MODULATION AND DEMODULATION OF STEERABLE ULTRASOUND BEAMS FOR AUDIO TRANSMISSION AND RENDERING

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ABSTRACT

Nonlinear effects in ultrasound propagation can be used for generating highly directive audible sound. In order to do so, we can modulate the amplitude of the audio signal and send it to an ultrasound transducer. When played back at a sufficiently high sound pressure level, due to a nonlinear behavior of the medium, the ultrasonic signal gets self-demodulated. The resulting signal has two important characteristics: that of becoming audible; and that of having the same directivity properties of the ultrasonic carrier frequency.

In this paper we describe the theoretical advantages of single-sideband (SSB) modulation versus a standard amplitude modulation (AM) scheme for the above-described application. We describe our near-field soundfield measuring experiments, and propose steering solutions for the array using two different types of transducers, piezoelectric or electrostatic, and the proper supporting hardware.

1. INTRODUCTION

The nonlinear effects generated by high-level ultrasound were studied already in the 1960's by Westervelt [1]. Westervelt found that the nonlinear effects made the beam act as a distribution of sources, creating what he called a *parametric array*; if two collimated sound beams travel in the same direction, the nonlinearity of the propagation medium will generate components at the sum and the difference of the frequencies. The sound beam will propagate with the same (high) directivity of the primary beams. Berktay, few years later [2], introduced the concept of envelope, showing the link between the primary sound and the scattered sound generated by the nonlinear effect. The scattered sound RMS pressure is proportional to the second time-derivative of the squared envelope of the primary sound.

Berktay's contribution is still used as the easiest, but still effective, model for this application. Westervelt's work was theoretical and Berktay's studies were limited to underwater applications. Bennet and Blackstock [3] in 1975 showed the actual existence of parametric arrays in air, being able to measure the 5 kHz difference frequency, generated by the nonlinear interaction of two highly collimated primary beams at respectively 18.6 kHz and 23.6 kHz. In 1983 Yoneyama and Fujimoto [4] presented an important application of these nonlinear effects, which is used as a basis for the present study: the *parametric loudspeaker array*. They claim that modulating the amplitude of an ultrasonic carrier with a broadband audio signal, and running the modulated signal through an ultrasonic transducer, the audio signal will 'self-demodulate' while preserving the directivity of the carrier beam. The RMS pressure of demodulated signal is proportional to the second time-derivative of the envelope of the AM signal, as predicted by Berktay [2].

Having in mind the application of a personal audio system, the distance between the listener and parametric array has been assumed to be of 1 meter or less, which means that the listener is in the near field of the array.

In Section 2 we discuss the SSB modulation and the standard AM used for parametric arrays of loudspeakers. Here we present some of the advantages and disadvantages of both. In Sections 3 and 4 we describe our experiments and the related measurement results of the beam pattern, while in Section 5 we will propose a solution on how to accomplish a suitable beam steering and in Section 6 conclusions are drawn and future perspectives are presented.

2. MODULATION SCHEMES

Standard amplitude modulation has always played and still plays a major role in parametric array studies. Among the modulation schemes, AM is one of the simplest to understand and implement.

It is well known that AM can be achieved by multiplying the sinusoidal carrier by a modulating (information-carrying) signal, which in our case is the audio signal. The spectrum of the modulated signal has two sidebands, symmetric with respect to the carrier frequency, and the carrier itself. The AM envelope can be thought of as the audio modulating signal. The receiver of the AM modulated signal is as simple as a peak detector.

Two things can be inferred from this. First of all, it is reasonable to use Berktay's work based on the envelope of the primary signal to have a simple and basic model for the difference frequency pressure; in AM, the envelope is the *difference* pressure signal since the envelope is a base band (*i.e.* audio band) signal.

Second, more important for commercial applications, nonlinear interaction happens between all the frequency components of the AM signal causing distortion.

We should distinguish between two types of distortion. One can be considered an intra-sideband distortion and the other inter-sideband distortion.

