The Use of Airborne Ultrasonics for Generating Audible Sound Beams*

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A device known as a parametric array employs the nonlinearity of the air to create audible sound from inaudible ultrasound, resulting in an extremely directive, beamlike wide-band acoustical source. This source can be projected about an area much like a spotlight, and creates an actual spatialized sound distant from the transducer. A basic theoretical analysis of the airborne parametric array is outlined and verified experimentally, and future challenges are discussed.

0 INTRODUCTION

The so-called *parametric array* [1] has been studied widely in the context of underwater sonar [2] and, to a lesser extent, in air [3], [5, ch. 8, pp. 233–261]. It exploits an effect known as *self-demodulation* to create extremely directive low-frequency sounds, which would otherwise require an enormous array of audible-frequency transducers. Self-demodulation occurs when nonlinearities of a compressible medium (such as water or air) cause high-frequency wave components to interact. This interaction produces new frequencies at the combination of sums and differences of their individual frequency components, in a process akin to AM demodulation. In the context of airborne acoustics, this allows the creation of audible sound from the interaction of ultrasonic waves.

The audible sound, which are predictable and therefore (ideally) controllable, have the desirable property of being highly directional, similar to the ultrasound beam. This directionality allows both the distant targeting of specific listeners and the projection of sound against a distant surface—like a spotlight—creating a truly spatialized acoustic source. In addition, because sound is distributed only within a narrow column, the device can generate substantial audible sound despite the relatively weak effect of self-demodulation.

The first published experiment demonstrating a parametric array in air [3] proved it was possible, but the device was not intended for reproducing complex audio signals. Much later a device [4] that employed several hundred piezoceramic transducers operating a 40 kHz, AM modulated with an audio signal, was used to reproduce nearly full-bandwidth audio. A similar but smaller device was recently introduced (described in [6]), which employed 60 piezoceramic transducers, also with a carrier of 40 kHz.

These early devices exhibited extreme distortion, a result of using simple AM modulation of the audio signal. Likewise, extremely intense ultrasound (up to 140 dB SPL) has been used at frequencies very close to the audible range, bringing their safety into question.

By preprocessing the audio signal one can ideally eliminate the harmonic distortion. However, because preprocessing introduces an infinite set of harmonics, all of which must be reproduced by the transducer, the ability to reduce distortion is limited by the bandwidth of the transducer [7].

In this engineering report a basic theoretical analysis of the parametric array is presented. In particular the efficacy and limitations of preprocessing to reduce distortion are analyzed. The performance of a prototype transducer—the first that allows a controlled investigation of preprocessing—is evaluated.

1 THEORY

1.1 Nonlinear Acoustics and Self-Demodulation

The equations of nonlinear acoustics, discussed in detail elsewhere [1], [5], [8], [9], do not have general analytical solutions. For most situations they require a reliance on numerical simulation. However, we can arrive at a reasonable approximation by following Berk-

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tay's analysis [2], which states (subject to his simplifying assumptions) that a collimated primary wave consisting of an AM-modulated wave of pressure $p_1(t) = P_1 E(t) \sin(\omega_c t)$, where P_1 is the amplitude of the primary beam pressure, $E(t)$ is the modulation envelope, and ω_c is the carrier frequency, will demodulate, creating a secondary wave p_2 given by

$$p_2(t) = \frac{\beta P_1^2 A}{16\pi\rho_0 c_0^4 z\alpha} \frac{\partial^2}{\partial t^2} E^2(\tau)$$

where $\beta = (\gamma + 1)/2$ is the coefficient of nonlinearity ($\beta_{air} = 1.2$), γ being the ratio of specific heats, ρ_0 is the ambient density of the medium, c_0 is the small-signal wave propagation speed, A is the beam's cross-sectional area, z is the axial distance, α is the absorption coefficient of the medium (at frequency ω_c), and $\tau = t - z/c_0$ is lag time. As an example, at $c = 348$ m/s, $\rho_0 = 1.18$ kg/m^3, $\alpha = 0.7$, and $A = 0.2$ m^2, a 130-dB ultrasound wave modulated with a 1-kHz signal should produce about 66 dB of audible sound at 1 m.

