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EVALUATION OF BEAMFORMING SYSTEMS

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ABSTRACT

The growing number of commercially available beamforming systems and the quite large number of end users and application fields confronts all potential customers with the task of comparing the systems of different suppliers as well as with the problem of evaluating the quality of beamforming solutions in general. This article will point out some of the main problems connected to this practically often difficult task and it also develops some proposals for a more systematic evaluation of acoustic array beamforming systems. Some useful quantities and evaluation methods which seem appropriate for a possible classification of beamforming systems will be proposed and discussed in detail. This discussion shall serve as a first contribution for a future standardisation and hence for the further qualification of acoustic beamforming as a measurement method.

1 INTRODUCTION

More than 10 years ago, with the introduction of the acoustic camera the first commercially available array systems for sound source localisation and analysis based on the delay-and-sum beamforming principle entered the market, confronting array system developers as well as potential customers with the question: **How shall the performance of such systems be evaluated and verified adequately?** To complicate the matter further, beyond phased arrays there are other methods of sound source analysis such as special solutions of laser vibrometry, nearfield acoustic holography or even acoustic mirrors, which are based on different physical principles and have consequently not been very helpful to answer this question.

As long as the various beamforming systems (phased array systems, respectively) have mainly been special solutions of individual research institutions (e.g. DLR / NLR / NASA) or of qualified acoustic labs within larger companies (e.g. Daimler AG, DB, Boeing and others), the beamforming method per se was not questioned too much. Instead, the ongoing scientific discussion concerned the question whether the results achieved were representing the reality in a plausible manner, being quantitatively repeatable and statistically reliable.

Among the experts in the field, this question has soon been answered in favour of the beamforming method, the more the less practical alternatives were available, e.g. to identify and measure noise sources of an airplane in flight, a vehicle passing by or in an aeroacoustic wind tunnel.

2 THE BEAMFORMING SYSTEM EVALUATION PROBLEM

2.1 The traditional approach – a narrowband view

Nowadays, there are many suppliers of commercial array systems, and especially non-expert users are facing a multitude of confusing information about the "pros" and "contras" of beamforming, which makes an objective evaluation of such systems very complicated for them. On the other hand, the general acceptance of beamforming methods within the acoustics community is far better today than it was about six to ten years ago, so the situation is much more comfortable now.

While the value of the microphone array beamforming method as such is not questioned any longer, the detailed practical evaluation of the method is still dominated by a rather questionable approach stemming from the wide spread "narrowband view" of traditional acoustics. Again and again, in real life we are confronted with the following evaluation procedure: Take a single sound source (preferably the worst and cheapest plastic PC speaker available, and preferably one equipped with a separate low/mid-range chassis and tweeter), place it somewhere in front of the array and feed it with a sinusoidal signal in order to determine the source location and the sound pressure values within the acoustic map for that single source and for that single frequency. Those "measurements" are then repeated for individual frequencies or for very narrow frequency bands, proceeding from the lowest frequency values to the highest frequencies in the audio band. The results are the following: the source location is hit quite well, the level values are appropriate for a dot-like source, but the map dynamic and the resolution depend strongly on the signal frequency. For low frequencies the dynamic and the resolution are rather bad (what a surprise!), for the mid frequency range those values improve and the array performs quite well, whereas for very high frequencies the resolution may improve further but the map dynamic gets worse again due to HF-aliasing effects (grating lobes). So far, so good. Nothing is really new here, all this is to be expected, it is long known from classic array theory [1] and it can easily be verified by analytic calculations for very simple array geometries or by means of simulation of the point spread functions for the more complicated ones.

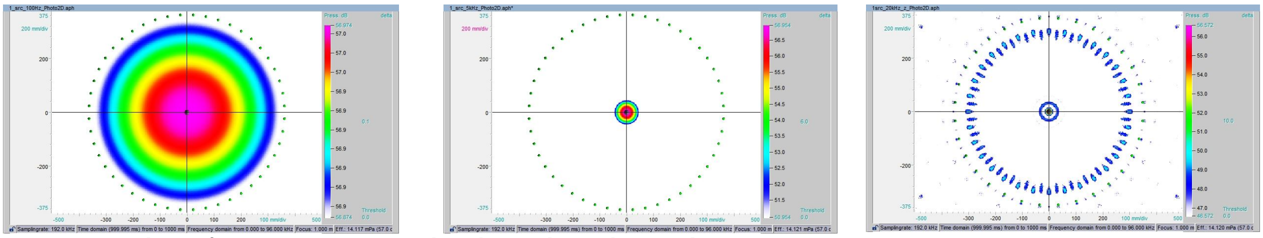

Fig. 1 Acoustic Photos of one Source, 100 Hz (0.1 dB), 5kHz (6 dB), 20kHz (10 dB)