The intra-sideband distortion is caused by different frequency components that subtract each other within a specific sideband. Although its impact is not negligible, this type of distortion has not been studied in this work. Inter sideband distortion, on the other hand, is the interaction between the frequency components of two different sidebands. The two sidebands interact and more distortion is added to the demodulated signal. This inter sideband distortion can be reduced. Kite *et al.* [6] propose a pre-processing algorithm to reduce the inter sideband distortion. This algorithm is used in most published applications of the parametric array loudspeaker. It is interesting for the goals of this study to understand *what* distortion this algorithm reduces and *how* it does this.

Kite's algorithm is based on Berktay's model [2,6], as it simply applies the *opposite* operations present in Berktay's model. According to the model, a collimated wave consisting of an amplitude modulated wave of pressure

$$p_1(t) = P_1 E(t)\sin(\omega_c) \qquad (1)$$

where , P_1 is the amplitude of the carrier signal, $E(t)$ is the modulating envelope and ω_c is the carrier angular frequency, will demodulate after the nonlinear phenomenon in an audible sound wave having the following pressure:

$$p_d(t) = \frac{\beta P_1^2 A}{16\pi\rho_0 c_0^4 z\alpha}\frac{\partial^2}{\partial t^2}E^2(\tau) \quad (2)$$

where $\beta = (\gamma + 1)$ is the coefficient of nonlinearity (β_{air}=1.2), γ is the ratio of specific heats, A is the radiating area of the transducer, ρ_0 is the density of air, c_0 is the small-signal wave propagation speed, z is the axial distance, α is the absorption coefficient of air for the carrier frequency. Expression (2) shows the dependency of the demodulated audio signal pressure on the envelope of the amplitude modulated signal. The envelope is a baseband signal for definition. Expression (2) is an important formula, presented by Pompei in [5], because it is the adaptation of Berktay's model to parametric loudspeaker arrays, where the modulating signal is a broadband audio signal. Pompei's work is remarkable because it contributed to one of the few commercial products based on the parametric array. Pompei bases his study on Berktay's model and on the usage of AM modulation and Kite's algorithm. The present work, as it will be shown later on, makes different choices than Pompei's approach, [5].

Kite's preprocessing algorithm applies 1) a double time integral to compensate for the double time derivative and 2) a square root to compensate for the quadratic nonlinearity in Berktay's model. The carrier then multiplies the pre-processed audio signal. Applying in sequence Kite's algorithm and Berktay's solution shows that the overall effect is a strong attenuation of the lower sideband of the AM spectrum: this is due to the double derivation and to the double integration. Hence Kite's pre-processing reduces inter sideband distortion.

With this in mind, we think that a better solution would be to use single-sideband (SSB) modulation

A more efficient way to avoid inter-sideband distortion is to adopt a SSB modulation scheme. The main characteristic of SSB is the presence of just one sideband, either the lower sideband (LSB) or the upper sideband (USB). Standard SSB used in telecommunications is without carrier, but in the parametric array application one needs to add the carrier to generate the difference frequencies in the audio band. Thus we used a so-called SSB-WC signal (single-sideband with carrier), which means we added the carrier (acoustically or electrically) just after the standard SSB modulation.

Having just one sideband, we do not need to be as concerned with inter-sideband distortion, therefore we do not need to use Kite's pre-processing. Moreover it may be interesting to be able to choose which sideband to use according to the frequency response characteristic of the transducer array and the absorption of the transmitting channel. One more rather obvious advantage of a SSB approach is that it uses the available bandwidth more efficiently than the double sideband approach.

In the next section we will show some measurements done to compare AM and SSB performance.

3. AM AND SSB PERFORMANCE MEASUREMENTS

In order to comparatively test the effectiveness of the AM and the SSB scheme, a set of measurements were done to determine the sound pressure levels (SPL) of the various signals at different distances from the array. For these measurements we used an octagonal array made of 120 piezoelectric transducers, each having a resonance frequency of 38 kHz. The array was moved away from a B&K 4138 microphone, used to measure the sound pressure as a function of distance; the minimum distance is 5 cm and the maximum 1 m. Because of the microphone's wide and flat frequency response we were able to measure both the audio and ultrasonic band signals simultaneously. The signal sensed by the microphone was analyzed using a HP spectrum analyzer which allowed us to automatically filter the measurements, avoiding any spurious signal that could have influenced the results, as pointed out by Blackstock *et al.* in [3].

