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Audio Projection: Directional Sound and Its Applications in Immersive Communication

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The ability to control sound radiation patterns in entertainment, gaming, communication and personal messaging is becoming an important differentiating feature in many commercial products. A common feature in these systems is to create a highly-directional sound field to targeted audiences by forming a tune-in zone (or personal audio) for a group of people. There are several ways to generate the directional sound field, which include (i) using a sound dome that projects sound to a convex surface to focus sound wave to the listeners below the sound dome; (ii) using a loudspeaker array that adjusts its phase-amplitude difference among the loudspeakers to spatially steer the audible sound beam in a horizontal plane; and (iii) modulating an audible sound signal onto an ultrasonic carrier and projecting the modulated signal via special types of ultrasonic emitters to generate the parametric array through the air, in such a way that audible sound can travel in a column of sound beams. This latter group of loudspeakers is commonly called the parametric (or ultrasonic) loudspeakers.

When using loudspeaker array to steer the audible sound beam described in (ii), the dimension of the loudspeaker or loudspeaker array must be significantly greater than the audio wavelength (i.e., more than a meter in diameter) in order to achieve good directivity at low frequencies less than 200 Hz. This approach of creating a focused sound beam incurs high cost and cannot be generated by a small loudspeaker. In contrast, the parametric loudspeaker in (iii) is able to generate a highly-directional sound beam for a low-frequency sound wave whose wavelength is much larger than the parametric loudspeaker's diameter. Therefore, the small-sized ultrasonic emitter in the parametric loudspeaker is able to produce a highly-directional sound beam with no vibrating cone as compared to conventional loudspeakers. Figure 1 shows two types of ultrasonic emitters used in parametric loudspeakers. The first type, which is shown in Figure 1(a), consists of several small piezoelectric transducers (PZTs) to form a bigger

[FIG1] Different types of ultrasonic emitters used in parametric loudspeakers: (a) consists of an array of small ultrasonic piezoelectric transducers; (b) a single piece of piezoelectric film that is being shaped and secured to an enclosure.

ultrasonic emitter. These small PZTs must be carefully matched in amplitude and phase near its resonance frequency. Typical resonance frequency of this PZT is around 40 kHz, and has a narrow bandwidth of 4 to 6 kHz. The second type uses a piezoelectric film ultrasonic material, as shown in Figure 1(b). A commercially available piezoelectric film-based parametric loudspeaker [15] is reported to have a bandpass frequency from 400 Hz to 16 kHz.

However, conversion efficiency between electric power and acoustic radiant power of the above piezoelectric-based transducer is rather poor due mainly to the impedance mismatch between electric-acoustic domain. An impedance-matching solution proposed by Kamakura in [32] is to add in a coil (or inductor) in parallel with the piezoelectric transducers to achieve power reduction of approximately 30%.

HISTORICAL DEVELOPMENT AND CURRENT DEVELOPMENT

In 1963, Westervelt [1] described how an audible difference-frequency signal is generated from two high-frequency collimated (or aligned) beams of sound. These high-frequency sound beams are commonly referred as primary waves. The nonlinear interaction of primary waves in mediums (such as air and water) produces an end-fire array of virtual acoustics sources that is referred to as the parametric

array. The primary interest for the parametric array focuses on the difference-frequency signal that is being created along the axis of the main beam (or virtual end-fire array) at the speed of sound. This phenomenon results in a sharp directional sound beam of audible signal. Berkay [2] extended Westervelt's analysis to spherically and cylindrically spreading sources to derive a simple expression for predicting the far-field array response. In contrast, the only way to produce an end-fire array of audible acoustics sources is to use a large array of conventional loudspeakers lining up directly in front of each other in the shape of a long column. But this approach is very costly, bulky, and impractical. Therefore, parametric array provides a practical mean of projecting a very narrow sound beam in air.

The use of the parametric array in air was first verified experimentally by Bennett and Blackstock [3]. Since then, the parametric array in air has been developed and the device that generates this phenomenon is generally referred to as a parametric loudspeaker. In 1982, Yoneyama *et al.* used the parametric loudspeaker, which is made up of 547 PZTs and a modulation circuit, to generate broadband audio [4]. They introduced the term 'audio spotlight' for audio applications of the parametric array. Their experiments revealed that the demodulated sound wave generated by the nonlinear acoustic phenomena has a very sharp directivity pattern. However, the demodulated signal suffered from high harmonic distortion, low electrical to acoustic conversion and poor frequency response.

In 1984, Kamakura *et al.* [5] reduced the distortion with double-sideband amplitude modulation (DSBAM) by preprocessing the modulating signal. Similarly, Pompei introduced a practical device in 1998 [6], which adopted the preprocessing technique proposed by Kite *et al.* [7]. Their approaches involved square-rooting the modulating signal, which reduced harmonic distortion and improved frequency response as compared to the DSBAM, but at the cost of requiring very wide bandwidth (> 10 kHz) ultrasonic emitters. About the same time, Croft and Norris [8] reported a similar device with proprietary algorithms and emitters, and commercialized their devices.

Several recent studies of the parametric loudspeakers [9], [10], [11] have resulted in new insights and observations. In addition, new processing techniques [12], [13], [14] have also been developed to further reduce the distortion, enhance the perceptual quality of the parametric loudspeakers, and to provide

constant beamwidth for the possible range of beamsteering angles. These signal processing techniques will be introduced in the following sections.

Several parametric loudspeakers [15], [16], [17], [18] are now commercially available.—Parametric loudspeakers are deployed for directing private messages in museums, art galleries, shopping malls, libraries, theme parks, and other public areas. A portable parametric loudspeaker for mobile phones was also reported in 2005 [19]. This handheld device uses a pair of 16 PZTs to project a directional stereo sound to the user at a distance of 0.5 meters. In the later sections, we will discuss some new usages and deployment of parametric loudspeakers to create a more immersive listening environment for communication, gaming, and entertainment applications.

