



On the use of audible sound from modulated ultrasound in indoor spaces

David Ortega , Jesús Carbajo, Pedro Poveda, Jaime Ramis

Department of Physics, Systems Engineering and Signal Theory - University of Alicante
dop6@gcloud.ua.es

Abstract

The use of highly directional sound sources in indoor spaces is of great interest for those cases in which a focused acoustic field is desired. A good example of such scenarios are museums or art galleries, the major problems of using classic loudspeaker systems being the reverberant sound and the background noise, which may not only worsen the comfort but also the intelligibility when reproducing recorded speeches. Alternatively, attendees are asked to use headphones to follow the exposition, the main drawback being the difficulties that arise to simultaneously interact with a tour guide or other people. This work explores the generation of audible sound from modulated ultrasound, also called “sound from ultrasound”, to achieve a directive low-frequency sound source and overcome some of the above limitations thus improving the overall experience in these spaces. For this purpose, a parametric array consisting of a set of ultrasonic transducers was prepared and measured in laboratory facilities to analyze its frequency response and directivity characteristics. Additionally, experiments in a museum were carried out to assess the *in situ* performance of such devices when compared to conventional audio systems. Preliminary results show the great capabilities of this technology and encourage exploring its use in many indoor applications.

Keywords: parametric speaker, nonlinear acoustics, high directivity, indoors audio systems, sound spotlight.

1 Introduction

Conventional speakers, mostly based on electrodynamic transducers, offer a directivity that depends on the radiated wavelength and the source size. The device known as parametric speaker drives out this dependency offering a constant directivity, even for low and middle frequencies where conventional speakers show an omnidirectional radiation. Furthermore, these devices can focus the sound along a path, presenting a radiation pattern with a narrow main lobe and almost no side lobes.

The theoretical foundations of the parametric speaker were explained in 1963 by Westervelt [1] for underwater applications. It was Yoneyama [2] in 1983 who applied Westervelt’s theories to create a new loudspeaker concept after the publication made by Blackstock and Bennet suggesting air radiation applications [3]. Since the work of J. Pompei in [4] reducing the audible distortion of the parametric speaker, many practical applications of parametric speakers have arisen such as those for audio spatialization in [5] and [6], bus pedestrian warning systems in [7], mobile phones loudspeakers in [8] or even for active noise cancelling applications as studied in [9]. Although there is an increasing number of publications suggesting PAL applications in indoor spaces, we consider there is a lack of information regarding case studies that transform suggestions into experiments, thus leading to the work presented here. This paper serves as an approach for the evaluation of the behavior of a proof-of-concept parametric speaker implementation when placed indoors. The high directivity shown by these devices suggests their application on information delivery audio systems such as those used in museums or art galleries. They could even replace or complement traditional audio information systems where headphones and electrodynamic speakers are widely used. As indoor sound

involves a reverberant field, this project studies the influence of such reverberation in the general sound field to assess the possible disturbance produced to other visitors when one sound source is used.

2 Sound field indoors

When a sound source is placed indoors, the emitted sound waves will reflect on the surfaces of the enclosure, leading to the creation of a reverberant field. The characterization of rooms according to this reverberant sound field is usually done through the reverberation time (RT), a measure of how much time is needed for a 60dB reduction in sound pressure level once the sound emission is interrupted. Sabine proposed a simple expression to calculate the RT assuming a diffuse field as

$$RT(s) = \frac{0.161V}{A}, \quad (1)$$

where V is the room's volume in m^3 , and A is the equivalent absorption area of the room's surfaces in *sabines*. Equation (1) illustrates the dependence of the RT on the surfaces' sound absorption, reducing the RT as the sound absorption increases.

The reverberant field can be decomposed, according to the room impulse response, into early reflections and late reflections, the former being favorable for speech intelligibility as stated in [10] whilst the latter worsens it. When a sound source is placed in an enclosure, there will be a distance where the energy of the direct field will be equal to the energy of the reverberant field; the critical distance r_c defined as

$$r_c = \sqrt{\frac{QA}{16\pi}}, \quad (2)$$

where Q is the source's directivity factor.

