Directional Steering of Audio Signal Using Parametric Array in Air

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Abstract

Directional Audio is very recent technology that creates focused beams of sound .By projecting sound to one location, specific listeners can be targeted with sound without other nearby hearing it.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.Development of parametric loudspeaker in many public places 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 electronic electronically.

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1. Introduction

The controllable audible sound can greatly reduce noise pollution in public places. For example, in the library, a persononal announcement system can help to communicate with a group of people without disturbing others.The directional sound system in museums can provide localized sound for those who wish to hear it, and quiet for everyone else.

The parametric array has been studied widely in the context of underwater sonar and to a lesser extent in air.It exploits an effect known as self-demodulation to extremely directive low-frequency waves, which would otherwise require an enormous array of ultrafrequency transducers.Self-demodulation occurs when non-linearities of a compressible medium cause high frequency wave components to interact.This interaction produces new frequencies at the combination of sums and differences of their individual frequency components.With the development of high-power transducers and signal processing techniques,the parametric array has been exploited for audio applications. By using the nonlinear interaction of sound beams, Yoneyama *et al.* designed a novel directional parametric loudspeaker in 1983. Array processing techniques[39] 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.

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2. Prior art

In 1963, Westervelt described how an audible difference frequency signal is generated from two high-frequency collimated beams of sound. These high-frequency sound beams are commonly referred to 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. Berktay [4] 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 way of projecting a very narrow sound beam in air. The use of the parametric array in the air was first verified experimentally by Bennett and Blackstock [5]. 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 [6]. 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. [7] reduced the distortion with double-sideband amplitude modulation (DSBAM) by preprocessing the modulating signal. Similarly, Pompei introduced a practical device in 1998 [8], which adopted the preprocessing technique proposed by Kite et al. [9]. 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 [10] reported a similar device with proprietary algorithms and emitters and commercialized their devices. Several recent studies of the parametric loudspeakers [11]– [13] have resulted in new insights and observations. In addition, new processing techniques [14]–[16] 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

In this paper, we shall highlight the steerable audio system using parametric array in air and validate its performance through subjective tests. This paper is organized as follows. Section 3 presents the theoretical overview for directing audio. Section 4 and 5 gives design approach of loudspeaker array. Section 6 provides the simulation results to validate proposed beamsteering algorithm. Section 7 discusse some practical limitations for the real time implementation. Lastly, Section 8 concludes this paper

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3. Theory

3.1 Non-linear Acoustics

A nonlinear restoring force on a displaced molecule generate sum and difference (combination) tones.This concept was first introduce by Helmholtz. These were not subjective tones but ones that actually existed in the air. He said that the springs that keep air molecules spaced apart exhibit a non-linear restoring force characteristic that manifests itself at higher displacement amplitudes. Helmholtz's theory and formulas predicted results that initially seem to match what Helmholtz had measured. Since two primary frequencies (in our case, ultrasonic ones) are generating new frequencies in the air, the shape of the primary wave must change as it propagates.Since two primary frequencies (in our case, ultrasonic ones) are generating new frequencies in the air, the shape of the primary wave must change as it propagates. Fourier tells us that any wave can be described with a series of sines and cosines. If one emits two high-amplitude sine waves, as in the Helmholtz example above, new frequency terms appear, and the shape of the wave train changes. The accepted mechanism for this propagation distortion is explained by A.L. Thuras, R.T. Jenkins, and H.T. ONeil of Bell Labs in a 1934 paper called Extraneous Frequencies Generated in Air Carrying Intense Sound Waves.The following explanation is taken from this paper, and from a similar paper by L.J. Black, A Physical Analysis of Distortion Produced by the Non-Linearity of the Medium, J. Acoust.(1940). It turns out that if equal positive and negative increments of pressure are impressed on a mass of air, the changes in the volume of the mass will not be equal. The volume change for the positive pressure will be less than the volume change for the equal negative pressure.

This phenomena may be unfamiliar to those in the relatively linear acoustics field of audio. The wave equation which is customarily used in the solution to acoustical problems is valid for small signal propagation only. The assumption involved in the derivation of the small signal wave equation is that the maximum displacement of the air particles, x, be small compared to the wavelength λ; x < λ. In other words, the pressure fluctuations are so small that the specific volume appears to be a linear function of pressure. When this is not satisfied, a plane wave or even a spherical wave propagated in the medium will not preserve its shape. As a result, the magnitude of the fundamental decreases and the magnitude of the distortion increases with propagation distance. A simple explanation of this phenomenon is given by L.J. Black. Each part of the wave travels with a velocity that is the sum of the small signal velocity and the particle velocity. The maximum condensation in a wave is at the point of maximum pressure and this portion of the wave has the greatest phase velocity. The fact that the phase velocity is greater at the peak of the wave than at the trough results in a wave whose shape changes continuously as it is propagated.

