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Development of an acoustic filter for parametric loudspeaker using phononic crystals

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ABSTRACT

The spurious signal generated as a result of nonlinearity at the receiving system affects the measurement of the difference-frequency sound in the parametric loudspeaker, especially in the nearfield or near the beam axis. In this paper, an acoustic filter is designed using phononic crystals and its theoretical simulations are carried out by quasi-one- and two-dimensional models with Comsol Multiphysics. According to the simulated transmission loss (TL), an acoustic filter is prototyped consisting of 5×7 aluminum alloy cylinders and its performance is verified experimentally. There is good agreement with the simulation result for TL. After applying our proposed filter in the axial measurement of the parametric loudspeaker, a clear frequency dependence from parametric array effect is detected, which exhibits a good match with the well-known theory described by the Gaussian-beam expansion technique. During the directivity measurement for the parametric loudspeaker, the proposed filter has also proved to be effective and is only needed for small angles.

1. Introduction

The accurate measurement of the difference-frequency sound in the parametric loudspeaker [1–9] is vital for understanding the principle of the parametric array and also the practical design in audio engineering, as well as the implementation of the newly-developed Volterra-based preprocessing method [10,11]. However, it will become a problem especially in the nearfield or near the beam axis because of the spurious signal [1] generated as a result of nonlinearity at the receiving system caused by the finite-amplitude ultrasonic waves (referred as the primary waves). A few methods have been presented to measure the difference-frequency sound of parametric array in the literature by adopting an acoustic filter mounted in front of the receiving transducer to suppress the levels of the primary waves [1,12–18]. Their characteristics related to the acoustic filter are listed in Table 1, including its material, the sound pressure level (SPL) reductions, i.e., the transmission losses (TLs) for both the primary wave(s) and the audible sound. As can be seen from Table 1, using these proposed acoustic filters, the spurious signal was reduced or eliminated due to the reduction of the primary waves at the receiving system; however, the measured difference-frequency sound was also affected. It can also be derived that for a good parametric loudspeaker characterization, a suitable acoustic filter should attenuate over 20 dB at the carrier frequency with a −10 dB bandwidth of around 20 kHz, with TL ≈ 0 for the difference-frequency sound. It should be sufficient in most of the measurements in the parametric loudspeaker for eliminating the spurious signal. Recently, two other methods to replace the acoustic filter in the measurement setup are also presented. Ju et al. proposed a technique by taking advantage of the sensitivity characteristics of the condenser microphone to reduce the levels of the primary waves with a large grazing angle [19]. But it will also affect the sensitivity in the audible range, especially when the signal's frequency is higher than 10 kHz. Based on the phase-cancellation method and the Gaussian beam expansion technique [4,5,20,21], we proposed an alternative method for measuring the difference-frequency sound accurately without using any traditional acoustic filter [22]. But it is only suitable for the axial measurement. Therefore, a simple structure to measure both on and off axial difference-frequency sound is much preferred, which can reduce the levels of the primary waves without significantly affecting the difference-frequency sound. It is even better that the structure's parameters can be theoretically predictable and easily adjustable according to the required characteristic for both the audible sound and the primary waves.

During the last two decades, phononic crystals (PCs) have attracted considerable interest mainly because it can create a phononic band gap, over which there can be no propagation of elastic waves in the structure. The existence of the band gap gives these

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Table 1
Characteristics of the acoustic filter in the literature.

Author(s)	Material of the filter	SPL reduction	
		Primary wave(s)	Audible sound
Bennett and Blackstock [1]	A 2.5 mm thick clear plastic	About 20 dB	3.5 dB
Lucas et al. [15]	A sheet of SOAB rubber (1.3 cm thick)	16 dB	2.5 dB
Kamakura et al. [14]	A sheet of air-pad	Over 20 dB at 40 kHz	Not given
Havelock and Brammer [12]	Four layers of 3.18 mm felt	Over 20 dB at 30 kHz	3–5 dB
Toda [16]	Four polymer layers	30–40 dB between 30 kHz and 40 kHz	±5 dB between 10 Hz and 10 kHz
Humphrey et al. [13]	30 mm thick polyurethane panel	Strong attenuation at 300 kHz	Little attenuation
Wygant et al. [17]	A thin sheet of Saran film	Not given	Not given
Ye et al. [18]	A 16-mm-diameter aluminum plate	15 dB	±5 dB

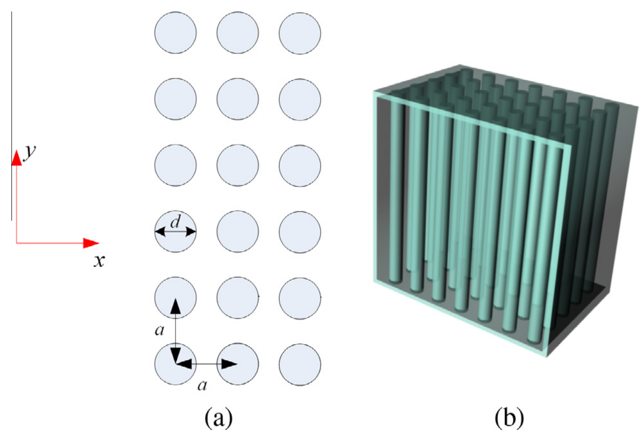

Fig. 1. (a) Classic structure of 2D PC with two components, (b) front view of proposed filter model.