The intelligibility depends mainly on the reverberation time and the signal-to-noise ratio, the latter depending on the source's directivity factor. Thus, the intelligibility will increase whenever the listener is placed within a circle of radius r_c from the sound source, hence using high directive sources will improve the signal-to-noise ratio and, in turn, the overall intelligibility.

3 Sound from ultrasound

Highly directional audible sound can be generated from ultrasound waves nonlinear interaction, leading to the effect that parametric speakers take advantage of; the so-called *autodemodulation* process.

3.1 Generation of sound from ultrasound

Nonlinearities are responsible for multiple effects such as shock waves, audio distortion or intermodulation phenomena. The latter produces new spectral components from the interaction of two or more different frequency waves. The nonlinear by-product of intermodulation at the difference frequency of the interacting primary waves is the one that the parametric speaker takes advantage from, resulting in audible sound creation from the ultrasound interaction in air. H.O. Berkay in [11] developed an analytical approach for the resultant audible pressure p_2 as a function of some design parameters of the parametric speaker as follows,

$$p_2 = \frac{\beta P_0^2 a^2}{16\rho_0 \alpha c_0^4 r} \frac{d^2}{d\tau^2} E^2(\tau), \quad (3)$$

where P_0 is the carrier wave pressure, a is the transducer radius, β is the nonlinear parameter (1.2 in the air), ρ_0 is the medium density, c_0 is the medium sound speed, α is the absorption coefficient at the carrier frequency in Np/m, r is the distance from the source and $E(\tau)$ is the envelope of the modulation. Equation (3) can be understood as the transfer function of the parametric speaker, considering the radiating system as well as the transmission channel, so it shows the main limitations of this phenomena: (i) the quadratic component $E^2(\tau)$ predicts an audible inherent distortion produced from the nonlinear interaction, (ii) the second time derivative anticipates a high-pass filter effect in the frequency response of the parametric speaker, and (iii) there is a strong dependency on the carrier's sound pressure and the array size to obtain high audible sound pressure levels.

The signal emitted by the parametric speaker must contain the spectral components that will interact nonlinearly with each other in the air, producing the auto-demodulation phenomena. This can be done by means of modulated signals where the amplitude, frequency, or phase of a carrier signal is changed according to a modulating signal, thus leading to amplitude, frequency, or phase modulation respectively. Some of these modulation schemes use the concept of *preprocessing*, where the modulating signal is modified to reduce the harmonic distortion produced by the auto-demodulation process in (3). Here, we evaluate the effect of some modulations in terms of frequency response and directivity. The simplest amplitude modulation including sidebands at both sides of the carrier frequency is the *Double Side-Band Amplitude Modulation (DSBAM)*, having a carrier wave $c(t)$ of angular frequency ω_c , and amplitude A_c being modulated by a signal of frequency ω_s and amplitude A_s as follows

$$y_{DSB}(t) = [1 + mA_s \cos(\omega_s t)]A_c \cos(\omega_c t) = [1 + ms(t)]c(t), \quad (4)$$

where m is the modulation index. The signal's envelope $E(t) = [1 + ms(t)]$ can be modified to compensate the quadratic term predicted by Berktaý in (3), leading to the *Square Root Amplitude Modulation (SRAM)*

$$y_{SRAM}(t) = \sqrt{1 + ms(t)} \sin(\omega_c t). \quad (5)$$

Here, we will be also considering the modulation proposed in [12] which also uses *preprocessing* techniques to reduce distortion and considering the finite transducer bandwidth resulting in the *Modified Amplitude Modulations (MAM)*

$$y_{MAM1}(t) = [1 + ms(t)] \sin(\omega_c t) + \left[1 - \frac{1}{2}m^2s^2(t)\right] \cos(\omega_c t). \quad (6)$$

Regarding nonlinear modulations, frequency and phase modulation can also be used with parametric speakers with

$$y_{FM} = A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t)), \quad (7)$$

for the frequency modulation, where β is the frequency deviation, and

$$y_{PM}(t) = A_c \cos(\omega_c t + \delta s(t)) \quad (8)$$

for phase modulation, where δ is the phase deviation.

