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A review of parametric acoustic array in air

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ABSTRACT

In this review paper, we examine some of the recent advances in the parametric acoustic array (PAA) since it was first applied in air in 1983 by Yoneyama. These advances include numerical modelling for nonlinear acoustics, theoretical analysis and experimentation, signal processing techniques, implementation issues, applications of the parametric acoustic array, and some safety concerns in using the PAA in air. We also give a glimpse on some of the new work on the PAA and its new applications. This review paper gives a tutorial overview on some of the foundation work in the PAA, and serves as a prelude to the recent works that are reported by different research groups in this special issue.

1. Introduction

The discovery of the parametric acoustic array (PAA) has come a long way (almost half century ago) since it was first theoretically analyzed by Westervelt [1] in 1963. It has since moved from theory and experimentation to implementation and application. Despite several useful characteristics of the PAA, such as high directivity, small size, and very small sidelobes, we are only just beginning to witness some interesting innovations from commercial companies in deploying the PAA for audio and speech applications. However, there are still some technical challenges concerning ultrasonic emitter performance, conversion efficiency, power consumption, acoustic modelling and measurement, and digital signal processing that are related to the PAA in air. This review paper provides a quick overview of some of the important milestones achieved in the field of the PAA, with the emphasis on creating the PAA in air. It also serves as a preamble to some of the latest works that are reported in this special issue.

This paper is organized as follows. A theoretical framework of the PAA is first introduced in the next section. This is followed by a brief description of the theory of audio parametric loudspeakers in Section 3, and Section 4 outlines several interesting developments of signal processing and modulation techniques for parametric loudspeakers. Section 5 highlights some of the current state-of-the-art implementation platforms used in parametric loudspeakers. The issue of safety in parametric loudspeakers is also highlighted in Section 6. Finally, Section 7 concludes this review paper.

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2. Theoretical framework of the parametric acoustic array

When two sinusoidal beams are radiated from an intense ultrasound source, a spectral component at the difference frequency is secondarily generated along the beams due to the nonlinear interaction of the two primary waves. At the same time, spectral components such as a sum-frequency component and harmonics are generated. However, only the difference-frequency component can travel an appreciable distance because sound absorption is generally increased with frequency, and amplitudes of higher-frequency components decay greatly compared with the difference frequency. The secondary source column of the difference frequency (secondary beam) is virtually created in the primary beam and is distributed along a narrow beam, similar to an end-fire array reported in antenna theory [1], as shown in Fig. 1. Consequently, the directivity of the difference-frequency wave becomes very narrow. This generation model of the difference frequency is referred to as the PAA.

The basic idea of the parametric array was originally conceived by Westervelt about 50 years ago based on the scattering of sound by sound [1]. When two primary waves of frequencies f_1 and f_2 ($f_2 > f_1$) are fully confined beams, he found that the angle, at which the sound intensity of the difference frequency $f = f_2 - f_1$ is reduced by one-half (−3 dB), is given approximately by

$$\theta_h \approx \sqrt{2\alpha_T/k}, \tag{1}$$

where k is the wavenumber of the difference-frequency wave, and α_T is the total sound absorption coefficient of the primary waves. When $f_1 \approx f_2$, $\alpha_T \approx 2\alpha_1$, where α_1 is the absorption coefficient of the primary wave of frequency f_1. Eq. (1) illustrates how narrowing

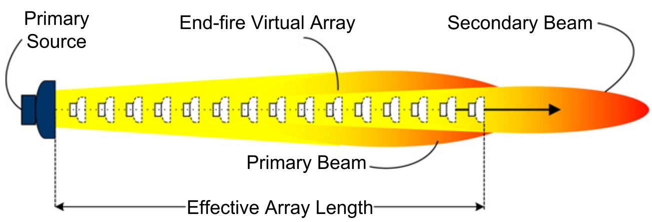

Fig. 1. Generation of audible sound beam through the PAA.

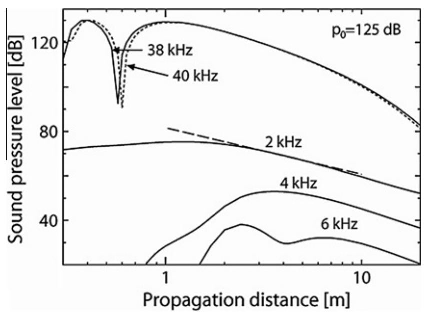

Fig. 2. Axial sound pressure curves for the primary and secondary waves (theory). The two primary waves of 38 kHz and 40 kHz from a sound source of 10 cm in radius produce a 2-kHz difference frequency wave in the air. The initial source pressures of the primaries are set to 125 dB. The dash line denotes an extrapolated −6 dB/dd line.

the secondary beam is realized by decreasing the primary frequencies and/or by increasing the secondary frequency. Interestingly, the directivity of a parametric source is thus independent of the source aperture. This may be contrasted with linear theory, which states the directivity of a sound source is dependent on two parameters, namely driving frequency and source aperture. Strictly, however, the primary beam is dependent on the source aperture of primary waves, especially when the wavelength of the difference wave is comparable with or smaller than the source aperture [2,3]. As an example, let the primary frequency be around 40 kHz and the difference frequency be 2 kHz. The absorption coefficient of airborne ultrasound is approximately 0.15 Neper/m at 40 kHz in ordinary room conditions. Hence from Eq. (1), θ_h is predicted as 7.4°. In contrast, a large aperture source of at least 68 cm in radius is needed by linear theory to realize such a narrow audio beam at 2 kHz.

As described above, the most remarkable acoustic property of the parametric array is its sharp directivity at low frequencies. Additionally, side-lobes, which usually exist for a directive sound source, are suppressed considerably. Furthermore, small changes in the high primary frequencies produce large relative changes in the low secondary frequency. In other words, the parametric sound is equivalent to a low-Q source even if high-Q ultrasound sources are used.

In order to accurately evaluate parametrically generated sound fields, diffraction as well as nonlinearity in primary waves are taken into account. The most useful and traditional model equation for such field evaluation is the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation, which combines nonlinearity, dissipation, and diffraction of a directive sound beam in the same order of magnitudes [4]. This model equation is described as:

$$\frac{\partial^2 p}{\partial z \partial t'} = \frac{c_0}{2}\nabla_\perp^2 p + \frac{\delta}{2c_0^3}\frac{\partial^3 p}{\partial t'^3} + \frac{\beta}{2\rho_0 c_0^3}\frac{\partial^2 p^2}{\partial t'^2}, \quad (2)$$

where p is the sound pressure, c_0 is the sound speed, ρ_0 is the medium density, δ is the sound diffusivity that is related to sound absorption, and β is the nonlinearity coefficient, which is equal to 1.2 for air. Moreover, $\nabla_\perp^2 = \partial^2/\partial x^2 + \partial^2/\partial y^2$ is a Laplacian that operates in the x–y plane perpendicular to the axis of the beam (z axis), and $t' = t - z/c_0$ is the retarded time. It is quite cumbersome to solve analytically the KZK equation even when nonlinearity is weak [3]. Especially, when nonlinearity is moderate or strong, we resort to numerical computation methods such as a finite difference scheme to obtain the solution. Since the KZK equation is derived under the parabolic approximation, we have to pay attention to its applicability; the upper limit of beam angle is restricted to the paraxial region that is within about 20° from the z axis [5].

