Development of Parametric Loudspeaker:

A Novel Directional Sound Generation Technology

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

Fig. 1 Example photos of directional sound applications

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 imitation of optical focusing lens, and it can be traced back to ancient China, where ancient buildings have architecture 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 limitation of placement and sound field may not be tightly confined to a sweet spot.

Another method of directional sound generation is by utilizing loudspeaker array. Its directivity is controlled by phases of loudspeakers' outputs in the array. With digital array processing techniques, loudspeaker array has certain flexibilities in adjusting and adapting the beam pattern and steering direction compared to the sound domes. However, one major disadvantage of loudspeaker array is that it requires larger array scale in the order of several meters to obtain higher directivity for projecting low-frequency audible sound. For example, Hiroshi Mizoguchi combined loudspeaker array with visual face detection and tracking in a novel human-machine interface, called invisible messenger. The first invisible messenger system is implemented with 16 loudspeakers. But in his latter work, 128-chanel surrounded speaker array 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 wave beyond 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 deploy in many applications. It is recently reported that small-volume high-frequency ultrasonic transducers have been designed and fabricated by micro-electro-mechanical technology, known as MEMS ultrasonic transducers, which may make parametric loudspeaker portable in the future. However, due to the current limitation in ultrasonic transducer technology to reproduce high fidelity low-frequency response, the parametric array is limited to speech based applications such as public address system, bill board advertisement, *etc*.

Principle of Parametric Loudspeaker

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

The phenomenon of 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 parametric array. It is noted that only the difference frequency is able to be perceived by human ear. These generated frequency waves are attenuated in air and decay more rapidly for higher frequency components and at increasing distant from the speaker. So the difference-frequency wave, which is lower in frequency and perceived by human, 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 hear by human beings.

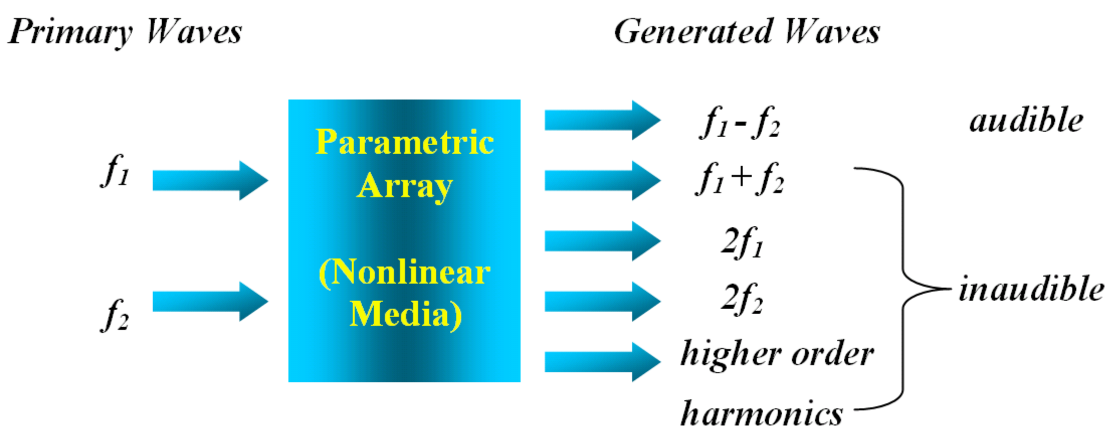

Fig. 2 Nonlinear interaction process in parametric array

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, wavefront are approximately planar. After Rayleigh distance, wavefront becomes more spherical and attenuate 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 virtual source.