The important result here is that the amplitude of the secondary (demodulated) beam is proportional to the second derivative of the *square* of the modulator envelope. This second-derivative factor produces a slope in the frequency response of 12 dB per octave, while the squaring adds significant distortion when using simple [double-sideband (DSB)] AM modulation, as we shall see.

1.2 Modulation and Preprocessing

Prior devices have used this simple AM modulation technique to drive the parametric array, using envelope function $E(t) = 1 + mg(t)$ (DSB), where m is the modulation depth and $g(t)$ is the audio signal. Upon demodulation, the result is the second derivative of the sum of two waves, one proportional to the original signal $2mg(t)$ and another proportional to its square, $m^2g^2(t)$, as observed by Yoneyama and his collaborators [4].

Therefore distortion is very strong and produces significant audible artifacts unless $m << 2$. Of course, lowering m lowers the distortion, but doing so proportionally lowers the amplitude of the intended signal, which is undesirable. An alternative that has been suggested [6], [7] is to simply integrate twice (equalize) and take the square root, resulting in target envelope $E'(t)$, which can be called "preprocessed AM,"

$$E'(t) = [1 + \textstyle\iint g(t)\,dt^2]^{1/2}\,.$$

This is still less than ideal, because the square-root operation creates an infinite series of harmonics which (for a pure tone) decay by a factor of -12 dB per octave. All of these harmonics must be reproduced if distortion is to be eliminated. The maximum reduction of distortion is thus directly related to the bandwidth of the device. This limitation was explored through simulations by Kite et al. [7].

However, because the harmonics introduced by the square-root operation attenuate as they increase in frequency, a wide-band ultrasound source can perform quite well. As an example, the ultrasound response of the prototype loudspeaker (described in the next section) is plotted with a signal intended to produce a 1-kHz tone $[\sqrt{1 + \sin(2000\pi t)}\sin(6 \times 10^4 \cdot 2\pi t)]$ in Fig. 1. A large number of the necessary harmonics are still within the response of the transducer, indicating that distortion can be reduced.

2 EXPERIMENT

An experiment was conducted to measure the frequency response, directivity, and distortion of reproduced signals using a prototype wide-band parametric array. This section describes the experimental procedure and the measured characteristics of the ultrasonic field and audible sound.

2.1 Procedure

The transducer array was mounted on a turntable in an anechoic chamber, and a high-frequency linear microphone (B&K 4138) was positioned at a distance of 3 m, aligned with the center of the array. It should be noted that standard audio microphones are not suitable for these measurements, as their own nonlinearities can create demodulation at the microphone element itself, causing incorrect measurements [3].

For each angle a series of tone bursts was transmitted and simultaneously recorded. The burst set consisted of pure-tone ultrasound bursts (from 40 to 90 kHz, in 2-kHz intervals) and ultrasound bursts modulated with pure audio tones (one-third-octave bands, 400 Hz to 16 kHz). Both AM modulation and preprocessed AM modulation (with the equalization and integration omitted) were used. The modulation depth was kept at unity. The received bursts were then windowed (Hanning) for

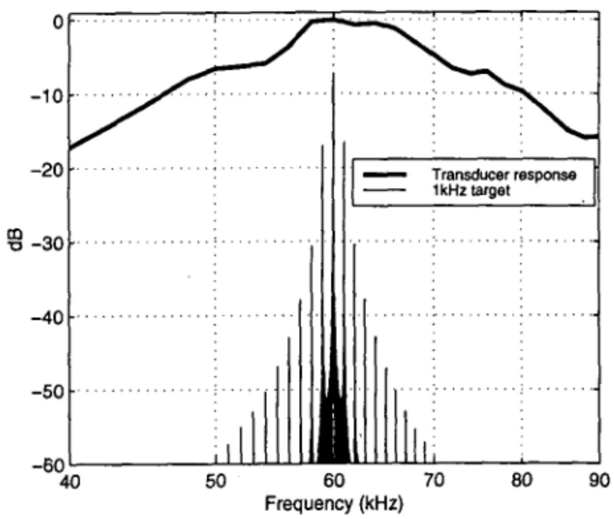

Fig. 1. Bandwidth of prototype array and transmitted signal. Lower curve—spectrum of audio signal which self-demodulates to 1-kHz pure tone; upper curve—experimentally measured frequency response of prototype transducer array. Most of the necessary spectrum is within the transducer response, indicating that the tone can be reproduced with relatively little distortion.

analysis. The array was rotated at intervals ranging from 5° for angles greatly off axis (> 15 °) to 1° near the axis (< 10°). Input and output sampling rates were both 250 kHz, and the microphone noise floor was approximately 40 dB SPL.

