TIME-DOMAIN BEAMFORMING USING ZEROPADDING

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ABSTRACT

Zeropadding is an approved method in signal processing to get additional samples or to suppress undesirable signals without changing essential signal properties. Beamforming in the frequency domain also uses zeropadding to transform the time signals into the frequency domain.

The advantage of beamforming in the time domain compared with beamforming in frequency domain is the providing of high time resolution additional to positioning resolution (acoustical movie). But by use of high resolution in the time domain in combination with non stationary, impulse containing signals the delay and sum beamformer produces overlaps between acoustic events that are well separated in time domain. The analysis of short and small impulses can be complicate or impossible. An acoustic movie shows this overlays by apparently inversely moving patterns that can hide noise sources completely.

This paper describes these moving patterns as an artifact of the delay and sum algorithm. Some examples show the limitations of spatiotemporally resolution.

An enhanced algorithm using zeropadding suppresses the patterns and improves the spatiotemporally resolution considerably. A comparison of analyses with and without zeropadding shows the advantages of the enhanced algorithm. It enables new areas of applications, e.g. in architectural acoustics.

1 INTRODUCTION

Acoustic movies created by time-domain beamforming are state of the art for some years now. They are very useful to analyze non stationary noise emissions. Although the time signals of microphones are divided in short time intervals, for any interval an acoustic photo is created. Time intervals can also be overlapped, then we have a better time resolution and a longer integration time. By using very short time intervals (100 samples or under), non stationary noise emissions and a planar virtual plane to create acoustic imagines we can observe a remarkable effect. Fig.1 shows the time functions of noise pressure received from a ring array (48 microphones), emitted from a loudspeaker which is centred in a plane in 2 m distance from the ring array. The virtual plane dimensions are 2m x 1.50m. The impulse of a loudspeaker was received from all microphones simultaneously because the distance between source and all microphones is equal. Fig. 2, 3, 4 show the results of the delay and sum beamformer from different time intervals. The length of all time intervals is 150 samples, the start point of the first interval is at sample 34300, the second and third interval are each moved with 50 samples to the left. A movie composed from these frames displays an inversely moving wave front, collapsing in the centre of the picture where the loudspeaker is situated.

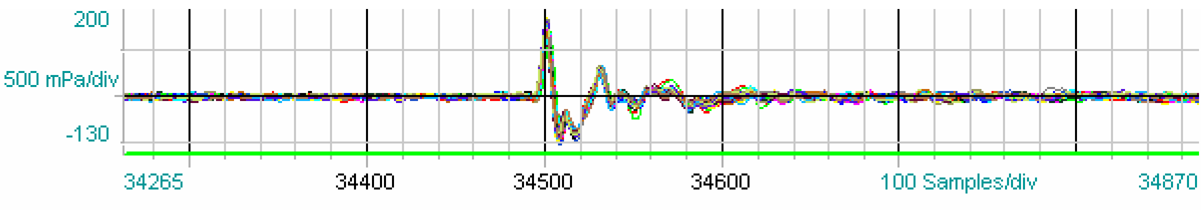


Fig. 1. 48 time functions of noise pressure received from a ring array

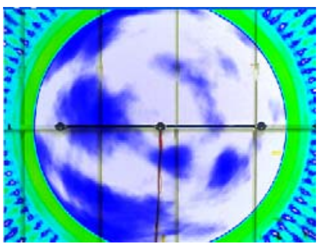


Fig.2. Acoustic photo from time interval 34300-34450 samples

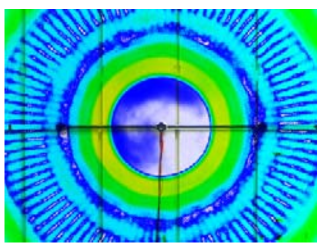


Fig.3. Acoustic photo from time interval 34350-34500 samples

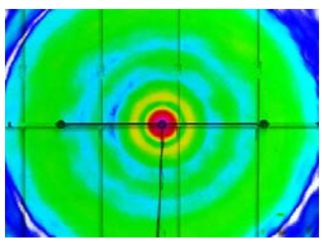
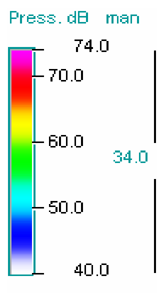


