. The static pressure inside the cavity is decreased in order to obtain a partial vacuum, wich will stretch the membrane and give it a spherical shape (Fig. 2). The spherical shape allows us to make use of the 3-axis piezoelectricity of the film and so increase the efficiency of the transducer.

The transducer response is a coupling between the cavity and the membrane response. It depends on the cavity dimensions and the vacuum quality inside the cavity. When the cavity radius decreases, the vibrating surface decreases and so the resonance frequencies increase. When the vacuum quality increases, the membrane tension increases, as do the resonance frequencies.

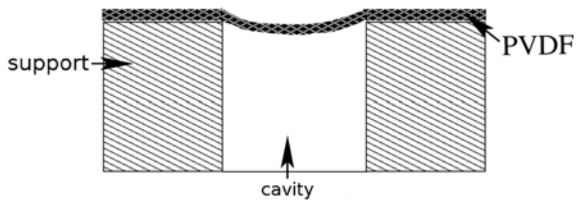

Fig. 2: PVDF transducer

In order to describe the transducer response precisely, a model will be made. One problem is the shape of the membrane, because the differential equation that describes the movement, cannot be resolved analytically. Another problem is the coupling between the membrane and the cavity; moreover the cavity response cannot be described with the lumped constant method, because its dimensions and the used wave length are approximately the same.

The loudspeaker is an array of transducers of this kind. A support with an array of cavities is used, and the PVDF film is placed on the support. The advantage is that, as there is only one film, all the active elements are in phase.

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References

[1] Westervelt, P. J.: Parametric Acoustic Array. *J. Acoust. Soc. Am*, Vol. **35** (1963), No. 4, p. 535–537.

[2] Berktay, H. O.: Possible Exploitation of Nonlinear Acoustics in Underwater Transmitting Application. *J. Sound Vib.*, Vol. **2** (1965), No. 4, p. 435–461.

[3] Yoneyama, M., Fujimoto, J.: The Audio Spotlight: An Application of Nonlinear Interaction of Sound Waves to a New Type of Loudspeaker Design. *J. Acoust. Soc. Am*, Vol. **73** (1983), No. 5, p. 1532–1536.

[4] Hamilton, M. F., Blackstock, D. T.: *Nonlinear Acoustics*. Academic Press, 1998.

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