3.2 The parametric array loudspeaker

The parametric speaker benefits from the lowest 2nd order intermodulation product produced when two ultrasound waves at different frequencies interact nonlinearly with each other, thus being able to radiate audible sound with the directivity properties of short-wavelength ultrasound waves. Safety regarding long exposure to the high-pressure ultrasound waves emitted by these arrays has been study in [13] with the HyperSound® model, declaring these devices as safe. The radiation pattern shown by such loudspeakers, which avoids the omnidirectional propagation of sound, could prevent disturbing other listeners when used as a PA system in indoor spaces. The easiest implementation of parametric speakers consists of a piezoelectric ultrasound transducers plane array, although there are implementations with concentric geometries or electrostatic films.

To assess the parametric speaker performance indoors as well as to compare this type of devices with conventional speakers, we developed a proof-of-concept parametric array with 9 rows and 5 columns; 45 piezoelectric transducers (TCT40) welded in parallel on a PCB as shown in Figure 1.

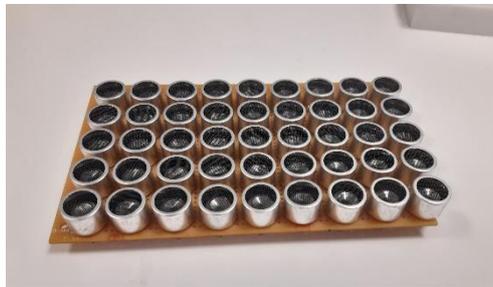


Figure 1. Proof-of-concept parametric speaker with 45 piezoelectric transducers.

4 Results and discussion

In this section, the main characteristics of the parametric speaker will be assessed. Eventually, a case study is shown to evaluate the performance of the parametric speaker when placed indoors.

4.1 General characterization

The frequency response of the proof-of-concept parametric speaker is shown in Figure 2.

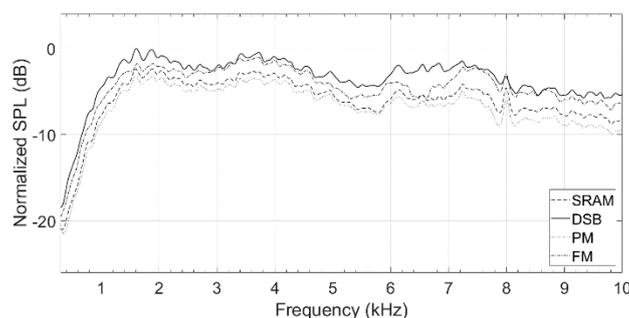


Figure 2. Frequency response comparison between different modulations of the parametric speaker.

As predicted by (3), the low-frequency response of the parametric speaker is affected by a high-pass filter effect. The parametric speaker shows a plain tendency for the middle frequency range, inviting for its

application in voice delivery systems. For this purpose, it is interesting to assess the total harmonic distortion levels of each modulation as well as its sensibility, thus leading to the plots in Figure 3 and Figure 4. The THD was measured at 1 meter with a 1 kHz pure tone as modulating signal. Figure 3 shows the benefit of *pre-processing*, having the SRAM the lowest THD value as expected, while the angular modulations exhibit a high amount of THD.

