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Technical note

An airborne parametric array

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ABSTRACT

The development of a prototype parametric speaker comprised of 37 piezo-speakers, also known as "acoustic flashlight" is described. The sound pressure along the axis was measured for both the primary (38.5 kHz ultrasonic) signal and the secondary (demodulated) signal. Comparison with the theory for an equivalent piston radiating at the same frequency and the simulation of the piezo speakers as simple sources in the same hexagonal arrangement was made. The absorption coefficient and the radiation patterns of the primary and secondary signal were also measured. All the measurements and the simulation agree with the corresponding theories. The demodulated (audio frequency) signal behaves in the near field of the speaker in the same way (that of an equivalent piston) as the primary (ultrasonic frequency) signal, which results in the inheritance of its high directivity.

1. Introduction

This sound source, fed by an ultrasonic signal modulated in amplitude by an audio signal, has the particularity that due to a nonlinear acoustic propagation in the air the modulating audio signal is recovered. The prototype generates an audio signal of plane waves traveling in one channel or beam that inherits the high directivity of the ultrasonic carrier. The beam can be directed, as one would do it with a laser beam, toward a listener, a wall, etc., without the sound spreading in all directions, but traveling like if it were in a virtual channel.

The first reports relating to the extra frequency components of distortion resulting from the propagation of sound waves were theoretically and experimentally initiated in the 1930's [1,2]. Westervelt [3] presented, in 1963, a theoretical model of what is called a parametric array that emits ultrasonic waves of large amplitude, where these waves are demodulated into an audible sound of great directionality due to nonlinear propagation effects. Thus, an ultrasonic beam modulated by a frequency of 1 kHz produces a beam with directionality similar to the ultrasonic carrier. Berktay [4] presented a quasilinear analysis of the parametric array and gave expressions for predicting the sound pressure amplitude of the demodulated wave valid outside the zone of interaction. Bennett and Blackstock [5] presented a demonstration of a parametric array that worked in the air. Yoneyama et al. [6] presented an experimental realization of a system of 547 loudspeakers which the authors called an "acoustic flashlight". Croft et al. [7] presented a survey of parametric systems. Moon [8,9], presented a stepped plate bi-frequency source for generating a difference frequency sound with a parametric array. This consists of a vibrating disk, similar to those used in acoustic levitators that can vibrate at two specific frequencies simultaneously. Mellert and Shwarz-Röhr [10] showed that the secondary signal is produced in the air, but also that the receiving transducer adds to the audio signal if the device (a microphone or an ear) is hit by the ultrasound beam. Currently, there are commercial versions of these systems. Applications can be manifold: in communication systems at distances of the order of one meter to several tens of meters. In this work, we report the construction and evaluation of a parametric speaker system consisting of 37 ultrasonic transducers. The system incorporates the electronics to generate the ultrasonic carrier and the amplitude modulation by an external audio signal from an iPod or similar audio gadget. One of the main problems with the sound from parametric arrays has been the distortion and its possible reduction (see for example [11,12]). No effort was made to reduce distortion; we hope to address this problem in the near future.

2. Theory

An ultrasonic beam at angular frequency ω_o, modulated by an audio signal $E(t)$ is radiated into the air. The pressure of the primary signal (amplitude modulated wave) P_p is given by Yoneyama et al. [6]:

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$$P_p = P_0 E(t) e^{-\alpha x} \sin\left(\omega_o\left(t - \frac{x}{c_0}\right)\right), \tag{1}$$

where P_o is the initial pressure, $E(t)$ is an envelope function, α [m^{-1}] is the pressure attenuation coefficient, related to the atmospheric attenuation coefficient A [dB/m], by $A = 8.68\,\alpha$ (there exists some confusion in the literature relating to these definitions, which are clarified by Wenmaekers et al. [13,14]), c_0 is the sound speed at ambient temperature, x is the axial distance traveled. The pressure of the secondary signal (demodulated signal) according to Berktaýs theory is:

$$P_d = \left(\frac{\beta P_0^2 a^2 m}{16\rho_0 c_0^4 x\alpha}\right)\frac{d^2E(t)^2}{dt^2}, \tag{2}$$

where β is the non-linearity coefficient of air, a is the effective radius of the source, ρ_o is the medium density, and $E(t)$ is the envelope function. This expression applies strictly to the axis in front of the speaker and in the far field.

Assuming a sinusoidal envelope $E(t)$ of modulating frequency ω, of the form

$$E(t) = 1 + m\sin\left(\omega t - \frac{\omega x}{c_0}\right), \tag{3}$$

where m is the modulation index, the pressure P_d can then be obtained by squaring and differentiating this expression twice with respect to time, and the result (introducing a trigonometric identity) substituted into Eq. (2). This yields the amplitudes of the pressures, P_d of the demodulated fundamental frequency Eq. (4), and of its first harmonic P_d^1, Eq. (5):

$$P_d = -\left(\frac{\beta P_0^2 a^2 m\omega^2}{8\rho_0 c_0^4 \alpha x}\right)\sin\left(\omega\left(t - \frac{x}{c_0}\right)\right). \tag{4}$$

It is to be noted, that this pressure amplitude is proportional to the square of the modulation frequency (which results in a drop of 12 dB/octave), so that at low frequencies, the system is not very efficient.