Fig.4. Acoustic photo from time interval 34400-34550 samples



2 INVERSELY MOVING WAVE FRONT - ARTIFACT OF BEAMFORMING

2.1 Causes of inversely moving wave fronts

The cause of this effect is the delay and sum beamforming algorithm. The virtual plane is divided into rows and columns (pixel). The algorithm is calculating the delays for each pixel to each microphone. For the given situation with a ring array and a planar virtual plane, the run time delays between a possible source situated on the border or a corner and the array are greater than the run time delays between a possible source situated in the middle of the virtual plane. To show the effect of this different delays on the calculation with short time intervals, we examine the calculation of sound levels for two points, based on the situation in Fig. 1 and Fig. 2 and the marked time interval in Fig. 5.

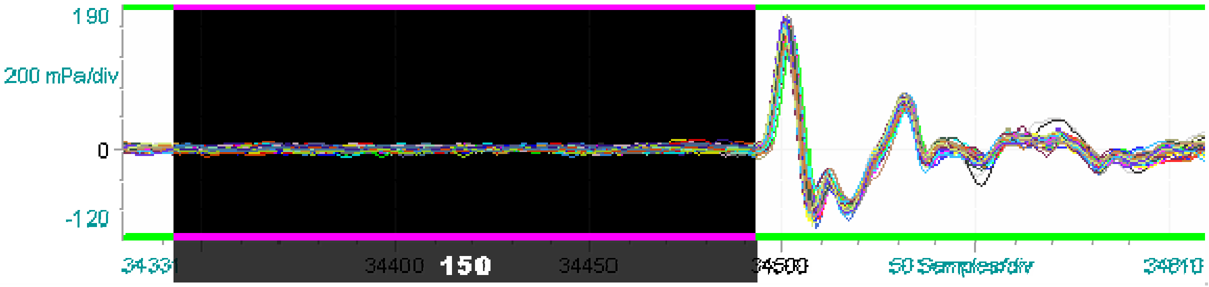


Fig. 5. 48 time functions of noise pressure received from a ring array, a time interval of 150 samples is marked in black

At first, we calculate the sound level for a pixel located in the middle of the plane where also the source is situated (Fig. 6). The algorithm is moving each time function of each microphone reversely with the corresponding delays (Fig. 7), adds all time functions and divides by number of channels (Fig. 8). For the pixel in the middle of the plane all delays are small and equal. The result is the correct superposition for the source in the middle of the plane, but the impulse sent by the loudspeaker is outside of the marked time interval. The level of this pixel (root mean square value of beamformer result function from Fig. 8) is low (30 dB) and displays the correlated background noise.

At second, the algorithm calculates the sound level for a pixel located on the bottom - right border of the plane for the same time interval. There is no sound source, but the delays for this location are very long and different. The received time functions are moved with the corresponding delays to compensate the run time delays of a possible sound source on this location (Fig. 8). Thereby, the impulse sent by the loudspeaker is moved into the marked time interval. There is no correlation, because the real source isn't located on the upper-right border, but the sound level of superposition of these non correlated signals is higher (50 dB) than the superposition of the correlated signals from a point in the middle (Fig. 9).

This situation is equal for all points that are calculated around the ring array and the reason for creating a circular "wave front". If the marked time interval in Fig. 5 moves to the right, the "wave front" is inversely moving and "collapsing" into the source. An absolute circular "wave front" is typical for using ring arrays, a planar virtual plane and point sources. But we can observe similar results by using other arrays, e.g. sphere arrays, spirals or star arrays. The "wave front" is then divided in a different pattern, but this pattern is also inversely moving and "collapsing" into the source by moving the time interval.