2.2 Ultrasonic Field

The transducer array (≈ 350-mm diameter) was designed to produce a large, uniform, collimated beam of ultrasound. The result of a numerical simulation of the propagation of the carrier frequency (60 kHz) is shown in Fig. 2. For clarity, amplitudes below 113 dB are not plotted. The high-intensity region in the beam becomes pronounced at about 2 m and remains quite well collimated thereafter, although with some small sidelobes. The estimated array length $L = 1/(2\alpha)$ is about 2 m at 60 kHz, and since measurements are done at a distance of 3 m, it is not certain that we are truly in the far-field region where the equations describing demodulation are satisfied.

The transducers making up the array were designed to have a large bandwidth and gently tapered slopes, to give a relatively flat audio response without the need for equalization. The ultrasonic response of the system (including amplifier) is shown in Fig. 3 for small angles, demonstrating a usable bandwidth of well over 50 kHz. There is some nonlinearity in the transducer itself (although it is not yet characterized), which causes it to output a small amount of audible sound.

A plot of the measured ultrasonic directivity at 60 kHz is shown in Fig. 4; the array exhibits a -6-dB beamwidth of about 3°

2.3 Audible Sound

Two aspects of the audible sound are of interest: 1) its on-axis characteristics, in particular the efficacy of preprocessing for distortion reduction, and 2) its directivity.

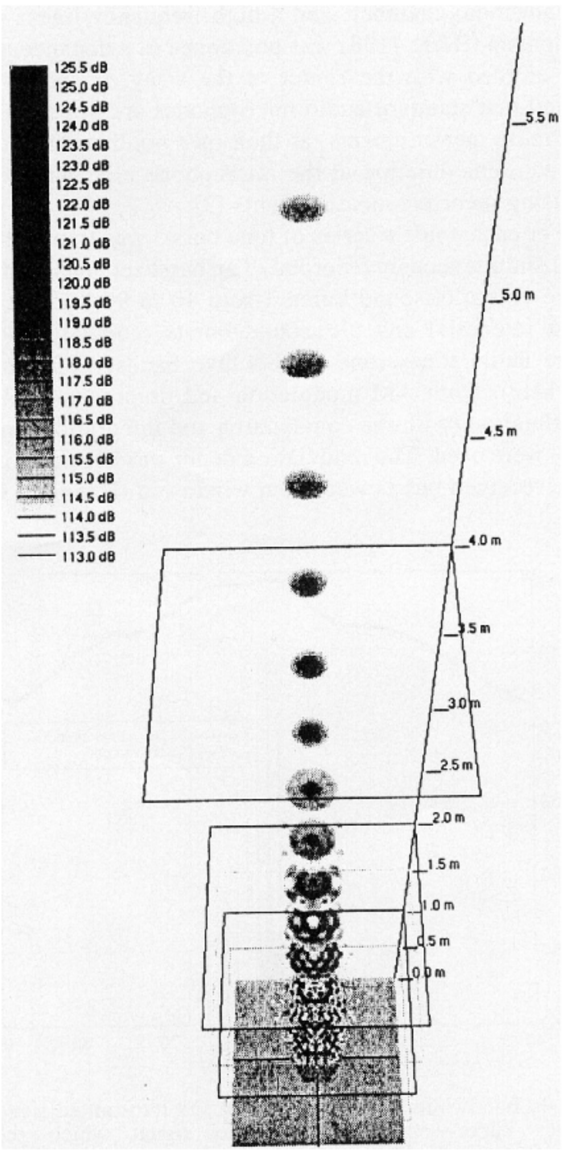

Fig. 2. Numerical simulation of 60-kHz carrier propagation, calculated by direct array modeling based on characteristics of individual elements. Coherent beam formation occurs at about 2 m from array.