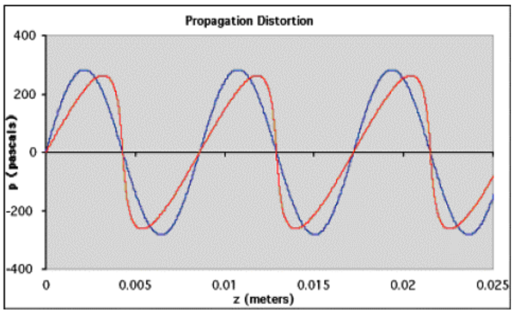

Fig. 1. Pressure fluctuations of a wave

The blue line represents a pure sinewave (a single-frequency signal); the red line represents the shape of the same wave after it has propagated through the non-linear medium for a time. A high-amplitude sinewave tends to form into a sawtooth wave as it travels. The sawtooth wave contains odd and even harmonics. The 2nd harmonic is fully half the amplitude of the fundamental. This means that strong harmonics are created during the propagation of a high intensity tone. In the two-tone case, f1 and f2, it can be shown that the harmonics of each will appear, as will the sum and difference frequencies, f1+f2 and |f1-f2|. This is the most simple case of a parametric acoustic array.

3.2 Non-linear Model

In 1963, Berktay's analysis led to a simplified model, which is known as the Berktay 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 amplitudemodulated ultrasonic carrier as follows:

$$p_2(t) \approx \frac{\beta p_0^2 a^2}{16 \rho_0 c_0^4 z \alpha_0} \frac{d^2}{d\tau^2} E^2(\tau) \qquad (0.1)$$

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 , and $E(\tau)$ denotes the modulation envelope of the ultrasonic carrier. Equation (1.1) shows that the demodulated signal is proportional to the size of the ultrasound source, 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 a distance can be achieved by increasing the values of these three parameters. The Berktay'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.

3.3 Audio Preprocessing

Signal pre-processing has three aims: amplitude modulation, distortion reduction, and transducer response compensation. In this paper, we consider the transducer to have an ideal response, so we are only interested in amplitude modulation and distortion reduction.

If classical amplitude modulation is used, the envelope function is given by $f = 1+ms(t)$, where $s(t)$ is the audible signal to be transmitted. The Berktay equation (Eq. (1.1)) shows that the demodulation wave is proportional to the second derivative of f^2, therefore to obtain $s(t)$ as a self-demodulation wave in the far field we have to use the modulation function

$$s_1(t) = \sqrt{1 + \iint s(t)dt^2} \qquad (0.2)$$

This processing is the ideal. However the square root of a signal has an infinite spectrum, while a transducer has a limited bandwidth. In practical terms, in order to have a self-demodulated wave with a satisfactory distortion rate, the transducer must have a bandwidth that is at least four times larger than the highest frequency of $s(t)$. This constraint can be difficult to fulfil when complex signals, e.g., music, have to be transmitted. Another solution is to use a single side band amplitude modulation (SSB). In this case the self-demodulated signal has less distortion than when classical amplitude modulation is used without other processing. To decrease the residual distortions, it is possible to simulate the self-demodulation, extract the distortions and correct the signal, as shown in Fig. 2. The advantage of this correction is that it does not increase the necessary bandwidth and can be used in iterative processing. These two processing methods give good results, but both have constraints: if the transducer has sufficient bandwidth the first method can be used, but if the transducer bandwidth is narrow and the calculation time is not a problem, then the second method is

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better.

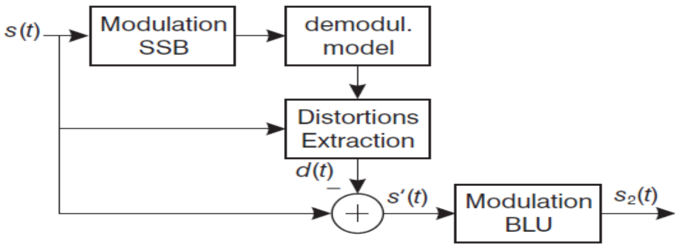

Fig 2 Correction of distortion

3.4 Beam Forming

A beamformer[32] is a spatial filter that processes the data obtained from an array of sensors in a manner that serves to enhance the amplitude of the desired signal wave front relative to background noise and interference. Signals from a particular angle or a set of angles are enhanced by constructive combination and noise from other angles is rejected by destructive interference. Spatial discrimination capability depends on the size of the spatial aperture, as the aperture increases, the discrimination improves. For computation of the delays the sensors need to be represented in a three dimensional co-ordinate system, i.e. each sensor position is represented by a 3D vector.