One of the current technological limitations in parametric loudspeaker, which prevents it from competing with other high-fidelity conventional loudspeakers, is the narrow frequency response of commercially-available ultrasonic transducers. Some recent advancements in ultrasonic transducer technology [20], which include the realization of a wider bandwidth and flat frequency response with narrow directivity beam pattern, shows that parametric loudspeakers have the capability of creating high-fidelity directional audio reproduction. In addition, recent development on micromachined ultrasonic transducers [21] for directional sound projection also has potential in developing small-sized emitters for personal sound reproduction in mobile devices. With these recent developments in signal processing algorithms and transducers' technology, parametric loudspeakers show potential in wide-spread applications. This paper will present processing techniques to achieve a seamless immersive audio reproduction.

We shall start by highlighting how this new type of parametric loudspeakers can be deployed in some applications that require private or personal listening zone. These applications serve as motivation in developing sound projecting devices that can be extended to other types of applications. This is followed by an introduction of how sound can be generated in air via parametric arrays, and describe the nonlinear wave equation that governs sound propagation. Unfortunately, this nonlinear process in air introduces severe distortion, thus requiring signal pre-processing techniques to reduce the distortion. In addition, signal processing can also be applied to shape

and steer the directional sound beam electronically, and enhance the perceived sound quality due to the poorer frequency response of the ultrasonic transducers. Finally, we will provide some insight into new trends of directional sound projection, especially in the area of spatial sound projection, where signal processing will play a key role in this application. Several limitations and challenges will also be highlighted at the end of this paper.

APPLICATIONS OF DIRECTIONAL SOUND

The parametric loudspeaker is able to project high-directivity sound beam to a distance ranging from half a meter to few hundred meters, depending on the diameter of the ultrasonic emitter, sound pressure level delivered at the ultrasound emitter, and sound beam's carrier frequency. The directional sound beam can be targeted (or projected) to a listener or a group of listeners to create a "private listening zone" along its propagation path; and can even be projected against a reflective surface (or wall), forming a "virtual loudspeaker" at the point of reflection. Due to this directional property, sound proximity can also be conveniently placed near the listener. Furthermore, multiple sound beams from different sources can interact to form an audible sound spot [29]. These unique features make the parametric loudspeakers different from conventional omni-directional loudspeakers, and provide many opportunities to create new types of applications. Below is a list of applications for parametric loudspeakers.

- (i) **Safety Applications:** Parametric loudspeakers can be used to guide visually impaired people to cross streets, avoid obstacles, and avoid straying into dangerous areas. In another application, this loudspeaker can be used to direct a long-distance sound message to swimmers who are too far from shore, or as a first-level warning system to warn intruders entering a restricted zone. It can also be used as a crowd control device by directing commands to an individual who is about to cross into a dangerous area.
- (ii) **Digital Entertainment:** In automotive entertainment, individual entertainment units can be equipped with parametric loudspeakers to give the choice of his or her favorite music or video, without disturbing others. Museum patrons looking for audio descriptions of a particular exhibit can stand in a tune-in zone to receive the information, while others standing away from the zone will not be bothered. This personalized

loudspeaker creates virtual headphones in the listening zone without the discomfort through wearing actual headphones. Parametric loudspeakers can also be used in gaming. For example, parametric loudspeakers can enhance the realism of gaming by giving multiple players their own game sounds or tips personalized to his/her situation during game play.

- (iii) **Advertising and Theme Restaurants:** Directional sound can be very useful in projecting product information for potential customers who are interested in a particular product without disturbing other shoppers. The current digital advertisement using LCD monitors are mainly silent because the noise in this digital advertisement will rise to uncomfortable levels when using conventional loudspeakers that allow sound to spread in all direction. By confining sound via parametric loudspeakers, we can better catch the attention of customers passing into an advertisement zone. Furthermore, by incorporating a video camera and person identification software, directional message can be aimed to only those who fall under the demographic profile of a consumer for the product. Several special-theme restaurants are now installing directional loudspeakers in each table to project the music chosen by the customers to create a personalized dining ambience. Customers can even bring in their own audio players to plug into the system to share the latest songs with friends during the meal. Trade shows can also use the parametric loudspeakers to draw people into their exhibition booths. This directional message within the tune-in zone is also capable of significantly reducing the noise level of the surroundings, thus focusing the message clearly. In addition, immersive soundscapes can be created more vividly using this type of audio projection. For example, we can more accurately pinpoint the direction of sound sources in an environment by using parametric loudspeakers. This feature of directional sound forms a more accurate 3D spatial perception for immersive gaming, theme park and omni-theatre environment.
- (iv) **Multilingual Teleconferencing, Meeting and Movie Viewing:** Multilingual teleconferencing or meetings are commonly hosted in a large meeting room with different groups of international participants. Translation of languages can be channeled to different parametric loudspeakers and

projected to those participants sitting in their preferred language zones. This setup eliminates the need to wear headphones and provides a more seamless communication environment. A similar application allows different languages to be projected simultaneously into different defined tune-in zones for watching movies in the same room.

- (v) **Art Works:** Parametric loudspeakers can add another dimension to the sound art performance. Artists can now manipulate sound projection much the same as how light can be manipulated to create special effects. For example, directional sound can be synchronized with light projection to spotlight at certain spots, creating a moving light and sound artwork. Walls can be fitted with parametric loudspeakers to create directional sound beams of different properties (loudness, pitch, timbre) based on the proximity and movement of the participant standing in front of the acoustic wall. In another example, sound messages delivered from parametric loudspeakers can be directed to follow a visitor within the tracking zone covered by cameras.