But as the next logical step, our bold tester has the following ingenious idea: Now he takes two sound sources (preferably the worst, cheapest and most mismatched pair of plastic PC speakers he can find), places them at a distance of a few decimetres to each other in front of the array, and repeats the frequency stepping procedure exactly as described above, feeding both speakers with the same sinusoidal signal, maybe at best adding a dedicated phase shift between the two. This time the results surprise him even more: the “beamforming method” fails badly in most of all the visited frequency bands, hence the actual system under test must be no good at all! There is only a very, very limited, very narrow-band region at medium frequencies, where the system is able to deliver results which can be understood and interpreted adequately by the (non array expert) end user. Completely unexpected by the clever tester, of course what actually happens is that the two (always perfectly correlated) sine signals superimpose with each other, and the simple delay-and-sum-beamforming just shows up a virtual single source somewhere in the room between the speakers or even worse it will eventually show much more than two sources at now completely unintelligible positions, depending on array geometry, measurement distance, frequency and phase shift of the two individual true sources and their distance. In fact, no different basic results would be achieved had the two cheap plastic PC speakers been replaced by high end, level and phase matched, ideal monopole sound radiators each emitting a perfect sine.

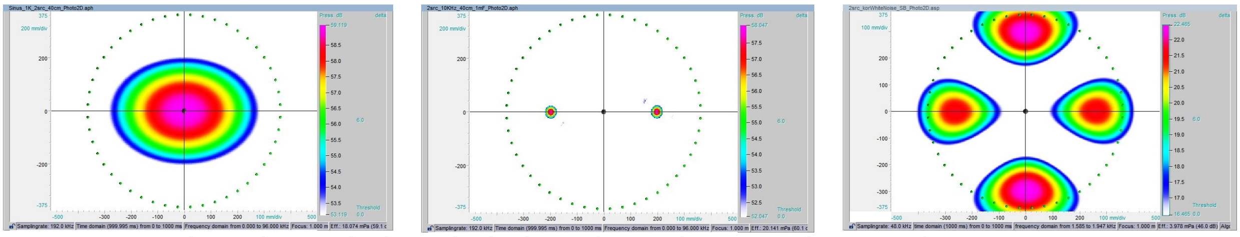

Fig. 2 Two sources: 1 kHz, 10 kHz, correlated narrowband (coherent white noise between 1.585 kHz to 1.947 kHz)

Sharp as a razor, in the sequel two conclusions are drawn by the tester: The beamforming system delivering the best mapping within the limited, narrow frequency band of interest might still be somewhat useful (but only as a narrowband method, as was certainly confirmed by the test results), but even more important: a near field acoustic holography system must be much better, because it can still resolve the two sources reliably. What is completely overseen with such kind of tests is the circumstance that the two speakers and the sinusoidal signal form used are in fact already constituting a mini-system for “sound field synthesis” (even if just a very primitive, stereo one). Unfortunately, it is often not recognized that a beamforming system in such cases shows exactly what has formed up in reality: at the location of the immission, a more or less complicated superposed sound field results, composed of the contributions of the original sources, and a simple delay-and-sum method can of course no longer separate the quantitative contributions of the individual sound sources when those source signals are strongly coherent. What is also often overseen in such “tests” is that with two sine signals of the same frequency, one source signal is always an amplitude scaled and phase shifted version of the other, they are linearly dependant and so inevitably those two signals will always be perfectly “coherent” or “correlated”, even if correlation in the stronger sense is only a statistical concept (meaning average strength of linear dependency between random variables) and thus is not defined for pure sine signals. Using two sinusoids of different frequencies does not change this situation much, because the linear dependency effect is now caused by the resulting sum and difference frequencies of the superimposed signals and the superposition of those frequencies with the original signal, leading to even more confusing results for an uninformed end user. Further irritation of users is caused by the fact that for every given array geometry with a discrete, finite number of microphones there will be some frequency ranges where sources can be well separated, but there will also always be some other frequency ranges where the sources can not be localised at all.