The fundamental characteristics of parametric sound in air are demonstrated in Figs. 2 and 3. An ultrasound source, whose circular aperture is 10 cm in radius, radiates bifrequency waves of 38 kHz and 40 kHz with the same pressure amplitudes of $p_0 = 50$ Pa (125 dB *re*. 20 μPa) at the source surface. Fig. 2 shows the axial pressure profiles of the primary waves and the parametric sound of the difference frequency at 2 kHz [6]. Note that the amplitude of the parametric sound increases with propagation, attains the maximum at about 1.5 m from the source, and then decreases gradually. Unfortunately, the pressure level of the difference-frequency wave is generally 40 dB or more lower than the level of the primary waves, except in the farfield. Additionally, unwanted harmonic sounds, such as a 4-kHz component, are prominently generated in a field that is a few metres away from the source. Hence, it is of importance to reduce such harmonic distortions and cross-modulation distortions as much as possible in designing parametric loudspeakers.

Fig. 3 shows the comparison of pressure distributions produced by a parametric array and an ordinary piston source under the conditions that the source radii are both 10 cm and audible

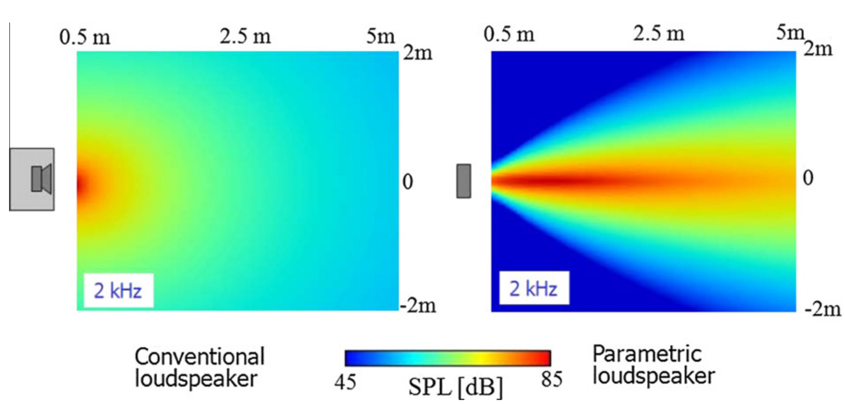

Fig. 3. Sound pressure distributions of a parametric array (right) and an ordinary sound source (left). Both the sources have the same apertures of 10 cm in radius, radiating sounds at a frequency of 2 kHz. The primary frequencies are 40 and 42 kHz.

frequencies of 2 kHz. It is stressed that the directivity of the parametric array is dramatically sharper than that of the ordinary source.

3. Realization of parametric loudspeakers

Since the discovery of the PAA by Westervelt, more than several hundred papers and technical reports on the topics have been published so far from theoretical and experimental points of view. However, almost all the papers before 1975 were focused on exploratory examinations and underwater applications [7,8]. In 1965, Berktay provided a simple expression on the pulse produced by the self-demodulation of a pulsed carrier [9], serving as the basis of predicting the far-field array response of a parametric loudspeaker. The expression states that the modulated signal (or audio difference frequency) pressure $p(t')$ along the axis of propagation is proportional to the second time-derivative of the square of the envelope of the amplitude-modulated ultrasound carrier as follows:

$$p(t') \approx \frac{\beta p_0^2 a^2}{16\rho_0 c_0^4 z \alpha_0} \frac{d^2}{dt'^2} E^2(t'), \tag{3}$$

where p_0 is the pressure source amplitude, a is the source radius, α_0 is the absorption coefficient of the ultrasound carrier, and $E(t')$ is the modulation envelope function of the carrier. Eq. (3) shows that the demodulated signal is principally proportional to the size of the ultrasound source, the pressure amplitude of primary wave, and the envelope function form. However, the high-frequency carrier reduces the amplitudes of parametric sounds due to the increase of carrier sound absorption. Therefore, higher audible sound pressures at a distance can be achieved by changing the values of the above four parameters. The application of Berktay's model is limited to cases where the primary source pressure is relatively low so that the parametric array is determined by small-signal absorptions of the primary waves. Later, Merklinger advanced Berktay's analysis to the case where the ultrasound carrier is sufficiently intense to cause nonlinear attenuation [10]. His expression is stated as

$$p(t') = \frac{S p_0}{4\omega\pi c_0 z} \frac{\partial^2}{\partial t'^2} \left[E(t') \tan^{-1} \left\{ \frac{\beta\omega p_0 E(t')}{4\alpha \rho_0 c_0^3} \right\} \right], \tag{4}$$

where S is the aperture area of an ultrasound source and is given as πa^2 for a circular aperture with radius a. In Eq. (4), the solution is approximately subjected to $p_0 \ll 4\alpha\rho_0 c_0^3/\beta\omega$ as

$$p(t') \propto p_0^2 \frac{\partial^2}{\partial t'^2} E^2(t'). \tag{5}$$

This solution indicates that the amplitude of produced parametric sound is proportional to the square of the envelope and is the same form as Berktay's expression. Whereas, when nonlinearity is strong and the condition $p_0 \gg 4\alpha\rho_0 c_0^3/\beta\omega$ is satisfied, the expression becomes

$$p(t') \propto p_0 \frac{\partial^2}{\partial t'^2} |E(t')|. \tag{6}$$

We note that the parametric sound is proportional to the amplitude of the carrier itself. It is therefore necessary to appropriately take account of the demodulation processes being changed from Eq. (5) to Eq. (6) in response to the primary wave amplitude when designing suitable parametric loudspeakers [11]. The following section presents the signal processing and modulation techniques that are derived from the Berktay's model (Eq. (3) or Eq. (5)).