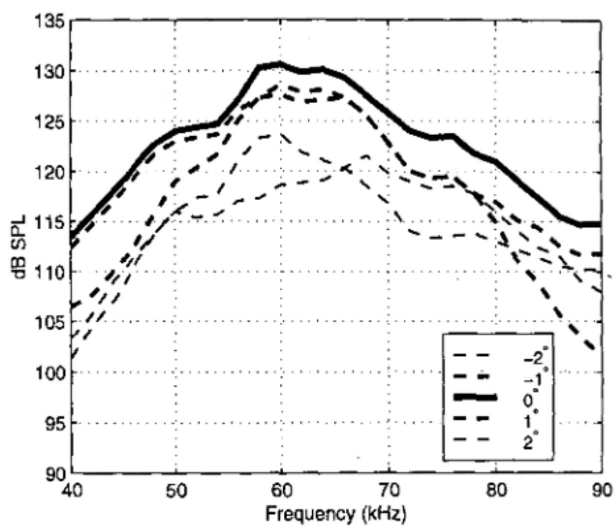

Fig. 3. Ultrasonic frequency response at 3 m for various angles, measured using pure tone bursts at 2-kHz intervals. Wide bandwidth and sharp directivity are apparent.

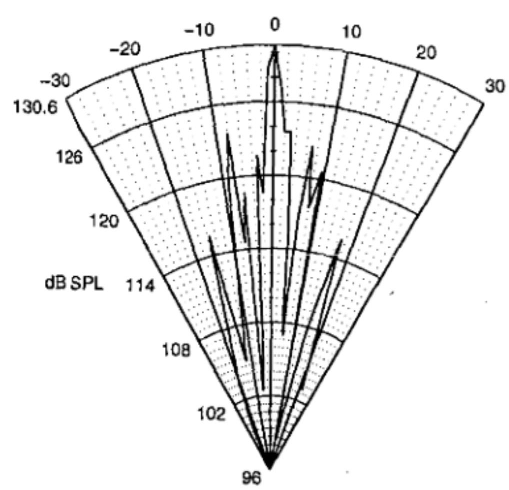

Fig. 4. Directional response of 60-kHz carrier at 3 m, measured at 1° intervals near on axis, and at larger intervals for off-axis measurements.

2.3.1 Response and Distortion

The audible sound was measured by inspecting peaks in the spectrum of the received burst for each angle of interest. Because the noise floor was approximately 40 dB for most frequencies, but was much greater below 400 Hz, we limited our analysis to frequencies of at least 400 Hz and SPLs greater than 40 dB.

The on-axis audible sound, including two harmonics, is shown for both traditional AM and preprocessed AM in Figs. 5 and 6, respectively. A relatively flat response is observed over most of the audible range, with some emphasis observed at very high frequencies.

From the first three harmonics identified, estimates of the total harmonic distortion (THD) were computed and are shown in Fig. 7. There is a substantial improvement when using preprocessing; the THD is lowered to only a few percent (approaching that of a traditional loudspeaker, which is typically near 1%).