PRINCIPLES AND CHARACTERISTICS OF PARAMETRIC LOUSPEAKERS

As described above, the parametric loudspeaker operates in the nonlinear region of sound propagation in air to generate a highly-directional sound beam. The nonlinear effect occurs when an ultrasonic signal (beyond the range of human hearing) at a high sound pressure level (or amplitude) is transmitted through air via a group of ultrasonic transducers (or an ultrasonic emitter unit, as shown in Figure 1(a)). The consequence of this nonlinear sound propagation is its ability to generate (or self demodulate) audible sound as difference frequencies among the main frequency components of the input signal. For example, as shown in Figure 2, a 40 kHz and 42 kHz ultrasonic wave (commonly known as the primary waves) are combined and sent to the ultrasonic emitter. Due to the nonlinear property of air triggered by the high amplitude ultrasonic signal, several frequency components (secondary waves) are generated and the only audible frequency is the difference of the primary waves (in this example, 2 kHz). In other words, the nonlinear distortion in air causes sound wave to vary in shape. In this case, the sine waves are transformed into sawtooth waves since sound pressure peaks travels faster than troughs, thus creating harmonic partials, as

[FIG2] Creation of primary, sum, difference, and multiple harmonic frequencies through the parametric array in air.

shown in Figure 2. Therefore, if we know how the air affects the sound waves, we can predict exactly what new audible frequencies will be generated in the air. The nonlinear model of air propagation is explained in the next section.

Instead of generating multiple ultrasonic frequencies to create an audible signal in air, a better approach is to modulate an audible signal onto a single high-amplitude ultrasonic carrier. In this way, audible sound can be generated in a tight column, as shown in Figure 3. This column of virtual audible sources forms an end-fire array of audible sources that add up in phase along the axis of propagation. Therefore, along the propagation axis, the audible sources get louder at greater distances from the emitter. Outside the propagation axis, the virtual audible sources are out of phase and interfere destructively. From Figure 3, the audible sound waves take on the directivity of the ultrasonic wave and the generation of these virtual sources extends to an effective array distance (or absorption length) within the near field. This effective array distance is defined as the distance whereby the sound pressure level of the ultrasound falls (or acoustic energy of ultrasound is absorbed by the atmosphere) below the utilization limit of nonlinearities. Outside the effective array distance, the virtual end-fire array ceases to exist, and audible sound will not be generated within a tight beam. Instead, the audible sound starts to attenuate at -6 dB per doubling distance outside this effective array distance. Unlike the omni-directional sound radiation generated by a conventional loudspeaker that is attenuated by -6 dB for every doubling of the distance starting from the loudspeaker

[FIG3] Generation of audible sound beam through the parametric array in air.

surface, the audible sound generated by a parametric loudspeaker becomes louder as the distance (within absorption length) increases, until the ultrasound no longer causes nonlinear effects in air in the far field.

NONLINEAR MODELS OF PARAMETRIC LOUSPEAKERS

The main difficulty in studying the behavior of the parametric array is the nonlinear interaction of sound. The theoretical analysis of nonlinear sound interaction involves complicated mathematical models [1], [9], [22] that do not have simple analytical solutions.

In 1963, Berklay [2] extended Westervelt's analysis to cover cylindrically, spherically and collimated primary plane waves. His analysis led to a simplified model, which is known as the Berklay far-field model, is widely used to approximate the nonlinear sound propagation. His model provided a simple expression, which can be used to predict the far-field array response of the parametric loudspeaker. The expression states that the demodulated signal (or audible difference frequency) pressure $p_2(t)$ along the axis of propagation is proportional to the second time-derivative of the square of the envelope of the amplitude-modulated ultrasonic carrier as follows:

$$p_2(t) \approx \frac{\beta p_0^2 a^2}{16 \rho_0 c_0^4 z \alpha_0} \frac{d^2}{d\tau^2} E^2(\tau) \quad (1)$$

$$\propto \frac{d^2}{d\tau^2} E^2(\tau),$$

where β is coefficient of nonlinearity for air, p_0 is pressure amplitude at the ultrasound source, a is source radius, ρ_0 is ambient density, c_0 is small-signal sound speed, z is coordinate along the axis of the beam, α_0 is absorption coefficient in air, τ is retarded time $t - z/c_0$, and $E(\tau)$ denotes the

modulation envelope of the ultrasonic carrier. Equation (1) shows that the demodulated signal is proportional to the size of the ultrasound source (i.e. the ultrasonic emitter in this case), a ; the pressure amplitude of primary wave, p_0 ; and the amplitude of the envelope function, $E(\tau)$. Therefore, a higher audible (demodulated) sound pressure at distance can be achieved by increasing the values of these three parameters.

The half power directivity (i.e., the angle where the mainlobe's amplitude drops by 3 dB) can also be computed [9] as

$$\theta_{hp} = \frac{4}{\sqrt{k_d R_a}} \text{ radians}, \quad (2)$$

where k_d is the wave number of the demodulated signal and R_a is the absorption length. Since the beamwidth is inversely proportional to the demodulated frequency, the beamwidth becomes narrower as the demodulated frequency increases and vice versa. Contrary to common understanding, increasing the carrier frequency results in broader beamwidth and shorter absorption length due to the fact that the higher carrier frequency will be attenuated faster than the lower carrier frequency. It is also interesting to note that the half-power beamwidth is independent of the diameter of the ultrasonic emitter.

The Berklay's model is able to predict the performance of the parametric array in air, and provides important guidelines in designing suitable parametric loudspeakers for different applications. The following section presents the signal pre-processing and modulation techniques that are derived from the Berklay's model.

