Highly Directional Multi-Beam Audio Loudspeaker

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Abstract

This work presents an audio beam loudspeaker that possesses the capability of emitting multiple audio beams into different directions at the same time. Ultrasonic loudspeakers applying parametric array technology are used to generate highly directional audible sound beams. The loudspeaker itself appears as a single device and can be mounted under the ceiling of a room in order to supply selected room segments with highly directional audio information such as music or speech.

Due to the fact that parametric array technology requires very high ultrasound levels, high-power transducers have to be used. Because phased array technology applied on those transducers for beam steering does not work successfully, an alternative approach based on pre-steered segments is presented in this work.

Index Terms: directional audio, multi-beam, parametric array

1. Introduction

Parametric array technology uses a parametric ultrasound loudspeaker that allows sending out a highly directional audible sound beam which can be projected onto a target like a light beam emitted by a spotlight. This device, for example, can be used in meeting or lecture rooms to deliver personal speech information to certain persons without disturbing other participants. Another application is giving information to museum visitors standing in front of an exhibit with different languages at different listening positions.

Generating audio sound beams with parametric arrays is based on transmitting a highly directional ultrasound carrier wave that is modulated by an audio signal. For this, an array consisting of many ultrasound transducers is used. If the audio beam has to be sent into various directions, beam steering is required which is usually done by applying phased array technology. But due to the transducers' directivity, this conventional approach can only be used combined with simultaneously tilting the transducers into the desired direction [1]. In order to deliver different beams at once, several units have to be used simultaneously.

This work presents a new design of an audio beam loudspeaker that enables us to send multiple beams with different information into different directions at the same time – without using mechanically moving parts or phased array technology. This is done by dividing the overall transducer array into several pre-steered acoustic segments pointing already into different directions, so that one may also describe the system as a *profiled acoustic device* (PAD). This approach makes more efficient use of ultrasonic energy and also leads to simpler signal processing and mechanical construction.

2. Ultrasound loudspeaker background

2.1. Generation of audio sound beams

Conventional approaches for directional audio use large loudspeaker arrays with dimensions far greater than audio wavelengths. In contrast, parametric ultrasound loudspeakers are capable of generating highly directional audible sound beams using emitter apertures that are small compared to audio sound wavelengths and therefore compared to conventional approaches as well. For example, using common technology to achieve sufficient directivity at audio frequencies, one has to use a loudspeaker with several meters in diameter what is far away from being reasonable. In contrast, taking a well-designed ultrasound loudspeaker system, only 20 cm in diameter will suffice to achieve similar directivity.

For this purpose, ultrasound transducers are arranged in an array so that the total array aperture is still small compared to the audio wavelength, but large in relation to the ultrasound wavelength (approximately 8 mm at 40 kHz), thus high directivity is gained for the carrier frequency. Using this array, a highly directional ultrasound carrier beam is now emitted into air, amplitude modulated by the audio signal $s(t)$:

$$p_1 = p_0\left(1 + ms\left(t - \frac{x}{c_0}\right)\right)\sin\left(\omega t - \frac{x}{c_0}\right)e^{-\alpha x} \tag{1}$$

where p_1 is the primary (ultrasound) pressure at a distance x from the ultrasound loudspeaker, p_0 is the ambient pressure, c_0 is the small signal sound velocity, α is the dissipation factor and m is the modulation index.

At high SPL (>110 dB [2]), nonlinear properties of sound propagation in air come into effect and lead to selfdemodulation of the audio signal, resulting from intermodulation of the carrier and the sidebands as primary waves. Due to this intermodulation process, air molecules are excited to oscillate at the audio frequency and therefore can be regarded as virtual sources. This process takes place inside a column within the primary beam where ultrasound pressure level is high enough to utilize nonlinear effects for audio generation. The virtual sources are distributed along the propagation axis of the primary beam with source density q [3 – 5]:

$$q = \frac{\beta p_0^2}{\rho_0^2 c_0^4}\frac{\partial}{\partial t}\left(ms\left(t - \frac{x}{c_0}\right) + \frac{1}{2}m^2 s^2\left(t - \frac{x}{c_0}\right)\right)e^{-2\alpha x} \tag{2}$$

with β as nonlinearity parameter and ρ_0 as ambient medium density.

The distance between the ultrasound loudspeaker and a virtual source oscillating at the audio frequency defines its audio phase, thus the virtual sources as a whole can be regarded as an endfire array: All demodulated elementary audio waves are in phase only for the primary beam direction and therefore add up yielding high directivity. Due to destructive interference, they weaken or even cancel each other out for every other direction.