artificial materials potential applications such as the filtering of elastic waves. There are two mechanism related to PCs, i.e., the Bragg's scattering mechanism [23,24] and the local resonant mechanism[25,26]. It is known that the PC based on the local resonant mechanism is more suitable for the lower frequencies compared to the conventional PC based on the Bragg's scattering mechanism. In our study, we only focus on how to suppress the levels of the primary waves in the ultrasonic range, and therefore, we only need to investigate how to apply the PC based on the Bragg's scattering mechanism to the measurement in the parametric loudspeaker. Studies on the Bragg scattering mechanism [23,24] have shown that the center frequencies of the gaps are given by the Bragg's condition,

$$f = nv/2a \ (n = 1, 2, 3, \ldots), \tag{1}$$

where v is the elastic wave velocity of the matrix material, and a is the distance between the scatterers, i.e., the lattice parameter. An advantage of PCs is that, by varying a, it is possible to attain peaks of attenuation in a selected range of frequencies.

In this paper, an acoustic filter using PCs is proposed to eliminate the spurious signal in the measurement of the parametric loudspeaker by suppressing the levels of the primary waves without significantly affecting the levels of the audible sound. Both simulations and experiments are carried out to verify the performance of the proposed filter. To further validate the measurement results after applying our proposed filter, the numerical simulations based on the Gaussian-beam expansion technique for the parametric loudspeaker is performed. The performance of the proposed filter in the directivity measurement of the parametric loudspeaker is also investigated.

The rest of this paper is organized as follows. In Section 2, the quasi-1D and 2D models of our proposed acoustic filter are established in Comsol and the numerical simulations are carried out to determine the optimal parameters for the measurement in the parametric loudspeaker. A prototype of our proposed acoustic filter is developed, and its performance is verified experimentally in Section 3, both on- and off-axis. Finally, the conclusions are drawn in Section 4.

2. Theoretical modeling and numerical simulations

The finite element method has been utilized to calculate its TL in the commercial Comsol Multiphysics simulation software (version 4.3a) by simulating the pressure field without and with the proposed PC. Because of a large number of meshes (>10^9) needed to solve the wave equation across the domain in three-dimension (3D), a 3D model is not practical. Hence in the present study, only quasi one-dimensional with infinite length along y-direction and finite length along x-direction (quasi-1D), and two-dimensional (2D) numerical simulations are performed to estimate the performance of the proposed filter.

A classic structure of 2D PC including two elastic materials with different mechanical properties is shown in Fig. 1(a), where d is the diameter of the cylinder. A front view of our proposed filter model with 2D PC system is illustrated in Fig. 1(b), which comprises circular cylinders embedded in air. The central frequency of the first Bragg band gap is

$$f_c = c/2a, \tag{2}$$

where c is the sound speed of the air. Because the expected central frequency of the Bragg band gap is around 40 kHz, which is the carrier frequency in our parametric loudspeaker, we roughly define the lattice parameter a = 4 mm.

The quasi-1D simulation model in Comsol is illustrated in Fig. 2 with the number of lattice N where the periodic condition is adopted to reduce the computation time. The material of the cylinder is aluminum alloy with the density of 2780 kg/m^3, the Young's modulus of 73.1 GPa, and the Poisson's ratio of 0.33. The environment fluid is the air, where the density is 1.2 kg/m^3 and the sound speed is 343 m/s. The normal acceleration boundary condition with 1 m/s^2 at the left edge is used as the radiation condition. The plane wave radiation condition has been applied at both the left and right edges to produce a plane wave. In the case of the rigid cylinders in the PC, sound hard boundary conditions have been applied, i.e., the normal component of the velocity of the air particles is zero in the walls of the cylinders because of the high acoustic impedance mismatch at the air–solid boundaries. The frequency range is from 0.1 kHz to 60 kHz, with an interval of 0.1 kHz. Hence, the minimum wavelength is 0.57 cm, and the maximum element size of the mesh is set to be 0.09 cm, which is taken as 1/6 of the minimum wavelength.

The effect of the size of the cylinder on the TLs is illustrated in Fig. 3(a) with N = 5, where the cylinder's diameter is from 1.6 mm

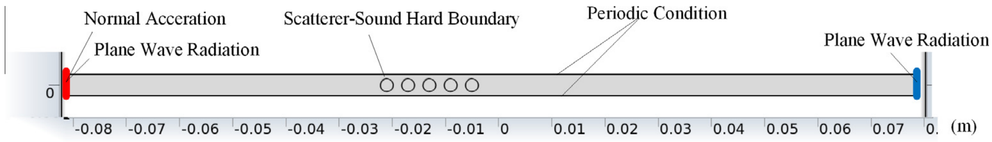

Fig. 2. Quasi-1D model of proposed filter in Comsol.

