

# A Review of Array Techniques for Noise Source Location

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In recent years the increase of computer data processing power has facilitated the use of several multi-channel techniques for noise source location and ranking. These techniques include Spatial Transformation of Sound Fields (STSF) and Inverse Boundary Element Method (IBEM) for reconstruction of the sound field on the source surface from near field measurements and Beamforming for far field measurements. STSF and IBEM are suited for the low to medium frequency range whereas Beamforming is suited for the medium to high frequency range.

The array methods are briefly introduced and a discussion of the advantages and limitations of the different techniques is presented. Benefits of combination of different techniques are also considered.

**Keywords:** Measurement, Analysis, Noise Source Location, Holography, IBEM, Beamforming, Engine Cycle

## 1. INTRODUCTION

Modern array measurement techniques for noise source identification rely on high performance multi channel data acquisition hardware and on the data processing capabilities provided by present day computers. Common to these methods is both the problem of administrating the enormous amount of data generated and to setup, detect and calibrate the many channels (typically around one hundred or more). In order to be useful in a real life measurement situation in a laboratory or test cell it is therefore mandatory that the array measurement is supported by software which provides efficient data management and tools for channel-measurement- and calculation-setup. Also the mechanical design and cabling of the arrays onto which the transducers are mounted is an issue of great practical importance. The assembly of the arrays must be easy and flexible and the cabling must be as simple as

possible and preferably with a logical grouping of the channels that facilitates identification of occasional hardware faults.

The planar Near-field Acoustic Holography (NAH) methods STSF and Non-Stationary STSF (NS-STSF) use sound field data measured over a planar surface to estimate a 3D mathematical model of the sound field, [1-4]. From this model calibrated values of the entire 3D sound field can be calculated. Today, STSF and NS-STSF are well-established techniques, which are also

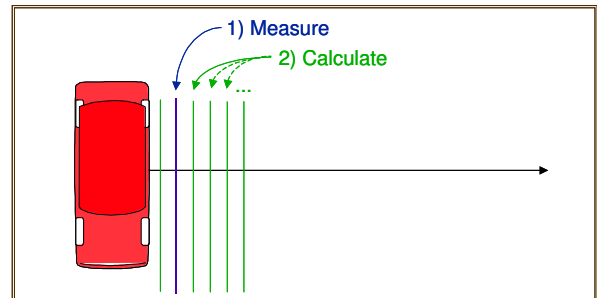


Fig. 1. Spatial Transformation of Sound Fields (STSF) provides 3D sound field mapping from 2D data using Near-field Acoustical Holography.

commercially available.

The Beamforming technique provides a directional source map (or acoustical picture) as seen from the array position, [5-6]. Unlike NAH no complete sound field model but a relative contribution map is provided. Beamforming is a well-known technique but until recently commercially available systems for noise source location have had limited performance due to the chosen array design.

The principle of Inverse BEM (IBEM) is the same as that of planar NAH, except that conformal measurement and calculation surfaces are used instead of planar surfaces. As a result, source maps can be obtained directly on the surface of conformal source geometries, [7]. The price to be paid is longer calculation time and the need to input the source geometry.

In the following sections STSF, Beamforming and IBEM are described in greater details. Finally the methods are compared and suggestions for useful combination of the techniques are given. As the purpose of the present paper is to provide an overview, an in-depth mathematical treatment of the subject is avoided. The interested reader is referred to the literature.

## 2. SPATIAL TRANSFORMATION OF SOUND FIELDS (STSF)

The planar NAH techniques (STSF and NS-STSF) involve a two dimensional measurement with a microphone array in a plane close to a noise source and then transformation of the scan plane data to any parallel plane, either closer to or further away from the noise source under test, see Fig. 1.

The basic condition to be fulfilled (or approximated) is that the volume between the measurement and calculation planes plus the half space further away from the source must be homogeneous and source free, [1]. As a consequence, the calculation plane can be closer to the source than the measurement plane, but only until it touches the nearest point on the source. If the calculation plane is

moved further in the direction of the source, then some blurring will occur around the regions where it intersects the source.

Both input and output are complex frequency domain data. The input is the pressure in the measurement plane. The pressure in a parallel plane closer to or further away from the source equals the measurement plane pressure spatially convolved with a 2D propagation kernel. This convolution is converted to the multiplication with a transfer function through a 2D spatial Fourier transform, or in practice a 2D FFT. The NAH calculation is very fast because it is FFT based.

Basically, the 2D Fourier transform in the measurement plane requires knowledge of the pressure in the entire plane. In practice, we measure only within a restricted 2D spatial window, but in order to limit the spatial window effects, the measurement area must cover the entire sound source, plus a certain solid angle away from the source.

