

On the Physics of Imaging Reconstruction for Acoustic Images and Films in the Time Domain¹

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Introduction

An analysis of interference systems [H93] led to what has been the best-kept secret in the field of Acoustic Photo- and Cinematography [H05] since 1993 [H942]: the *Interference Transformation* (also named as *Mask Algorithm*), which is presented here for the first time.

In addition to receiving the *Otto von Guericke Prize* in 2001 and the *Berlin-Brandenburg Innovation Award* in 2003, the development of the Acoustic Camera [AK1] was nominated for the *German Future Prize* (Deutscher Zukunftspreis) in 2005.

The terms "Acoustic Camera" and "Acoustic Photo- and Cinematography" both originated from this development. Today, acoustic cameras are used extensively throughout the automotive industry worldwide.

Signal Delay

Sound waves are subject to a time delay as they travel between a sound source i and a microphone k . In the case of an acoustic camera (involving sound pressure levels of up to 130 dB in the far field), it is sufficient to assume a linear relationship defined by the velocity v between the distance r_{ik} and the propagation time τ_{ik} . By employing a Cartesian coordinate system, this propagation time can be determined with sufficient accuracy using the pythagorean distance formula.

$$(1) \quad \tau_{ik} = \frac{r_{ik}}{v} = \frac{1}{v} \sqrt{(x_i - x_k)^2 + (y_i - y_k)^2 + (z_i - z_k)^2}$$

In other words, every location i in the image field has a specific set of n delay times - corresponding to n microphones (indexed by k) - that uniquely characterize it. We shall refer to these n delay times as a delay vector (a column or row matrix) or as the mask M_i for location i (image point i) $M_i = (\tau_{i1}, \dots, \tau_{in})$, see Fig.1.

For a typical acoustic image for example with a resolution of 200×140 pixels, 28,000 delays - multiplied by the number of microphones - are used for one image.

Considered a standard delay element $+\tau$. It shifts the argument of a time function by a delay of $+\tau$, resulting in a time function $f(t-\tau)$, and vice versa,

$$(2) \quad f(t+\tau) \rightarrow \boxed{+\tau} \rightarrow f(t) \rightarrow \boxed{+\tau} \rightarrow f(t-\tau)$$

At a microphone k , the source signal from a point-like sound source f_{ik} consequently arrives as a delayed time function $f_{ik}(t-\tau_{ik})$.

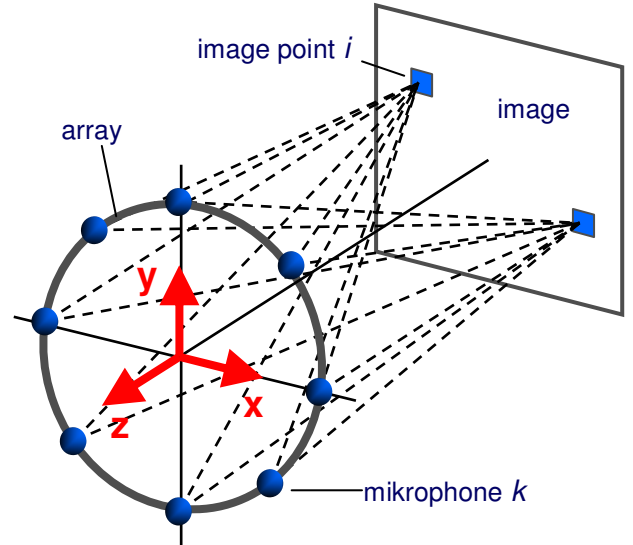


Figure 1: Each image point i has different delays relative to different microphone positions. These characterize the delay mask vector M_i for the location i .

Interference Transformation (Reconstruction)

The central idea behind it is to reverse these τ_{ik} delays through appropriate compensation, in order to approximate the time function at the source location. In this way, the delays of all delayed paths $f(t-\tau)$, can be compensated step by step roughly in the form $f(t-\tau+\tau) = f(t)$.

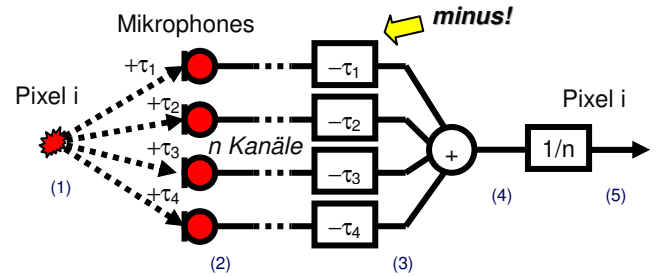


Figure 2a: Principle of the interference transformation, illustrated using an image point (pixel i). Negative delays $-\tau$ are required for compensation.

Steadily increasing CPU speeds from 1993 onwards made it feasible to implement **negative delays** $-\tau \sim f(t+\tau)$ in software, initially on parallel computing systems (8-node T805 transputer array, later a 16-node Parsytec PowerXplorer) and, beginning 1995, on Intel i486/Pentium-based systems running on Windows 95.