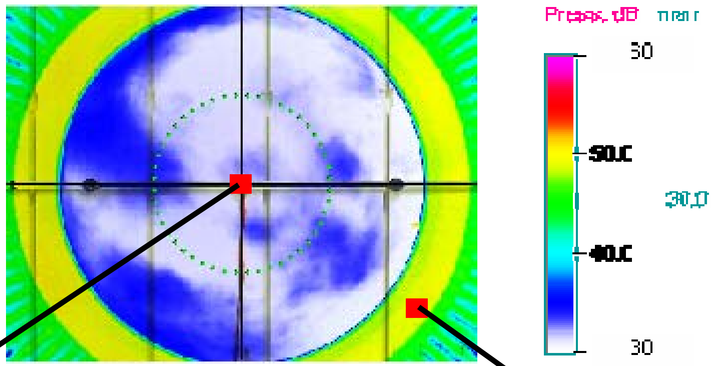


Fig.6. Acoustic photo based on timefunctions and marked interval of Fig. 5. The ring array is drawn by green points

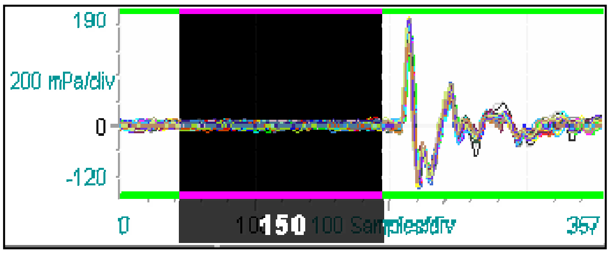


Fig. 7. Delay compensated time funtions for center point

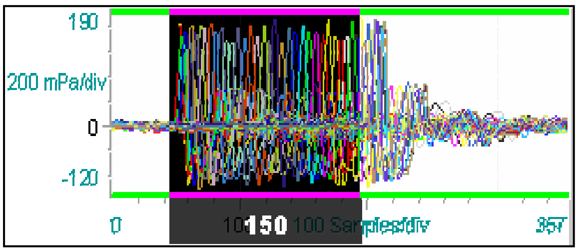


Fig. 9. Delay compensated time funtions for a point situated on border right

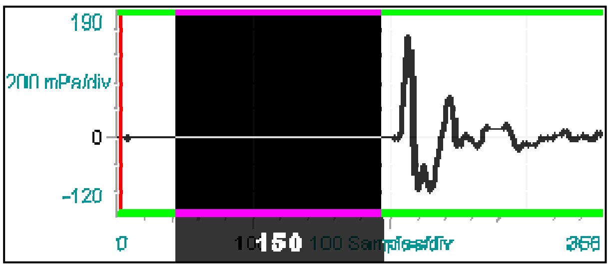


Fig. 8. Sum- and delay beamformer result for center point

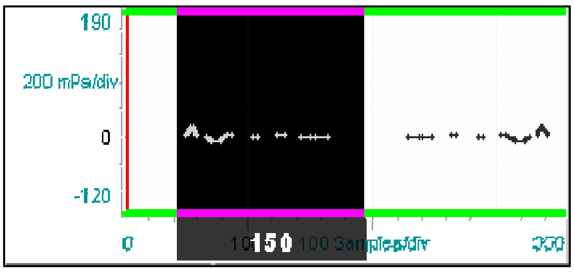


Fig. 10. Sum- and delay beamformer result for a point situated on border right

2.2 Problems and disadvantages

Apparently inversely moving wave fronts are firstly an irritation in the analyses of non stationary sources. On the other hand - in an acoustic movie they are collapsing at the position where the real source is situated, and a source localization is possible.

In many cases of acoustic analyses we have a situation like that shown in Fig. 11. An acoustic emission from an object, e.g. a glass, is initiated by an impact. The emission is measured with a ring array and can be divided into two sections: the first time interval represents the immediate emission by the impact, the second time interval shows the acoustic emission from the stimulated object. The second emission has a lot of more power than the first emission, but to locate the impact we have to analyse the first emission only. With a traditional beamforming method, the detection of the location of this impact is impossible. The result of this experiment is shown in Fig. 16 - a circular "wave front". Due to the reverse delays, the signals of the more powerful second emission are moved into the marked time interval and overlay the low initial emission.