What we are facing here in general is covering a huge problem lying much deeper, namely this problem is rooted in the traditional way of model building in acoustics itself. The single sinusoidal wave or narrowband "wave packets" have established themselves as such a strong and basic concept in the teachings and textbooks, in the measurement practice and in the theoretical calculations of acoustics that there is hardly any reflection on the adequacy of this modelling concept for a given application case at all. While using a single frequency signal seemingly simplifies the testing procedure, especially in the technical fields there is a lot of different, non-tonal signal forms (broadband signals such as noise and pulses) which are not less important. The historical origin of the very early beamforming methods in active, single frequency based sonar and radar applications as well as the widespread use of plane wave models (adequate enough for the true far field case) might have contributed to the still common classification of beamforming as being only a narrowband and farfield method. But also the first successful applications in the field of aeroacoustics and in airplane related research that have later been performed by various renowned research institutions [2] have often still been limited to the localisation of narrowband sources, either determined by the mainly tonal characteristics of the sound sources themselves, but also caused by the limited computing resources that have been available during the very early, formative years of the beamforming array technology. In the result, the acoustics community still considered beamforming mainly as a narrowband method. Also the more recent deconvolution methods that have been developed within the aeroacoustics community (e.g. CLEAN-SC, DAMAS etc.) for the better quantification of sound sources of turbines, airplanes and in wind tunnels are at their basic level only "single line" frequency domain deconvolution methods which can be applied for a broader frequency range only with considerable additional numerical afford. Especially for DAMAS [3] and related methods, the computational demand in the general broadband case is very high, and the solutions to the ill posed inversion problem are often not unique.

2.2 A different point of view – beamforming seen as a broadband method

Contrary to the widely adopted "narrow band view" described above, it is our opinion that beamforming is ***not*** a method for the investigation of sole frequencies or of narrow frequency bands [4]. Rather, the opposite is the case: especially where the modelling assumptions for acoustic waves are very appropriate, the delay-and-sum beamforming approach must inevitably fail. Vice versa, the localisation of a a "wave" is only possible, if the wave model assumption fails. In this case, the so called "wave" is not of an infinite extent any longer, it now has a beginning and an end, which we want to locate and which can easily be found. So, the best results with the beamforming approach are achieved in the broadband case.

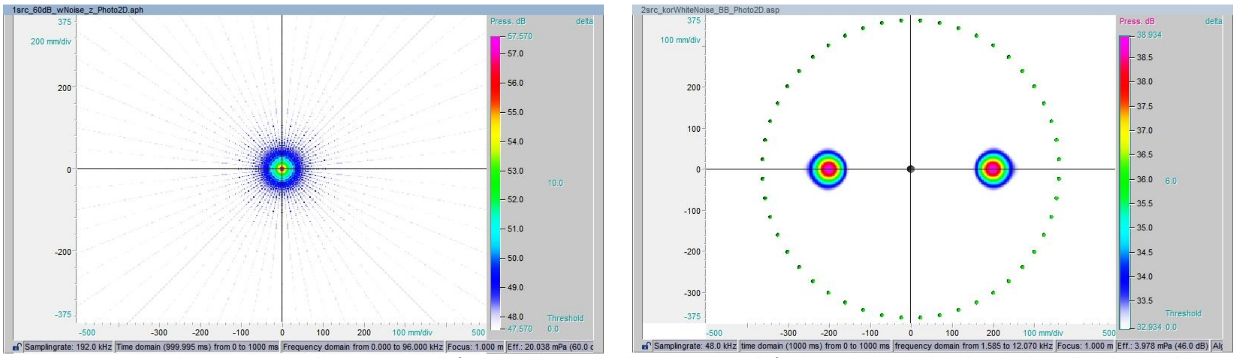

Fig. 3 One source, white noise, time domain (over all frequencies), 10 dB dynamic range (left), 2 sources, correlated noise, broadband 1.585 kHz to 12.070 kHz (right)

In the general case of broadband signals, the model assumption of a “wave” does not lead to a simplification of the localization task. E.g. if the signal is a single pulse in the time domain, then this pulse can only be described in the frequency domain by a packet of nearly infinitely many individual waves of different frequencies. The transformation (e.g. a discrete Fourier transform) into a system of orthogonal sinusoidal base functions does not simplify our task in this case but rather makes it much more complicated. In time domain beamforming, no longer phase differences but instead relative run time differences are determined, which is possible very precisely and leads to a very good spatial source localization. The detailed array geometry is then of much less importance for the overall dynamic and for the quality of the mapping, whereas also a large number of broadband sources can be localized. The quality of the broadband mappings will increase the more the smaller the signal correlation between the sources is and the more pulse-like the signal form of those sources is.