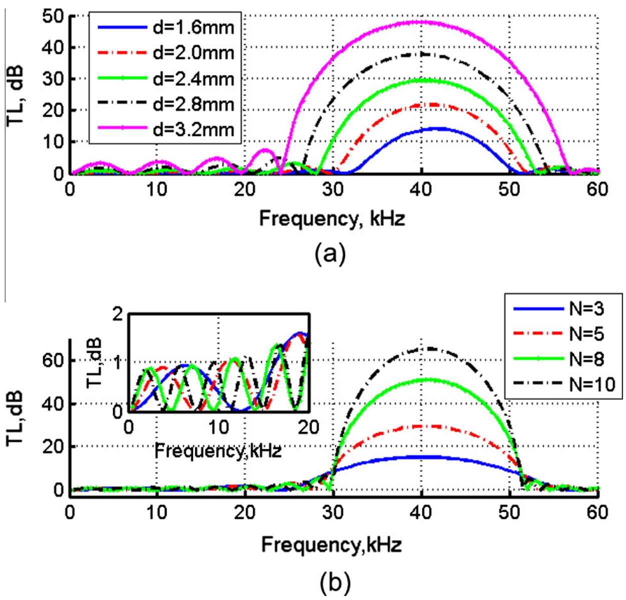

Fig. 3. Simulation results of proposed filter using quasi-1D model in Comsol (a) PC's TL with $N = 5$, (b) PC's TL with $d = 2.4$ cm.

to 3.2 mm at an interval of 0.4 mm. The Bragg band gap is obviously observed for all *d*s. The TL around the designed center frequency of 40 kHz clearly shows band gap formation with the wavelength of the incoming acoustic wave comparable to the lattice parameter, which means a high attenuation at this selective frequency band is occurred as a consequence of multiple scattering phenomena. The TL for pre band gap formation clearly demonstrates that at low frequencies the PC system behaves as a homogeneous material and acoustic wave propagation is almost unaffected by the periodic structure. This is due to the lattice parameter being much smaller than the relevant wavelength. The TL for post band gap formation becomes smaller because the wavelength of the acoustic wave is smaller than the lattice parameter of the PC system.

From Fig. 3(a), it can be seen that when the cylinder's diameter increases, both the TLs for the whole frequency band and the bandwidth of the Bragg band gap increase. The center frequency of the Bragg band moves slightly toward the lower frequency when the cylinder's diameter increases. Since the purpose of adopting the acoustic filter in the parametric loudspeaker is to increase the TLs in the ultrasonic frequency and meanwhile to lower the TLs in the audible frequency, a trade-off should be made between the TLs in these two regions. From Fig. 3(a), it can be observed that Bragg band gap for $d = 2.4$ mm larger than 10 dB is between 30 kHz and 51 kHz, with a peak of almost 30 dB at around 40 kHz. Meanwhile, the maximum TL in the audible range for $d = 2.4$ mm is less than 2 dB. Therefore, d is chosen to be 2.4 mm, which can basically satisfy the requirement of the measurement in the parametric loudspeaker mentioned in Section 1.

The effect of the number of lattice N on the TL has been investigated and its result is demonstrated in Fig. 3(b), with $d = 2.4$ mm. The Bragg band gap is still easily observed for all *N*s, and the trend is same as the above study. The band gap becomes smaller with N because the transitional zones between 0 and 10 dB become steeper when N increases, but the bandwidth for TL larger than 10 dB is around 21 kHz and nearly unchanged. More ripples appear with N before and after the Bragg band gap. The TL in the Bragg band gap greatly increases with N and its maximum is almost proportional to N. The position of the center band gap keeps almost unchanged at around 40 kHz. The TLs in the audible range remains relatively constant with their maximums of about 1.6 dB and almost have the same envelop, as shown in the inset of Fig. 3(b). After considering the actual requirement in the measurement of the parametric loudspeaker as mentioned above, N is chosen to be 5.

Therefore, from the quasi-1D simulation results in Comsol, the optimal parameters for PC that satisfy the requirement in most of the measurements related to the parametric loudspeaker are $a = 4$ mm, $d = 2.4$ mm, and $N = 5$. It is also noted that an extremely high attenuation in the primary waves can be achieved by adjusting the parameters of the structure.

A 2D numerical investigation is also carried out to further investigate the performance of the acoustic filter using PCs, and its model in Comsol is illustrated in Fig. 4, which composes of 5×7 cylinders with almost the same boundary conditions as those in quasi-1D simulation. The main difference between these two models is that the filter in the 2D model consists of a finite number of scatterers, compared to the quasi-1D model, which adopts the periodic condition to simulate the infinite number of cylinders in the y-direction. To approach the practical situation, where all the cylinders need to be mounted on a holder, two same rectangular frames with 1 mm thick and 2 cm width using the same material as the cylinders are placed both at the bottom and top sides.

The TL distribution with the frequencies and the positions is demonstrated in Fig. 5 for both x- and y-axis. Fig. 5(a) is the TL at $x = 1$ cm, where y is from -3 cm to 3 cm, and Fig. 5(b) is the TL at $y = 0$ cm, where x is from 0 to 3 cm, both at an interval of 1 mm. A similar trend as in the quasi-1D model can be found in Fig. 5, where a clear Bragg band gap can be observed, and the reason is the same as that explained in the quasi-1D model. Because of a finite number of scatterers in the 2D model, the Bragg band gap can only be observed within the region near the filter. When the observation point is far from the filter, there is almost no attenuation, which is especially obvious in the y-axis, as shown in Fig. 5(a).

A TL comparison between the above quasi-1D and 2D models is demonstrated in Fig. 6, where the TLs located at $x = 1$ cm and $y = 0$ cm are chosen as the representative of the 2D model. It is noted that they have almost the same trend, i.e., the TLs in audible frequencies are relatively smaller and in ultrasonic frequencies near the center of the band gap are larger. It also shows that the TLs for the 2D model are smaller than that for the quasi-1D model for most of the frequencies. The maximum difference between these two models is around 2 dB in the audible frequency range and 10 dB in the ultrasonic range. Hence, the TLs in both the quasi-1D and 2D models are desirable in the measurement of the parametric loudspeaker to reduce or even eliminate the spurious signal in the receiving system.