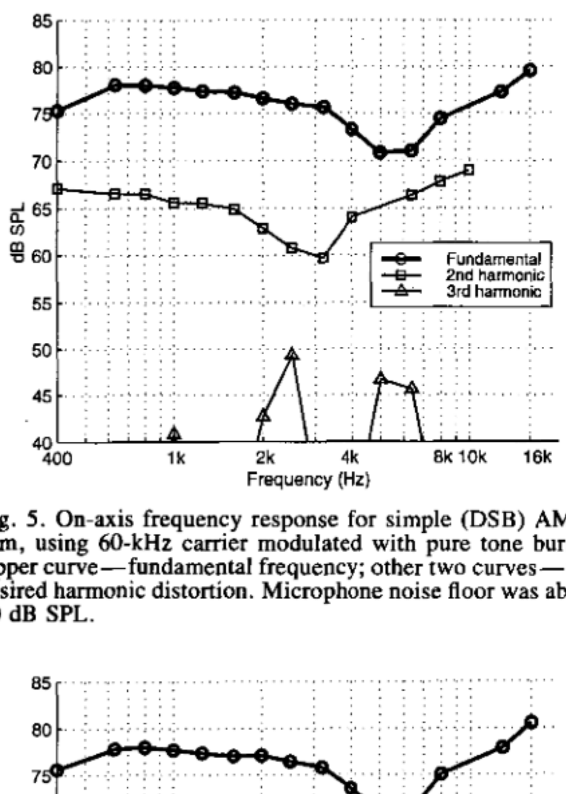

Fig. 5. On-axis frequency response for simple (DSB) AM at 3 m, using 60-kHz carrier modulated with pure tone bursts. Upper curve—fundamental frequency; other two curves—undesired harmonic distortion. Microphone noise floor was about 40 dB SPL.

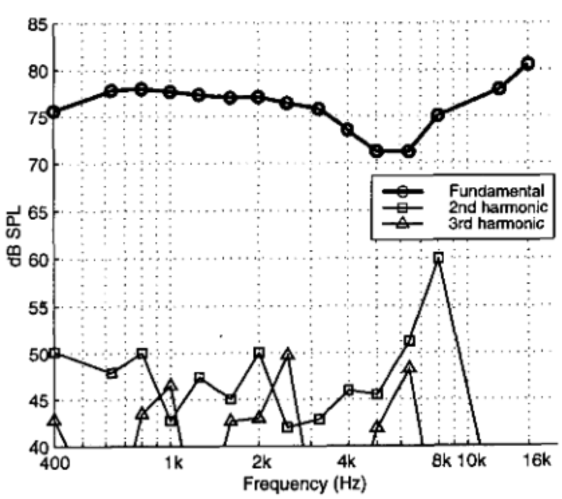

Fig. 6. On-axis frequency response for preprocessed AM at 3 m, using 60-kHz carrier modulated with preprocessed (see text) pure tone bursts. Upper curve—fundamental; other two curves—undesired harmonic distortion. Note that distortion is substantially less than in simple AM case.

The dip in the response, and the related increased distortion at 5–8 kHz, as seen in Figs. 5–7, may be attributed to the small dips and asymmetry in the ultrasonic response at 60 kHz ± 5–8 kHz. Because the cancelation harmonics provided by the square root are not uniformly distributed on both sides of the carrier, these extra harmonics may even increase distortion. The audible output is proportional to the square of the primary amplitude, which exaggerates any flaws in the ultrasonic response. This highlights the importance of creating a broad, symmetric ultrasonic response for minimizing distortion.

2.3.2 *Directional Response*

As the audible sound is created by the ultrasound, it follows that the audible sound propagates in a similar fashion, that is, in a narrow beam. Directional plots for various frequencies in the preprocessed case are shown in Fig. 8. Note that even for very low frequencies, an extremely small beam angle (3°) is apparent. For comparison, the beam angle for a loudspeaker (baffled piston) of this size would range from about 120° at 400 Hz to 10° at 4 kHz, finally reaching 3° at about 14 kHz.

One curious result is that directivity *decreases* for higher frequencies. This is again due to nonlinearities in the transducer itself, as such effects may be more pronounced at higher frequencies. Any additional output from the transducer that is directly audible will behave like a traditional piston source, weakening the hyperdirectivity caused by parametric generation.

3 CONCLUSIONS AND FUTURE WORK

An airborne parametric array built from properly designed broadband ultrasonic transducers has been shown to produce, compared to traditional loudspeakers of similar size, extremely directive beams of sound over a broad range of audible frequencies. The efficacy of preprocessing to reduce distortion has been demonstrated to reduce harmonic distortion to less than 5% over most audible frequencies.