SIGNAL PRE-PROCESSING AND MODULATION TECHNIQUES

As shown in Figure 2, the nonlinear interaction of two sinusoidal primary waves (ω_{c1} and ω_{c2}) in air produces several secondary waves (also known as the demodulated signals), such as $\omega_{c1} \pm \omega_{c2}$, $2\omega_{c1}$, $2\omega_{c2}$, and other multiples of the primary waves and their combinations. Most of these demodulated signals are inaudible ultrasonic by-products that are absorbed in air and decay at a faster rate than the audible sound $\omega_{c1} - \omega_{c2}$ in the directional sound column, where $\omega_{c1} > \omega_{c2}$. Therefore, to generate broadband audio using parametric loudspeaker, we can amplitude modulate an audio signal onto an ultrasonic carrier [4]. 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 preprocessing and modulation techniques.

AMPLITUDE MODULATION

In 1993, Yoneyama et al. [4] proposed a parametric loudspeaker system which used the conventional AM or the DSBAM. The modulation envelope for their parametric loudspeaker system is given as $E(\tau) = 1 + mg(\tau)$, where m is the modulation index and $g(\tau)$ is the input signal. The block diagram of the DSBAM is shown in Figure 4, where $\sin(\omega_0\tau)$ is the ultrasonic carrier.

For a single tone input, the modulation envelope becomes $E(\tau) = 1 + m\sin(\omega_1\tau)$, where ω_1 is the angular frequency of the single tone input. Using Berkta's far-field prediction expressed in (1), the demodulated signal becomes

$$p_2(t) \approx -\frac{\beta p_0^2 a^2 \omega_1^2}{8\rho_0 c_0^4 z \alpha_0} [m\sin(\omega_1\tau) - m^2 \cos(2\omega_1\tau)]. \quad (3)$$

Several observations are obtained from (3). The demodulated signal comprises of $m\sin(\omega_1\tau)$ and $m^2 \cos(2\omega_1\tau)$, which are the input signal (refers as the desired signal for the rest of this paper) and the distortion, respectively. Both the sound pressure

[FIG4] Block diagram of the double sideband amplitude modulation.

level of the desired signal and distortion are proportional to m and m^2 , respectively. Equation (3) shows a nonlinearity gain of ω_1^2 , which results in a 12 dB/octave highpass ramp in the demodulated signal. Due to this highpass filter effect that attenuates the low-frequency contents of audible signals, the audio signal must first be compensated in the pre-processing stage of the parametric loudspeaker system. A simple approach is to perform a lowpass filtering (or double integration of modulation envelope) ramp at -12 dB/octave starting from zero frequency up to the audio range at the expense of reducing the dynamic range at higher frequencies. However, audio signals can be partially compensated by the ultrasonic emitter, which generally has a high resonance frequency response with a slope of -4 dB/kHz. Several studies [5], [8], [14] have also shown that complete compensation is not required, and parametric loudspeakers can be inherently compensated by the ultrasonic emitters' own frequency response.

Generally, the performance index used in parametric loudspeaker is the total harmonic distortion (THD). This index indicates the amount of harmonic distortion of the system with a single tone input and is expressed as

$$\text{THD} = \sqrt{\frac{T_2^2 + T_3^2 + \dots + T_{n-1}^2 + T_n^2}{T_1^2 + T_2^2 + T_3^2 + \dots + T_{n-1}^2 + T_n^2}} \times 100\%, \quad (4)$$

where T_1 and T_i are the amplitude of the fundamental frequency (ω_1) component and the higher harmonics at $i\omega_1$ (for $i = 2, 3, \dots, n$), respectively. By substituting the amplitude of the desired signal and distortion in (3), the THD for DSBAM is reduced to

$$\frac{m}{\sqrt{m^2 + 1}} \times 100\%.$$

Figure 5 summarizes the THD versus 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

[FIG5] Total harmonic distortion vs. modulation index.

increasing distortion. By reducing the modulation index, we can tradeoff sound pressure level of the demodulated signal with lower distortion, which is not desirable for practical applications. Therefore, DSBAM is seldom used as the modulation technique for parametric loudspeakers. In the following sections, several modified amplitude modulation techniques which achieve high demodulated sound pressure level with reduced distortion, will be presented.

SQUARE-ROOT AMPLITUDE MODULATION

Based on the Berkta model expressed in (1), a squaring effect of the modulation envelope occurs in the self demodulation process. This leads to audible harmonic distortion in the demodulated signal from parametric loudspeaker. A direct solution to remove this distortion is to include a square-root operation in the modulation envelope. This operation serves as a predistortion step which yields a demodulated signal that is close to the input signal. This technique was proposed by several researchers [5], [6], [7], and is commonly known as the square-root amplitude modulation (SRAM). A block diagram of SRAM is shown in Figure 6.

For a single tone input, the modulation envelope of SRAM becomes $E(\tau) = \sqrt{1 + m \sin(\omega_1 \tau)}$. An optional double integration can be carried out before the square-root operation to avoid the 12 dB/octave highpass ramp. This double integration can be regarded as a form of equalization to achieve a demodulated signal with uniform response. However, equalization generally reduces the peak output levels to the lowest levels within the band of interest (lower

[FIG6] Block diagram of square-root amplitude modulation.

and higher sidebands for the case of AM and SRAM). Since the frequency response of the ultrasonic emitter approximates a bandpass filter with its center frequency occurs at the resonance frequency of the emitter, partial compensation of the 12 dB/octave highpass ramp can be achieved by the emitter. Hence, a simplified implementation of the parametric loudspeaker does not require the double integration.