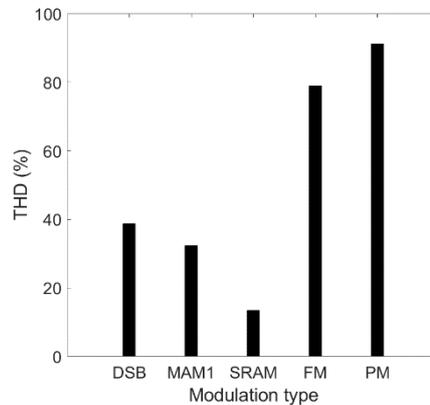


Figure 3. Total harmonic distortion measured at 1 meter and 1 kHz with the parametric speaker. Amplitude modulations have a modulation index of 1, frequency modulation has a frequency deviation of 1 kHz, and phase modulation has a phase deviation of π .

Sensibility in Figure 4 was measured at the same conditions of THD with a reference carrier wave pressure of 100 dB at 40 kHz. The proof-of-concept implementation of the parametric speaker is able to reach a maximum of 58 dB SPL with the nonlinear modulations, resulting in a compromise between THD and sensibility when choosing the modulation scheme. Note that no acoustic filter has been used for the measurements as suggested in [14] to prevent ultrasound disturbances in the microphone.

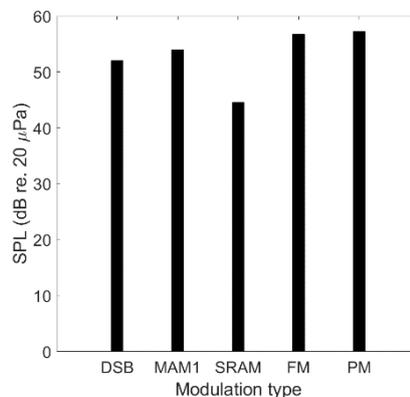


Figure 4. Sensibility of the parametric speaker measured at 1 meter and 1 kHz with a reference of 100 dB at the carrier wave (40 kHz).

4.2 Directivity characteristics

Both, parametric and conventional sources' directivity was measured by means of a turntable in a semi-anechoic environment with a DSB modulated MLS signal sampled at 96 kHz for a 180° angle range with a step of 2.5° between measures. To observe the main advantage of parametric speakers, that is the high directivity when compared to conventional electrodynamic speakers, Figure 5 shows a polar plot comparing both devices at 1 kHz.

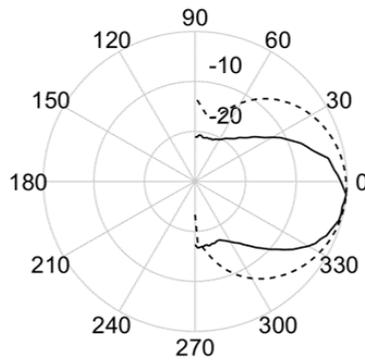


Figure 5. Directivity comparison at 1 kHz and 1 meter of the proof-of-concept parametric speaker with a DSB modulation (continuous line) and a conventional 5” electrodynamic speaker (dotted line) with an equivalent effective surface.

Figure 6 illustrates the directivity results for different modulations at 1 kHz and 1 meter, showing that the parametric speaker directivity isn’t much affected by the modulation scheme being used.

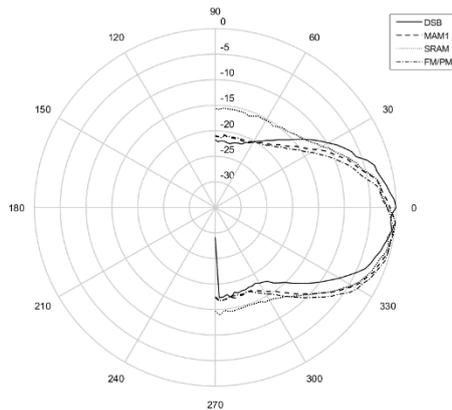


Figure 6. Directivity comparison of different modulation schemes radiated with the parametric speaker at 1 kHz and 1 meter.

In order to evaluate the directivity behaviour at a wider frequency range, Figure 7 and Figure 8 show a frequency against azimuthal angle plot so the directivity behaviour of both devices can be assessed at once.