The pressure amplitude of the first harmonic of the demodulated signal is:

$$P_d^1 = \left(\frac{P_0^2 a^2 m^2 \omega^2}{8\rho_0 c_0^4 x\alpha}\right)\cos\left(2\omega\left(t - \frac{x}{c_0}\right)\right). \tag{5}$$

The percent harmonic distortion (HD) of the demodulated signal due to its first harmonic can be defined as:

$$\%HD = \left(\frac{P_d^1}{P_d}\right) \times 100 = 100\ \mathrm{m} \tag{6}$$

The modulation index is obtained from the relationship between the maximum (A_{max}), and minimum (A_{min}) amplitudes of the modulated carrier, as follows:

$$m = \left(\frac{A_{max} - A_{min}}{A_{max} + A_{min}}\right) \times 100. \tag{7}$$

The modulation index (m) can have values between 0 and 1, low values of m results in reduction of the sound pressure level of P_d because it is proportional to m; a low value of m will also decrease harmonic distortion of the demodulated wave since the first harmonic is proportional to m^2.

The demodulation occurs over a length of intensity absorption [15], $\eta = 2\,\alpha$ [m^{-1}]:

$$l_\eta = \frac{1}{\eta}. \tag{8}$$

The ultrasonic (primary) signal decays rapidly, and the audio (demodulated) signal travels relatively long distances. Thus, absorption is a fundamental parameter. Along the length of absorption, nonlinearities add in phase, as happens in an end-fired array, and thus, this parameter determines the output sound pressure amplitude [15]. The emitted sound can then be visualized as an output of the sum of low-frequency waves by virtual sources placed along the axis; the nonlinear effects, however, cease to be noticeable approximately at the so-called Raleigh distance, given by:

$$R_l = \frac{\pi f_c a^2}{c_o}, \tag{9}$$

where f_c is the frequency (primary) of the source, and a is its effective radius.

Assuming the air to be at 24 °C with 40% relative humidity, then the sound waves propagate with a speed c_o = 343.9 m/s. A theoretical atmospheric attenuation coefficient of $A \approx 1.4$ dB/m (α = 0.16 m^{-1}), at around 38.5 kHz and an altitude of 2, 200 m above sea level, thus obtains. The theoretical shock formation distance for the primary signal, is given by [14] as $\rho_o\ c_o^3/2\ \pi\ \beta\ f_c\ P_o$, which gives a shock distance greater than 13 m.

3. Description of the system

The parametric array system comprises an array of 37 (URCM-R38.5K1), ultrasonic transducers that radiate the modulated acoustic field. The phase of each element was checked and all elements were connected in such a way that the radiated field had in each transducer the same phase. A phenolic card with a hexagonal form and with the exact space for the transducers in a compact packaging in a hexagonal pattern was made, the system is shown in the photograph in Fig. 1. The electronic circuits used are fairly common: a power amplifier, an amplitude modulator, a preamplifier and a sinusoidal generator. The prototype utilizes circuits and elements that may be readily available and that are inexpensive. The block diagram of the complete system is shown in Fig. 2.

The XR-2206 integrated circuit generates a sine wave of 38.5 kHz, which corresponds to the carrier signal, this frequency can be varied to compensate for temperature changes that can somewhat change the resonance frequency of the piezo-transducers. To provide the necessary audio input to the voltage modulation circuit, an operational amplifier TL071 preamplifier with a gain of 5 is used. To amplitude modulate the carrier signal a MC1496p balanced modulator circuit was used, in AM modulation configuration (DSB). To amplify the modulated signal, a TDA

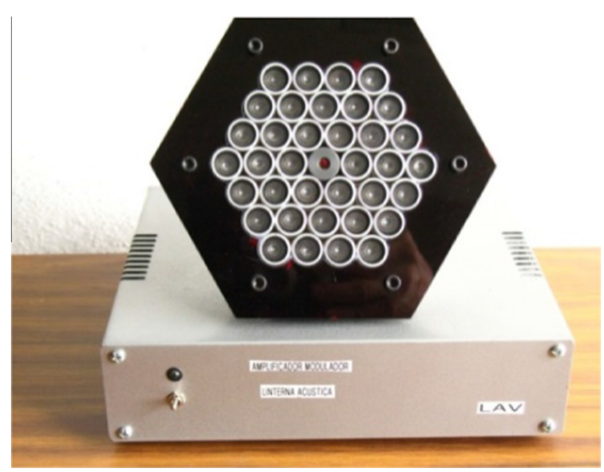

Fig. 1. A photograph of the parametric source with the electronics housing is shown. In this case, the central source was substituted with a laser diode from a laser pointer. The data measurements were obtained with the central source in place.