The signal source was an iBook G4 connected to an Edirol external audio card. Two `Matlab` scripts were written to generate AM and SSB modulation of the carrier by a 1 kHz tone. The carrier frequency was 38 kHz in both cases and a sampling frequency of 192 kHz was used in the D/A-conversion. The modulated signal was then sent to a wide-band NAD amplifier and from there to the piezoelectric array.

The measurements showed interesting results confirming our expectations and helping us plan future work.

As a first step the near-field distance of the array was checked. The near-field distance is given approximately by the following equation

$$n = \frac{\pi a^2}{\lambda} \qquad (3)$$

For a circular array having a radius of 6 cm (a) and a signal carrier of 38 kHz, the equation gives a near-field limit distance of 1.25 m, which corresponds to the farthest distance used by us in our experiments.

In general, the near-field is characterized by an irregular pressure pattern. Our measurements show a quite regular sound pressure level in the 0.1 – 1 m range distance between the transducer and the microphone in contrast to expectations, as shown in Figure 1, 2 and 3. The shape of the sound pressure level curve as a function of distance is very similar to the one obtained by Blackstock *et al.* [3], which is reasonable, since their measurements were also done in the near-field; notice that Blackstock *et al.* were somehow forced to perform the measurement in the near-field, since the measuring room was not large enough to be in the far-field, while we wanted to study the near-field behavior of the parametric array.

