Development of a parametric loudspeaker: A novel directional sound generation technology

Chuang Shi and Woon-Seng Gan

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Sound is one of the most important mediums used by people to deliver information. Regarding personal privacy in modern society, the generation of directional sound is emerging significantly. For example, in the library, a personal public announcement system can help to communicate with a group of listeners without disturbing others. In a museum or art gallery, an introduction of exhibitions can only be heard by those standing in front of the exhibit. There are also similar needs for private messaging in vending and dispensing machines, exhibition booths, and billboards. Some application photos of parametric loudspeakers are shown in Fig. 1. This sound confinement can greatly reduce noise pollution in public places where messaging sound levels can be overwhelming. Another application of directional sound is in multilanguage teleconferencing, where different languages can be broadcasted to different participants in a common room without any physical partition or the need for headsets.

Generation of directional sound

Several techniques of directional sound can be adopted to implement directional sound, and each has its benefits and drawbacks. One straightforward idea comes from the imitation of an optical focusing lens, and it can be traced back to ancient China, where ancient buildings have architectural features that acoustically focuses sound in an area. In 1995, Brown from the United States patented his acoustic imaging sound dome, which can reflect and focus audible frequency wave to a small listening area. However, this dome feature has certain limitations of placement and the sound field may not be tightly confined to a "sweet spot."

Another method of directional sound generation includes utilizing a loudspeaker array. Its directivity is controlled by phases of loudspeakers' outputs in the array. With digital array processing techniques, a loudspeaker array has certain flexibilities in adjusting and adapting the beam pattern and steering direction compared to sound domes. However, one major disadvantage of a loudspeaker array is that it requires a larger array scale in the order of several meters to obtain higher directivity for projecting low-frequency audible sound. For example, Hiroshi Mizoguchi combined a loudspeaker array with visual face detection and tracking in a novel human-machine interface, called invisible messenger. The first invisible messenger system was implemented with 16 loudspeakers. But in his latter work, a 128-channel surrounded speaker array

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was adopted to generate a more effective sound spot.

Unlike the loudspeaker array, whose directivity is determined by array aperture and frequency, a new type of directional loudspeaker, known as the parametric loudspeaker, is able to project low-frequency sound with a small-size ultrasound emitters array. Parametric loudspeaker uses ultrasonic waves beyond the human hearing range as a unidirectional carrier to deliver audible sound to desired locations with precision. The most significant advantage of generating directional sound by parametric array is that the ultrasound emitters array can be built to different sizes to achieve different focal lengths that can be more readily deployed in many applications.

It has been recently reported that small-volume, high-frequency ultrasonic transducers have been designed and fabricated by microelectromechanical technology, known as MEMS ultrasonic transducers, which may make parametric loudspeakers portable in the future. However, due to the current limitation in ultrasonic transducer technology to reproduce a high-fidelity, low-frequency response, the parametric array is limited to speech-based applications such as public address systems and billboard advertisements.

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Principle of a parametric loudspeaker

The fundamental theory of a parametric loudspeaker is based on the principal of the parametric array, which was discovered and explained by Westervelt in 1960, at a meeting of the Acoustical Society of America. In 1975, Bennett and Blackstock proved that a parametric speaker can work with air as the transfer medium by sending 18.6 kHz and 23.6 kHz collimated beams and observing the 5 kHz difference frequency wave.

Fig. 1 Directional sound applications in a library and at a vending machine.

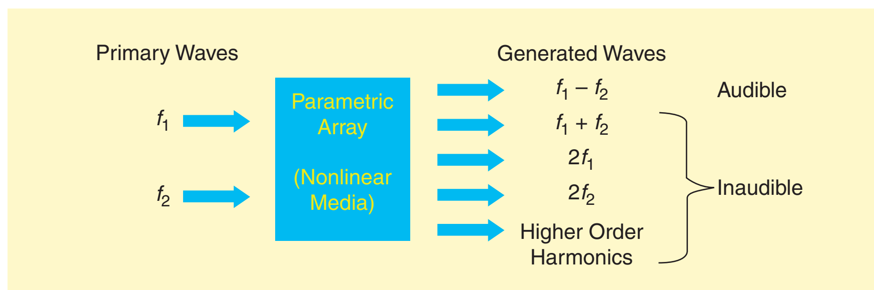

Fig. 2 Nonlinear interaction process in parametric array.