The use of a discrete planar set of measurement points can be seen as a spatial sampling of the sound field. In order to avoid spatial aliasing, the sampling interval must be less than half a wavelength (at the highest frequency of interest). In the far-field region this is a sufficient requirement, because the propagation of the sound field from the source surface to the measurement plane has removed all the higher spatial frequencies (representing evanescent waves). Effectively the propagation acts as a 2D spatial low-pass filter, which can be utilized as an anti-aliasing filter. Using a measurement grid spacing equal to the measurement distance will ensure that spatial aliasing components are suppressed at least 20 dB. The resolution on the source surface can be shown to be approximately half a wavelength towards high frequencies and approximately equal to the measurement distance towards low frequencies.

Because the processing is performed by 2D spatial FFT, the measurement points must constitute a regular rectangular grid.

A consequence of the above requirements

is that many measurement points will be needed, if the source dimensions are much larger than one wavelength at the highest frequency of interest. Provided the source is stationary and provided only averaged sound field data are required, then a scanning of a small array can be performed to synthesize a larger measurement area. This solution is adopted in the STSF method, [2].

### 3. STATIONARY NOISE: CROSS-SPECTRAL HOLOGRAPHY

Basically, this STSF technique involves a two-dimensional scan measurement with a microphone array in a plane close to a noise source, [2]. By measurement of the Cross-spectra between a set of references and the Cross-spectra from each scan position to each of the references, a Principal Component representation of the sound field is extracted, which can be employed for both NAH and for the Helmholtz Integral Equation. The sound source must be stationary during the measurement period in such a way that all the necessary Auto- and Cross-spectral measurements are repeatable to a sufficiently high degree of accuracy. The Principal Component sound field representation includes only those parts of the sound field, which are coherent with the references. This can be an advantage, because uncorrelated background noise can be suppressed, and because sound fields from uncorrelated partial sources can be separated. But if the total sound field from a complicated sound source with many independent source mechanisms is to be investigated, then it may be a problem to select a sufficient set of reference signals.

#### 3.1 Complete description of the sound field in three dimensions

The Maps of pressure, particle velocity and intensity (active and reactive) can be obtained. The ability of the STSF technique to provide any near-field and far-field information is just a consequence of the fact that the measured Cross-spectral sound field data provide a complete mathematical model

of the sound field over a region extending from the source surface to infinite distance. Therefore it is also possible to determine which parts of the noise source are the most significant contributors to the far-field radiation pattern.

The Simulation of Source Attenuation capability enables the near-field particle velocity map to be modified and for example the far-field radiation pattern to be recalculated based on the modified data.

#### 3.2 Measurement technique

A practical STSF system includes a set of reference transducers, an array of scan microphones, an automated positioning system (i.e. a robot), a multi-channel data acquisition system and a computer with the STSF software.

### 4. NON-STATIONARY NOISE: TIME DOMAIN HOLOGRAPHY

Non-Stationary STSF (NS-STSF) is an implementation of Time Domain Holography (TDH), [3-4]. Because it is a time domain method, all signals need to be measured simultaneously. TDH processes a recorded acoustic event that consists of simultaneously measured pressure time histories from a planar surface covering the sound source. As a consequence, the scan principle used in STSF cannot be used – a full-size microphone array is needed, which is not realistic when the source dimensions exceed 8-10 wavelengths. The two big advantages of NS-STSF are that there are no requirements for stationarity and that full time resolution is retained throughout the NS-STSF measurement and the Time Domain Holography calculations. Thus, while STSF provides detailed information about *where* noise is radiated, NS-STSF can show both *where* and *when* noise is radiated.

#### 4.1 Principles of NS-STSF

TDH maps all sound field descriptors (pressure, particle velocity, intensity, etc.) in the near field, not only as a function of

physical location, but also as a function of time. A TDH measurement can be seen as a sequence of snapshots of the instantaneous pressure over the array area, the time separation between subsequent snapshots being equal to the sampling interval in the A/D conversions. Similarly, the basic output of TDH is a time sequence of snapshots of a selected acoustic quantity in a calculation plane parallel with the measurement plane. Envelope Detection by use of the Hilbert Transform is used in order to avoid the classical detector limitations.

When NS-STSF is applied for engine noise analysis, two crankshaft tacho signals are normally recorded simultaneous with the array signals – one high-resolution tacho for precise measurement of the crank angle rotation, and another for identification of a reference angular position, i.e. a key phasor. These two tacho signals allow the crank angle at the time of each snapshot to be identified in addition to the engine RPM. With the NS-STSF system it is therefore possible to visualize radiation, and at the same time, relate it to the different events during an engine cycle.