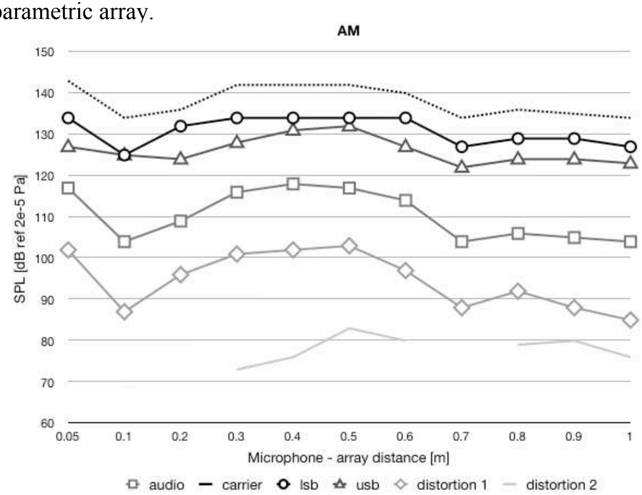

Figure 1: *0.1 – 1 m range SPL for AM.*

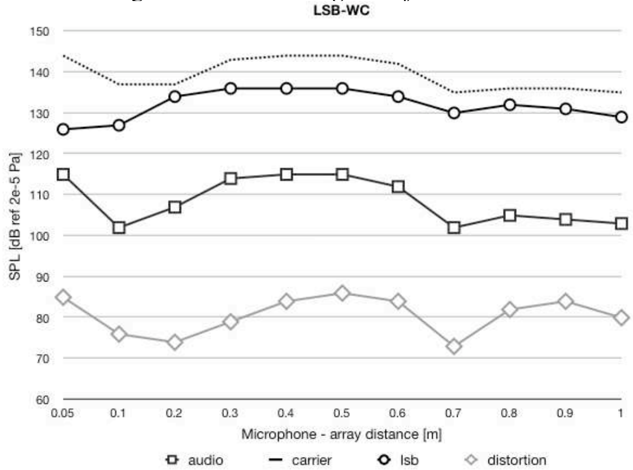

Figure 2: *0.1 – 1 m range SPL for LSB-WC.*

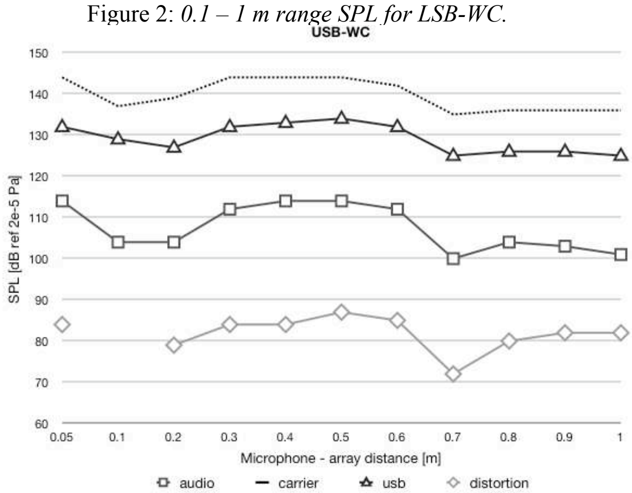

Figure 3: *0.1 – 1 m range SPL for USB-WC.*

The curves of the sound pressure level as a function of distance, shown in Figure 1, 2, 3, are very similar whether using AM, LSB-WC, USB-WC. We can notice the presence of distortion in AM: this was expected. Actually this distortion should be due just to the inter-sideband demodulation, since each sideband had only one frequency component. This means that single frequency modulated SSB-WC should show no distortion. The distortion present in our SSB-WC measurements is due to the piezoelectric transducers which, when required to generate the high sound pressure level (the carrier SPL was around 140 dB), also generate some distortion. This is confirmed by the fact that our SSB scheme shows less distortion than the AM scheme, since AM suffers of both inter-sideband distortion and transducer distortion. The distortion components appear in the HP spectrum analyser monitor as lines in the audio band, parallel to the demodulated 1 kHz tone's spectrum. Referring to Figures 1-3, *distortion 1* corresponds to a frequency component at 2 kHz (inter-sideband distortion), while *distortion 2* (transducer distortion) corresponds to a frequency component at 3 kHz. In reality the piezoelectric transducers generate more distortion components; the majority of these components have a lower SPL than *distortion 1* and *distortion 2*, thus they are not easily readable on the monitor of the HP spectrum analyser.

Two conclusions can be drawn from the results of this measurement. One is that using AM or SSB does not change the demodulated audio signal SPL (which is what we are mainly interested in) and both modulation schemes have a similar audio signal SPL profile, as a function of distance, in the near-field. As discussed in section 2, we also wish to choose SSB because of its other advantages. Moreover this particular kind of piezoelectric array does not really fit this application.

3.1. The transducers

The construction of the piezoelectric transducers used for the octagonal array is shown in Figure 4. This type of transducer depends on the resonance properties of the piezoelectric material itself (the PZT layer in the figure), the radiating cone (which is needed to match the high impedance of the PZT material to the much lower impedance of air), and the air cavity. Correctly used, this approach results in a fairly wide-band transducer (typically about 5 kHz), depending on the choice of resonance frequencies.

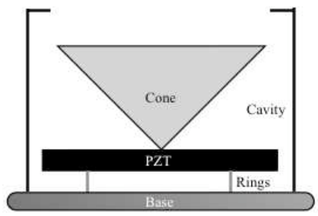

Figure 4: *Construction of the piezoelectric transducer used for the octagonal array.*

Because of its bandpass characteristic, and because of the high pressure inside the air cavity, this type of transducer is likely to produce noticeable linear and non-linear distortion in audio applications such as the one considered in this work.

The problem can be solved using other types of transducers, such as electrostatic Sell-type transducers. Electrostatic Sell-type transducers, as shown in Figure 5, have an intrinsic high efficiency due to their better impedance match between the thin diaphragm foil layer and the air, which is better than the impedance match between a piezoelectric material and the air. Moreover, such electrostatic transducers can be both broadband (about 20 kHz or more) and directive.