The phenomenon of the parametric array was described by Westervelt as: “two plane waves of differing frequencies generate, when traveling in the same direction, two new waves, one of which has a frequency equal to the sum of the original two frequencies and the other equal to the difference frequency.” Fig. 2 shows the creation of the sum and difference frequency waves, as well as higher harmonics of primary waves from the parametric array. It is noted that only the difference frequency is able to be perceived by the human ear. These generated frequency waves are attenuated in air and decay more rapidly for higher frequency components and at increasing distances from the speaker. The difference-frequency wave, which is lower in frequency and perceived by humans, is less abated by the air absorption. Therefore, after a short distance of propagation, only the audible waves in the sound beam remain sufficient amplitudes to be heard by humans.

There are two important distances to be considered in a parametric array, namely, Rayleigh distance and absorption length. Rayleigh distance is defined as the distance from the array at which there is a transition from a near-field region to a far-field region. Within Rayleigh distance, wavefronts are approximately planar. After Rayleigh distance, the wavefront becomes more spherical and attenuates more rapidly at a rate of –6dB per double distance.

Absorption length is defined as the distance beyond which the nonlinear interaction no longer exists. The absorption length is also called the effective array length (see Fig. 3), determining the extent of the distance traveled by the ultrasonic carrier before it ceases to generate any more audible sound sources. Effective array length is also explained as the range of end-fire virtual audible sources. Intermodulation process inside the primary beam excites air molecules to oscillate at the audio frequency, and the oscillation is regarded as a virtual source.

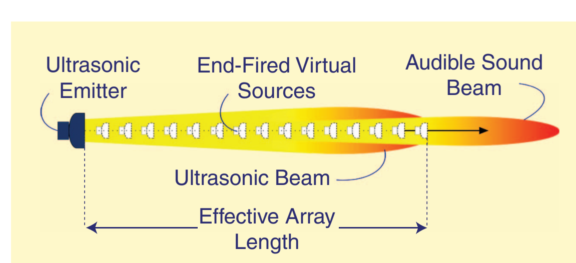

Fig. 3 Geometric model of the parametric array.