4. Signal processing and modulation techniques

When employing the PAA principle for directional sound, the chosen primary wave usually lies beyond the human hearing range, typically at around 40 kHz, which is amplitude-modulated by audio signals. Thus, amplitude-modulated ultrasound wave has a carrier, upper and lower side-band components, resulting in reproduction of the audible sound in air due to the nonlinear interaction of the carrier and each side-band in the ultrasound beams. Needless to say, the directivity of the reproduced audible sound is very sharp owing to the characteristic of the parametric array. It shall be shown that the sound pressure level and harmonic distortion of the demodulated signal are proportional to the modulation index, therefore care must be exercised to determine the modulation index used in amplitude modulation (AM). We will discuss different types of modulation techniques that can reduce the distortion introduced by the self-demodulation process in the following subsections. A single-tone analysis is used to evaluate the performance of reducing distortion for different pre-processing and modulation techniques.

4.1. Double sideband amplitude modulation

In 1983, Yoneyama et al. [12] proposed a parametric loudspeaker system, which used the conventional AM or the double sideband amplitude modulation (DSBAM). The modulation envelope for their parametric loudspeaker system is given as $E(t') = 1 + mg(t')$, where m is the modulation index and $g(t')$ is the input signal. The block diagram of the DSBAM is shown in Fig. 4, where $\sin(\omega_0 t')$ is the ultrasonic carrier.

However, this original PAA system would cause high total harmonic distortion (THD). This phenomenon was validated by their experiments, where the second harmonic was of similar level to the fundamental signal in the case of a single-tone input for high m. Fig. 5 summarizes the THD vs. modulation index m for DSBAM. This figure shows that DSBAM is not a preferred technique because it incurs high distortion at high m. Moreover, a high modulation index is required to produce a demodulated signal with desirable high sound pressure level at the expense of increasing distortion. By reducing the modulation index, there is a tradeoff between sound pressure level of the demodulated signal and lower distortion, which is not desirable for practical applications. Therefore, DSBAM is seldom used as the modulation technique for parametric loudspeakers, except for the case where DSBAM is employed to evaluate the acoustic performances of the PAA. In the following sections, several modified amplitude modulation techniques, which achieve high demodulated sound pressure level with reduced distortion, will be presented.

4.2. Square-root amplitude modulation

Based on the Berktay's model expressed in Eq. (3), many other attempts [13–22] have been implemented to improve the quality of the demodulated signal. In 1984, Kamakura et al. [13,14] presented a square-root AM (SRAM) method that applied an envelope function:

$$E(t') = \sqrt{1 + mg(t')}. \tag{7}$$

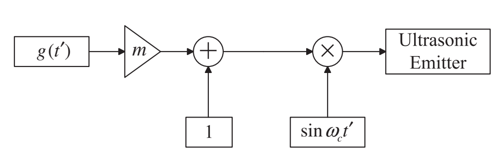

Fig. 4. Block diagram of the double sideband amplitude modulation.

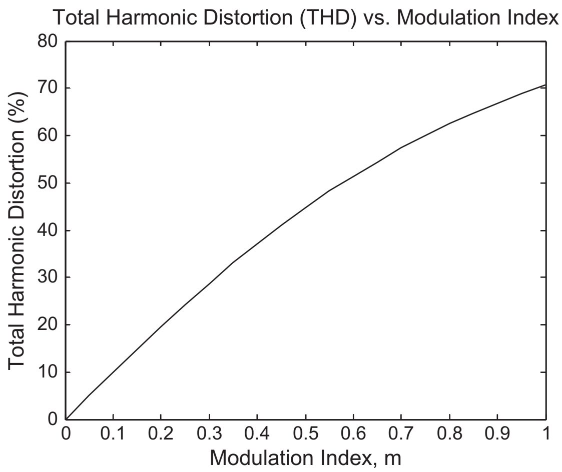

Fig. 5. Total harmonic distortion vs. modulation index.

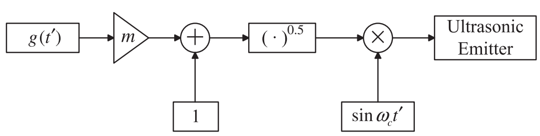

Fig. 6. Block diagram of square-root amplitude modulation.

The block diagram of the SRAM method is demonstrated in Fig. 6. Compared to the conventional DSBAM method, lower THD values have been achieved. However, the ultrasonic emitter with large bandwidth is required to generate the infinite harmonics introduced by the square-root operation. The same technique was also adopted by Pompei [15] and Kite et al. [16].

4.3. Single-sideband amplitude modulation

Another modulation method was proposed to reduce both the distortion and power consumption in driving parametric loudspeaker based on the single-sideband amplitude modulation (SSB-AM) [17], as shown in Fig. 7. The major advantage of the SSB-AM method is that it produces a similar envelope as that in the SRAM method, with only half the bandwidth of the SRAM method. In the case of two primary waves, there is no difference between the envelope produced by the SSB-AM and the SRAM methods, as shown in Fig. 8. This feature in SSB-AM implies that it is necessary to use a high bandwidth emitter, as in the case of SRAM, to achieve low THD performance. However, in case where multiple primary waves or a broadband signal (such as speech) modulates a single primary carrier, envelope error occurs.

Hence, Croft et al. [18,19] proposed a recursive SSB-AM (RSSB-AM) method to approximate the envelope generated by the SRAM method, as shown in Fig. 9d. The RSSB-AM method consists of the SSB modulator and the nonlinear demodulator (NLD), as shown in Fig. 9a and b, respectively. The role of the NLD is to calculate the square of the envelope, which models the nonlinear acoustic propagation in air by assuming that the second-time derivative effect in the Berktay's model can be perfectly compensated. These two blocks are combined into a distortion model (DM) in Fig. 9c and subtracted from the original input $g(t) = g_0(t)$ to obtain the distortion $d_i(t)$ at the output of the ith stage, where $i = 1, 2, \ldots, q$. The distortion is progressively reduced by each DM stage, and higher reduction is achieved by cascading several DM stages. Because of the high complexity of the RSSB-AM method, a high-speed processor must be employed to achieve real-time performance. A detailed investigation on the analytical performance of the RSSB-AM method was given in [20], where an RSSB-AM method with optimal parameters were presented for directional speech reproduction.

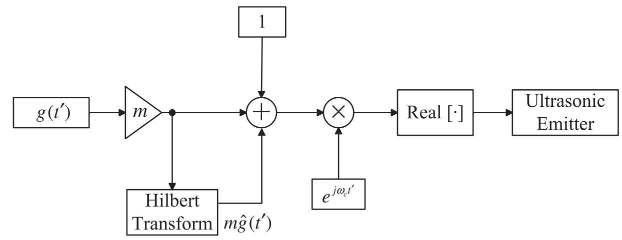

Fig. 7. Block diagram for single sideband modulation with carrier.

4.4. Modified amplitude modulation

A different modulation technique known as the modified AM (MAM) method, which is a class of hybrid AM and SRAM methods based on the orthogonal amplitude modulation, is proposed in [21,22]. The block diagram is shown in Fig. 10. It has the flexibility to scale the relative bandwidth requirement to match the bandwidth of the ultrasonic emitters, and also provides complexity scaling.