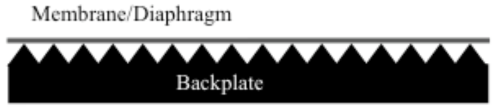

Figure 5: *Construction of the electrostatic transducer*

The design showed in Figure 5 is a convenient Sell transducer design, which can be produced by micro-machining, proposed by Oksanen *et al.* [7]. A transducer of this or similar design would allow us to use electrostatic transduction to transmit a more broadband SSB modulated signal at a higher frequency. Working at frequencies in the order of 150-200 kHz, it would allow us to use SSB-WC with the desired 20 kHz audio bandwidth.

4. NEAR-FIELD SOUND PATTERN MEASUREMENT

Since we are interested in the near-field properties of the array, we did both a `Matlab` simulation and physical measurement of the SPL characteristics of the beam.

The near-field is known to present an irregular pressure profile, until the far-field is reached. In the far-field the pressure decreases monotonously, proportional to the inverse of the distance, while in the near-field the coherent interaction between the sound pressure contributions from various parts of the array generate a complex interference pattern.

The simulation and the measurements show how the sound field behaves at various cutting planes, orthogonal to the sound beam and parallel to the plane of the array.

The numerical simulation assumes that each transducer in the array is made of a larger number of square patches whose sides are λ/6 wide. Each patch is assumed to radiate as a monopole giving a sound pressure contribution Δp as shown in by the following equation:

$$\Delta p = \frac{\rho c j \Delta U}{2\pi c d} e^{-jk(d-a)} \tag{4}$$

where d is the distance between the transducer and the observation point and U is the volume velocity. Figure 6 shows the simulation for a cutting plane ad 0.5 m. Side-lobes are very evident.

The measurements were done using a *robotic arm* that moves the microphone along a pre-defined matrix of observation points. At each point a measurement is done using MLSSA soft- and hardware. The standard MLS sequence provided by MLSSA was used as the audio input signal to the system. The pseudo-random broadband noise generated by MLSSA was filtered using a low-pass filter having a cutoff frequency of 5 kHz to avoid aliasing. This is necessary because the MLSSA system bandwidth was set to 25 kHz and the AM carrier was 38 kHz, necessary to operate the piezoelectric transducers.

The filtered signal was then fed to a Leader function generator to obtain standard AM modulation. From the modulator, the modulated signal was again fed into a NAD amplifier, from which the signal then is fed to the matrix of the all-in-phase piezoelectric transducers.

The scanned matrix of observation points was 60 cm by 60 cm large, having 30 lines with 30 measurement points per line, which means a spatial resolution of 21 mm. The matrix was chosen to be fairly large in order to measure the entire sound beam, estimate the directivity of the audio beam and study the field around it. The measurement was done in laboratory surroundings; this resulted in reflections and scattering from all the surfaces and objects in the room. In the post-processing of the data, however, the time vector measured by MLSSA was windowed to use only the direct signal sensed by the microphone. The reverberant response of the room was entirely eliminated by this windowing process.

Figure 7, 8 and 9 show the SPL at the 2 kHz frequency of the demodulated MLS signal at 0.1 m, 0.5 m and 1 m respectively.

The audio sound beam has high directivity and a small diameter, the width can be estimated from the measurements to be about 15 - 20 cm at the 1 meter distance.

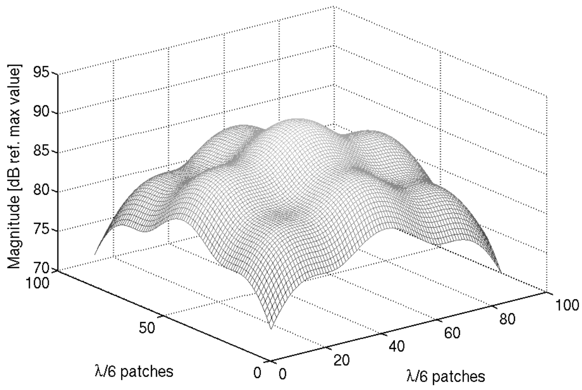

Figure 6: *Sound field simulations at 0.5 m. Array modeled as a matrix of λ/6 wide patches emitting according to formula (2). Studied frequency: 2 kHz.*