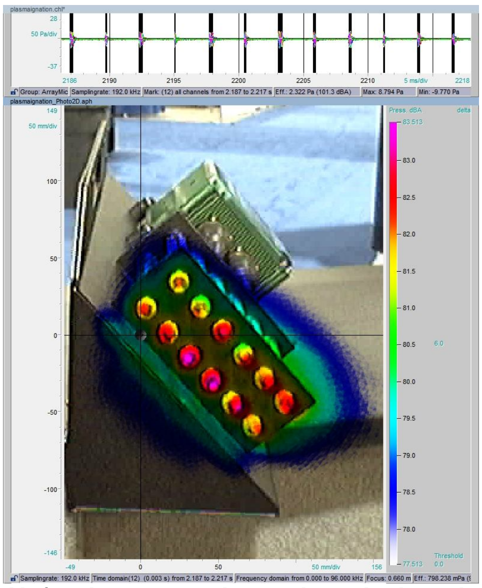

Fig. 4 Field of 12 plasma ignition plugs, single impulse shorter than 100 µs

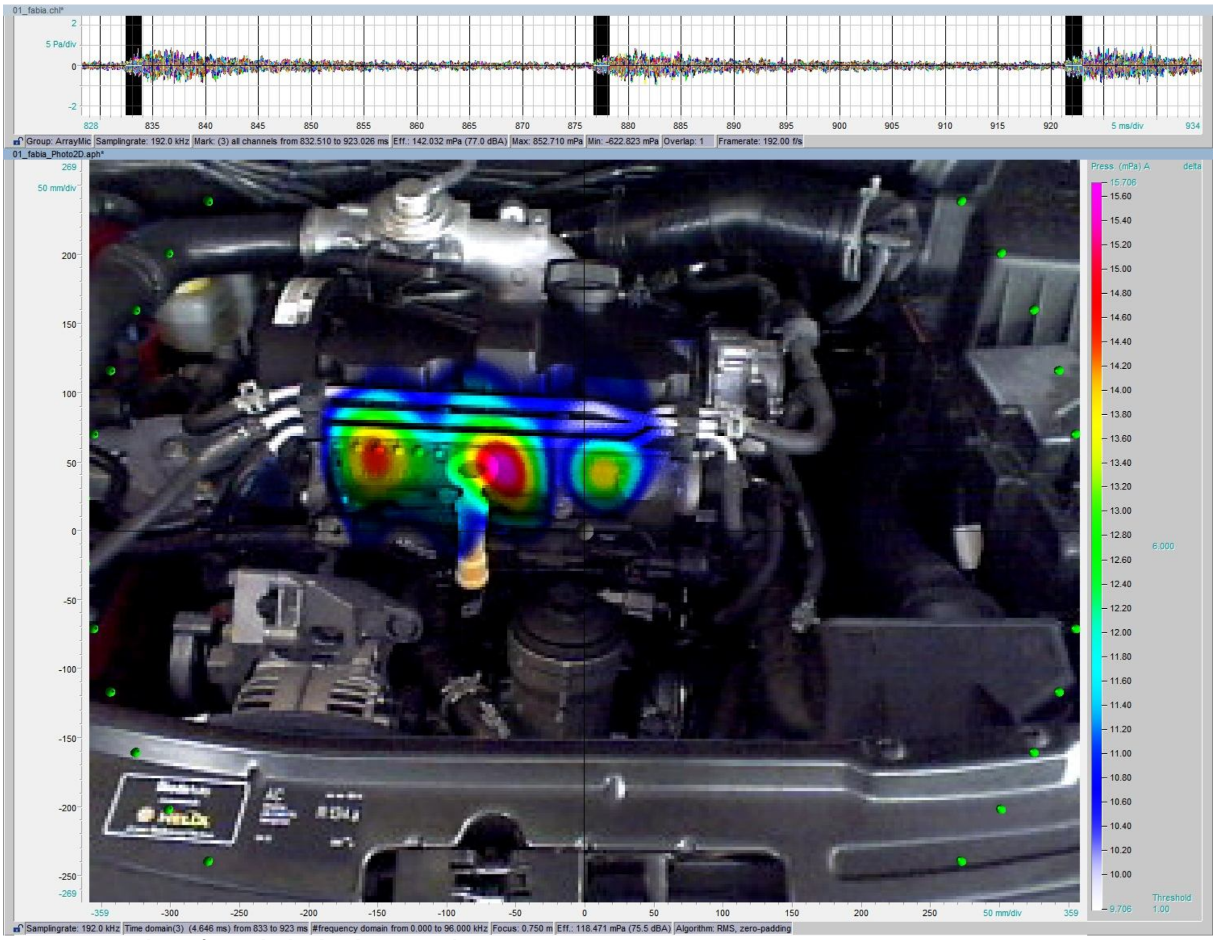

Fig. 5 Acoustic Photo of a 3-cylinder diesel engine at the ignition moment

The only conclusion we can draw from this is that acoustic array beamforming is a broadband solution. In fact, this confirms with all our practical experiences over more than a decade with both time domain and frequency domain beamforming. The true strength of the beamforming method lies in the localization and analysis of broadband, pulse-like sources. This perception allows to show up a possible way for the general evaluation of beamforming systems.

3 PROPOSAL FOR THE EVALUATION OF BEAMFORMING SYSTEMS

The practical testing of beamforming systems should not be performed at the sharp edge of the method to its very failing (narrow band case), but instead it should be performed within a region where the strengths and advances of the method can be evaluated more objectively (broadband case). Surely, frequency limits and restrictions can still be (and should still be) analyzed, but because those limits are mainly determined by the detailed channel number and array geometry as well as the (maybe rather narrow) frequency band of interest, it is also always possible to design a special case optimum narrow band array configuration for a specific