5. Implementation Issues

In this section, we look into some of the implementation issues on developing the parametric loudspeakers in analog and digital circuits. The parametric loudspeaker system consists of three main parts, namely, signal processing, amplifier, and ultrasonic emitter, as shown in Fig. 11. Some of the early parametric loudspeakers used analog circuit elements to perform the signal processing block. Fig. 12 exemplifies a typical analog circuit for implementing the PAA system [23]. A peak level detector, which is fed by the input signal, is employed to prevent overmodulation and ensure the envelop signal generated from the adder is always positive. The peak level detector can be designed with a voltage comparator (LM111), and the adder is simply achieved by using an operational amplifier. Next, the envelope signal is pre-processed by applying a square-root operation using a four-quadrant analog multiplier (AD734). This pre-processing step has been previously explained in Section 4.2 to reduce the distortion produced during the nonlinear interaction of the air propagation. The carrier signal is generated using a high-frequency, precision waveform generator (MAX038). In the final step, the carrier signal is multiplied with the pre-processed signal to form the modulated signal. The analog development of a parametric loudspeaker has the advantages of low cost and simple structure. However, the analog circuit is less flexible, less stable, and it is inefficient for implementing complicated functions and operations using analog components, such as a Hilbert filter in the SSBAM scheme. These disadvantages have limited the development of high performance parametric loudspeaker.

A digital signal processor has the ability to process more complicated operations without incurring extra cost, size, and power consumption compared to the analog circuit system. A field programmable gate array (FPGA) is an attractive digital processing platform to implement the parametric loudspeaker due to its flexible configuration and high performance. FPGA has a programmable feature of logic cells, which can be interconnected to perform signal processing algorithms either in sequential or parallel operation. Thus, the FPGA can be configured for high-speed parallel processing with the tradeoff of more logic gates used.

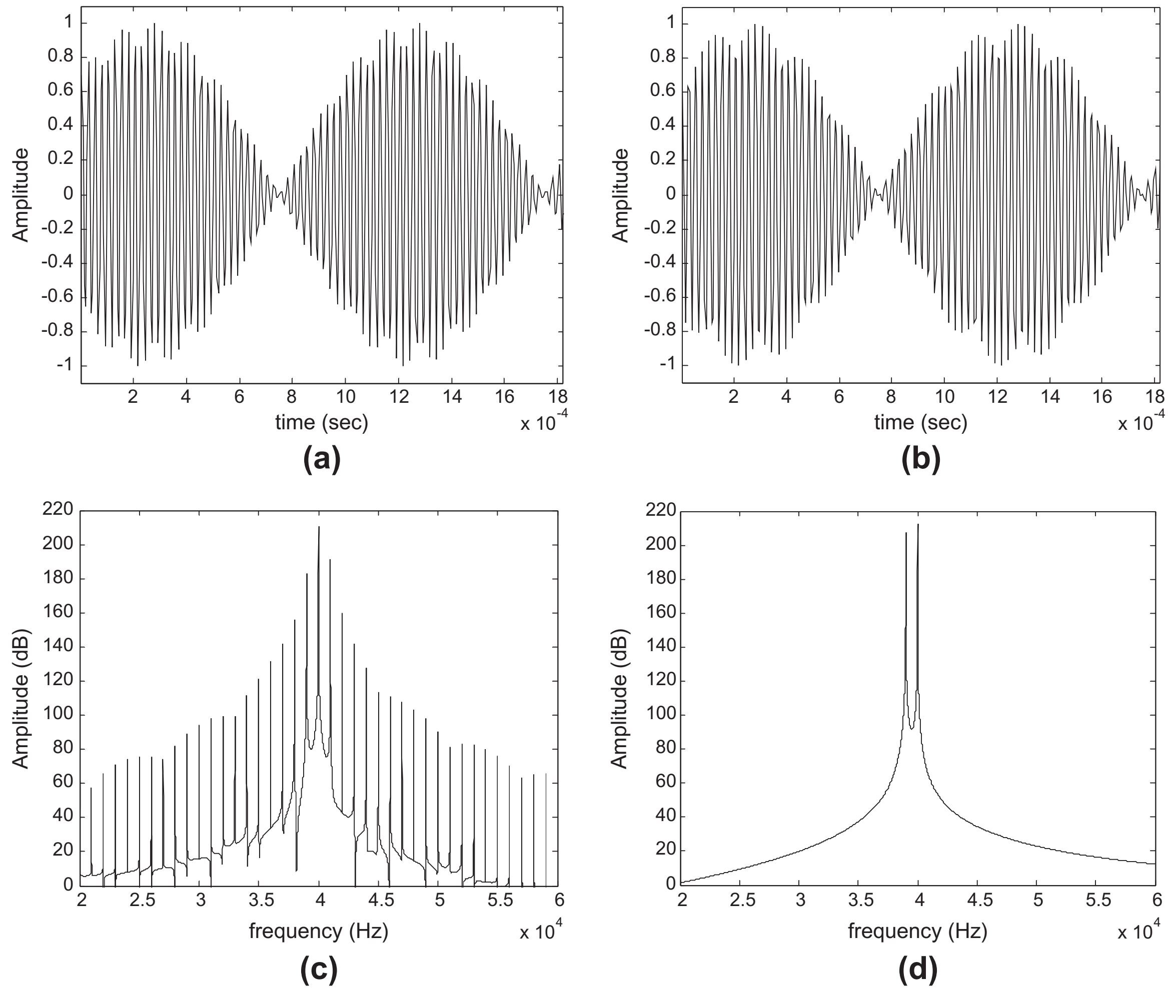

Fig. 8. Time domain plots of modulation envelopes of (a) SRAM, and (b) SSB; frequency spectra of the modulated signal of (c) SRAM and (d) SSB.

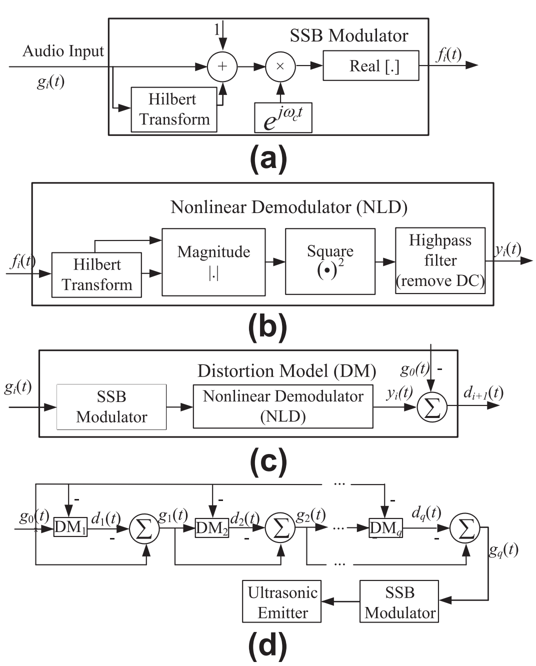

Fig. 9. Block diagram of the RSSB-AM and its components. (a) SSB modulator, (b) NLD, (c) single DM stage, and (d) cascade of DM stages to form the RSSB-AM.