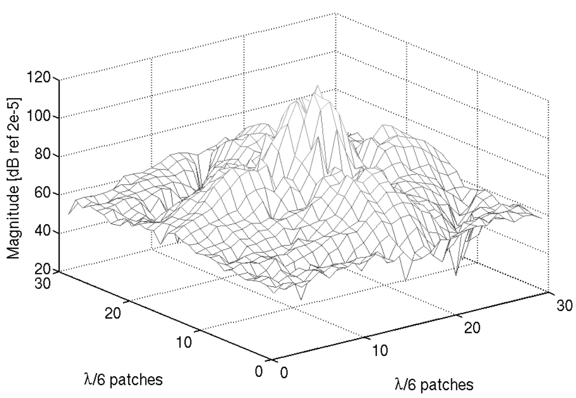

Figure 7: *Sound field measurement at 0.1 m. The 2 kHz frequency is extracted from the MLS sequence and plotted.*

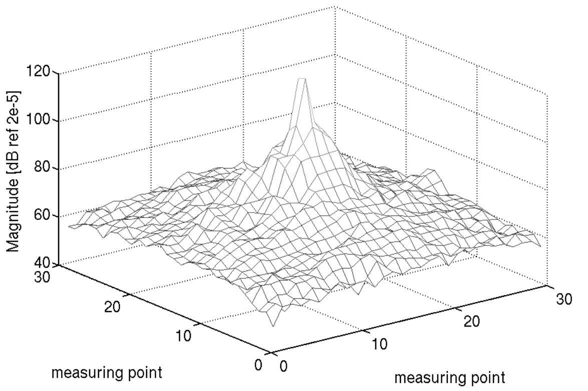

Figure 8: *Sound field measurement at 0.5 m. The 2 kHz frequency is extracted from the MLS sequence and plotted.*

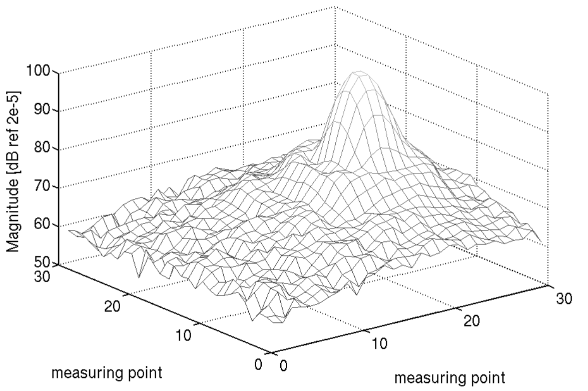

Figure 9: *Sound field measurement at 1 m. The 2 kHz frequency is extracted from the MLS sequence and plotted.*

Another good result is that in spite of our being in the near-field, the sound field is quite even and does not show interference pattern behavior, in contrast to the numerical simulation, which however was done for a single frequency only. With a better model of both the array and of the air nonlinearity even more accurate simulated results may be achieved.

Notice that the reason of such a good measurement might also be the usage of a rather large microphone, 1/8-inch in particular, causing a spatial integration of the sound field. A measurement with a probe smaller than λ/6 may be interesting to do.

5. BEAMSTEERING

The Figures in Section 4 show the characteristics of a super-directive audible sound beam generated by an experimental parametric array audio system, the properties of which are determined by both hardware and digital signal processing. Further digital signal processing could help improving the performance of this parametric array audio system.

A DSP-controlled array generating a super-directive audible sound beam, can be used, for example, to point the beam in a different direction, and the directions can be changed instantaneously, unlike with a mechanically governed array. Classical beamforming theory can be used and implemented also for parametric arrays.

Known techniques to create steerable beams are based on arrays of transmitters, each driven with a different but coherent signal. Each signal differs from another by a suitable time delay, which is chosen in order to sum the contribution from each loudspeaker in order to for example obtain a focusing effect. If the time delay is suitably chosen one can steer the sound beam in different directions.