Consider a group of M sensors located in the space, whose position vector is given by($\vec{r_i}$); i = 1,2…:M. Each($\vec{r_i}$) is a 3D vector representing the x, y, z co-ordinates of the sensor, with reference to the origin.

$$\text{Let}\, s(n)= e^{j2 f_n / f_s} \qquad (0.3)$$

be the complex sinusoid prpagating through the medium with a unit direction vector u . The time delayed signal received at the ith sensor is

$$x_i(n)=\exp[j(2fn/fs+kr_iu)];i=1,2\ldots.M \qquad (0.4)$$

where k=2πf/c

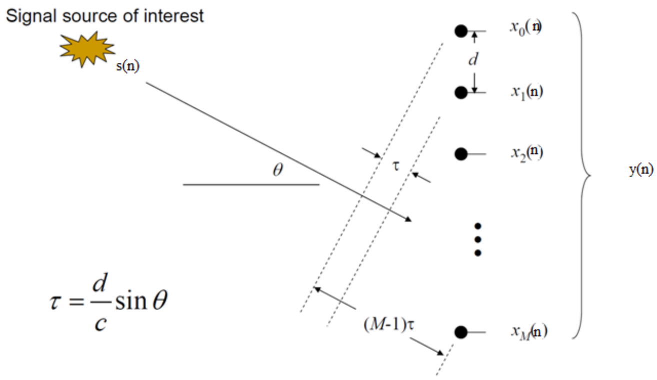

Fig 3 Plane wave impinging on a uniform linear array

3.4.1 Conventional Beamforming

The simplest approach to beamforming is conventional delay and sum beamforming. The under lying idea is very simple: if a propagating signal is present in an arrays apperture, sensor outputs ,delayed by appropriate amounts add together ,reinforce the signal with respect to noise or waves propagating in different directions. The delays that reinforce the signals are directly related to the length of time it takes for the signal to propagate between sensors.Fig .4 shows the block diagram of the conventional delay and sum beamformer in time domain. Here, inputs from each sensor are shifted so that the signals are aligned in time and are then added. However the delays are generally not integer multiples of the sampling period T sec,we cannot form sums that involve sensor signals delayed by non integer multiple of T .To reduce the aberrations introduced by delay quantization, we can interpolate between the samples of the sensor signals.

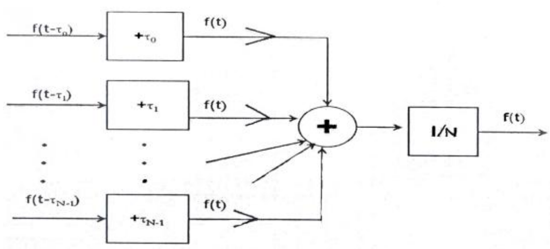

Fig 4 Conventional Delay Sum Beamforming

3.4.2 Adaptive Beam forming

Adaptive beamforming optimizes a collection of weight vectors to localize targets via correlation with the data in a noisy environment. These weight vectors

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generate a beampattern that places nulls in the direction of unwanted noise (i.e., signals, called interference, from directions other than the direction of interest). In contrast to conventional beamforming where the weight vector is a constant and independent of incoming data ,Adaptive Beamforming (ABF) algorithms use information about the cross spectral density matrix (CSM) to compute the weights in such a way so as to improving the beamforming output.

Minimum variance distortion less response is an adaptive algorithm which will minimise the output in all directions subject to the condition that gain in the steering direction is unity. The steering direction is the bearing that the array is steered towards to look for a particular incoming signal. This algorithm gives optimum performance by steering nulls in the direction of interference and also offers better performance in the case of correlated noise sources.Fig 5 shows the comparison between conventional and delay sum beamforming steered at 60 and 55^0.