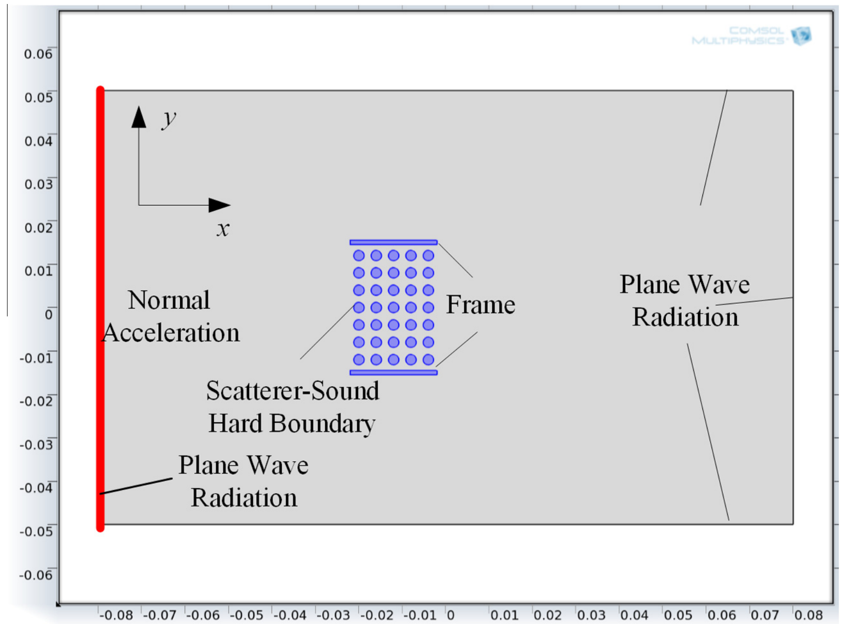

Fig. 4. 2D model of proposed filter in Comsol.

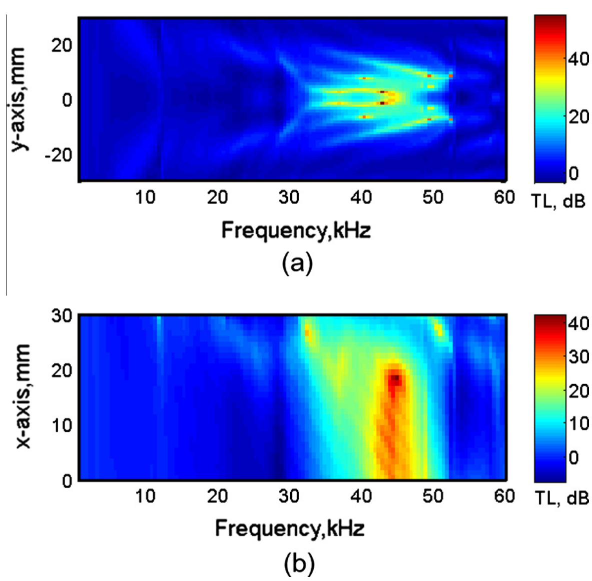

Fig. 5. Simulation results of TL distribution with frequency and position using 2D model in Comsol (a) at $x = 1$ cm (b) at $y = 0$ cm.

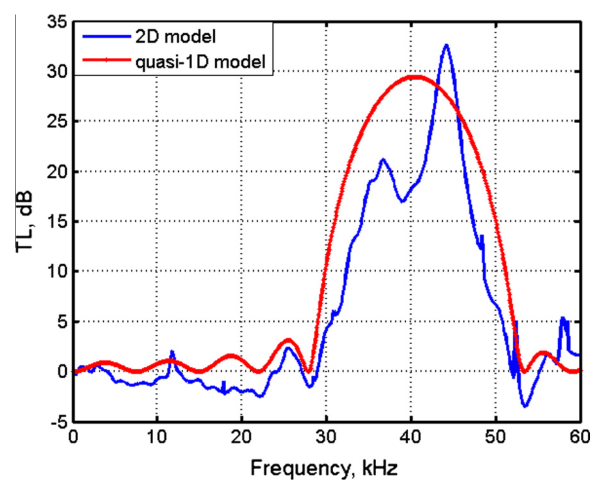

Fig. 6. TL comparison between quasi-1D and 2D models of proposed filter.

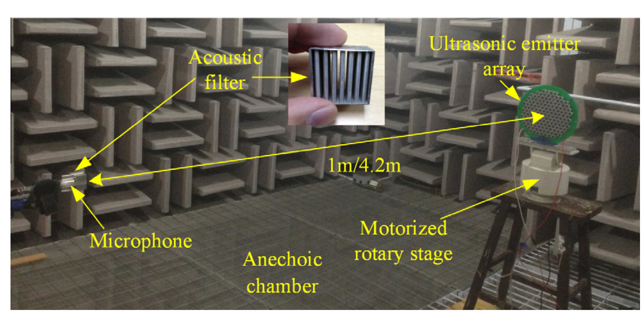

Fig. 7. Experimental setup.

3. Experiments and discussions

According to the above simulation results, an acoustic filter using PCs, consisting of 5×7 cylinders, has been prototyped as shown in the inset of Fig. 7, where $a = 4$ mm and $d = 2.4$ mm. The material of the cylinder is aluminum alloy, which is kept the same as in the simulations. The filter has the dimension of 29.2 mm $\times$ 30.2 mm $\times$ 20 mm in total, which is relatively small, compared to those filters listed in Table 1, and can be easily mounted in front of the microphone.