narrow band application or for a specific end user. On the other hand, good array geometries for broader frequency bands already exist such as spiral geometries which can also be designed according to a users needs.

In the following, some proposals for methods and useful quantities for a better comparison and evaluation of beamforming array systems are given.

Test of level accuracy and dynamic within the acoustic map

- measurement of a nearly point-like source (no speakers allowed!) with a broadband-signal and defined level (e.g. a reference sound power source according to ISO 3741, ISO 3747 and/or ISO 6926) and comparison of the map results with the values of a sound level meter at the position of the array center under good freefield conditions (anechoic chamber)
- position of the source is determined in the map center and near the map edges, the results should be within the accuracy range of level meters of a given accuracy class (averaging of time stable and calibrated array microphones is expected to give a good level information)

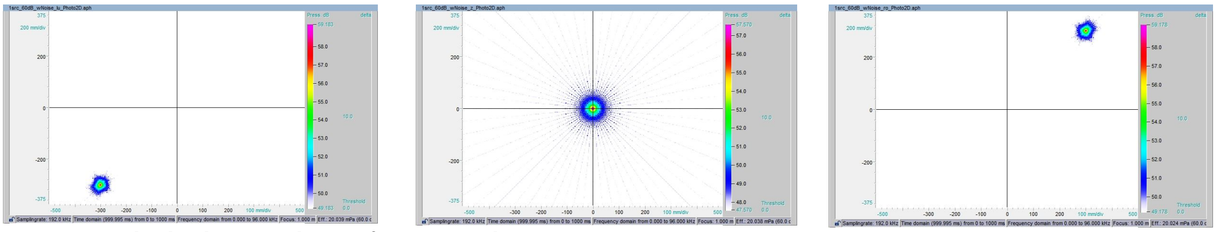

Fig. 6 Position, level and mapping dynamic for 1 source (white noise)

- measurement with two point-like broadband reference sources, level and position of the sources are to be determined and visualized according to a quality grade which is to be defined for this purpose in future standards