Consider an angle θ in respect to the normal to the plane where the M-elements array lies and assume the inter-element spacing to have a value of *d* meter, it is possible to focus and steer the sound beam of θ degrees if the signal from the *m*th element is delayed of

$$\tau_0^m = \frac{md}{c}\sin\theta \qquad (5)$$

The far-field array response can be expressed as

$$H(k,\theta) = \frac{1}{M}\sum_{m=0}^{M-1} w_m e^{-j\omega\tau_0^m} e^{j\omega(md/c)\sin\theta} \qquad (6)$$

where w_m is the amplitude weight for each element, and ω is the frequency of interest.

Gan *et al.* in [8] use this theory, which is the *classical delay-and-sum* beam-forming technique, to develop an algorithm to steer the difference frequency audible tone generated by self-demodulation using SSB. The proposed algorithm is to be implemented on a DSP board. We thought about some developments in the actual state of the art.

We may use the fact we are actually working in the array's near-field. *Delay-and-sum* algorithms allow us to focus on a given direction if we are working in the far field, however, in the near-field we can focus the array on a particular spatial location. Assuming r^o as the distance between the spatial location and the centre of the array, r^o_m as the distance between the spatial location and the *m*-th element, x^o as the spatial location of the centre of the array and x^o_m as the spatial location of the *m*th sensor we can apply the following delays:

$$\frac{r^o - r_m^o}{c} = \frac{r^o}{c}\left[1 - \sqrt{1 + \frac{|x_m|^2}{(r^o)^2} - 2\frac{x^o \cdot x_m}{(r^o)^2}}\right] \qquad (7)$$

to focus to the spatial location defined by the vector r^o. This can be used to increase the ultrasonic power locally in the beam to generate louder self-demodulated audio than otherwise possible. The same property could also be achieved by using geometrical focusing.

One possibility would be to build the transducer over a parabolic surface to direct the radiation to a chosen point. The other is based on the idea of a *zone plate*, which in its standard use, is a device to focus light using diffraction instead of refraction (like lenses). The idea is to place rings of transducers at different radii in order to get constructive interference at the focus.

6. CONCLUSIONS AND FUTURE PERSPECTIVES

Our measurements show good results regarding the near-field behavior of the parametric loudspeaker, which have not been thoroughly investigated in previous studies. The theoretical advantages of single-sideband (SSB) modulation versus a standard amplitude modulation (AM) are shown to hold in practice. The use of standard, commercially available piezoelectric transducers leads to problems of narrow bandwidth, distortion, and large sidelobes. Future work must focus on better adapted transducers, arrays, and beamsteering methods to circumvent these problems, and will require a multidisciplinary approach scientifically and technologically.

To implement the parametric array loudspeaker algorithm, a possible solution already investigated [9] is the use of field-programmable gate-arrays (FPGAs), digital logic chips with some interesting features. These are based on parallel signal processing, they are reprogrammable, they can change their functions from time to time, they have a shorter time to market. In 2004 Karnapi *et al* [9] successfully implemented an AM-based parametric loudspeaker system in a FPGA, using Kite's algorithmic distortion reduction.

In our proposed system the FPGA would be used to perform both the SSB-WC modulation and the array beamsteering. As mentioned previously, no distortion reduction algorithm is needed distortion if SSB-WC is used.

For the beamsteering many different solutions are possible. These can be roughly divided into two categories, implementation of amplitude shading and time delays 1) by DSP, as described in the previous section, or 2) by *mechanical* techniques.

The first solution for beamsteering requires a matrix of transmitting elements that are addressable and algorithmically controlled. Piezoelectric or electrostatic transducers can be adapted to these needs.

The second solution can be itself further divided into two subgroups, depending whether we work on the transducer or on the sound beam. So, a) with mechanical means it would be possible to move the surface on which the transducer is built, or b) it would be possible to have a mirror that reflects the sound beam to a chosen direction.

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