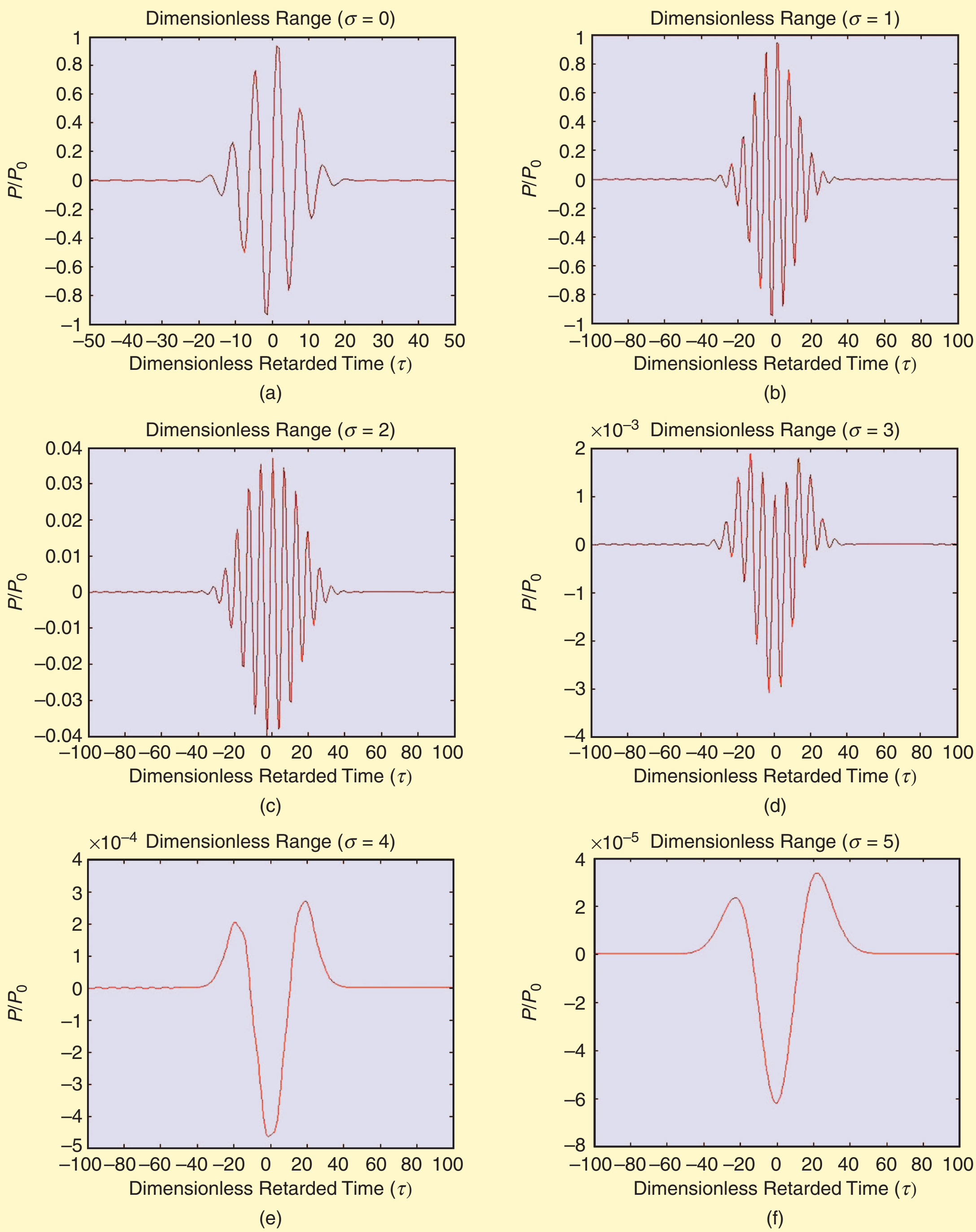

Fig. 4 Axial waveforms at different dimensionless ranges.

Models of nonlinear acoustics

Several model equations have been explored to describe the propagation of finite-amplitude sound beams produced by the parametric array. The Westervelt equation (1963) is one of the fundamental equations of the parametric array, and approximates the full second order wave equation.

In 1971, the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation was derived from the Westervelt equation. The KZK equation assumes the sound to localize in the vicinity of the propagating axis. It accurately describes the propagation of finite-amplitude sound beams combining the effects of diffraction, absorption, and nonlinearity under a parabolic approximation, and expressed by the first, second, and third terms on the right hand side of KZK equation, respectively:

$$\frac{\partial^2 p}{\partial z \partial \tau} = \frac{c^3}{2}\nabla_\perp^2 p + \frac{\partial}{2c^3}\frac{\partial^3 p}{\partial \tau^3} + \frac{\beta}{2\rho c^3}\frac{\partial^2 p^2}{\partial \tau^2}.$$

Here p is the sound pressure; z is the coordinate along the beam propagation direction; $\tau = t - z/c$ is the retarded time, and c is the small signal sound speed. Furthermore, p, δ and β are the density, dissipation factor corresponding to thermoviscous absorption, and the nonlinearity coefficient of the medium, respectively. $\nabla_{\perp}^2$ is the Laplacian operator that operates on the X-Y plane perpendicular to the axis of the beam.

The numerical solution of the KZK equation can show us the effect of waveform distortion (self-demodulation), which is caused by the nonlinearity of air. Fig. 4 shows the simulation result of axial waveforms at different dimensionless ranges, whose original signal is a short tone burst with Gaussian envelope. As the wave propagates, the carrier wave is being damped out, and the self-demodulated waveform is formed simultaneously. The relatively high absorption prevents higher harmonic spreading to farfield. If dissipation factor increases, the primary wave is more rapidly absorbed, and thus the fully demodulated waveform is generated closer to the source.

Although the KZK equation is one of the most efficient models of parametric array, it is too complex to use in practice. Berktay provided a simple expression that can be used to predict the farfield array response on the propagating axis. It is stated that the demodulated waveform along the axis of propagation is proportional to the second-time derivative of the square of the envelope of the primary signal:

$$p_2 = \frac{\beta P_0^2 a^2}{16\rho c^4 z\delta}\frac{d^2}{d\tau^2}E^2(\tau),$$

where a, P_0 are source radius and pressure amplitude at source, respectively; and is the modulation envelop. The other notations are defined identically as in the KZK equation.