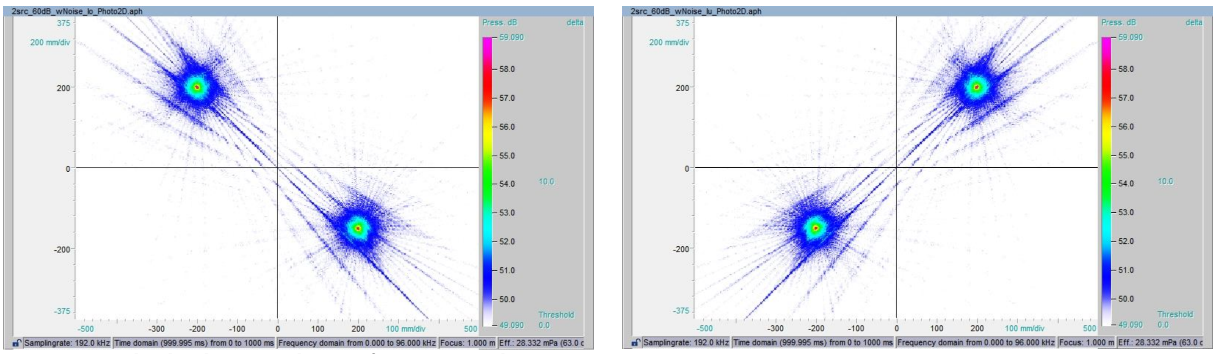

Fig. 7 Position, level and mapping dynamic for 2 sources (white noise)

Test if the position accuracy is independent of the source position in the map

- sources are point-like non correlated broadband noise sources and/or short time pulses
- those sources are arranged in a plane at defined locations (e.g. a square pattern of 13 x 13 source points)

- image defects/aberrations can easily be determined and documented from knowledge of the exact positions

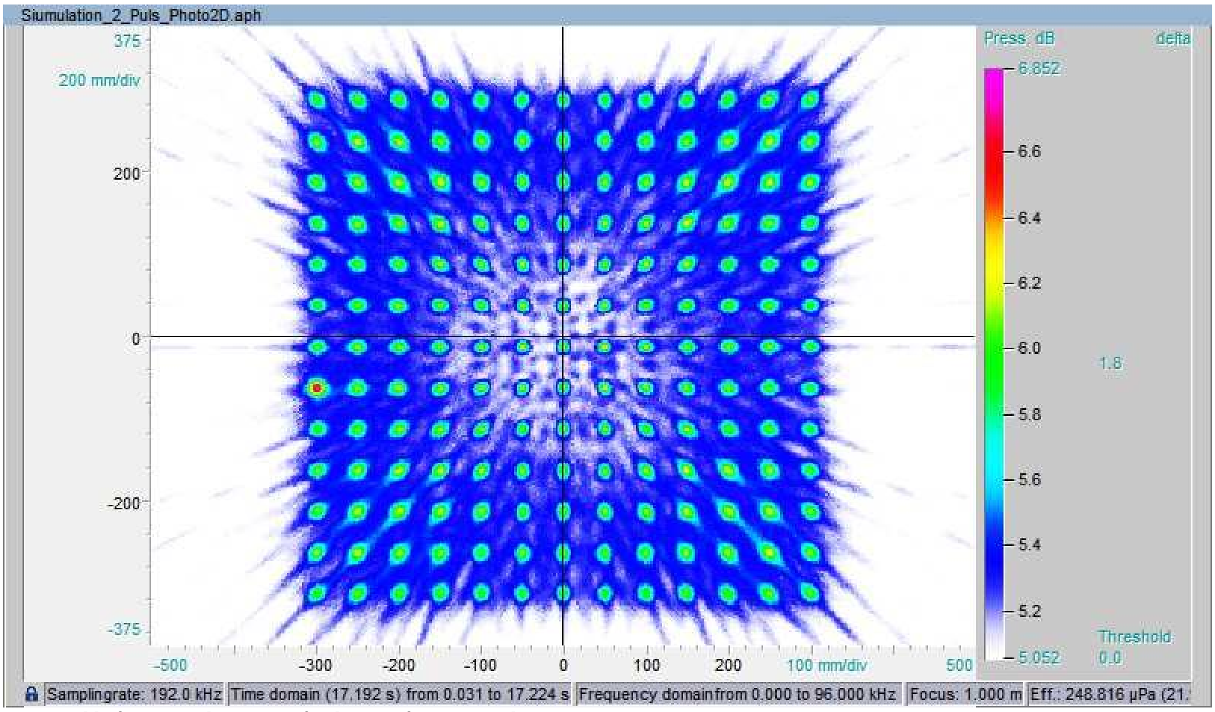

Fig. 8 Field of 169 impulse sources for the test of position and mapping error

- perform the same test for different distances of the array to the sources within the plane
- perform same testing for varying distances between the source positions
- this will lead to the appearance of grating lobes at a certain dynamic range (e.g. 3 dB or 6 dB) and at a certain upper limit frequency