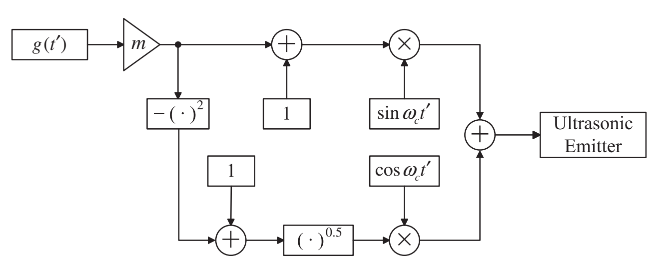

Fig. 10. Block diagram of the MAM method.

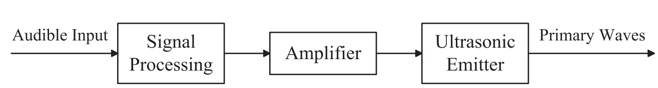

Fig. 11. Block diagram of parametric loudspeaker.

Fig. 13 shows the generic block diagram for implementing digital pre-processing and modulation in the parametric loudspeaker [24]. A dynamic range control processor consists of a compressor and a limiter. By compressing the dynamic range of the input speech or musical signal with a constant gain, the reproduced sound can be perceived clearly without noticeable distortion. The limiter is used to prevent the input signal from clipping, which

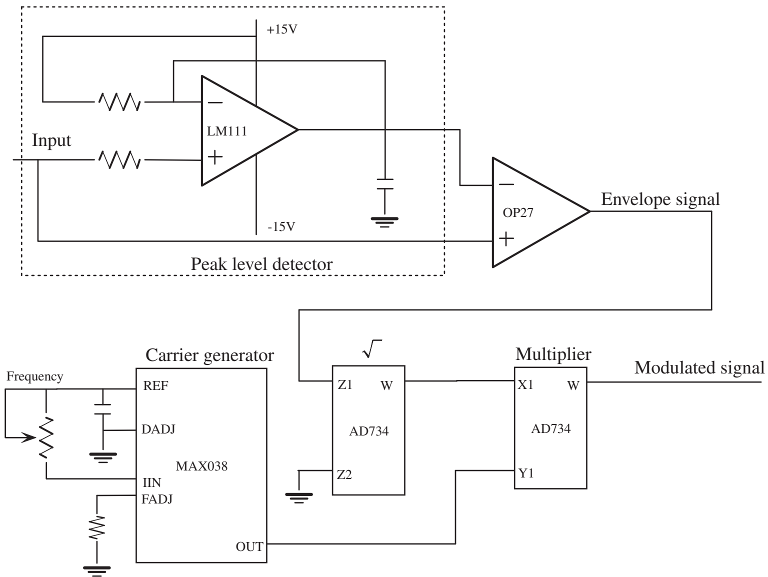

Fig. 12. Analog implementation of the PAA system.

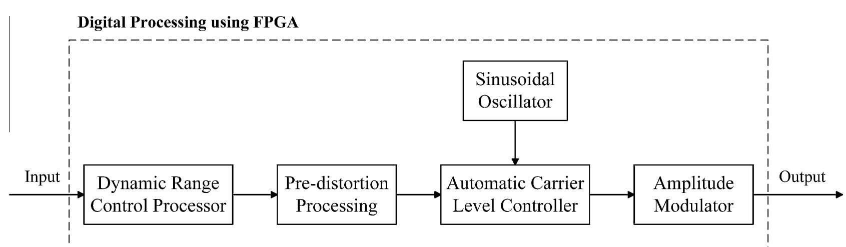

Fig. 13. Digital processing of parametric loudspeaker.

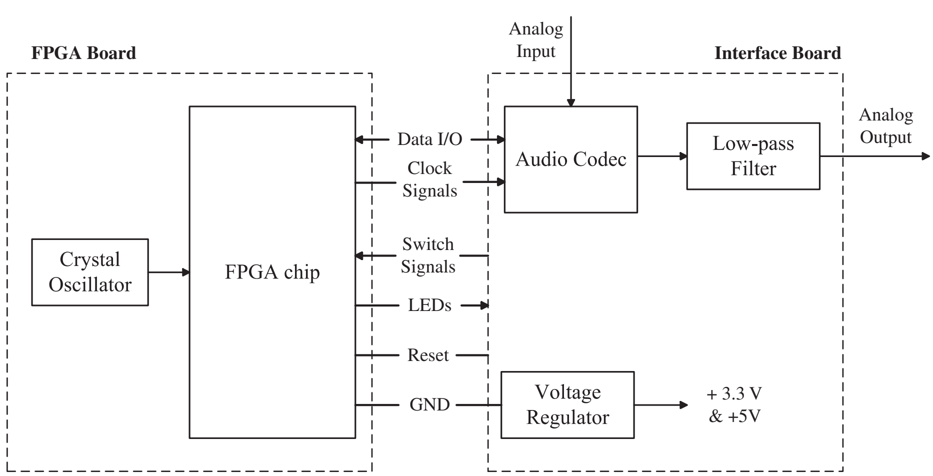

Fig. 14. Interconnection between FPGA board and interface board.

can cause perceivable distortion in the audible output. The output of the dynamic range control processor is passed to the pre-distortion processing block. This block pre-processes the input signal to reduce the amount of distortion produced from the higher-order nonlinear interaction under nonlinear acoustics propagation. The pre-processing methods have been previously discussed in Section 4.2. Next, an automatic carrier level controller performs automatic gain control to the carrier signal based on the input level. The gain is calculated to proportionally scale the oscillator output with respect to the input level, which can result in significant reduction in power consumption. The sinusoidal oscillator, which can be implemented using either a recursive algorithm or a look-up table, is used to generate the required (single or orthogonal) carrier signal(s). The final process modulates the dynamic carrier signal using the pre-processed audible signal.

