30 May - 1 June 2006 Tampere, Finland

TRANSIENT NOISE SOURCE LOCALIZATION

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ABSTRACT

Conventional beamforming systems for noise source localization are mostly based on frequency domain calculations. Additionally, narrowband assumptions concerning signal characteristics are often made to simplify the calculations, i.e. to be able to express a single time delay as a sole phase shift in the frequency domain as well. If necessary, a broadband system is then modeled as a composition of many narrowband ones. This traditional way is completely dominated by the classical acoustician's „harmonical signal superposition" thinking.

Especially transient signals are not narrowband at all, but often of high interest in practical applications. Unfortunately, they need extraordinary many superposed harmonics to be adequately modeled in the Fourier domain. Conversely, the algorithm used in the acoustic camera software works entirely in the time domain, so it is inherently a broadband approach. It is shown that those time domain calculations make problem solutions much easier for transient signals.

Practical application examples from technical fields up to exotic topics as the localization of bats in biological field research are demonstrated. It is also pointed out that it is possible to undersample in the spatial domain if there are enough signal bandwidth and appropriate time domain oversampling.

1 INTRODUCTION

During the past few years, a new technology for visualising sound sources ("acoustic camera") has gained practical importance in real-world applications for fast localization of the dominant noise sources of machines and equipment. In the farfield approach, such systems are based on the well known "delay-and-sum"-beamforming method. A microphone array will be successively focused to the individual points on a measurement plane by compensating for the relative runtime delays between the microphone channels and adding up the shifted time signals coherently. Normalization by the channel number gives a reconstructed time function for which interesting parameters as the sound pressure level can be determined. This way, a mapping of the sound pressure distribution in the measurement plane of a distant object can be calculated. In present state of the art systems, an automatic overlay of an optical photo or an extracted edge picture of the object with the calculated acoustic colour map is provided.

This paper will give an overview of the advantages and disadvantages of calculating the beamforming algorithm in the time domain and in the frequency domain. Though completely equivalent in mathematical theory, there are certain meaningful differences between both domains from a practical viewpoint. Especially for short-time events the simple time domain formula is often better suited than the frequency domain approach broadly used throughout the traditional array signal processing literature. It will be shown that a high signal bandwidth can generally be beneficial for a successful beamforming.

2 DELAY-AND-SUM-BEAMFORMING

2.1 Calculation in the time domain

The simplest approach, the straightforward calculation of a delay-and-sum-beamformer in the time domain, is also the historically oldest method [1].

In Equation (1), t denotes time, M is the number of microphones in the array, the w_i are (optional) shading weights, the $f_i(t)$ are the recorded time functions of the individual microphones, and the Δ_i are the appropriate relative time delays, calculated from the absolute run times $\tau_i = |\mathbf{r}_i| / c$ by subtracting the minimum over all τ. The symbol c denotes the speed of sound in air and $|\mathbf{r}_i|$ is the geometrical distance between microphone number i and the actually calculated focus point $\mathbf{x}$. In practice, this is equivalent to the use of a spherical wave model which has proven useful because it automatically translates to the model of a plane wave for larger distances. The reconstructed time function at location $\mathbf{x}$ is calculated as:

$$\hat{f}(\mathbf{x},t) = \frac{1}{M}\sum_{i=1}^{M} w_i f_i(t-\Delta_i). \tag{1}$$

Despite its extreme simplicity, the delay-and-sum method has shown its practical usability in a very wide range of acoustic localization and trouble shooting applications.

This direct calculation in the time domain especially offers some often unnoticed advantages for transient signals which have naturally very broad spectra in the frequency domain.

2.2 Calculation in the frequency domain

In traditional acoustics, theoretical considerations are dominated by wave models. In most cases, signals are not directly used in the time domain, but they are regarded as a superposition of harmonic basis functions in the complex frequency domain. This viewpoint has been strengthened in the past by the availability of fast fourier transform algorithms and powerful digital signal processors and computers.

Using the linearity and the shifting theorem of the Fourier transform, equation (1) can of course easily be calculated in the frequency domain as well. Taking the Fourier transforms of the individual microphone signals now yields the reconstructed spectral function:

$$\hat{F}(\mathbf{x},\omega) = \frac{1}{M}\sum_{i=1}^{M} w_i F_i(\omega) \cdot e^{-j\theta_i}. \tag{2}$$

Here, the $\theta_i = \omega\Delta_i(\mathbf{x})$ denote the location-dependent phase angles by which the component of every partial frequency ω of a signal has to be shifted in order to achieve the exact compensation of the relative runtime delays of the individual microphone channels in the superposition of all spectral components according to Eq. (2). The terms $exp(-j\theta_i)$ dependent on focus direction in Eq. (2) are usually summarized into a vector that is therefore called the "steering vector".