Ideally, SRAM leads to a demodulated signal free of distortion. However, small piezoelectric ultrasonic emitters generally have a 3-dB bandwidth of less than 6 kHz [4], [21], and this limited bandwidth prevents SRAM from producing a distortion free demodulated signal. This practical limitation is seldom mentioned in literature and is first highlighted by Kite *et al.* [7] and further investigated by Tan *et al.* [14]. Figure 7 summarizes the THD performance of SRAM and DSBAM for a 1 kHz tone. Comparing SRAM with DSBAM, it is clear from Figure 7 that there is significant reduction of THD using SRAM under high modulation index and relative bandwidth, where the relative bandwidth is defined as the ratio of the ultrasonic emitter's 3-dB bandwidth and its resonating frequency. From a practical viewpoint, an ultrasonic emitter with a 6 kHz 3-dB bandwidth and 40 kHz resonant frequency would have a relative bandwidth of 15%. From Figure 7, it is also noted that SRAM requires ultrasonic emitters with a large bandwidth (more than 20%) to reproduce higher order harmonics introduced by the square-root operator, otherwise THD performance degrades beyond 10% for high modulation index. On the other hand, THD performance of DSBAM does not change when the relative bandwidth is higher than 10%. This is due to the fact that the only distortion in the demodulated signal for DSBAM is the second harmonic of the input signal; hence THD performance of DSBAM saturates at a relative bandwidth higher than 10% and a wider bandwidth emitter has no effect on its performance.

[FIG7] Total harmonic distortion against the relative bandwidth of the ultrasonic emitter for two modulation methods of different modulation indices.

SINGLE SIDEBAND MODULATION

In the preceding section, we have established that SRAM produces the ideal modulation envelope which leads to a distortion free demodulated signal, if and only if an ultrasonic emitter having infinite bandwidth is used. According to the Berkta solution in (1), the demodulated audio signal is proportional to the squared envelope, and not the spectrum of the modulated carrier. Interestingly, single sideband (SSB) modulation produces a modulation envelope that is very similar to SRAM, but without infinite harmonics. This feature in SSB modulation implies that we do not need to use high bandwidth emitter, as in the case of SRAM, to achieve low THD performance. Figure 8 illustrates one of the possible techniques to generate SSB and the modulated carrier is expressed as

$$p_1(t) = m[g(\tau)\cos(\omega_0\tau) \pm \hat{g}(\tau)\sin(\omega_0\tau)] + \sin(\omega_0\tau), \quad (5)$$

where $\hat{g}(\tau)$ is the Hilbert transform of the input signal $g(\tau)$. In Figure 8, the addition or subtraction sign of combining the in-phase component with the quadrature-phase component determines the use of lower sideband modulation (LSB) or upper sideband modulation (USB).

Figure 9 compares the modulation envelope of the SRAM and SSB (upper sideband) modulation in time and frequency domains. A single tone input at 1 kHz and a carrier at 40 kHz is used in this illustration. The

[FIG8] Block diagram for single sideband modulation with carrier.

modulation envelope of the SRAM and SSB modulation are similar in time domain as shown in Figure 9(a) and 9(b), but the frequency spectrums of these modulation envelopes are very different, as observed from Figure 9(c) and 9(d). The spectrum of the modulation envelope from SRAM contains many harmonics, whereas the spectrum of the modulation envelope from SSB modulation only contains the carrier (40 kHz) and the upper sideband (41 kHz). Fortunately, the nonlinear effect in air only considers the time-domain envelope, therefore, SSB modulation results in a very low THD (less than 5%) without the requirement of high bandwidth ultrasonic emitters. To date, SSB modulation is commonly employed in some commercial products [18]. Other extension of the SSB technique has also been reported in [13] to recursively predict the in-band distortion and subtract this predicted component from the original input signal before performing SSB. This SSB extension is known as the recursive p th-order equalization in parametric loudspeakers.

MODIFIED AMPLITUDE MODULATION

A new modified amplitude modulation (MAM) scheme proposed in [14], [23] uses the quadrature amplitude modulation technique to reduce distortion introduced in DSBAM, and has the flexibility to scale the relative bandwidth requirement to match the bandwidth of the ultrasonic emitters. The block diagram of the MAM scheme is shown in Figure 10.

Similar to the SSB modulation, MAM consists of an additional carrier $\cos(\omega_0 t)$ which is orthogonal to $\sin(\omega_0 t)$. By considering $g_1(\tau) = 1 + m g(\tau)$ and

[FIG9] Time domain plots of modulation envelopes of (a) SRAM, and (b) SSB; frequency spectrum of the modulated carrier of (c) SRAM and (d) SSB.

$g_2(\tau) = \sqrt{1 - m^2 g^2(\tau)}$, the modulation envelope becomes $\sqrt{2}\sqrt{1 + mg(\tau)}$. This modulation envelope is similar to the one used in SRAM with the exception of an additional factor of $\sqrt{2}$. However, the square root operation in $g_2(\tau)$ produces infinite harmonics. Using Taylor series, $g_2(\tau)$ can be rewritten as

$$\sqrt{1 - m^2 g^2(\tau)} \approx \sum_{i=0}^q \frac{(2i)!}{(1-2i)i!^2 4^i} m^{2i} g^{2i}(\tau), \quad (6)$$

for $|m^2 g^2(\tau)| < 1$,

where polynomial approximation equates to the ideal envelope for $q = \infty$. By using different degrees of expansion order in (6), different approximations of the modulation envelope $\sqrt{2}\sqrt{1 + mg(\tau)}$ are obtained. Higher degree of series expansion leads to higher bandwidth requirement since higher harmonics are

produced and yields lower distortion in the demodulated signal. Let $g_2(q, \tau)$ denote

$$\sum_{i=0}^q \frac{(2i)!}{(1-2i)i!^2 4^i} m^{2i} g^{2i}(\tau), \text{ where } q \text{ is a positive integer.}$$

A higher order q results in more predistorted components to reduce distortion introduced by nonlinear property in air. The modulated ultrasonic wave of the MAM shown in Figure 10 is given by

$$g_T(q, \tau) = g_1(\tau) \sin(\omega_0 t) + g_2(q, \tau) \cos(\omega_0 t) \\ = \sqrt{g_1^2(\tau) + g_2^2(q, \tau)} \sin\{\omega_0 t + \tan^{-1}[g_2(q, \tau)/g_1(\tau)]\}. \quad (7)$$

This quadrature scheme in MAM can be viewed as DSBAM with an orthogonal carrier modulating a predistortion term $g_2(q, \tau)$. With this selection of $g_1(\tau)$ and $g_2(q, \tau)$, the envelope of $g_T(q, \tau)$ approaches $\sqrt{2 + 2mg(\tau)}$ when $q \rightarrow \infty$. Hence, it is

[FIG10] Block diagram of modified amplitude modulation.

clear that $g_2(q, \tau)$ effectively removes the distortion of DSBAM. Unlike SRAM whose performance degraded significantly if narrow bandwidth ultrasonic emitters are used, MAM with different order q offers flexibility in controlling the bandwidth of the pre-distortion term.