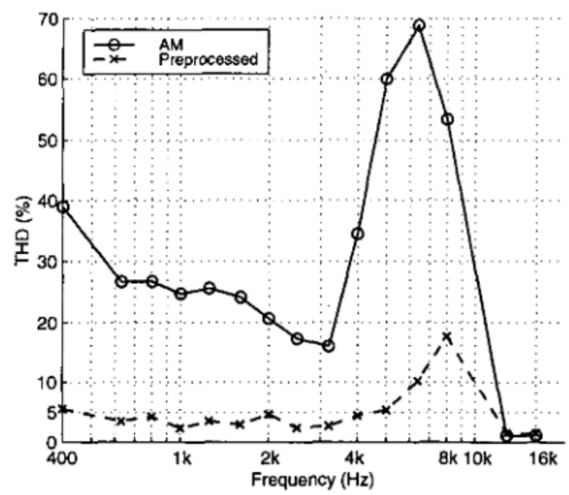

Fig. 7. Total harmonic distortion (THD), simple AM and preprocessed AM, calculated from data shown in previous figures. Note that THD is only about 5% for preprocessed case.

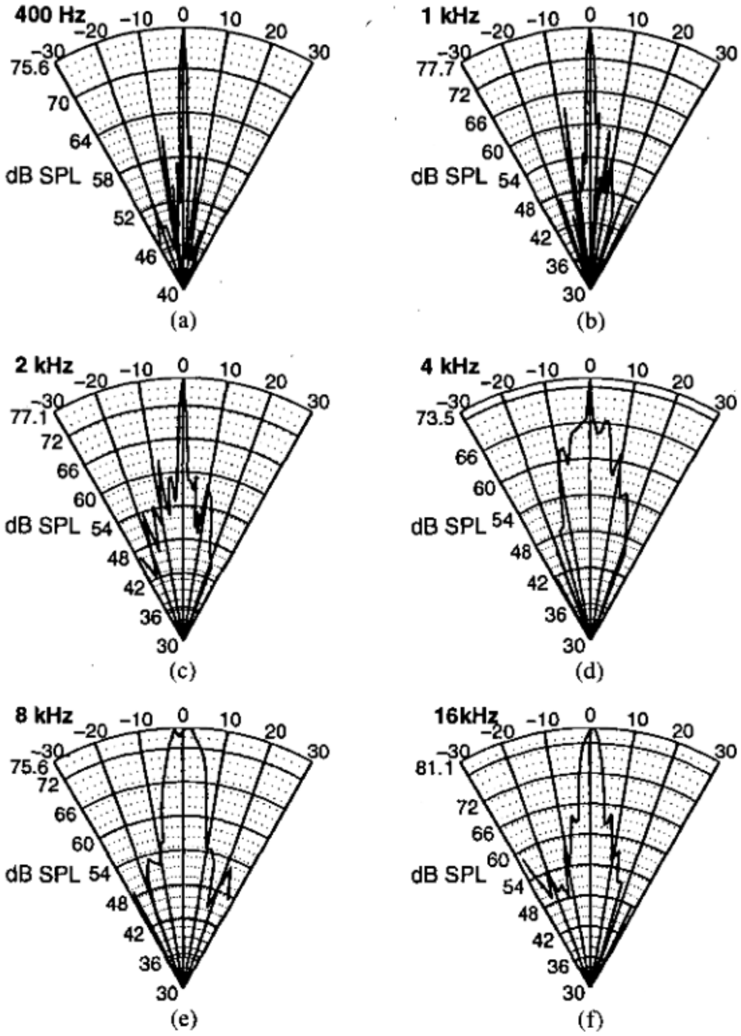

Fig. 8. Directional responses for audible sound, measured at 3 m. Amplitude of demodulated fundamental audible tone at several frequencies using preprocessed AM, plotted as a function of angle.

Future research will involve numerical simulation of the entire demodulation process—which so far cannot be solved analytically—as well as the theoretical and practical investigation of beam steering. Additional transducer development will further improve the directivity and harmonic distortion of the array.

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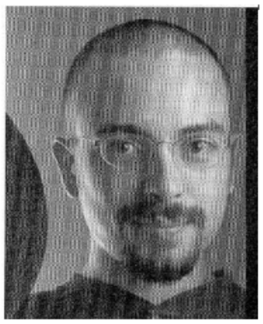

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Being acutely aware of the limitations of traditional loudspeakers, Mr. Pompei had the idea of using ultrasound as an acoustic projector, and is now developing such a device at the MIT Media Lab while continuing his education in pursuit of a Ph.D.

Mr. Pompei is a member of the Audio Engineering Society.