A preliminary experiment to examine the performance of the proposed filter is carried out in an anechoic chamber of Institute of Acoustics, Chinese Academy of Sciences (Beijing, China), which has the dimensions of 4.8 m (length) $\times$ 3.2 m (width) $\times$ 6.5 m (height). The cut-off frequency is 70 Hz, and the background noise level is 13 dBA. The temperature is 29.6 °C with a relative humidity of 64%. One hexagonally shaped ultrasound emitter array, consisting of 147 commercial ceramic transducers (Type ZT40–16, Shanghai Nicera Sensor Co., Ltd., China), is used in this experiment. The level of the measured primary wave at 40 kHz is around 136 dB (rel. to 20 μPa) measured with GRAS 46BE 1/4 in. microphone at 1 m from the ultrasonic emitter array.

A prepolarized free-field 1/2-in. microphone type B&K 4189 (Brüel & Kjær, Denmark) is used to measure both the audible sound and the primary waves, which is placed at two positions: 1 m and 4.2 m, where the latter is the maximum observation distance limited by the size of the anechoic chamber. The B&K 4189 is connected to a B&K analyzer platform (Brüel & Kjær, Denmark) with a steady state response (SSR) analyzer. The whole procedure mainly includes three steps, and each consists of two substeps with and without the filter installed in front of the B&K 4189. The first and second steps are to measure the filter's TL in the audible range and the ultrasonic range, respectively, and the third step is to investigate its performance in the parametric loudspeaker. A 4-in. loudspeaker (HiVi H4) is adopted as the sound source to measure the TL in the audible range, where the testing frequency spans from 0.1 kHz to 19.9 kHz with an interval of 0.1 kHz. By computing the difference between the frequency response of the source measured by the B&K 4189 with and without the acoustic filter, the TL in the audible range can be obtained. Same procedure is carried out to measure the TL in the ultrasonic range, where the ultrasonic emitter array mentioned above is used as the source, and the testing frequency spans from 20 kHz to 60 kHz with an interval of 0.1 kHz. A class-D power amplifier with very low phase shift and output noise, which is capable of amplifying up to 100 kHz, is used to channel the testing signal and drive the ultrasonic emitter array. It is needed to note that the nominal sensitivity of the B&K 4189 is only suitable for the audible frequency, and it will become inaccurate if this same sensitivity is applied to the ultrasonic range. Fortunately, the TL is a relative value, and it will not affect by the inaccurate sensitivity in the ultrasonic range. That is, the B&K 4189 can be used to obtain the TLs both in the audible and ultrasonic ranges. The characteristic of the acoustic filter used in the parametric loudspeaker is also investigated and its experimental setup is illustrated in Fig. 7, where the testing frequency spans from 0.5 kHz to 15 kHz with an interval of 0.1 kHz, and the same class-D amplifier used above is adopted to channel the modulated signal and drive the ultrasonic emitter array. In this study, for simplicity, the lower single side-band modulation with carrier scheme (LSB-WC) is adopted in the parametric loudspeaker, which is implemented in the Analog Devices ADSP-21469.

The smoothed TLs of the proposed filter at these two axial positions are demonstrated in Fig. 8, where the Bragg band gap can be easily observed. Their TLs have almost the same trend, where they are nearly the same in the audible range, and slightly different in the ultrasonic range. A high attenuation (>10 dB) can be observed between 31 kHz and 51 kHz with a maximum sound pressure attenuation level of 27 dB at a predicable frequency around 43 kHz. There are slight fluctuations in the audible range of the TLs. The undesired attenuation for the whole audible frequency is less than 2 dB both for these two positions, which is superior to any known acoustic filter adopted in the parametric loudspeaker retrieved by the literature search. For comparison, the 2D simulation result of the proposed filter located at $x = 1$ cm and $y = 0$ cm is also plotted in Fig. 8. A good agreement can be found between the experimental results and the 2D simulation results, especially in the audible frequency range. While in general the agreement is good, small discrepancy is observed and may be attributed to the following factors. Firstly, the filter is not long enough to qualify as a two dimensional problem. Our theoretical study deals with a two-dimensional problem; however, our experimental results can only be regarded as a quasi-two dimensional problem. The finite length of the scatterers will affect the performance of our proposed filter, and their edges may act as sound radiators [27]. Secondly, the incident wave is not strictly a plane wave, as it will also contain an oblique component. Lastly, the imperfections in the experimental system will also cause some discrepancy.

The measured difference-frequency signals of the parametric loudspeaker measured at these two axial positions are illustrated in Fig. 9 with and without the filter. To verify our results, the measured frequency responses of the difference-frequency signals by the acoustic filter proposed by Ye et al. [18] are also plotted in Fig. 9. It is needed to note that the frequency responses of both acoustic filters to the following experimental audio results have been compensated when the filters are adopted, which is referred as a system calibration step.