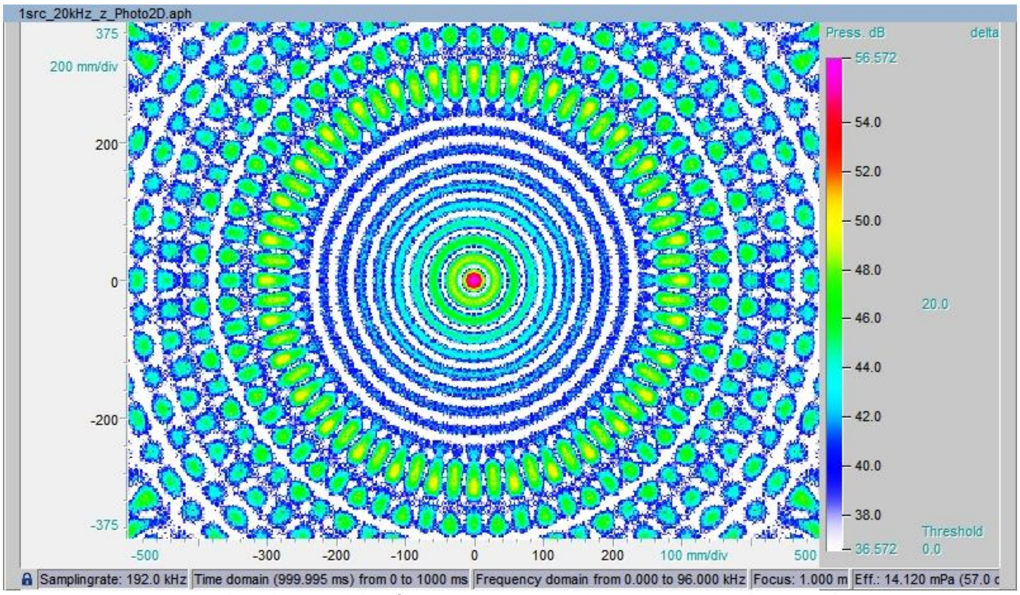

Fig. 9 Side and grating lobes for a ring array, 75cm diameter, 48 microphones, 20 kHz

- a lower frequency limit may be determined for a given dynamic range in the mapping (e.g. 3 dB or 6 dB)