A time-domain signal, delayed along its path from the source (1) to the microphone (2), is subsequently subjected to an inverse delay (or negative delay) via software (3), Fig.2a. Summing all channels (4) and dividing by the number of channels (5) yields an approximation of the original time-domain signal (1).

¹ Lecture hold at the 33. Deutsche Jahrestagung für Akustik, DAGA 2007, University Stuttgart, Lecture 255, March 21, 2007, 2:50 PM, Room 57.03

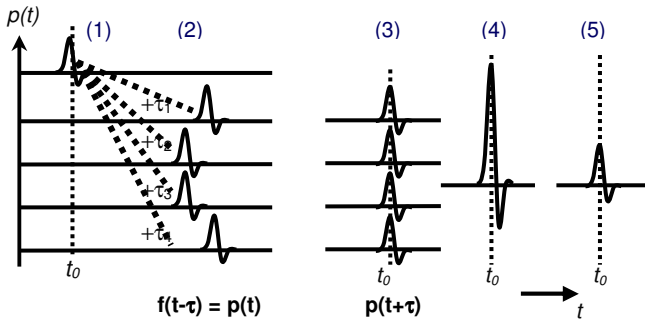


Figure 2b: Time functions for Fig.2a)

Expressed as a formula, we obtain the (approximated) result-time function $p_i(t)$ for each location (image point i) as the sum of the microphone time functions $p_k(t)$, each shifted by the propagation delay τ_{ik} normalized by the number of microphone channels n .

$$(3) \quad p_i(t) = \frac{1}{n} \sum_{k=1}^n p_k(t + \tau_{ik}) \quad (\text{Interf.-Reconstruction})$$

plus!

The RMS value $e_i(t)$ of the acoustic image is formed numerically at the level of the sampled values t .

$$(4) \quad e_i(t) = \frac{1}{T} \sum_{t=1}^T \sqrt{p_{it}^2} \quad (\text{RMS-value})$$

Analytically, one could call this an interference integral.

$$(5) \quad e_i(t) = \frac{1}{T} \int_0^T \sqrt{p_i^2(t)} d\tau \quad (\text{Interference Integral})$$

If we display all effective values $e_i(t)$ with colours (Fig.3), depending on the integration time T the result is either a *wave field* or an *acoustic image*. The shortest interference integral with a duration of $T=1$ (in samples) represents the fastest wave field itself.

In the software *NoiseImage*, the resulting time function for each pixel (Eqn.3) can be heard by shifting the mouse within the image area.

Equation (3) features a *plus sign* (representing a *negative, non-causal delay*). Unlike Beamforming approaches (interference projections) of the form $f(t-\tau)$ [J94], this method avoids overdetermination (in regions far away from the axis). Furthermore, it eliminates issues related to the Heisenberg uncertainty principle and wave number aliasing that typically can arise when transitioning into the complex domain (via FFT).

Beyond this, among the hundreds of studies in the field, the author was aware of only three similar approaches: Brühl and Schmitz [B93] and Michel [M97] also employ a compensatory approach of the form $f(t+\tau)$, though they do it in the frequency domain. Fink [F96] achieves a similar result by using an inverted time axis, $f(r,-t)$, so as PSI-Tools since 1995 allowed a time-inversion to calculate *mirroring interference-projections*.

Wave Field Properties

A characteristic feature of *interference reconstruction* (correctly oriented with *negative delays*) is a secondary wave field that - contrary to our understanding of physics - propagates inward as time progresses, with the *wavefront located on the inside*; Fig.3.

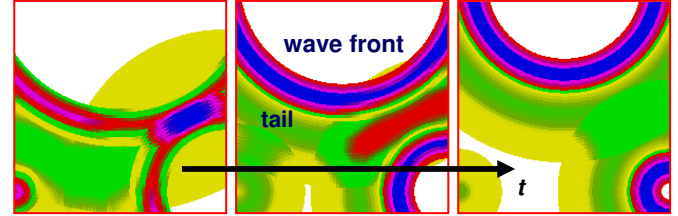


Figure 3: In contrast to beamforming, interference reconstruction shows an inwardly traveling wave field with the wave front inside, (simulation with PSI-tools)

Miscellaneous

Curiously, the original purpose of the invention was quite different. In 1993, the goal was to develop a demonstrator capable of illustrating the fundamental equivalence - from a neural perspective (in terms of temporal functions) - between hearing, feeling, and seeing, with the aim of correcting certain conceptual misconceptions in the field of neurocomputing (ANN). The concept envisioned bio-technical interaction, utilizing input and output devices that should communicate with the nervous system [H941], [H942].

Unfortunately, no sponsor could be found. Consequently, the software for the first *acoustic still images* was originally dubbed *Bio-Interface* (1994–1995), later named *PSI-Tools*. Ultimately, the sound map was automatically integrated onto the image feed from a USB video camera in the software solution *NoiseImage*, introduced 1999. Attempts by [D05] and [J06] to publish my algorithm for the first time unfortunately failed to capture the core concept correctly.

Sources

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