## 5. BEAMFORMING

The Beamforming is source location from medium-to-long distances using a, usually, sparse array geometry of microphones.

### 5.1 Principle of beamforming

The principle of Delay-and-Sum Beamforming is as follows: A planar microphone array is connected to a processing unit with individual delays and summation of the delayed signals, [5]. The delays of the individual microphone signals are chosen in such a way, that signals (plane waves) arriving from a chosen focus direction will add up coherently. Keeping the same focus direction (the same delays), signals arriving from other directions will provide lower output.

Having processed the microphone signals with the above Delay-And-Sum algorithm for

a set of focus directions that scan a mesh covering the source object, a contour plot of the source distribution can be obtained. To focus at a finite distance from the array, we assume a distribution of monopole point sources on the focus plane, and the delays are computed accordingly.

### 5.2 Beamformer properties

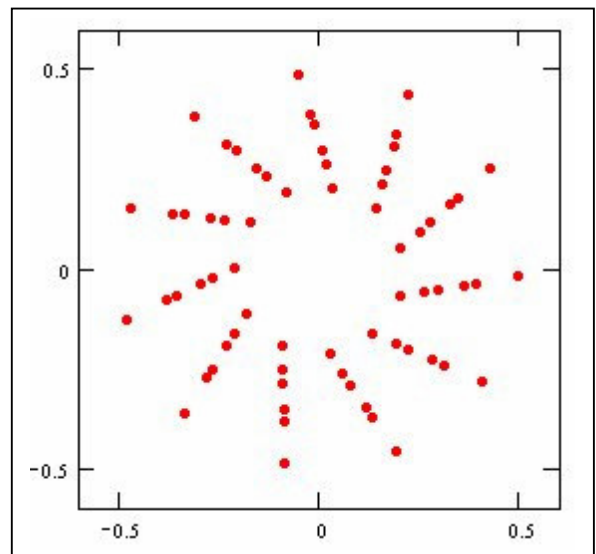
Considering the Directivity Pattern of the array the sensitivity will be highest in directions close to the focus direction. This peak in the directivity pattern is called the Mainlobe. The sensitivity in other directions should preferably be much lower. Smaller local peaks here are called Sidelobes. The Maximum Sidelobe Level for all directions and up to a given frequency can be identified as the highest peak next after the mainlobe, see reference [6]. The width of the mainlobe is seen to be inversely proportional to the array diameter  $D$ . Actually, the on-axis angular width of the mainlobe equals  $\lambda/D$ , where  $\lambda$  is wavelength, implying that the resolution  $R$  at a measurement distance  $L$  will be given by

$$R_{Axis} \cong L \frac{\lambda}{D} = \frac{L}{D} \lambda \quad (1)$$

The resolution at an angle  $\theta$  away from the on-axis direction can be shown to be

$$R(\theta) \cong \frac{R_{Axis}}{\cos^3(\theta)} \quad (2)$$

so clearly the resolution degrades very quickly



when the off-axis angle exceeds 30°.

### 5.3 Beamformer Design

The degree to which the Sidelobes are suppressed relative to the Mainlobe determines the beamformers ability to focus on a particular direction and reject unwanted noise from other directions. In beamformer design the primary objective is therefore to produce arrays, which have the lowest possible MSL in the frequency range of the arrays intended use. Beamformer performance is determined by the chosen array design.

Regular arrays, as used for STSF, will introduce severe spatial aliasing, when the regular microphone spacing exceeds half a wavelength. Such aliasing will show up as very high sidelobes – so-called grating lobes. The effect of very high sidelobes is ghost images in measured directional source maps.

Random array geometries are known to have poorer suppression of aliasing at low (spatial) frequencies, but avoid the sudden appearance of very high aliasing above a certain frequency. A random array can provide acceptable suppression of aliasing up to a much higher frequency. A new type of wheel array geometry, which has been patented by Brüel & Kjær, turns out to possess some of these properties of random arrays, still maintaining a very useful regular mechanical structure, see Fig. 2.

*Fig 2. 66-element wheel array used for car engine measurement. The geometry of this array has been optimized for minimum MSL up to 21 kHz. Over this frequency range the MSL is at least 10.4 dB below the mainlobe.*

## 6. INVERSE BOUNDARY ELEMENT METHOD (IBEM)

IBEM could be referred to as 3D STSF: Where STSF is bound to planar measurement and calculation surfaces in order to allow for the 2D FFT spatial processing, IBEM allows conformal measurement and mapping

geometry. The use of reference signals is identical to that in STSF, and scanning can also be performed, just in a much more free way, not being bound to a plane. Actually array scan positions can be arbitrary chosen in 3D space around the sound source.