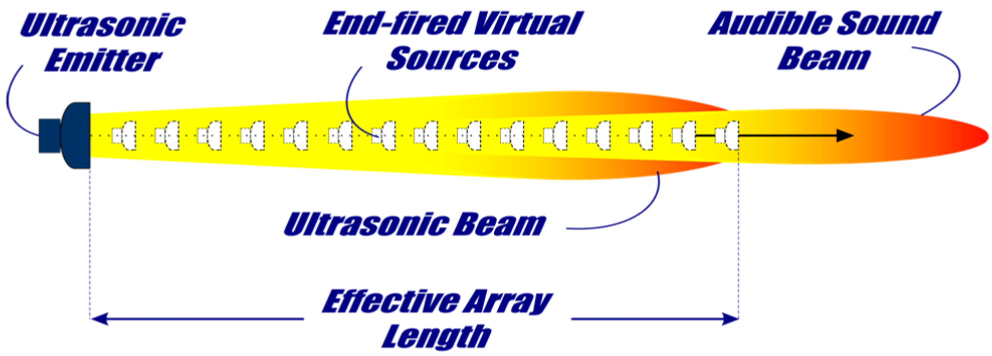

Fig. 3 Geometric model of the parametric array

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 parametric array, and approximates the full second order wave equation.

In 1971, 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:

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

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, ρ, δ and β are the density, dissipation factor corresponding to thermoviscous absorption, and the non-linearity 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.

Numerical solution of 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.