From above analysis, it is noted that the measured signal without any acoustic filter is a combination signal including both the difference-frequency sound from the parametric array effect and the spurious signal from the nonlinearity at the receiving system. As can be seen from Fig. 9(a), the spurious signal is severe at a distance of 1 m, i.e., the nonlinearity at the receiving system dominates the measurements close to the source. It is noted that the spurious signal becomes smaller for the higher frequencies mainly because of the high-Q characteristic of the ultrasonic emitter array. The frequency response of the ultrasonic emitter array measured at

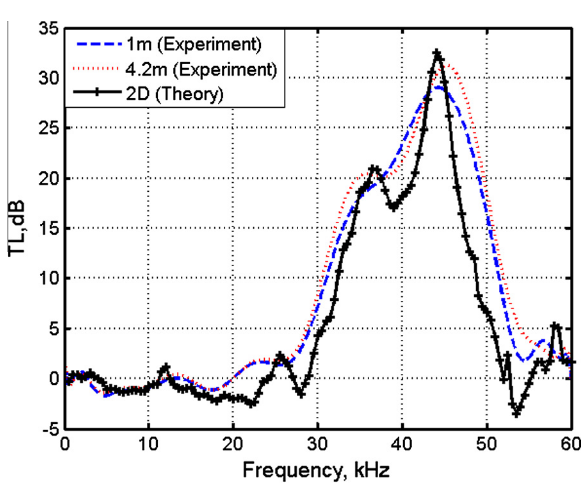

Fig. 8. TL comparison of the filter between simulation result and experimental results at two different axial positions.

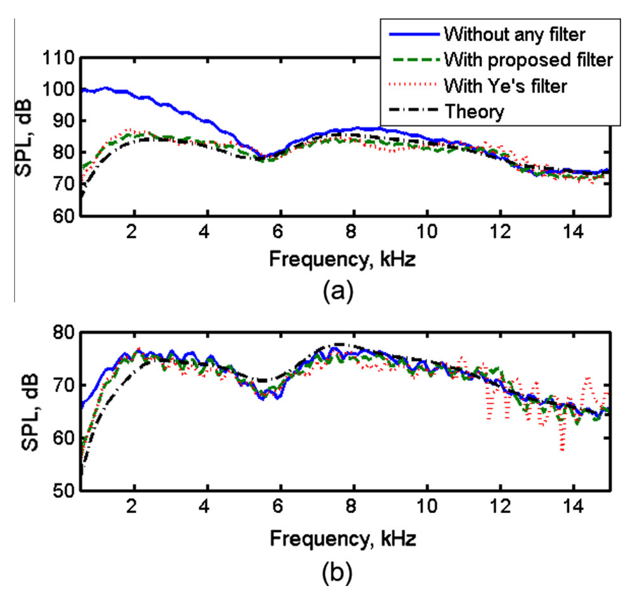

Fig. 9. Measured difference-frequency signals of the parametric loudspeaker at (a) 1 m and (b) 4.2 m (the same legend in (a) applies to all plots).

a distance of 1 m from the source by GRAS 46BE 1/4 in. microphone (GRAS Sound & Vibration, Denmark) is illustrated in Fig. 10. The response peaks at about 40 kHz, with a −6 dB bandwidth of only 4 kHz. Because the band-pass filter response of the ultrasonic emitter array reduces the levels of sideband frequency, i.e., the lower primary wave, it partially alleviates the nonlinearity of the microphone in the higher frequencies and so did the spurious signal. Therefore, the frequency response of the difference-frequency signal f_d before using the filter at 1 m can be roughly divided into two regions. When $f_d < 5$ kHz, the spurious signal dominates the effect of the parametric array. Beyond that region, the spurious signal is almost negligible and the frequency responses with and without the filter are nearly the same, and in that case, no filter is necessary for the measurement. Similar observation at a distance of 4.2 m can be found, but the region where the spurious signal dominates is lower than 2 kHz, is much smaller than that at the distance of 1 m, as shown in Fig. 9(b). The reason is explained as follows. It is known that the spurious signal is proportional to the product of the primary waves. Because of larger attenuation for ultrasonic waves by effects of absorption, diffraction and nonlinearity, which can be accurately described by the Khokhlov–Zabolotskaya–Kuznetsov (KZK) nonlinear parabolic wave equation [28], the primary waves are greatly reduced with the propagation distance, and so did the spurious signal. It has been proved that the difference-frequency signal first increases with the distance, attains its peak and then gradually decreases for the parametric loudspeaker with both the circular and the rectangular apertures because of the end-fire effect [29,30]. That means the difference-frequency signal has a slower change than the spurious signal with the propagation distance. It can be predicted the spurious signal will have a negligible effect on the combination signal when the distance is larger, and then there is no need to adopt any acoustic filter at a larger distance [17].

In order to further validate our results, the numerical simulation based on the Gaussian-beam expansion technique for the parametric loudspeaker is also performed and the results both at 1 m and 4.2 m are plotted in Fig. 9, considering the effect of the frequency response of the ultrasonic emitter array. The Gaussian-beam expansion technique has been applied to calculate the sound fields of the sound beam in the case of weak nonlinearity, which has been proved to be efficient in greatly reducing the computation time, especially for the second-order sound fields. In this study, the source is roughly modeled as a circular planar piston with an effective radius of 0.08 m. It is observed that the experimental curves with the filters show almost good agreements with the simulation

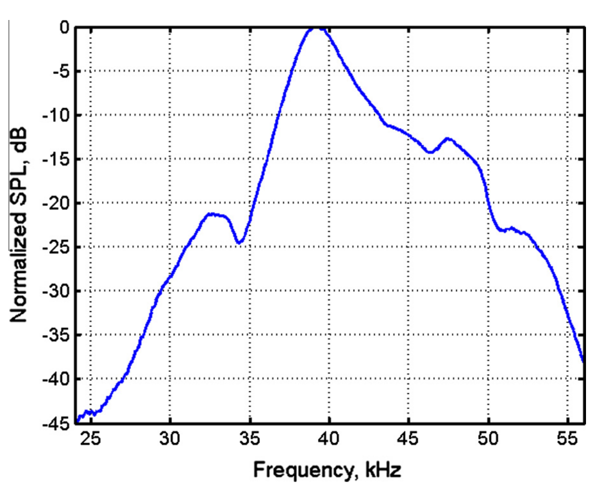

Fig. 10. Normalized frequency response of the ultrasonic emitter array measured at 1 m.