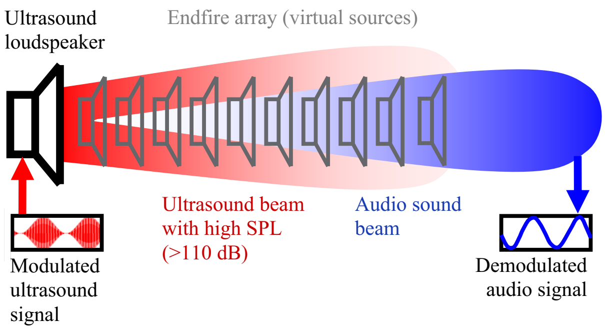

Figure 1 *Generation of an audio beam using an ultrasound loudspeaker.*

The audio wave equation is given by:

$$\nabla^2 p_2 - \frac{1}{c_0^2}\frac{\partial^2 p_2}{\partial t^2} = -\rho_0 \frac{\partial q}{\partial t} \tag{3}$$

Assuming the beam geometry to be circular, audio sound pressure level p_2 can be obtained from the following [3 – 5]:

$$p_2 = \frac{\beta p_o^2 a^2}{8\rho_0 c_0^4 \alpha x}\frac{\partial^2}{\partial t^2}\left(ms\left(t-\frac{x}{c_0}\right)+\frac{1}{2}m^2 s^2\left(t-\frac{x}{c_0}\right)\right) \tag{4}$$

The second time derivative in equation (4) implies a slope down to lower frequencies with 12 dB / octave in the audio frequency domain which theoretically results in a frequency response with emphasis on high frequencies and lack of lower ones. In practice, this can be partly compensated for example by active equalization. The squared part of *s* in p_2 will result in a harmonic distortion component at twice the modulating frequency whose level increases with the square of modulation depth *m*. It is due to interaction between the lower and upper sidebands and can be reduced by decreasing *m* what simultaneously reduces demodulation efficiency.

Because of using an ultrasonic frequency of 40 kHz, the carrier remains inaudible while only the demodulated audio signal can be heard. It shows remarkably sharp directivity which depends firstly on the distance to the loudspeaker and secondly on the loudspeaker size; a larger active area gives smaller beam width. Audio beam width decreases with distance within the generation zone which is defined by having ultrasonic SPL high enough to keep the endfire array active and generate audio sound [2]. Therefore, the distance between the loudspeaker and the user should not be less than say 50 cm in order to obtain tight audio beam width. This is useful as well in terms of medically safety because ultrasonic SPL may exceed 140 dB close to the loudspeaker. The device should therefore be operated with a proximity sensor to limit ultrasound level when humans are too close. At larger distances, ultrasonic SPL is reduced by dissipation and beam divergence.

Unfortunately, efficiency in terms of audio sound pressure level in relation to electric input power is very low, since approximately 140 dB of ultrasound pressure level is required to generate an audio SPL of 80 dB. Because parametric array technique has to generate those high ultrasonic SPL, it requires far more electric power than a common loudspeaker for generating similar audio SPL.

2.2. Approaches to audio sound targeting using ultrasound loudspeakers

Phased array technology uses point sources which emit circular waves, sending sound energy into all directions equally. By driving them with certain time delays, the elementary waves interfere to a new wave front which then propagates into the desired direction. As mentioned in section 1, applying phased array technology on ultrasound loudspeakers for beam steering approaches is usually inefficient because unidirectional transducers in terms of point sources are needed for beam steering, but only directional transducers are commonly available that deliver enough sound power to utilize nonlinear effects for parametric sound generation.

An alternative approach [1] shows the possibility to circumvent this aspect by using a hybrid system which consists of combined electronic (phased array) and mechanical (tilting) beam steering. There, additionally to being driven with matching time delays, the transducers are tilted simultaneously into the desired direction to compensate for the lack of omni-directionality. Consequently, tilting the transducers increases overall system complexity and effort. Additionally, noise is generated during the tilting process that may be disturbing, especially in silent environments.

The new approach to audio sound targeting in this work is based on using an ultrasound loudspeaker as a PAD. When a selected room segment has to be supplied with sound, just that sub array that points into the matching direction is driven. By doing this, only emitters are active that actually deliver sound energy into the desired direction instead of utilizing phased array technique and driving point sources that deliver energy into all directions equally.

Compared to phased array technique, the disadvantage of having non-point sources is turned into an advantage since sound energy has exclusively to be emitted into the desired direction instead of spreading it into all directions equally. This helps saving electric power consumption significantly since generating audio beams utilizing parametric array technology is characterized by very low efficiency.