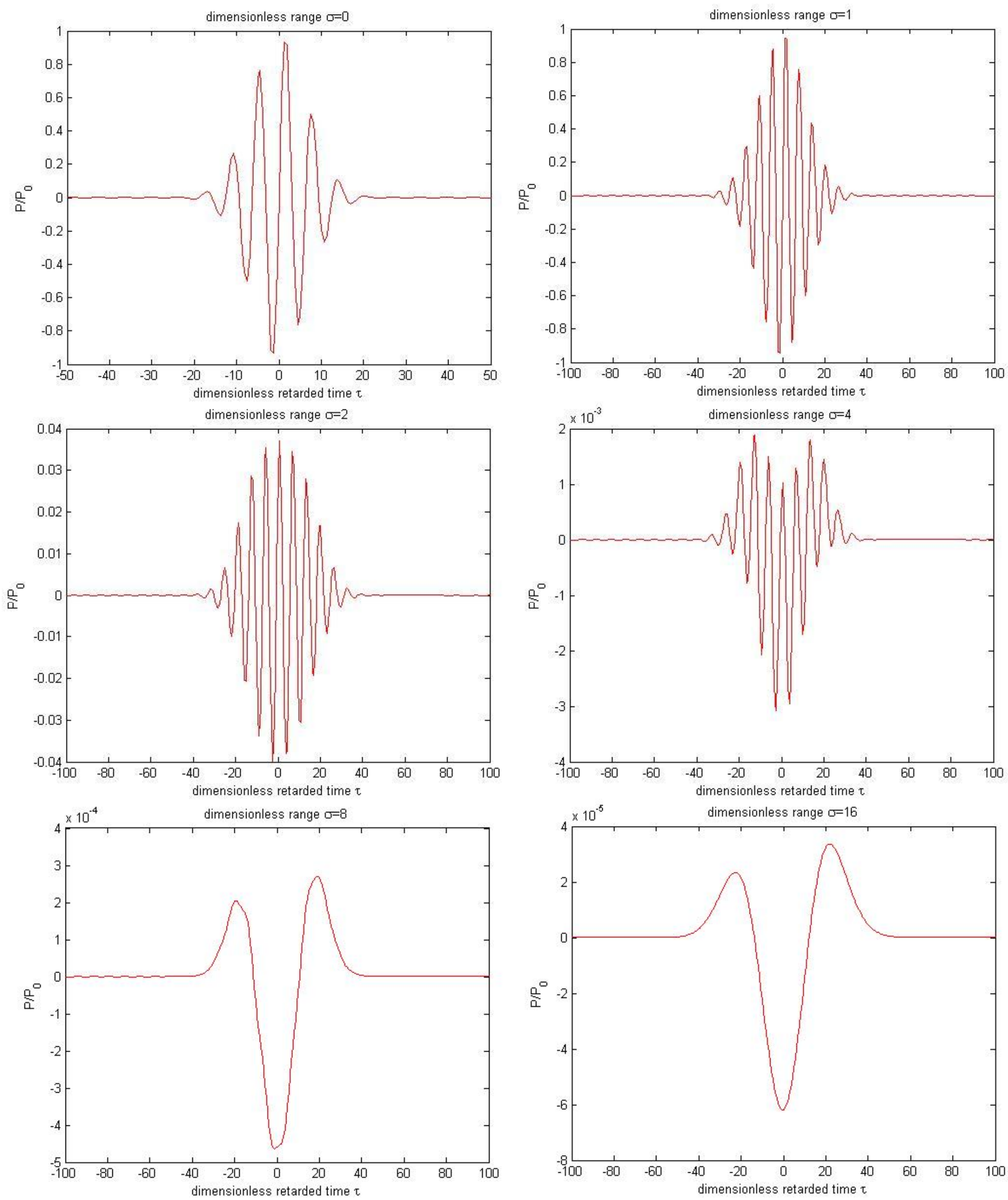

Fig. 4 Axial waveforms at different dimensionless ranges

Although 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 far-field 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:

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

where a, P_0 are source radius and pressure amplitude at source, respectively; and $E\left(\tau\right)$ is the modulation envelop. The other notations are defined identically as in the KZK equation.

Berktay derived his equation from the Westervelt equation. Since KZK equation and Westervelt equation are identical on the axis of farfield, it is not surprising that Berktay equation can also be derived from KZK equation. By quasilinear assumption, 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 parametric array. According to the Berktay's equation, a 12-dB/octave slope in the frequency response of 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. Because 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.

Prototyping of Parametric Loudspeaker

Based on the above theoretical analysis, we design and implement 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): 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 (PZT) or polyvinylidene fluoride (PVDF) materials. Due to the nonlinear interaction of the air, directional audible sound will be generated at a targeted sweet spot.

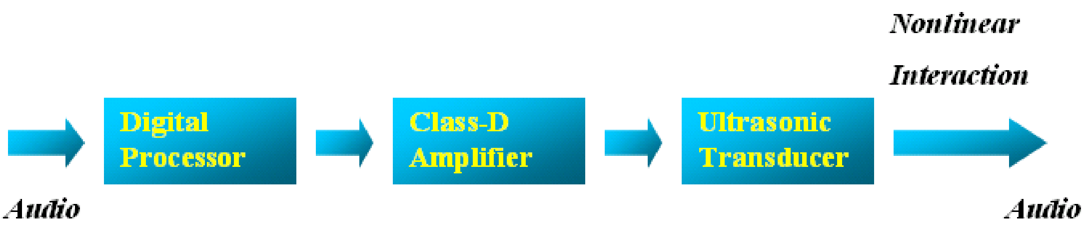

Fig. 5 Block diagram of Audio Beam System

The Audio Beam System has recently been 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. 6 Audio Beam System and its ultrasound emitters

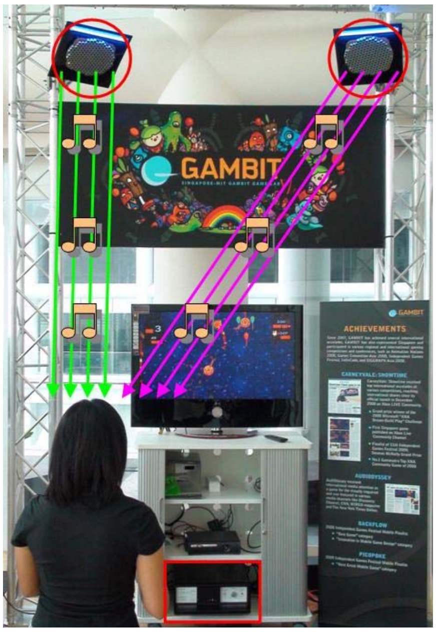

Fig. 7 Audio Beam System installed in a gaming booth

Conclusion

The development of theory on the parametric array in the air provides an attractive and challenge approach to generate directional sound. Parametric loudspeaker has many useful characteristics that allow a high directivity, controllable beam and reasonable implementation size, unlike conventional loudspeaker, which radiate sound wave in omnidirection and requires large array to generate a focused sound. Even though the bass quality of 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.

Acknowledge

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Read more about it

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