Fig. 14 shows the connection between the FPGA chip and the audio analog–digital convertor (ADC) and the ultrasonic-frequency digital-to-analog convertor (DAC). The FPGA chip performs the digital signal processing, as described in Fig. 13. An audio codec, which consists of a two-channel 16-bit 96 kHz ADC and 192 kHz DAC, is

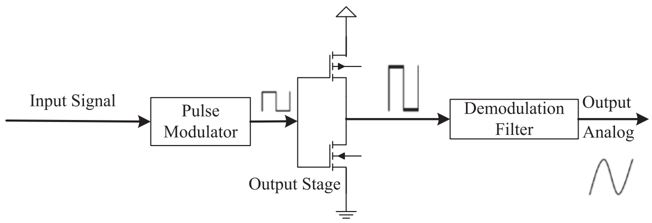

Fig. 15. Block diagram of Class-D amplifier for the PAA.

used in our digital implementation. This audio codec is suitable for emitting ultrasonic frequencies in the range of 30–50 kHz, which falls within the resonating frequency of most ultrasonic emitters used in parametric loudspeakers. A low-pass filter on the interface board performs signal conditioning and the voltage regulator provides a clean and stable DC supply across the board.

In order to create a parametric array effect in air, the modulated signal must be amplified before being fed into the ultrasound emitter. The Class-D power amplifier is widely used in the parametric loudspeaker due to its efficiency and small size. Fig. 15 shows the block diagram of the Class-D amplifier used in the parametric loudspeakers. In the preceding FPGA implementation, the digital modulated signal can be directly converted to a pulse train using the pulse modulator block, without the need of a DAC. Two pulse modulation techniques are commonly used: pulse width modulation (PWM) and pulse density modulation (PDM). PWM generates pulses with widths proportional to the input amplitude, while PDM generates pulses with fixed width but with the density of the pulse train dependent upon the amplitude [25]. PWM is more efficient than PDM, but generates higher harmonic distortion. In the Class-D output stage, the incoming pulses are amplified so as to drive the ultrasound emitter. Since the switching between the on and off stages is not ideal, the output stage will add distortion and noise to the signal channel. This problem can be significantly alleviated if an error correction technique [26] is adopted. Finally, the analog signal is derived by using a demodulation filter, which should be carefully designed to remove the high frequency products and maximize the power transfer.

6. Applications of the parametric acoustic array

The PAA has commonly been used in display kiosk, entertainment, communication, and personal messaging systems [27,28] due to its ability to create a highly directional sound beam using a small-surfaced emitter. In this section, we highlight some recent applications of the PAA that show promising potential in exploiting the unique features of the PAA.

Traditionally, in active noise control (ANC) systems, omni-directional loudspeakers are utilized as control sources to suppress the noise in the targeted area. However, the sound pressure level (SPL) of the noise at locations away from the target area may actually increase. To solve this problem, Brooks et al. [29] investigated the feasibility of using a parametric loudspeaker as a control source in ANC systems. Experimental results showed that an ANC system using a parametric loudspeaker could reduce the SPLs at the control points without increasing noise level in other areas. Kidner et al. [30] proposed a method of combining the PAA and virtual sensing techniques to create localized zones of quiet. Tanaka and Tanaka [31,32] presented a novel idea of using a steerable PAA based upon phased array theory with an optimal control law to track a moving target point without rotating the parametric loudspeaker mechanically. Furthermore, the same research group [33] proposed a novel method to use the reflected sound wave, which is produced by the parametric loudspeaker, on the target source to collocate both the primary source and the control source. Therefore, global noise control could be achieved because the distance between the two sources became theoretically nullified. In another work, Komatsuzaki and Iwata [34] compared the interfered sound field resulting from using the parametric loudspeaker and omni-directional loudspeaker as control sources to achieve local noise control. Three different constructions of the primary source and the control source were considered: opposed placement, orthogonal placement, and coaxial placement. Results showed that the parametric loudspeaker can be used as a control source to mitigate sound locally without increasing noise level in other areas.

In the area of personal communication, Nakashima et al. [35,36] mounted two parametric loudspeakers on a prototype mobile phone to deliver a private sound field (or personal sound) to the users. Each parametric loudspeaker consisted of 16 piezoelectric emitters. The SPL of the audible sound on the central axis was measured to be more than 70 dB at a distance of about 50 cm away from the phone. Because of the directivity of the PAA, the sound pressure difference between the two ears is approximately 15 dB, which is helpful in binaural sound reproduction.

Due to the high directivity of sound wave at low frequencies, the PAA has also been found to be suitable in detecting concealed objects, such as land mines and weapons. Haupt and Rolt [37] used the PAA to excite the buried mines from a safe distance. The vibration signatures of the mines were measured by the laser vibrometer to locate the position of the mines. In another work, Achanta et al. [38] presented a novel method to use the PAA to detect concealed weapons. The metallic and non-metallic materials under clothing, as well as abnormality in handheld devices, can be scanned and detected by the directional sound at low frequencies generated by the PAA system.

The PAA can also be deployed to measure the acoustic performance of materials. The application of the PAA in this field was introduced by Humphrey [39–41]. His group used a PAA as a sound source to determine the reflection loss and transmission loss of panels in a vessel which simulated the ocean environment. With the highly directional sound beams, the effects of diffraction from the panel edges were reduced. Castagnede's group used a PAA and a microphone to measure the absorption coefficient of material at normal incidence in air under the excitation of impulse signals [42,43]. Kuang et al. introduced the transfer function method into the process [44], which allowed measurements to be taken in situ. Moreover, this method can be combined with the phase-cancellation method proposed by Kamakura [45] to effectively remove spurious noise for accurate measurements [46].

Many theoretical and/or experimental data have also been previously reported for the PAA in underwater applications. The acoustic features of the parametric array in water are basically the same as that in air, with sharp directivity at low frequency and with suppressed side-lobes. The technique that uses parametric sound propagation, such as navigation, communication with other vessels, or detection of targets is sometimes referred to as the parametric sonar. The parametric sonar is one of the most matured technologies developed and evaluated in underwater acoustics, and its academic papers and technical reports are summarized in scientific monographs and reference materials [7,8]. Applications and challenges of PPA in shallow water sonar and sub-bottom profilers are also current research of interest.

7. Safety issues

In this section, we shall discuss how airborne ultrasound affects human body. Almost 99.9% of the ultrasound energy is reflected by the human skin due to the high impedance mismatch between air and the skin. Generally, the major effects of ultrasound exposure in

Table 1
Guidelines for the safe usage of ultrasound that are recommended in various countries. All values in decibels are upper limits for whole-day exposure. Table extracted from [47].