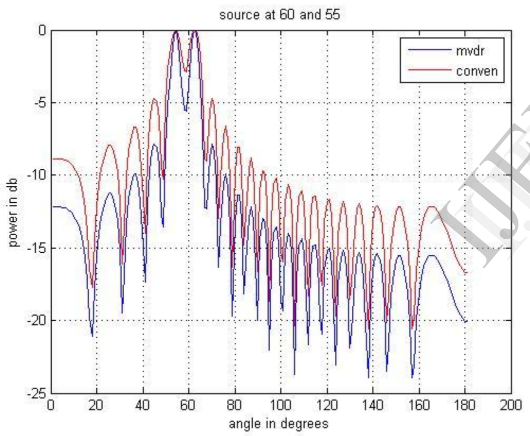

Fig 5 Conventional and MVDR beamformer output

4 Design of directional sound beam

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[23] , the far-field directivity function of the primary wave solution with a Gaussian source is derived as

$$D_1(k,\theta)=\exp[-\frac{1}{4}(ka)^2\tan^2\theta\,] \qquad (0.5)$$

where θ is the angle with respect to the axis of the beam and *a* is the source radius. For a bifrequency Gaussian source, the far-field directivity of the difference frequency (or demodulated signal) $D\_(\theta)$ can be described as the product of the primary waves' directivities of

$$D\_(\theta)=D_{1a}(\theta)D_{1b}(\theta) \qquad (0.6)$$

where $D_{1a}(\theta)$ and $D_{1b}(\theta)$ are the primary beam directivities at frequency ω_a (which is also the carrier frequency) and ω_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 ω_a is given by $D_1(k_a,\theta)H(k_a,\theta)$, where $D_1(k_a,\theta)$ is the aperture directivity, and $H(ka,\theta)=1/M(\sum_{n=0}^{M-1}\omega_{an}e^{j\omega(nd/c)\sin\theta})$ is the far-field array response. Here, *wan* is the *n*th emitter weighting for *n* = 0, 1, 2,… *M*-1. Similarly, the far-field directivity for primary frequency ω_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

$$D\_(\theta)=D1(ka,\theta)H(ka,\theta)D1(kb,\theta)H(kb,\theta) \qquad (0.7)$$

An algorithm has been proposed[29] in to control the sidelobe level of the demodulated signal's directivity, forming a beamformer with constant beamwidth for the difference frequency in parametric loudspeakers.

A single set of weights *wan* and weighting response vector $W_{bn}(e^{j\omega})$,*n*=1,2..,*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

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vector $W_{bn}(e^{j\omega})$ associated to the modulated broadband frequency is a frequency-dependent function to achieve a broadband beamformer. Beamsteering in parametric loudspeakers 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 6. Since SSB is used in this beamsteering structure, either the lower sideband modulation or the upper sideband modulation output can be derived from the nth digital-to-analog converter (DAC*n*). For LSB, the output from the nth DAC is given as

$$\varphi_{LSB,n}(nT)=0.5\{\omega_{an}\cos[\omega a(t-n\tau_{a0})]+ \omega_{bn}\cos[(\omega-\omega\_)(t-n\tau_{b0})]\} \qquad (0.8)$$

where ω_a and $\omega\_$ are the angular frequencies of the carrier frequency and difference frequency, respectively. The main design issue of the beamsteering algorithm is the selection of a set of effective delays for both carrier and sideband frequencies to determine the direction of beamsteering. Since the carrier frequency is a fixed single frequency, delay for the carrier frequency can be computed as

$$\tau_{a0}=\frac{d}{c}\sin\theta\_ \qquad (0.9)$$

where $\theta\_$ 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 (1.9), the delay for the sideband frequency, $\tau b0$, 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.

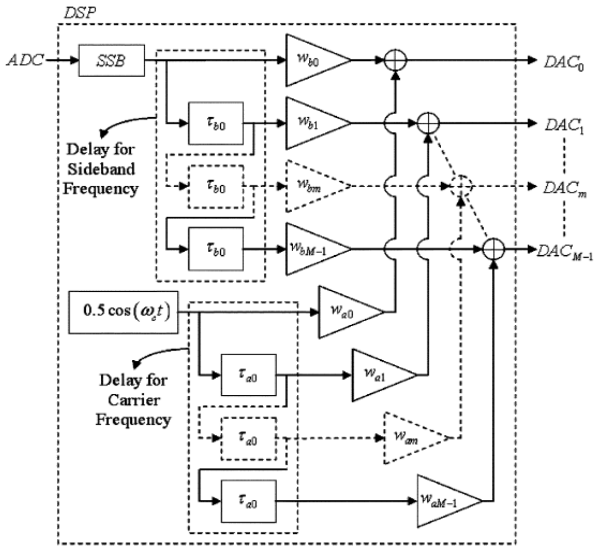

Fig 6 Proposed Beamsteering Algorithm

5 Transducer Array

Transducer arrays can be nearly any shape: linear, circular, rectangular, or even spherical[15]. A one-dimensional array allows beamforming in one dimension; additional array dimensions allow for 2-dimensional beamforming. Given the limited number of microphones and amount of time we have, a linear array is the best choice.