2.3 Problem solving with zeropadding

To avoid these problems, we apply the zeropadding method to time functions. At first, we cut the time interval of the interesting signals (Fig. 11). Then we calculate the maximum time delay which is expected in the scenery (dimensions of virtual plane versus positions of microphones). We can use this maximum time delay to calculate the number of samples that is an equivalent for the maximum shift operation by the beamforming algorithm. We extend all cut signals with this number of samples and set the values of those samples to zero (zeropadding, Fig. 12), so we have enough signal length to perform the beamformer operation. Only the signal parts in the marked time interval will go into the beamformer, outside parts of signals are replaced with zeros (Fig. 13).

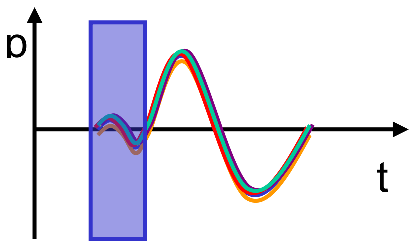


Fig. 11.Cut the interesting time interval

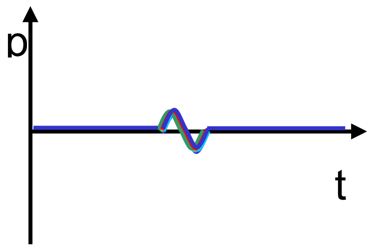


Fig. 12.Extend the interesting time interval according the maximum delay and fill with zeros

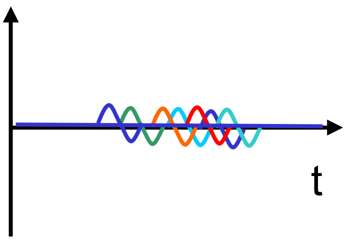


Fig. 13.Perform the beamformer

2.4 Limitations

The described method can lead to masking of other sources within the image field. This is especially the case when the relative delays between the individual points to calculate are large with respect to the delays between the microphones. An improvement is only possible by leaving the common time base. For each image point, beginning from a common start time (beginning of marked section) only the relative delays between the microphones are used, the run time between this image point and the array is dropped. But this method will lose the time relations between different events at different locations. Thus, "sharpening" the spatial resolution leads to a "blurring" in the time domain and vice versa.

2.5 Examples

First example for using zeropadding is the analysis of noise emissions of striking a glass. Fig. 12 shows the time functions, received by a ring array (diameter 0.36 m) Fig. 13, 14 and 15 show the results from several extremely short time intervals (length: 100 µs) using zeropadding, Fig. 16 shows the same analysis without using zeropadding. The use of zeropadding can not only localize the impact, but also shows different oscillation modes of the glass. Without zeropadding we get no utilizable results.

Fig.13 (under). Sound pressure received from a ring array (36 microfones, diameter 0.35m), emitted by striking a glass