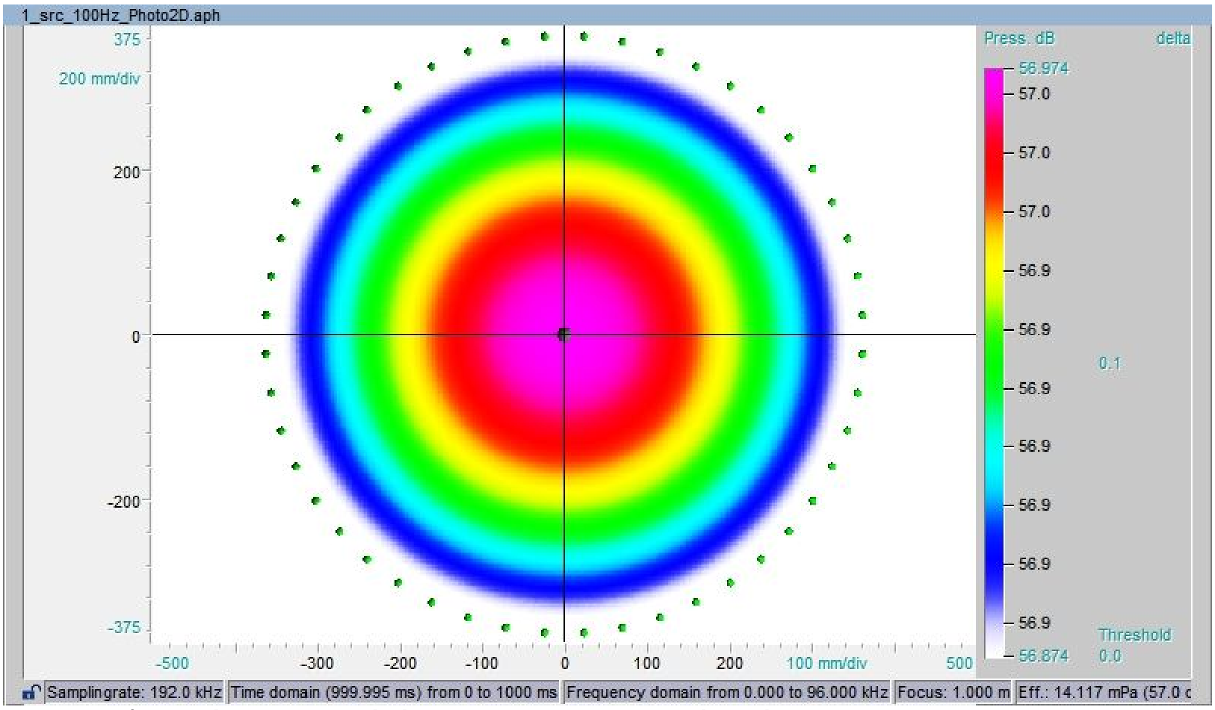

Fig. 10 Lower frequency limit, 100 Hz, 0.1 dB dynamic range, ring array 48 microphones, diameter 75cm

Test of geometric resolution

- determine the minimum distance between two non-correlated broadband noise sources which can be resolved, this also depends on an appropriate resolution criterion (Rayleigh, 3 dB main lobe overlap or something else to be defined), on the sampling rate and on image resolution (how many "acoustic pixels" within the map are to be used)

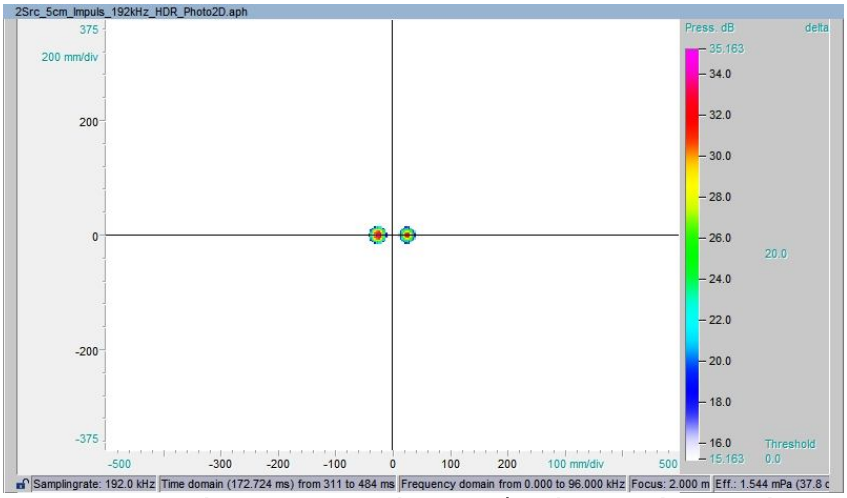

Fig. 11 Geometric resolution, 2 sources, 5cm distance, 2m focus distance, 20 dB mapping dynamic

The quality of beamforming systems tested in this way will depend on:

- the quality of the array microphones used
- the size and geometry of the array
- the precision of the microphone positions
- the sampling frequency of the data recording equipment
- the performance of the implemented beamforming signal processing (spatial resolution, aliasing suppression, integration of advanced deconvolution methods, error corrections for varying focus distances, other advanced signal post processing in the time or frequency domain)

4 CONCLUSIONS

The paper gives a critical review of the classical, “narrow-band” evaluation approach which is still widely applied in the practical comparison of beamforming array systems. It is shown that the widespread use of sinusoidal test signals which is appropriate in many other fields of acoustics will badly mislead array designers and end users if an objective evaluation of an individual beamforming system is the goal. The reason is that two monofrequent test signals of the same frequency are always perfectly linearly dependent (in case of two different frequencies the sum and difference terms also lead to linear dependence), and this strong signal coherence thus will forcibly drive the simple delay and sum beamforming method to its very edge of failing instead of objectively evaluating its true performance potential. Therefore, a different approach is needed which should basically consider beamforming as a broadband method instead. Based on this insight, some proposals for a better evaluation approach for array beamforming systems are given which may serve as a basis for more systematic and generally accepted testing procedures as well as for a possible future standardization of microphone array beamforming methods.

ACKNOWLEDGEMENT

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