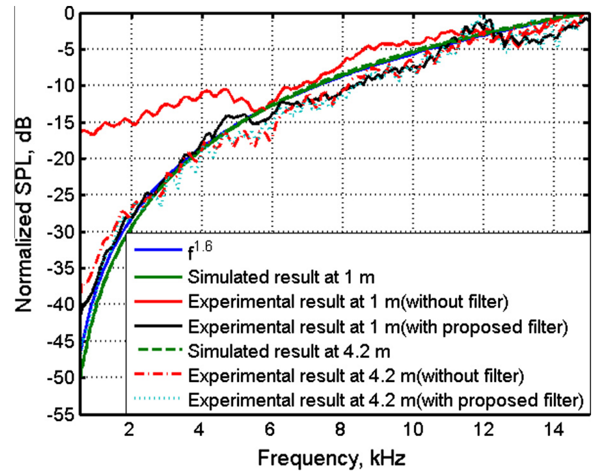

Fig. 11. Normalized frequency responses of the difference-frequency signal for parametric loudspeaker at 1 m and 4.2 m.

results for both these two distances. Besides the experimental measurement error, the main possible reason for the difference between the simulation and experimental results is the uneven velocity distribution at the source's surface in the experiments.

The results from our proposed filter agree very well with those from Ye's filter, and in certain higher frequencies, our filter can even be superior to the other filter. It means that the proposed filter using PCs is designed with better performance which has an extremely lower attenuation in the whole audible range and an acceptable attenuation in the ultrasonic range, and controllable parameters to achieve desired characteristic.

The normalized frequency responses of parametric loudspeaker measured without and with the acoustic filter for both two distances are illustrated in Fig. 11, where the effect of the ultrasonic emitter array has been removed. As mentioned above, the audible sound generated by parametric loudspeaker is proportional to f_d^n, where $1 \leqslant n \leqslant 2$ [17,18]. The index number n depends on the ratio of Rayleigh distance to absorption length, where $n = 2$ and $n = 1$ correspond to the Westervelt solution for a large ratio and the Berktay's solution for a small ratio, respectively. As shown in Fig. 11, the experimental results indicated $n \approx 1.6$, which was a good approximation of the difference-frequency dependence. The normalized numerical results both at 1 m and 4.2 m based on the Gaussian-beam expansion technique is also plotted in Fig.11 and almost fit with the curve of $f^{1.6}$, except that there is a slight difference in the lower frequencies (<4 kHz). Thus, from another point of view, it is possible to roughly predict the frequency response of the difference-frequency signal from the parametric loudspeaker using f_d^n.

For parametric loudspeaker, it is critical to measure its directivity accurately without any spurious signal. Therefore, it is needed to determine the performance of the proposed filter during the directivity measurement. Because of the sharp directivity of the primary waves and also the relationship between the spurious signal and the primary waves, only the characteristic of the filter within a small angle ($\leqslant 15^{\circ}$) will be investigated in this study. For a large angle, there will be no spurious signal because of relatively smaller primary waves. The ultrasonic emitter array was mounted on a motorized rotary stage with a precision of 0.0125°, which is computer-controlled. The experimental procedure for each angle is similar to the above experiments, i.e., the rotating angle is zero. The B&K 4189 is placed at a distance of 2 m from the source. The smoothed TLs with six rotating angles (i.e., 0°, 2.5°, 5°, 7.5°, 10°, and 15°) are demonstrated in Fig. 12. It is

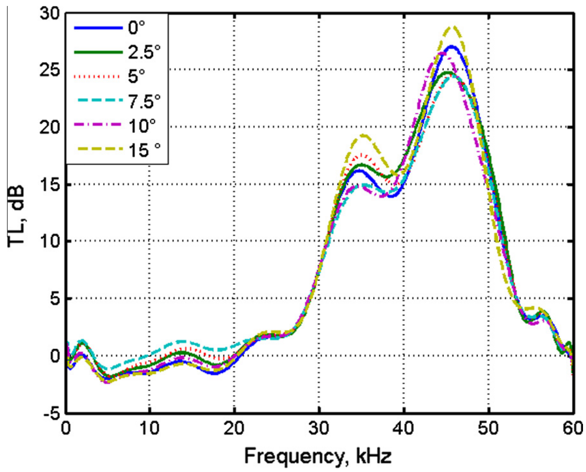

Fig. 12. Measured TLs of proposed filter for different angles.

observed that all the TLs for these angles have almost the same trend, where the band gaps can be easily observed. In this case, it is expected that the proposed filter should have a good performance even at the directivity experiment of the parametric loudspeaker.