2.3 Comparison of both domains

Undoubtedly, at first glance the calculation of a beamformer in the frequency domain offers some advantages. While the localization information is connected via complicated convolutional relations to the individual microphone signals in the time domain, this connection with the focus point information now reduces to a simple complex multiplication in the frequency domain. This decoupling of the microphone signal information and the location dependent information (steering direction) permits the independent use of the complex cross spectral matrix of the microphone signals for an advanced signal processing. This circumstance led to the development of a wide range of specialized beam-forming variants (adaptive beamformers, eigendomain methods etc.) and more sophisticated algorithms described in detail in the array literature, see e.g. [2].

On the other hand, working in the frequency domain has a subtle but obvious disadvantage: Because the frequency domain does not know time informations at all but *only* phase informations of harmonic basis functions (assumed to be infinitely long in theory), the calculation of Eq. (2) at every focus point will be very elaborate and tedious work especially in the case of broadband signals. Even for a single constant time shift Δ_i, the neccessary phase rotation θ_i will be a different one for every individual frequency component, thus enforcing the calculation of another phase angle for every individual spectral line separately.

At the times when the basic theory of beamforming has already been worked out, the then available computing power and memory sizes did not yet allow the fast calculation of such amounts of data needed for true broadband systems. This and the fact that early beamformers were strongly connected with active localization systems led to a strong dominance of narrowband beamforming algorithms throughout the classic array literature. Only in case of a single frequency component it is correct to regard a constant time shift as a sole constant phase shift as well. But in the sequel, the restriction to the narrowband case in the spectral domain (which was just a matter of normal historical development and not of any theoretical requirements) caused an unneccessary blocking of the acoustician's view onto the potential advantages of a direct implementation of Eq. (1) in the time domain.

But the calculation in the time domain basically offers great virtues for signals that are broadband anyway, as are the most sounds of technical machinery which only very seldom are composed of just a few tonal components but in reality are rather to be considered as a mixture of broadband noise and many different tonal signals. The same holds for short pulses and transient signals of any kind which are generally not very difficult to localize but exhibit very broad spectra which makes spectral domain calculations expensive again. Especially for transients, the calculation approach in the frequency domain unneccessarily transforms a quite simple problem to a complicated one. Transient noise source localization is a quick and easy task in the time domain.

The pros and cons of the calculation of the simple delay-and-sum-beamforming method in each domain are summarized from a practical viewpoint and resulting from our own measurement experiences in Table 1.

Table 1. Advantages and disadvantages of the calculation of the beamforming method in the time domain and in the frequency domain.

Property, Feature	Applicability of BF in time domain	Applicability of BF in frequency domain
signals: periodic, sinusoidal, narrowband, strongly tonal	bad (aliasing problems)	bad (aliasing problems)
signals: narrowband noise	mainly bad	mainly bad
signals: broadband noise, statistically stationary	very good	very good
signals: pulses, transients, statistically instationary	very good	mainly bad
short calculation times possible	good	not so good
applicability of advanced correction algorithms	not so good	very good

The Table 1 clearly shows: The beamforming method is, independent of the domain the calculations are performed in, generally not very well suited for narrowband signals at all. This is also the reason why classical third octave band analysis in the acoustic lab always will relegate the method to the fringe of failing instead of making visible its true potential capabilities. In practice, beamforming should be regarded as a broadband method, and in many cases, the indirection via the frequency domain is not needed at all for mere localization purposes.

3 APPLICATION EXAMPLES

3.1 Plasma ignition field

In Fig. 1, the time domain signal of a plasma ignition field as used in modern electronic combustion control systems is shown as an example of a technical signal exhibiting extremely short pulses.