3. System realization

To deliver targeted audio information to a meeting room, one firstly has to determine an appropriate spatial angle resolution. While a course distribution just gives too few independent listening zones, a fine resolution causes overlapping of target positions and therefore increases crosstalk levels between them. Therefore, audio beam widths and angle resolution have to be accommodated. Additionally, emitted sound beams have to be prevented from being reflected at neighboring arrays which has to be considered when constructing the system. We decided to build 13 segments in three concentric rings what makes the

PAD suitable to deliver targeted audio in small or medium sized meeting rooms. The loudspeaker is intended to be mounted under the ceiling thus a listening distance of 1.5 to 2 meters is achieved.

3.1. Sub arrays

In order to achieve similar beam widths, the sub arrays should consist of similar numbers of emitters. Due to the overall mechanical construction, arrays in different rings get different shapes so that beam widths still differ slightly but not remarkably. Due to the fact that the transducers differ slightly in resonance frequency and phase – probably because of production tolerances – they have been sorted out and matched within a sub array to assure homogenous beam geometry within each channel. They have been arranged in a hexagonal structure to achieve high packing density for generation of high SPL while occupying minimum area.

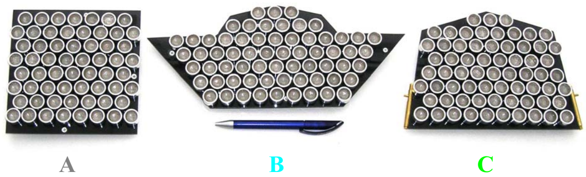

Figure 2 *Sub array of inner (A), second (B) and outer ring (C).*

3.2. Signal processing

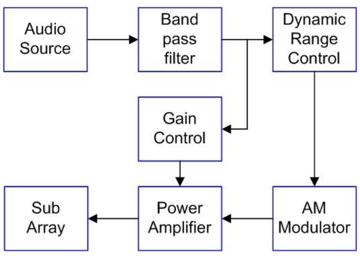

Figure 3 *Single channel block diagram.*

The loudspeaker contains 13 independent channels that are electronically equal – see fig. 3. An audio signal may either be routed to one single channel or several channels may be combined and supplied from the same audio source to obtain a larger coherent listening zone. The band pass filter cuts frequencies below 200 Hz, which cannot be sufficiently reproduced by the ultrasound loudspeaker but may cause distortion. Frequencies above 10 kHz are cut as well because the information content within audio, respectively speech reproduction is assumed to be redundant. Dynamic range control is needed to achieve constant modulation depth and prevent distortion due to overmodulation. The gain control section reduces the power amplifier's gain when incoming audio signals are too weak to be reproduced in a useful manner. This also helps reducing power consumption. Reduction of distortions that are implied by equation (4) has not been taken into account since speech intelligibility is not worsened remarkably. But if reduced audio volume is desired, we suggest reducing modulation depth since then distortion is reduced as well. Further approaches can be found in [6].

In contrast to [1], where a DSP has been utilized to perform signal processing and calculating phase delays, effort has been reduced significantly since standard analog components have been used.

3.3. Overall construction

Having the PAD mounted under a meeting room ceiling, the center element represents the inner ring (A) and points directly downwards. The four elements of the second ring (B) are arranged in angle of 30 degrees inwards, so that their beams cross each other before reaching their targets. The outer ring (C) consists of eight sub arrays pointing outwards. Their angles can be adjusted within 10 to 60 degrees to enable adapting the beam angles to various room geometries. This assures optimum flexibility.

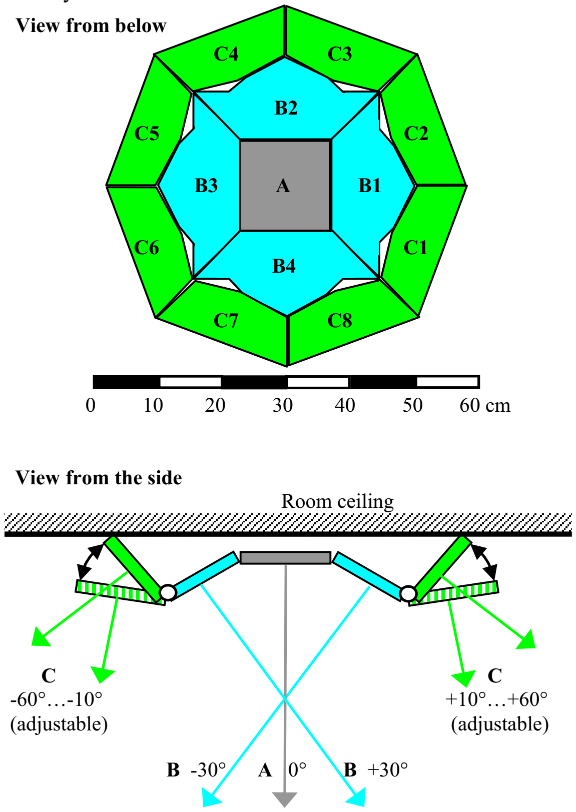

Figure 4 *Overall construction details; suppose the loudspeaker to be mounted under a room ceiling.*

4. System performance

4.1. Directivities and frequency response

Fig. 5 shows the directivity of a single sub array at a distance of 1 m. Directivity is generally sharper at high frequencies due to shorter wavelengths in relation to array size and longer generation zone compared to low frequencies [2].