In order to evaluate the performance of the proposed filter in the directivity measurement of the parametric loudspeaker, another experiment is also carried out and the measured frequency responses of the difference-frequency signal are shown in Fig. 13 for the above six different angles. It is easy to observe that when the rotating angle becomes larger, the difference between the combination signal and the difference-frequency signal becomes smaller, and when the rotating angle is larger than 7.5°, there is almost no difference between the measured signals with and without the filter. It means that in such a larger angle, there is almost no spurious signal as expected because the product of the primary waves becomes smaller and there is no need to adopt any acoustic filter. In this study with our ultrasonic emitter array, the acoustic filter is still effective in the directivity measurement, and when the rotating angle is larger than 7.5°, no acoustic filter is necessarily needed.

From above analysis and results, three major advantages of our proposed filter can be clearly observed and summarized as follows. The first one is that its performance can be roughly predicted by the theory. The structure's parameters can be theoretically predictable and adjustable according to the required characteristic for both the audible sound and the primary waves, which means a customizable filter by the given characteristic can be easily designed. The second one is that it has a small size, and it is easy to prototype and mount in front of the microphone. The characteristic of the proposed filter has been verified to be sufficient in the measurement of the parametric loudspeaker with such a small size in our study. The last is that a smaller attenuation for the interested audible frequency. There is even no need to do the system calibration step for some scenarios if the audible frequency response of the filter cannot be obtained without causing severe mistake. For example, the measured frequency responses of the parametric loudspeaker at 1 m with/without system calibration step, i.e., equalization (EQ) of above two filters are demonstrated in Fig. 14. It is easy to observe that both results (with/without EQ) almost match the same theoretical result. This is not the case when Ye's filter was adopted. Although the result with Ye's filter after the system calibration step is almost the same as that of

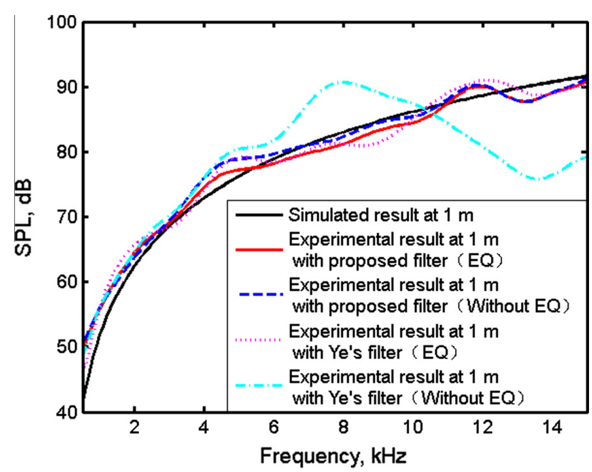

Fig. 14. Measured frequency responses of our parametric loudspeaker at 1 m with/without EQ for both two filters.

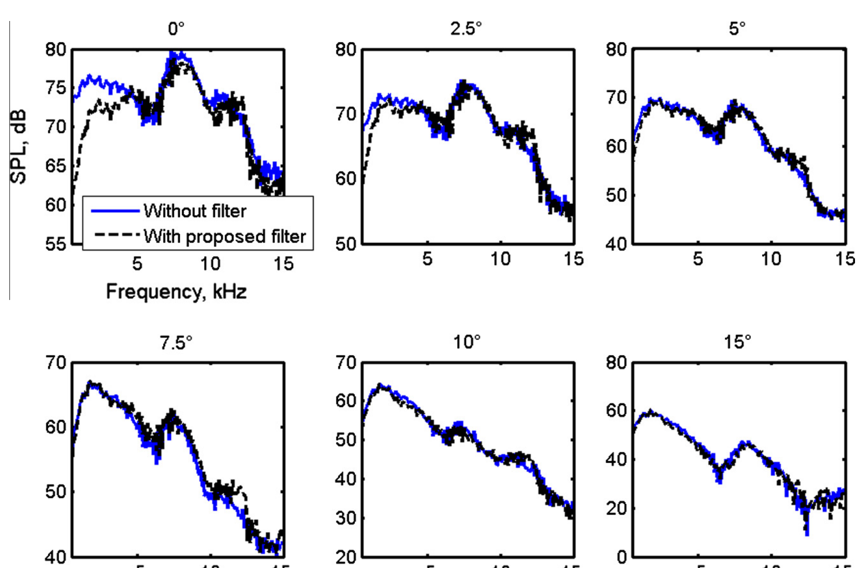

Fig. 13. Measured frequency responses of the difference-frequency signal with different angles for parametric loudspeaker (the same legend in (a) applies to all plots).

our filter, it is difficult to tell its trend when the system calibration step is not adopted in the frequency region higher than 4 kHz.

4. Conclusions

An acoustic filter using PCs has been proposed to eliminate the spurious signal in the measurement of the parametric loudspeaker. The parameter effect has been numerically investigated by both the quasi-1D and 2D models in Comsol. Based on the theoretical analysis, a feasible structure with optimal parameters has been obtained. According to the simulation results, an acoustic filter has been prototyped with 5×7 aluminum alloy cylinders with a small size and its TL has the -10 dB bandwidth of around 20 kHz with a maximum attenuation of 27 dB, without a significant effect on the audible sound. The experimental results show that the prototype has a good performance to reduce the spurious signal in the measurement of the parametric loudspeaker for both the axial propagation and the directivity measurement. A good agreement between the axial experimental results with the proposed acoustic filter and the theoretical results based on the Gaussian-beam expansion technique for the parametric loudspeaker has been obtained, which are both proportional to $f^{1.6}$ in this study.

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