Figure 11 shows the THD performance of MAM q (where $q = 1$ and 3) schemes and it has been shown in [14] that the THD performance does not significantly increase with higher orders ($q > 3$) of MAM. For a given value of q , the THD performance of the MAM q scheme is dependent on the available bandwidth of the ultrasonic emitters and the modulation index. Similar to SRAM, MAM q reduces THD values with increasing relative bandwidth, and the reduction of THD diminishes as the relative bandwidth increases beyond 10 %. However, lower THD is obtained with MAM q for lower relative bandwidth as compared to the SRAM. Therefore, this new modulation technique is particularly suitable for ultrasonic emitters with low relative bandwidth, and has the ability to scale up its order to match wider bandwidth emitters.

VIRTUAL BASS ENHANCEMENT

The conventional cone-based loudspeaker uses a large diaphragm movement to move air molecules in order to produce low-frequency sound; however, sound is gradually built up in air using parametric loudspeaker, which does not move large volume of air molecules. For this reason, parametric loudspeakers lack low-frequency reproduction capability (typical cutoff frequency at around 1000 Hz). This deficit makes it difficult for parametric loudspeakers to compete with conventional cone-based loudspeakers in terms of low-frequency sound reproduction. One way to rectify this

[FIG11] Total harmonic distortion of MAM1 and MAM3 schemes against the relative bandwidth of the ultrasonic emitter for different modulation indices.

problem is to augment the parametric loudspeakers with conventional loudspeakers or sub-woofers. In other words, we can channel mid and high-band frequency content to the parametric loudspeakers and leave the low-band frequency content to the sub-woofer. However, this approach incurs higher cost and requires additional space to house the sub-woofers, and it is not appropriate for portable devices. Another approach is to recreate the sensation of low-frequency tones by introducing a harmonic series of its overtones without the presence of the physical fundamental (low) frequency. This psychoacoustics phenomenon is known as the “missing fundamental” [24] and can be readily implemented using signal processing. A nonlinear function, which can be easily implemented digitally, is usually used to create the harmonic series of its overtones, which are added into the highpass filtered signal to create a perceptually bass-rich sound track [25]. Studies [26] on how different nonlinear functions can affect the low-frequency perception of sound have been reported and applied to parametric loudspeakers with some success. However, new transducer technology with larger diameters must be realized to achieve a better bass perception in order to compete with conventional loudspeakers in terms of low-frequency quality.

The preceding sections described some of the commonly-used pre-processing and modulation techniques that are currently used in state-of-the-art parametric loudspeakers. In the next section, we shall introduce signal processing techniques that can be

applied to the beamforming and beamsteering of sound beams produced by parametric loudspeakers.

BEAM CONTROL TECHNIQUES IN PARAMETRIC ARRAY

In general, the focused sound beam generated from the parametric loudspeaker can be mechanically shaped or steered [23] to a desired direction. Alternatively, array processing techniques can also be applied to form and steer the demodulated sound beam electronically. This electronic beam control is advantageous by allowing the mounting of the parametric loudspeakers directly on the wall without the need for a mechanical pan-and-tilt system. However, unlike the beam control techniques used in conventional loudspeaker arrays that generally applies different delays to different loudspeakers to generate the beamsteering and beamforming effects, the beam control of the parametric loudspeakers is more challenging since we have to consider the combined effects of the carrier frequency and the modulating signal. Currently, there is very little study in this area and some of the reported works are still in their preliminary stages. We will briefly describe some works on beam control and highlight problems that still need to be solved.

The beam pattern of the demodulated signal can be controlled by array processing techniques. To simplify the equation for studying the beam pattern, a primary (wave) source with Gaussian amplitude shading is assumed. By using quasi-linear theory [22], the far-field directivity function of the primary wave solution with a Gaussian source is derived as

$$D_1(k, \theta) = \exp\left[-\frac{1}{4}(ka)^2 \tan^2 \theta\right], \quad (8)$$

where θ is the angle with respect to the axis of the beam and a is the source radius.

For a bifrequency Gaussian source, such as the two primary frequencies depicted in Figure 2, the far-field directivity of the difference frequency (or demodulated signal) $D_-(\theta)$ can be described as the product of the primary waves' directivities [22] of

$$D_-(\theta) = D_{1a}(\theta) D_{1b}(\theta), \quad (9)$$

where $D_{1a}(\theta)$ and $D_{1b}(\theta)$ are the primary beam directivities at frequency f_a (which is also the carrier frequency) and f_b (which is the modulating frequency), respectively. Note that the bifrequency

Gaussian source ignores the frequency dependence of the attenuation coefficient in air.