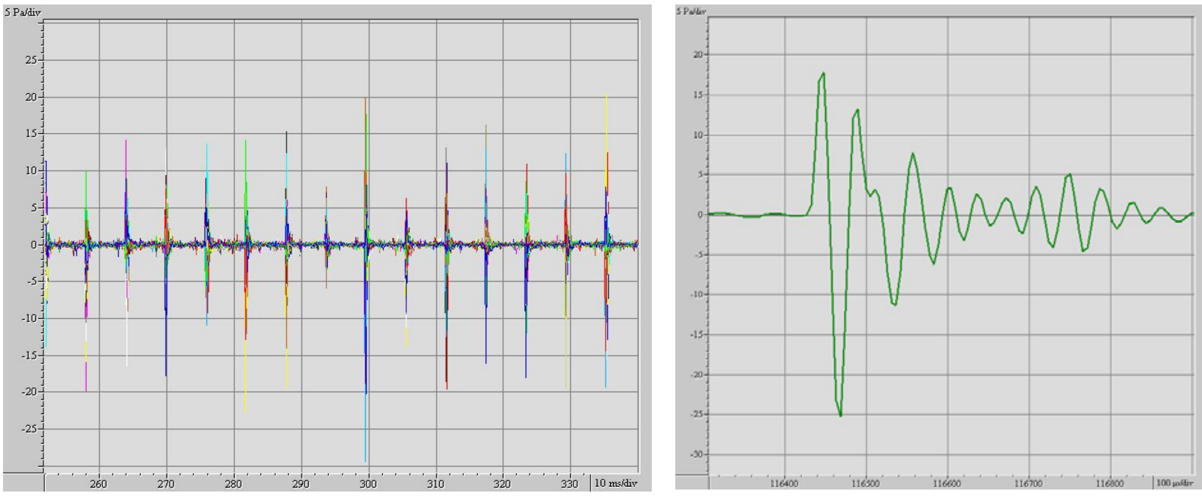

Fig. 1. Consecutive pulses of a plasma ignition field. Time spacing between pulses is 6 ms (left), but the duration of a single pulse is less than 100μs (right). The following peaks in the right picture are already reflections at the mounting plate of the individual sparking plugs.

In Fig. 2, the beamforming result of the time domain calculation is shown as an acoustic photo. The array used was our standard ring array with 32 microphones and with a diameter of 72,5 cm. No filtering was applied. The measurement distance was 66 cm, and the sampling rate for every microphone channel was 192.000 samples per second. Image contrast is 6 dB.

3.2 Localization of a flying bat

In Fig. 3, an acoustic photo of a flying bat is shown as an example for a non-technical application. The signal was bandpass-filtered between 25 kHz and 50 kHz before time domain calculation. Measurement distance was about 4 m, the array used and all other parameters were as in the technical example shown above.

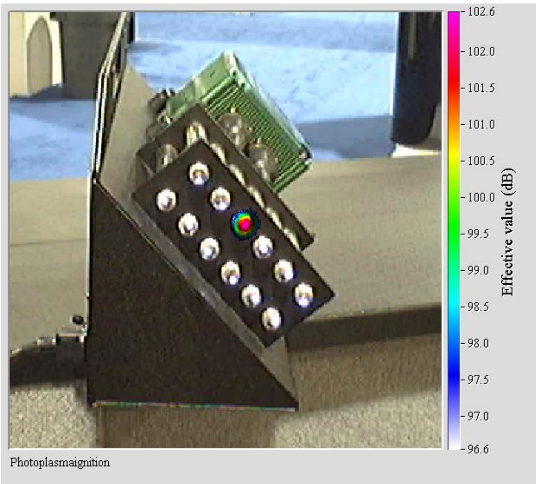

Fig. 2. Localization of an individual pulse of a plasma ignition field (Woodward company) with the acoustic camera.Contrast is 6 dB.

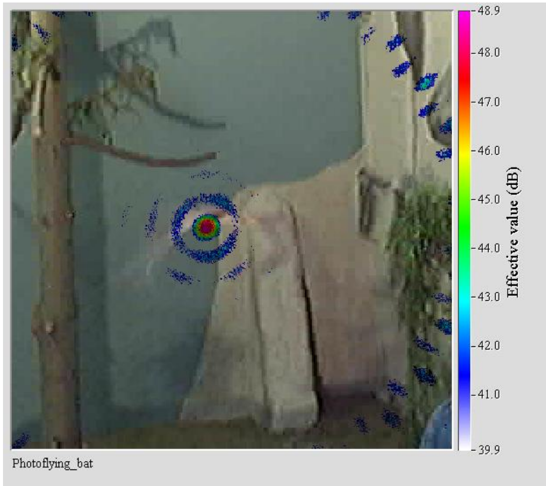

Fig. 3. Acoustic photo of a flying bat.Contrast here is 9 dB and clearly HF-Aliasing is visible, but localization is possible without problems.

4 CONCLUSIONS

Transient noise source localization is not a difficult task per se. For such signals, our practical measurement experiences show clear advantages of using the very simple but effective time domain delay-and-sum beamforming method over the traditionally preferred frequency domain calculations. The frequency domain beamformers have the same problems with very short-time signals as do the classical spectrograms: When the time resolution is fine enough to detect the signals precisely, in many cases the frequency resolution may already be quite bad. Windowing, overlapping and averaging are only of limited value here.

The reason for the difficulties of the frequency domain with very short pulses is clear. The wave model is just not an appropriate thinking approach in such cases. While transient signals may be described with very few parameters in the time domain, many parameters (nearly infinitely many harmonic basis functions) are neccessary to describe the same signal in the frequency domain. Contrary, even acoustic movies with high time resolution can be readily calculated in the time domain.

REFERENCES

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[2] H. van Trees: *Optimum ArrayProcessing.* J. Wiley & Sons, 2002.