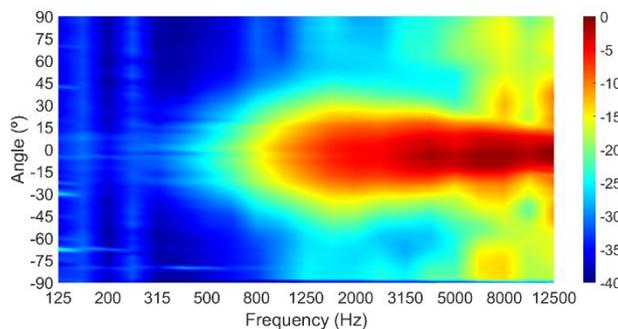


Figure 7. Directivity plot of the proof-of-concept parametric speaker (DSB modulation).

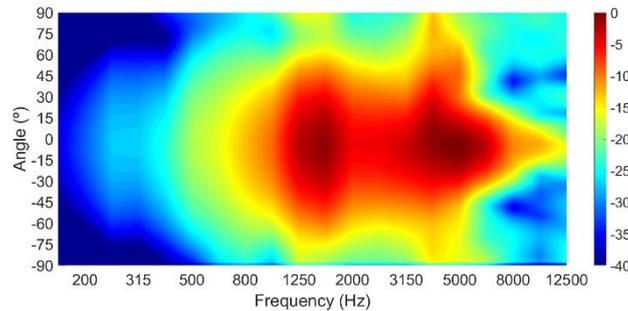


Figure 8. Directivity plot of the conventional electrodynamic loudspeaker.

As Figure 7 shows, the proof-of-concept device drives out the dependency on the source size and the radiated wavelength, showing a constant directivity tendency within frequency starting at 500 Hz.

4.3 Case study: *in situ* performance at Vilamuseu

To assess the performance of the parametric speaker when placed indoors, a straightforward experiment was carried on at the *Vilamuseu* facilities in La Vila Joiosa, Alicante. The parametric speaker was placed on an exhibition stand at 270cm height as shown in Figure 9, and seven listening points with a 1-meter separation were evaluated with the distribution shown in Figure 10. The microphone in point 4 is placed along the axis of the sound source, being this point the central measure with the highest received sound pressure level, thus the SPL of the rest of the points will be normalized to the maximum at point 4. In this way, we can assess the disturbance produced by both sound sources to the adjacent exhibition stands placed at points 1 and 7. The sound source reproduced an MLS signal, ensuring an equal SPL at point 4 for both the conventional and the parametric source.



Figure 9. Position of the sound source and microphone at the Vilamuseu exhibition stand.

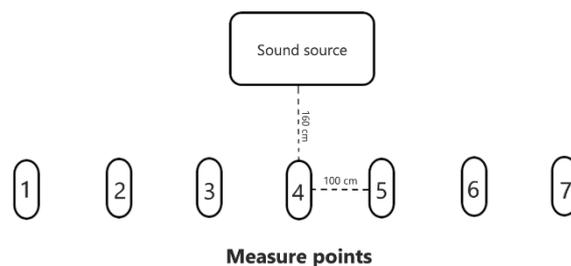


Figure 10. Measuring points scheme for the case study.

The distribution of the evaluation points in the museum facilities is shown in Figure 11.

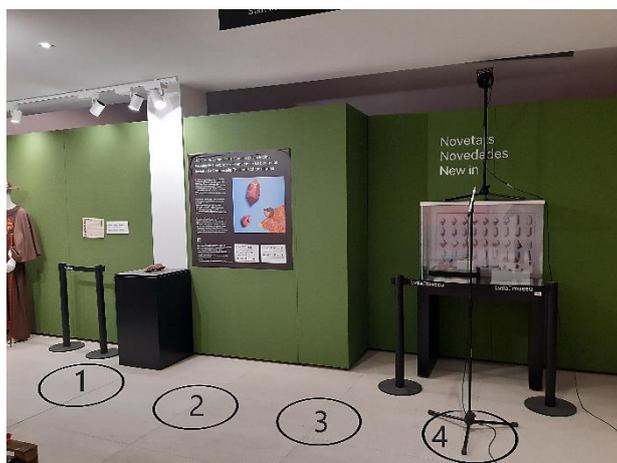


Figure 11. Measuring points locations in Vilamuseu.