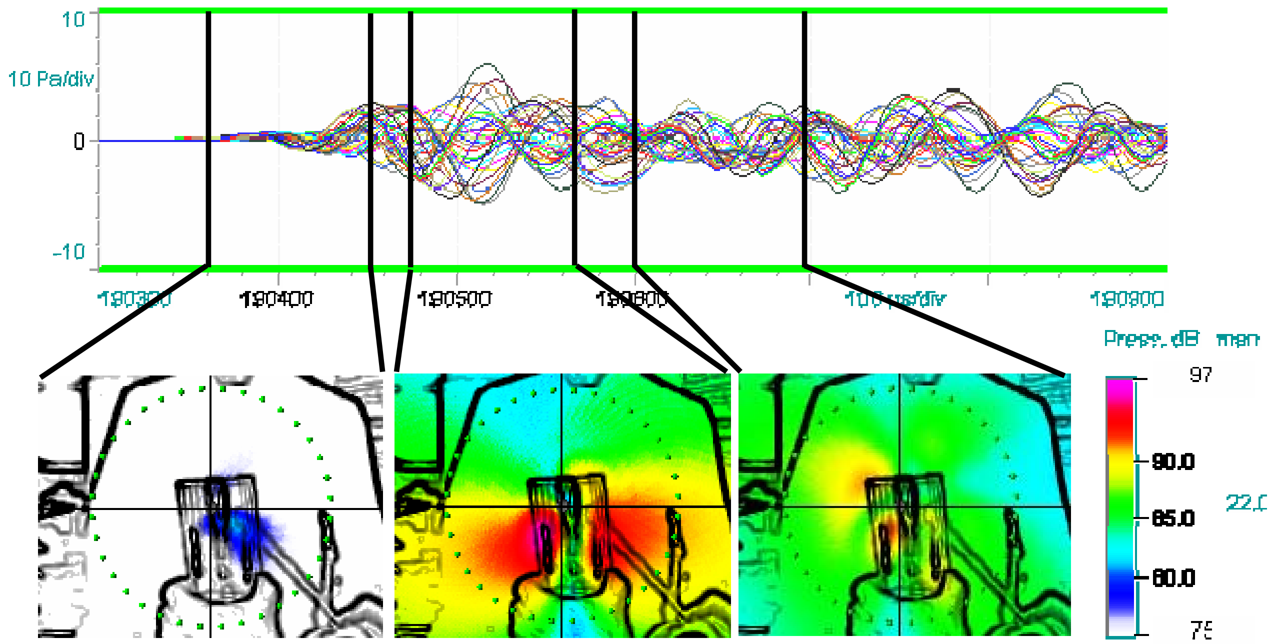


Fig. 13. Zeropadding: Strike localization

Fig. 14. Zeropadding: Oscillation mode 1

Fig. 15.Zeropadding: Oscillation mode 2

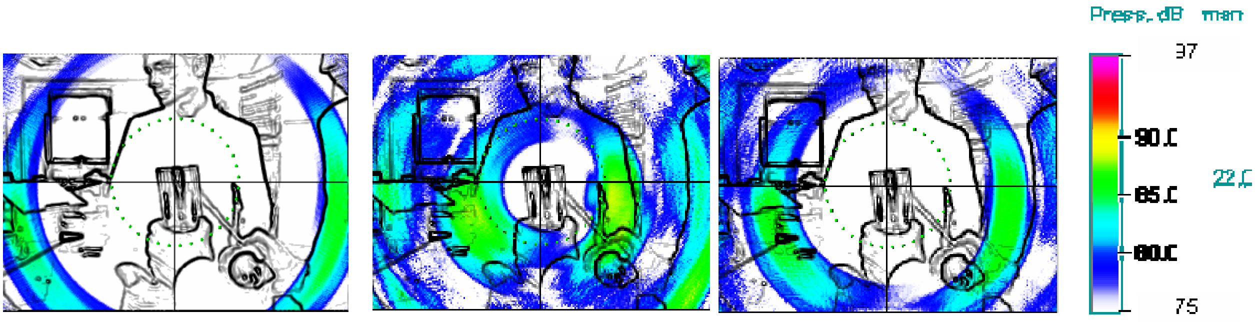


Fig.16. Without zeropadding. Same time intervals as Fig. 13,14,15

Second example shows a 3D analysis of a room where the right wall is stimulated with a hammer from outside. The localization of the impact is only possible with zeropadding. By 3D-analysis in halls or rooms we have three conditions that strengthen the problems described in section 2.2: At first, the direct emission of the impact on the wall is very short, because the velocity of sound in massive walls is higher than the velocity of sound in air. Therefore, we receive the direct emission of the impact only for a very short time, after that we receive the emissions from the stimulated wall. To locate the impact with beamforming we have to calculate a very short time interval (Fig. 17). Second, in many cases emissions of a stimulated wall are more powerful than the direct emissions of an impact. Beamforming without zeropadding is "smearing" the powerful emissions of the stimulated wall with the emissions of the impact - the impact is not shown in an acoustic photo. Thirdly, in rooms and halls the delays may be very long - more and more parts of signals outside of the marked time interval will be introduced into the beamformer calculation.