Berktay derived his equation from the Westervelt equation. Since the KZK equation and the Westervelt equation are identical on the axis of farfield, it is not surprising that the Berktay equation can also be derived from the KZK equation. By quasilinear assumption, the Berktay equation is an analytical solution to the KZK equation on the axis of propagation.

Distortion and preprocessing technique

These theoretical models reveal many essential characteristics of a parametric array. According to Berktay's equation, a 12-dB/octave slope in the frequency response of a parametric array is predicted. This has been verified by both numerical simulations and experiments. Therefore, it is recommended that a low-pass filter with 12-dB/octave transition is used to equalize the frequency response before amplitude modulation. On the right-hand side of the Berktay equation, second-order derivation of the square of the envelop function is involved. To eliminate the distortion introduced by these operations, Kite proposed a square-root amplitude modulation of twice integrations of the modulating wave. This preprocessing technique is commonly known as the square-root amplitude modulation. Since there is no theoretical models that can provide accurate descriptions of the entire nonlinear self-demodulation process, all these preprocessing algorithms are proposed under certain limitations. The trade-off of high fidelity and computation is still a challenge to the design of parametric loudspeakers.

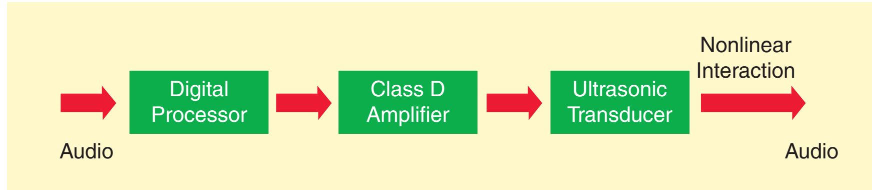

Fig. 5 Block diagram of ABS.

Fig. 6 ABS and its ultrasound emitters.

Prototyping of parametric loudspeaker

Based on the above theoretical analysis, we designed and implemented a directional sound projection (or parametric loudspeaker) system, known as the audio beam system (ABS). The prototyping of the parametric loudspeaker consists of three main components (see Fig. 5): a digital signal processor (DSP), amplifier, and ultrasound emitters. The programmable DSP is the main processing block of the parametric array that performs preprocessing, such as equalization, dynamic range control, carrier control, and our proposed modified amplitude modulation techniques. The special-designed class-D amplifier is used to adjust the voltage gain to drive an array of ultrasound emitters, which are made of lead zirconate titanate or polyvinylidene fluoride materials. Due to the nonlinear interaction of the air, directional audible sound will be generated at a targeted sweet spot.

The ABS was recently installed in a gaming booth in the Fusionopolis, a research and development complex located at the One-North business park in Singapore. We used two ABS speakers (ultrasound emitters, see Fig. 6) to project directional binaural sound from a gaining console to the gamer (see Fig. 7) standing in front of the gaming booth. This setup confines sound to a predetermined sound zone. A laser or LED device fitted to the ABS speakers can project a light beam with the same direction as the sound beam to indicate the sweet spot on the floor. The gamer within the zone can enjoy playing the game without disturbing the other visitors beyond the defined "tune-in" zone.

Fig. 7 ABS installed in a gaming booth.

Conclusion

The development of theory on the parametric array in the air provides an attractive and challenging approach to generate directional sound. A parametric loudspeaker has many useful characteristics that allow a high directivity, controllable beam, and reasonable implementation size, unlike the conventional loudspeaker, which radiates sound waves in omnidirection and requires a large array to generate a focused sound. Even though the bass quality of a parametric loudspeaker is not satisfied to reproduce music, it is anticipated that this problem can be solved by psychoacoustics technology. One the other hand, as updated advancements in digital processor and ultrasound emitters are becoming available, more preprocessing algorithms can be conducted in real-time, and more applications can benefit in the deployment of parametric loudspeakers.

Acknowledgment

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Read more about it

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About the authors

Chuang Shi (shic0002@ntu.edu.sg) is currently pursuing his Ph.D.'s degree in electrical and electronic engineering at Nanyang Technological University.

Woon-Seng Gan (ewsgan@ntu.edu.sg) is an associate professor of electrical and electronic engineering at Nanyang Technological University. He is a Senior Member of IEEE.

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