Consider a group of M weighted primary sources that are equally spaced, with d meters between adjacent sources. The far-field directivity of the weighted primary source array for frequency f_a is given by $D_1(k_a, \theta) H(k_a, \theta)$, where $D_1(k_a, \theta)$ is the aperture directivity given in (8), and

$$H(k_a, \theta) = 1/M \left(\sum_{n=0}^{M-1} w_{an} e^{j\omega \frac{nd}{c} \sin \theta} \right)$$

is the far-field array response. Here, w_{an} is the n th emitter weighting for $n = 0, 1, 2, \dots, M-1$. Similarly, the far-field directivity for primary frequency f_b is given as $D_1(k_b, \theta) H(k_b, \theta)$. Hence, in the case of beamforming Gaussian sources, the beam pattern for the audible demodulated signal can be estimated as

$$D_-(\theta) = D_1(k_a, \theta) H(k_a, \theta) D_1(k_b, \theta) H(k_b, \theta). \quad (10)$$

An algorithm has been proposed in [27] to control the sidelobe level of the demodulated signal's directivity, forming a beamformer with constant beamwidth for the difference frequency in parametric loudspeakers. This approach relies on a weighted linear array of $M = 2N + 1$ equally-spaced primary sources that is able to produce an audible frequency by using SSB modulation with a $f_a = 40$ kHz (carrier, as shown in Figure 8). A single set of weights w_{an} and weighting response vector $W_{bn}(e^{j\omega})$, $n=1, 2, \dots, M-1$ associated to carrier frequency and modulated broadband frequency, respectively, can be computed using the Chebyshev window weighting function with a specified amount of sidelobe attenuation. Note that the weight response vector $W_{bn}(e^{j\omega})$ associated to the modulated broadband frequency is a frequency-dependent function in order to achieve a broadband beamformer. A detailed guide in computing the weighting function is given in [27].

Different weighting configurations have been investigated and compared to the proposed constant beamformer technique. Figure 13(a) shows the beam pattern for the conventional M -channel parametric loudspeaker response for the difference frequency without any weighting applied to the carrier frequency and modulated broadband frequency. The highest sidelobe attenuation of the M -channel array response is -29.99 dB. In Figure 13(b), weight w_{an} is added to

[FIG12] M weighted parametric arrays with weight w_{an} and weighting response $W_{bn}(e^{j\omega})$.

channel n , which shows that the sidelobes can be attenuated at the expense of its beamwidth. The highest sidelobe attenuation is reduced to -52.53 dB. Figure 13(c) shows the array response for the difference frequency of the algorithm given in Figure 12. With the additional weighting response $W_{bn}(e^{j\omega})$, Figure 13(c) shows that the sidelobes can be further attenuated to -70.20 dB. Therefore, by introducing two sets of weighting to the carrier frequency and the broadband frequency, the sidelobe of the difference-frequency response can be greatly reduced while keeping the beamwidth constant over the broadband difference frequency.

Beamsteering in parametric loudspeakers [28] can be extended from the previous constant beamwidth beamformer structure by adding delays τ_{a0} and τ_{b0} to the carrier frequency and the sideband frequency, respectively, as shown in Figure 14. Since SSB is used in this beamsteering structure, either the lower sideband modulation or the upper sideband modulation output can be derived from the n th digital-to-analog converter (DAC _{n}). For LSB, the output from the n th DAC is given as

$$\phi_{LSB,n}(nT) = 0.5 \{ w_{an} \cos[\omega_c(t - n\tau_{a0})] + w_{bn} \cos[(\omega_c - \omega_-)(t - n\tau_{b0})] \}, \quad (11)$$

where ω_c and ω_- are the angular frequencies of the carrier frequency and difference frequency, respectively. The main design issue of the frequencies to determine the direction of beamsteering. carrier frequency is a fixed single frequency, delay for

[FIG13] Parametric array response of demodulated (difference frequency) signal using (a) conventional parametric array, (b) parametric array with w_{an} , (c) parametric array with w_{an} and $W_{bn}(e^{j\omega})$.

beamsteering algorithm is the selection of a set of effective delays for both carrier and sideband. Since the carrier frequency can be computed as

$$\tau_{a0} = \frac{d}{c} \sin \theta_-, \quad (12)$$

where θ_- is the desired steering angle of the difference frequency. As the steerable delay of the difference frequency is solely determined by the carrier signal, due to the product directivity principle given in (12), the delay for the sideband frequency, τ_{b0} , can be rounded to the nearest integer multiple of the sampling period that is closest to the desired steering angle for simple implementation. The objectives of the carrier frequency's weighting function, w_{an} , are to control the difference frequency's beamwidth and to attenuate the carrier frequency's sidelobes. The function of the sideband frequency's weighting function, w_{bn} , is to generate a flat directivity response over a range of angles across all audible frequencies such that the difference frequency's sound pressure level is the same for different steering angles. An additional objective of this weighting function is to attenuate the sideband frequency's sidelobes such that the generated difference frequency has lower sidelobe directivity. The detailed procedure for designing the weighting functions is presented in [28].

Simulations are performed to determine the effectiveness of the beamsteering algorithm in parametric loudspeakers. A total of $M = 32$ arrays of ultrasonic emitters are formed with inter-element spacing of $d = 4.9$ mm [28]. The effective source radius is set to $a = 3.85$ mm, and the speed of sound is set as $c = 344$ m/sec. The difference frequency's beamwidth is steered to 10° and 20° , as shown in Figure 15 (a) and (b), respectively. The difference frequency's directivity inherits the carrier frequency's directivity for angles within the sideband frequency's mainlobe. The highest amplitudes of the difference frequency's directivity for the above steered angles are reduced by -25.69 dB and -24.54 dB for the steered angles of 10° and 20° , respectively. This suppression of the difference frequency's amplitude may reduce the power conversion efficiency of the parametric loudspeakers, which is currently an open research topic.

DIRECTIONAL SOUND IN IMMERSIVE AUDIO

In the previous section, we have highlighted some commonly-seen applications whereby directional sound can enhance the "personal audio" experience. There are however, many more creative ways and

[FIG14] Block diagram of the beamsteering algorithm for parametric loudspeakers.

[FIG15] Difference frequency (10 kHz) beamsteering for (a) $\theta_- = 10^\circ$ and (b) $\theta_- = 20^\circ$.