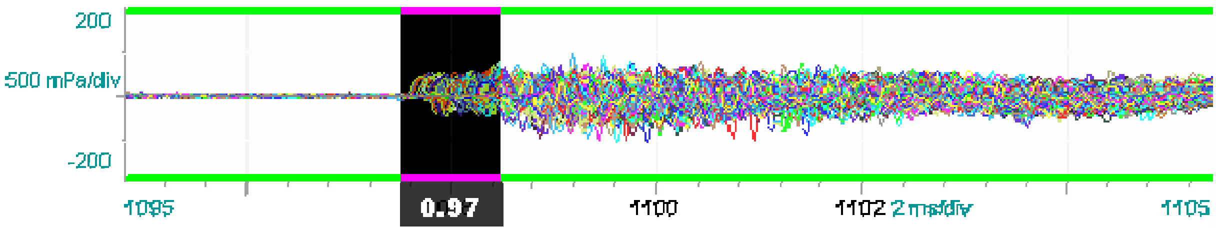


Fig. 17. Sound pressure received from a sphere array, 120 microphones,diameter 0.7m, situated in a room. Short time interval of impact emission is marked

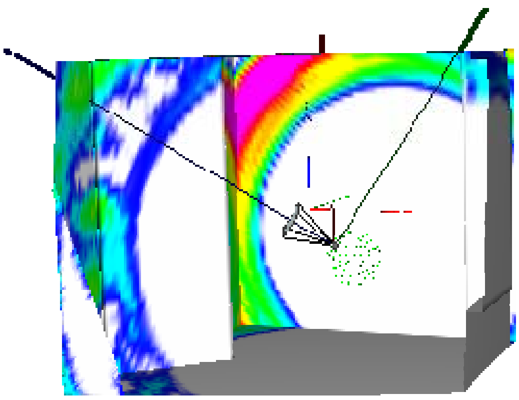


Fig.18. Beamformer result without zeropadding

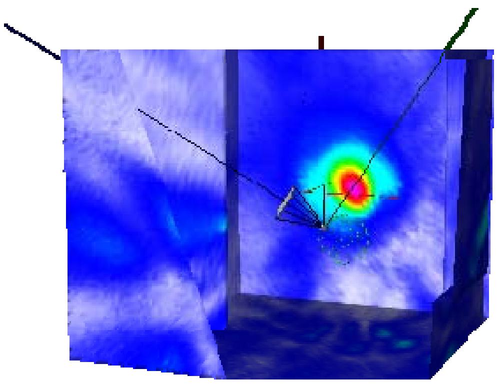


Fig.18. Beamformer result using zeropadding

3 CONCLUSIONS

Localization of nonstationary, transient signals using a delay and sum beamformer needs calculation of sound pressure values of different locations for very short time intervals. Depending on the geometry of room and array, the time delays the beamformer uses for the „backward“ delays in its calculations may become larger than the time interval to analyze itself. This is especially the case when the sound travel times from locations to calculate to the individual microphones are extremely different. Result is a “smearing” in the time domain, because the shifting of the signals with respect to each other signal parts can come to computation that do not belong to the time interval that should have been analyzed. This blurring in the time domain shows in acoustic movies as seemingly inversely moving wave fronts, that can mask quieter sources (e.g. impact sources) completely. These artefacts are not a property of a so called interference reconstruction [3], instead they are caused by the used delay and sum beamforming algorithm itself. Through the time domain “blurring” the analysis is restricted to time intervals which are at least as long as the wavelength to analyze plus the maximum time delay appearing in the scene.
A simple solution to this problem can be provided by zeropadding. The time interval that can be analyzed is then only limited by the wavelength of the signals under concern, disturbing inversely and inwards moving “wavefronts” disappear completely. This opens up new application fields, e.g. in room and building acoustics, in analyzing transient signals like impacts or of complex, fast moving mechanics.

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