Transducer spacing: The spacing of the transducer is driven by the intended operating frequency range. For spatial filtering, a narrower beam width is an advantage, because signals which are not directly from the intended direction are attenuated. A narrow beam width is analogous to a narrow transition band for a trandiretional filter. Lower frequencies will correlate better with delayed versions of themselves than high frequencies, so the lower the frequency, the broader the beam. Conversely, a longer array will result in a greater delay between the end microphones, and will thus reduce the beam width. At the same time, the spacing between microphones determines the highest operating frequency[33]. If the wavelength of the incoming signal is less than the spacing between the microphones, then spatial aliasing occurs. At the same time, the spacing between microphones determines the highest operating frequency. If the wavelength of the incoming signal is less than the spacing between the transducers, then spatial aliasing occurs[16,18]. The spacing between microphones causes a maximum time delay which,

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together with the sampling frequency, limits the number of unique beams that can be made.

maxbeams=2.Fs.timespacing (0.10)

The variable timespacing is the maximum amount of time it takes for sound to travel from one microphone to an adjacent microphone, as the case when the source is along the line created by the array.

6 Simulation Result

The carrier frequency of the lower sideband modulation is set to 40 kHz. The sampling frequency of the DSP board, is 160 kHz. The input of the beamsteering system accepts frequency from 500–19 500 Hz, with 500-Hz increment. A total of 32 elements is formed with interelement spacing 4.9mm(Fig. 7). The effective source radius is set to 3.85 mm. The speed of sound $c=344ms^{-1}$, and the difference frequency's beamwidth, is 10^0 . By following the design procedure[29] for weighting functions , the calculated weighting functions for the carrier frequency and the sideband frequency are shown in Table I.

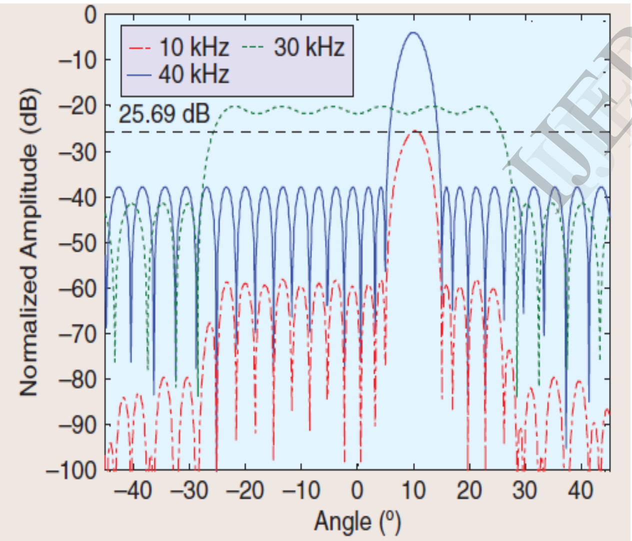

Fig 7 Difference frequency (10kHz) beamsteering for $\theta=10^0$

7 Limitations And Future Works

However, there are two major physical limitations of the parametric loudspeaker due to its sound generation principle. Firstly, during the self-demodulation process caused by the parametric array effect, higher harmonic components of the original sound are generated as by-products. Secondly, this self-demodulation process shows a high-pass filtering effect, resulting in a very poor bass quality of the parametric loudspeaker. Research over the last two decades have mainly been focused on reducing harmonic distortions using different preprocessing techniques and controlling the beam patterns of the parametric loudspeaker.

One way to rectify the second problem is to augment the parametric loudspeakers with conventional loudspeakers or subwoofers. In other words, we can channel mid- and highband 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 subwoofers, 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” [25] 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 . Studies 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 to compete with conventional loudspeakers in terms of low-frequency quality.

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.

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TABLE I
WEIGHTING FUNCTIONS FOR THE CARRIER AND SIDEBAND FREQUENCIES

m	Weighting Function		
	w_{am}	w_{bm} (LSB)	w_{bm} (USB)
0	0.2843	-0.0246	0.1047
1	0.1906	-0.1607	0.0014
2	0.2484	0.0078	-0.0445
3	0.3124	0.0495	0.0242
4	0.3817	0.0563	0.0363
5	0.4549	-0.0168	-0.0329
6	0.5303	-0.0850	-0.0436
7	0.6062	-0.0530	0.0661
8	0.6806	0.0646	0.0262
9	0.7515	0.1297	-0.0916
10	0.8168	0.0290	-0.0096
11	0.8746	-0.1642	0.1447
12	0.9231	-0.2103	-0.0472
13	0.9609	0.0718	-0.2397
14	0.9868	0.5885	0.2300
15	1.0000	1.0000	1.0000

8 Conclusion

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 highaccuracy 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 article, 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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