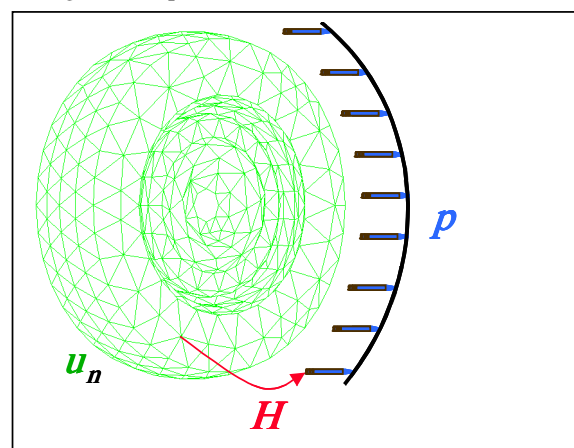
Before the IBEM calculation can be performed, a Boundary Element (BE) Model of the sound source must be defined. The surface vibration pattern  $u_n$  of the BE model is to be determined from the IBEM calculation. Use of the BE model allows determination of the transfer functions  $H$  from BE node velocities  $u_n$  to the pressure  $p$  at a set of microphone positions. These transfer functions  $H$  are arranged in a transfer matrix,  $H$ , allowing the vector of pressure data  $p$  at the microphone positions to be expressed in the following way,

$$\mathbf{p} = \mathbf{H}\mathbf{u} \quad (3)$$

where  $u$  is a vector containing the unknown BE node velocities  $u_n$ . We have to solve a system of linear equations to find the node velocities. Because the evanescent wave components are decaying from the source surface to the measurement area, the matrix  $H$  is always ill conditioned, and therefore regularization is necessary to obtain a stable and meaningful solution. Once the vector  $u$  of node velocities has been calculated, the entire radiated sound field can be estimated through use of the BE model, see Fig. 3.

Use of an automated parameter-choice method, such as the L-curve approach, means that the optimal amount of regularization is always used, see reference [7]. In other words, the best possible source reconstruction is obtained from a given set of noisy test data. The resolution obtained with IBEM will be like the NAH resolution.

Fig. 3 Principle of IBEM method



## 7. COMPARISON

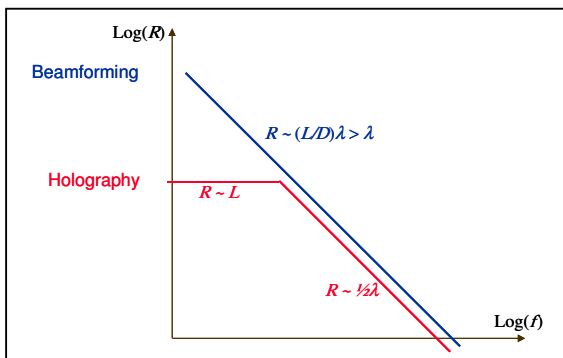
As stated previously, the NAH resolution is approximately half a wavelength at high frequencies and equal to the measurement distance  $L$  at low frequencies. The resolution obtained with beamforming is given in equations (1) and (2), see Fig. 4.

Here one needs the additional information that the measurement distance  $L$  should not be smaller than approximately  $0.8D$ , where  $D$  is the diameter of the beamformer array. So clearly NAH can provide much better resolution at low frequencies, through the use of a small measurement distance, while at high frequencies beamforming can provide almost the same resolution.

Beamforming has the advantage of not requiring the array to be larger than the source, and the related disadvantage of not providing maps of calibrated acoustical quantities – only relative contribution maps.

In the practical measurement situation combination of NAH and Beamforming can be beneficial. Especially for large objects it can be an advantage to firstly obtain with Beamforming a quick scan measurement covering the whole object, and secondly apply NAH for detailed analysis of the trouble areas identified by the Beamforming measurement. In case the radiated noise contains both low- and high-frequency components (e.g. 100Hz – 20 kHz) the whole frequency range can only be investigated by using NAH for the low-medium frequencies (100 Hz – 5 kHz) and Beamforming for the high frequencies (5 – 20 kHz).

Fig. 4. Comparison of Beamforming resolution with holography resolution



## 8. CONCLUSION

In this paper four different array techniques for noise source identification has been reviewed and compared. Also it has been discussed how the different techniques can supplement each other in a complete investigation the noise radiation from a source object.

## 9. REFERENCES

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