[FIG16] A pair of parametric loudspeakers installed in a gaming booth.

novel applications yet to be conceived that can potentially push this technology to the mass market with wider acceptance. In this section, we shall highlight some of the recent works related to directional sound for immersive audio application being carried out by the authors.

Recently, the parametric loudspeakers have been installed in the Fusionopolis, which is a research and development complex located at the One-North business park in Singapore. A pair of parametric loudspeakers was built to project directional binaural sound from a gaming console to the gamer (see Figure 16) standing in front of the gaming booth. This setup confines sound to a predetermined sound zone. The gamer within the zone can enjoy playing the game without disturbing others beyond the defined “tune-in” zone. In order to enhance the low-frequency ambience in gaming, a sub-woofer was also included in the parametric loudspeaker setup. Therefore, the omnidirectional gaming ambience is delivered from the subwoofer, while the directional sound cue in the gaming track is projected via the parametric loudspeakers.

An add-on to the above gaming setup is to incorporate a video camera to track the position of the gamer and to steer the parametric loudspeakers toward

[FIG17] Audio beam system fitted with head tracking software to track the movement of the gamer.

the gamer for relaying personal messages or gaming tips while playing multi-player games. A preliminary prototype of this steerable sound projection, as shown in Figure 17, is being built in the lab. This system links the video camera equipped with head tracking software to control the position of the parametric loudspeaker via a pan-and-tilt system. Another system is currently under development, which eliminates the need of the pan-and-tilt system through an electronic steering technique described in the previous section.

High-definition graphics in today’s gaming platforms have brought a new level of realism to gamers; however, for any gaming experience, surround sound and accurate 3D sound cues projection are crucial in completing the gaming experience, especially in first and third person shooters. Since such games put the gamers in the middle of all the action, both visual and auditory cues become crucial in creating a highly immersive experience. In particularly, 3D sound cues permit the gamers to not only see but to also hear their opponents/enemies in the games, hence enhancing the realism of the experience. The game audio generally comprises of the gaming soundtrack and sound effects. Usually, audio cues are embedded in the sound effects to enhance the realism of games. These sound effects are processed with 3D audio techniques, such as Direct Sound 3D in Windows. These techniques allow the game developers to position the sound effects potentially anywhere in a virtual space surrounding the gamer via stereo or surround loudspeakers; hence adding another dimension of realism in the game.

The degree of audio imaging (mainly the sound

[FIG18] I3DA system with directional and conventional loudspeakers.

effects) and spaciousness of the game audio are dependent on the directivity of the loudspeakers. If the loudspeaker is fairly directional, the game audio may seem to lack spaciousness due to little influence from the room acoustics. On the other hand, if the loudspeaker has wide dispersion, the game audio may be perceived as lacking sharpness in spatial imaging due to reverberant nature from the room acoustics.

To solve this problem, a novel and unique setup is proposed in [30], which comprises of conventional omni-directional and parametric loudspeakers to transmit the gaming soundtrack and sound effects, respectively. This system is called the immersive 3D audio (I3DA) system. As illustrated in Figure 18, this system exploits the high directivity of the parametric loudspeakers to recreate a high quality, sharp audio image and a highly immersive gaming experience that is intended by the game developers [31]. Also, it has been found from preliminary results that the parametric loudspeakers create a closer auditory image as compared to conventional loudspeakers. These observations enhance the gaming and entertainment experience. Furthermore, the parametric loudspeakers enable multiplayer games to be executed in close proximity without the need for headphones.

OPEN ISSUES AND CHALLENGES

Ever since the first development of the parametric loudspeaker by Yoneyama, we have seen tremendous progress in the technology and application of this type of directional speakers. However, there still remain

several technical challenges and limitation of deploying this audio projection technology efficiently.

One of the persisting challenges is to improve the efficiency of the parametric loudspeakers, and devising a good impedance matching circuitry can lead the way in overcoming this problem. Furthermore, transducer array configuration can also affect the radiation efficiency and work has been carried out to determine the best array configuration. In addition, improved emitter design with broader bandwidth is one of the key challenges in obtaining high quality directional sound projection, but some improvement may be possible by equalizing the modulated signal based on the emitter's response.

In addition, there are several research challenges on the beam control of parametric loudspeakers. These topics include the distribution and arrangement of the ultrasonic emitters forming different configurations of ultrasonic emitter arrays to enhance the directivity patterns of different frequency bands; complexity reduction using different array configurations; grating lobes elimination in parametric loudspeakers, and the phase response of the parametric array effect in air.

CONCLUSIONS

The parametric loudspeaker provides an effective means of projecting sound in a highly directional manner without using large loudspeaker arrays to form sharp directional beams. It can be augmented with conventional loudspeakers to create a more immersive audio soundscape. Deployment of parametric loudspeakers in many public places where private messaging can make a difference in attracting attention, conveying messages without needing headphones, and creating private listening zones to reduce noise pollution.

Digital signal processing plays a significant role in enhancing the aural quality of the parametric loudspeakers and array processing can help to shape and steer the beam electronically. In addition, other signal processing techniques can also be applied to add more flexibility and improve the performance of parametric loudspeakers. These developments rely heavily on the latest techniques in acoustics and audio signal processing to overcome some of the current limitations in nonlinear acoustics modeling and ultrasonic transducers' technology.

A useful feature in sound projection is to realize a high-accuracy digital beamsteering capability in air using an array of parametric loudspeakers. An in-depth study into the theoretical model of wave steering

capability in parametric array in air can provide some hints on how we can best steer the demodulated signal in an efficient manner. As seen from this paper, digital signal processing provides the main engine to achieve directional sound projection, and new digital processing techniques will be devised to provide a better quality, controllable audio beaming, and efficient sound focusing device in the future.

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