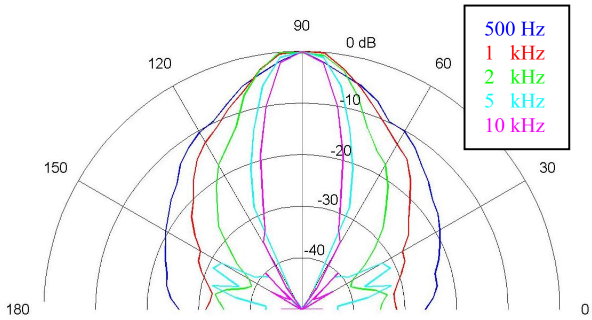

Figure 5 *Directivity of single sub array at 1 m distance.*

Regarding the whole loudspeaker device with the C arrays adjusted to 60°, Fig. 6 shows the resulting directivities for 2 kHz as if all sub arrays were working in one common plane. It can easily be seen that finer angle resolution will not be useful since beams of neighboring arrays overlap already at -8 dB. Actually, neighboring B and C arrays are shifted against each other by 22.5° thus – for example – the elements B3, C1 and C2 are practically direct neighbors but work in different planes. In this case, their beams overlap at -15 dB which generally should assure sufficient crosstalk level reduction.

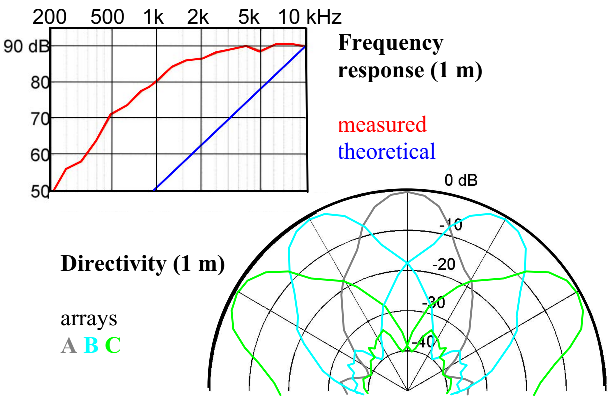

Figure 6 *Multi-beam directivity and frequency response.*

The theoretical slope down to low frequencies implied by equation (4) has been partly compensated by using high-Q transducers. Low frequency content that is located near the carrier due to modulation is emitted at much higher levels than high frequencies, yielding partial equalization and improved sound quality.

4.2. Sound distribution in a meeting room

Fig. 7 shows an example scenario examined within the CHIL project [7] where the multi-beam loudspeaker is mounted under a meeting room ceiling. The outer ring elements C8 and C1 have been adjusted to serve the region in front of the projection board with targeted audio, for example to deliver a lecturer with personal information. C4 points at the room entrance and is intended to welcome persons entering the room and deliver them with information about the ongoing meeting (e. g. if they come late) without disturbing other participants. If these persons then walk towards the projection board, their way might be automatically tracked and C2 and C3 become active. The B arrays deliver personal information to participants at the table. Arrays A and C5…C7 are not used in this example.

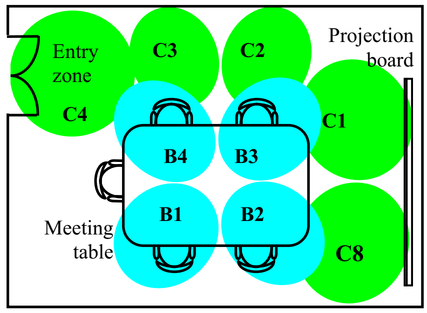

Figure 7 *Meeting room scenario with multi-beam loudspeaker mounted under the ceiling right above the middle of the table.*

5. Conclusion

A multi-beam audio loudspeaker has been constructed to be mounted under a meeting room ceiling with the aim to supply selected room segments with highly directional audio information. In contrast to former approaches, power consumption and effort in terms of complexity have been reduced significantly. Additionally, the device is highly flexible since spatial audio sound distribution can be adapted to various room geometries.

6. Acknowledgement

This work has been part of the FP6 project CHIL (Computers in The Human Interaction Loop, [7]) which is funded by the European Union. The project aims to facilitate everyone's daily life by having computers interact with humans in an indirect and unobtrusive way instead of having them directly occupying humans´ attention.

7. References

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