One-third-octave band centre frequency (kHz)	10	12.5	16	20	25	31.5	40	50	63	80	100
Source	Band level (dB)										
IRPA, 1984	–	–	–	75	110	110	110	110	110	110	110
Australia, 1981	75	75	75	110	110	110	110	110	–	–	–
USA, 1981	80	80	80	105	110	115	115	115	–	–	–
Canada, 1980	80	80	80	110	110	110	110	110	–	–	–
Sweden, 1978	–	–	–	105	110	115	115	115	115	115	115
USSR, 1975	–	75	85	110	110	110	110	110	110	110	110
Norway, 1978	–	–	–	–	—120 (octave)—			—120 (octave)—			–

practice are induced through the ear. This is especially obvious if the ultrasound level is extremely high and may cause unpleasant sensation, headache, fatigue, and nausea [47]. Moreover, the symptoms vary from person to person. These effects are temporary and are relieved by removing the ultrasound in most situations. It was reported that long hours of exposure may cause threshold shift in hearing loss [48]. However, there is no report that describes the cause-and-effect relationship between hearing loss and exposure of the ear to high frequency ultrasound.

For half a century, considerable efforts have been carried out to find the human exposure limits of ultrasound as carried out by various countries and several health organizations. Such guidelines have been presented for industrial workers using ultrasonic devices, such as cleaning, drilling, and soldering from health and safety protection points of view. Most of the guidelines recommend an upper limit of ultrasound between 100 and 115 dB for frequencies above 20 kHz as the admissible sound pressure level, which will not cause audiometric hearing loss [47].

Ultrasonic devices obviously produce intense ultrasound, whose frequencies are above 20 kHz. Additionally, these devices generate wide band noisy audible by-products, which are possible sources of annoyance for workers, rather than the inaudible ultrasound. Table 1 shows the maximum permissible sound pressure levels [47] for the safe usage of ultrasound in various countries and organizations. Basically, the pressure levels in the table are applied to continuous exposure for an 8-h working day. Some reports indicate that the exposure limit levels may be increased for shorter exposure durations, as shown in Table 2. For example, the International Radiation Protection Association (IRPA) committee recommends that the permissible levels may be increased by 3 dB for a duration of 2–4 h. However, the maximum limit prescribed by the Health Canada is 110 dB above 25 kHz and it is independent of duration [48,49].

In the parametric loudspeaker, the attained sound pressure level can reach 120 dB, especially in the nearfield. Depending on the dimension of parametric loudspeaker and its initial sound pressure, a pressure level of 120 dB can be attained at 1 m from the ultrasonic emitter, although such high pressure is limited to the on-axis beam. Generally, the pressure level decreases beyond the Rayleigh distance to less than 100 dB at 8 m. However, it is not appropriate to readily apply the permissible pressure levels in Table 2 as safety guidelines for operating the parametric loudspeakers, since the ultrasound waves emitted from the parametric loudspeakers consist of relatively narrow spectral bands around the carrier frequency. This is fundamentally different from ultrasonic devices that radiate wideband ultrasound as well as broadband audible sound in space. However, as the specific response to ultrasound frequencies is still unclear, the maximum permissible levels of 105–110 dB should be kept in mind even for parametric loudspeakers in short time operation because of questionable problems in auditory and non-auditory biological effects of intense ultrasounds around the carrier frequencies, such as 40 kHz. Recently, Lee et al. [50] suggested from a physiological point of view that the burden of a parametric loudspeaker is smaller than

Table 2
Supplement for exposure duration less than 8 h. Table extracted from [47].

Source	Duration (h)	Correction (dB)
IRPA, 1984	2–4	+3
	1–2	+6
	<1	+9
Sweden, 1978	1–4	+3
	<1	+9
USSR, 1975	1–4	+6
	¼–1	+12
	<¼	+18

a conventional loudspeaker in listening tests of short-sentence speech. Unfortunately, their report does not describe the relationship between ultrasound pressure levels and physiological effects of ultrasound on the auditory function of human being.

In summary, one of the important considerations in designing and developing a parametric loudspeaker is to examine how to reduce the carrier ultrasound to a safe pressure level without compromising the performance of parametric sound.

8. Conclusions

This review paper serves as a quick tutorial reference to readers, who are interested to further explore and extend this technology, and bring this technology to other application areas. This review outlined some of the important theoretical development of the PAA, and gave an overview of different technologies, implementation techniques, applications and safety issue for parametric loudspeakers. In particularly, we describe the current state-of-the-art technology in pre-processing and modulating the audible signal before sending to the ultrasonic emitters. However, these pre-processing and modulation techniques are mainly based on the Berktay's model, which is only approximated solution for the sound beam propagation in weak nonlinearity. It is the quest for future work to obtain a more accurate and simple-to-use model that can be used to derive a new pre-processing technique to equalize the nonlinear distortion. Also, the computation cost for implementing different pre-processing and beam control techniques must be taken into consideration in building the next-generation parametric loudspeakers, where new requirements, such as high audio quality, electronic steerable, and accurate audio beam control are integrated into a personal sound entertainment system and other novel applications, and some of which are being documented in this special issue.