Figure 12 shows the results for the electrodynamic loudspeaker normalized to the axial measurement. It is seen that the sound pressure level at exhibition stands points (points 1 and 7) at 3 meters from the central point, has a decay of 8 dB, which means that this type of source can't be used in focalized audio applications, producing an undesired disturbance to the adjacent stands.

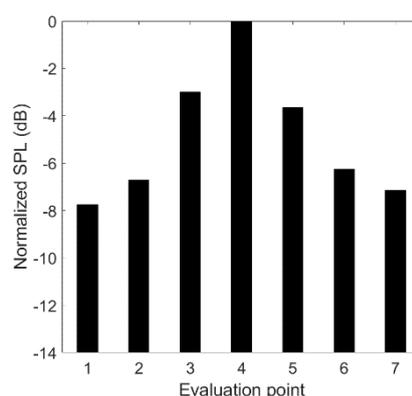


Figure 12. Normalized sound pressure level at the evaluation points when the conventional speaker is used as the main sound source.

On the other hand, Figure 13 shows the results obtained when the parametric speaker is used as the main sound source. Here, we can assess the great performance of this loudspeaker when compared with a conventional one, as it shows a near 25 dB decay at points 1 and 7, where another exhibition stand would be placed.

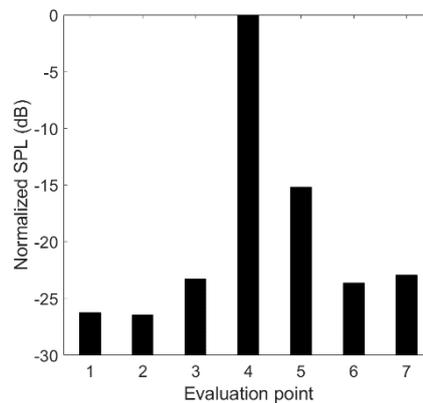


Figure 13. Normalized sound pressure level at the evaluation points when the parametric speaker is used as the main sound source.

Evaluation points 1 and 7 are placed at 62° from the central point, being the measured SPL at points 1 and 7 in Figure 13 a contribution of the parametric speaker direct radiation, that is very low for 62° as shown in Figure 7, and the reflections produced in the enclosure’s walls creating a reverberant field. The influence of this reverberant field should be considered if more than one parametric speakers are going to be working at the same time or if the sound reflections of a single parametric speaker are high enough to disturb other visitors. A final measure was taken where a narrow bandwidth recorded speech was reproduced leading to a 18 dB reduction between exhibition stands (points 1 and 4) with the parametric speaker, and a 7 dB reduction for the conventional one. Note that the reduction obtained with the parametric speaker can be negatively affected by the noise floor and the real reduction can be higher.

5 Conclusions

The proof-of-concept parametric array implementation has been proven to achieve a high directivity when compared to conventional sound sources. Its application in museum exhibit stands shows that it can deliver sound in a focused way, preventing the disturbance between consecutive stands. Although these devices offer a very narrow main lobe, the reflections from the enclosure’s walls can negatively affect other listening positions as well as the overall sound field. This fact must be considered when applying these devices in indoor sound applications by controlling the sound absorption at the main reflection points, which involves a deep study of the indoor sound field created with the parametric speakers. Further work includes this deep study of the reverberant field when one or more parametric speakers are used as the main sound sources as well as the study of possible solutions to diminish this effect leading to a feasible use of multiple parametric speakers at once in the same enclosure.

Acknowledgements

The case study to assess the ‘in situ’ performance was carried on at the *Vilamuseu* facilities in La Vila Joiosa, Alicante (Spain).

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