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References

[1] Westervelt PJ. Parametric acoustic array. J Acoust Soc Am 1963;35(4):535–7.
[2] Naze J, Tjøtta S. Nonlinear interaction of two sound beams. J Acoust Soc Am 1965;37(1):174–5.
[3] Hamilton MF. Sound beams. In: Hamilton MF, Blackstock DT, editors. Nonlinear acoustics. New York: Academic Press; 1997 [chapter 8].
[4] Hamilton MF, Morfey CL. Model equations. In: Hamilton MF, Blackstock DT, editors. Nonlinear acoustics. New York: Academic Press; 1997 [chapter 3].
[5] Jensen FB, Kuperman WA, Porter MB, Schmidt H. Computational ocean acoustics. New York: Springer Verlag; 2000 [chapter 6].
[6] Kamakura T, Aoki K, Sakai S. A highly directional audio system using a parametric array in air. In: Proc 9th Western Pacific acoustics conf, Seoul; June 26–28, 2006. p. 1–8.
[7] Novikov BK, Rudenko OV, Timoshenko VI. Nonlinear underwater acoustics. New York: AIP Press; 1987.
[8] Moffett MB, Konrad WL. Nonlinear sources and receivers. In: Crocker MJ, editor. Encyclopedia of acoustics. New York: John Wiley & Sons; 1997. p. 607–17.
[9] Berktay HO. Possible exploitation of nonlinear acoustics in underwater transmitting applications. J Sound Vib 1965;2(4):435–61.
[10] Merklinger HM. Improved efficiency in the parametric transmitting array. J Acoust Soc Am 1975;58(4):784–7.
[11] Aoki K, Kamakura T, Kumamoto Y. Parametric loudspeaker – characteristics of acoustic field and suitable modulation of carrier ultrasound. Electron Commun Jpn 1991;74(9):76–82.
[12] Yoneyama M, Fujimoto J, Kawamo Y, Sasabe S. The audio spotlight: an application of nonlinear interaction of sound waves to a new type of loudspeaker design. J Acoust Soc Am 1983;73(5):1532–6.
[13] Kamakura T, Yoneyama M, Ikegaya K. Development of a parametric loudspeaker for practical use. In: 10th International symposium on nonlinear acoustics, Kobe; 1984. p. 147–50.
[14] Kamakura T, Yoneyama M, Ikegaya K. A study for the realization of a parametric loudspeaker. J Acoust Soc Jpn 1985(June):1–18.
[15] Pompei FJ. The use of airborne ultrasonics for generating audible sound beams. J Audio Eng Soc 1999;47(9):726–31.
[16] Kite TD, Post JT, Hamilton MF. Parametric array in air: distortion reduction by preprocessing. In: Proc 16th ICA and 135th meeting of the ASA, vol. 2; 1998. p. 1091–2.
[17] Sakai S, Kamakura T. Dynamic single sideband modulation for realizing parametric loudspeaker. In: AIP conf proc; 2008. p. 613–6.
[18] Croft JJ, Spencer ME, Norris JO. Modulator processing for a parametric speaker system, US 20080063214A1; 2003.
[19] Croft JJ, Norris JO. Theory, history, and the advancement of parametric loudspeaker. White paper of American Technology Corporation; 2001.
[20] Ji P, Gan WS, Tan EL, Yang J. Performance analysis on recursive single-sideband amplitude modulation for parametric loudspeakers. In: IEEE international conference on multimedia and expo (ICME), Singapore; 2010. p. 748–56.
[21] Tan EL, Gan WS, Ji P, Yang J. Distortion analysis and reduction for the parametric array. In: 124th Convention of the audio engineering society, Amsterdam; 2008.
[22] Tan EL, Ji P, Gan WS. On preprocessing techniques for bandlimited parametric loudspeakers. Appl Acoust 2010;71(5):486–92.
[23] Pompei J. Sound from ultrasound: the parametric array as an audible sound source. PhD thesis. School of Architecture and Planning, Massachusetts Institute of Technology; 2002.
[24] Karnapi FA, Gan WS, Chong YK. FPGA implementation of parametric loudspeaker system. Microprocess Microsyst 2004;28(5–6):261–72.
[25] Stark S. Direct digital pulse width modulation for class D amplifiers. Master thesis. Department of Electrical Engineering, Linköping University, Sweden; 2007.
[26] Nielsen K. A review and comparison of pulse width modulation (PWM) methods for analog and digital input switching power amplifiers. Bang & Olufsen A/S, Denmark and Institute of Automation, Technical University of Denmark.
[27] Pompei J. Audio spotlight. Sound Commun 2009;55(7):54–7.
[28] Gan WS, Tan EL, Kuo SM. Audio projection: directional sound and its application in immersive communication. IEEE Signal Process Mag 2011;28(1):43–57.
[29] Brooks LA, Zander AC, Hansen CH. Investigation into the feasibility of using a parametric array control source in an active noise control system. In: Proc acoustics 2005, Busselton; 2005.
[30] Kidner MRF, Petersen C, Zander AC, Hansen CH. Feasibility study of localized active noise control using an audio spotlight and virtual sensors. In: Proc acoustics 2006, Christchurch; 2006.
[31] Tanaka N, Tanaka M. Active noise control using a steerable parametric array loudspeaker. J Acoust Soc Am 2010;127(6):3526–37.
[32] Tanaka N, Tanaka M. Active noise control using a parametric array loudspeaker. In: 17th International congress on sound and vibration, Cairo; 2010.
[33] Tanaka N, Tanaka M. Mathematically trivial control of sound using a parametric beam focusing source. J Acoust Soc Am 2011;129(1):165–72.
[34] Komatsuzaki T, Iwata Y. Active noise control using high-directional parametric loudspeaker. J Environ Eng 2011;6(1):140–9.
[35] Nakashima Y, Yoshimura T, Ohya T. Prototype of parametric array loudspeaker on mobile phone and its acoustical characteristics. In: 118th Convention of the audio engineering society, Barcelona; 2005.
[36] Nakashima Y, Yoshimura T, Naka N, Ohya T. Prototype of mobile super directional loudspeaker. NTT DoCoMo Tech J 2006;8(1):25–32.
[37] Haupt RW, Rolt KD. Standoff acoustic laser technique to locate buried land mines. Lincoln Lab J 2005;15(2):3–22.
[38] Achanta A, McKenna M, Guy S, Malyarenko E, Lynch T, Heyman J, et al. Nonlinear acoustic concealed weapons detection. Mater Eval 2005;63(12):1195–202.
[39] Humphrey VF. The measurement of acoustic properties of limited size panels by use of a parametric source. J Sound Vib 1985;98(1):67–81.
[40] Humphrey VF. The influence of the plane wave spectrum of a source on measurements of the transmission coefficient of a panel. J Sound Vib 1986;108(2):261–72.
[41] Humphrey VF, Robinson SP, Smith JD, Martin MJ, Beamiss GA, Hayman G, et al. Acoustic characterization of panel materials under simulated ocean conditions using a parametric array source. J Acoust Soc Am 2008;124(2):803–14.
[42] Castagnede B, Saeid M, Moussatov A, Gusev V, Tournat V. Reflection and transmission at normal incidence onto air-saturated porous materials and direct measurements based on parametric demodulated ultrasonic waves. Ultrasonics 2006;44(2):221–9.
[43] Castagnede B, Moussatov A, Lafarge D, Saeid M. Low frequency in situ metrology of absorption and dispersion of sound absorbing porous materials base on high power ultrasonic non-linearly demodulated waves. Appl Acoust 2008;69:634–48.
[44] Kuang Z, Wu M, Ye C, Yang J. Method for measuring the absorption coefficient of sound absorbing materials in situ. Acta Acoust 2010;35(2):162–8 [in Chinese].
[45] Kamakura T, Sakai S, Nomura H, Akiyama M. Parametric audible sounds by phase-cancellation excitation of primary waves. In: Acoustics08, Paris; 2008.
[46] Kuang Z. Investigation on the measurement of absorption coefficients in situ with the parametric loudspeaker. PhD thesis. Institute of Acoustics, Chinese Academy of Sciences; 2011 [in Chinese].
[47] Lawton BW. Damage to human hearing by airborne sound of very high frequency of ultrasonic frequency. Contract research report 343. HSE Books; 2001.
[48] Guidelines for the safe use of ultrasound: part II – industrial and commercial applications, safety code 24, Minister of Supply and Services Canada; 1991.
[49] Howard CQ, Hansen CH, Zander AC. A review of current ultrasound exposure limits. J Occup Health Safety Aust New Zeal 2005;21:253–7.
[50] Lee S, Katsuura T, Towatari-Umeno T, Shimomura Y. The effects of parametric speaker sound on salivary hormones and a subjective evaluation. Act Nerv Super Rediviva 2010;52:119–24.