8 Near Field Acoustics

8.1 Summary ......................................................... 2
8.2 Introduction .................................................... 4
8.3 What is important in visualization methods? .......... 6
8.4 Direct method .................................................... 8
   Bandwidth 8
   Simple mathematics, no errors, fast method 9
   Background noise 10
   Direct method can be used in diffuse sound fields 10
   Direct method is usable for non correlated noise sources 10
   Measurement errors of the direct intensity method 10
   Relation with source path contribution 12
8.5 Planar NAH; a Fourier method .......................... 13
   Microflown and Planar NAH 15
8.6 HELS and SONAH ............................................. 19
   Microflown in HELS and SONAH 20
8.7 IBEMS NAH; a Helmholtz method ..................... 21
   Microflown in IBEM 23
8.8 Direct method Measurements ............................ 23
   Acoustic problem finding of a micropump 23
   Helicopters interior noise testing [8] 24
   Sound field visualisation of a thin plate 25
8.9 Acoustic eyes (near field) ................................. 28
8.10 Acoustic noise source finder (based on [24]) .... 30
    Measurements 33
    Comparison between pressure and velocity signals 35
    Noise source finder, Mid-side technique 37
8.11 References .................................................. 38

Fig. 8.1 (previous page): Historical photo: first Microflown based NAH measurements of a Diesel engine at the Otto-von-Guericke-Universität Magdenburg. As can bee seen: cabling was an issue.

8.1 Summary

The aim of the methods that are discussed in this chapter is to be able to reconstruct and visualize sound pressure, particle velocity and sound intensity on a surface out of a near field measurement.

As long as the source to sensor distance is in the order of acoustic wavelength the source is considered in the near field. (If the sensor is very close to the source, it is considered in the very near field. See chapter 7: Vibration).

Most of the time these methods are used to find sound sources on engines, in cars or in airplanes. In such applications the ease of use, the accuracy and the time to find these sources is important. In these applications un-correlated noise sources (e.g., wind noise) and (diffuse) background noise may be expected.
In theory there are several visualization methods but only three methods are commercially available: Planar holography, beamforming arrays and the direct method [1]. IBEM is a method that has potential in some cases but is not commercially available yet [6].

Most of the methods mentioned above are scientific proven in an anechoic environment and under favorable conditions. This is a prove of concept but no prove of usefulness: in most practical situations background noise is influencing the results considerably.

Sound pressure based PNAH has a poor particle velocity reconstruction, in practice a limited bandwidth (300Hz-3kHz), a limited dynamic range (<20dB) and it is difficult to use for non coherent noise sources. It is based on a 2D Fourier transformation and very much data acquisition channels are needed [1], [4], [5].

Particle velocity based PNAH has a far better reconstruction capability and dynamic range (>20dB) than pressure based PNAH but it has the same (bandwidth and sound field) limitations as pressure based PNAH [4], [5].

SONAH and HELS are holography methods that do not involve 2D Fourier transformation and therefore have less extreme requirements on the sound field [20]. However similar reconstruction (pressure to velocity) problems can be expected [21].

![Real time hand held acoustic camera](image)

Fig. 8.2: A real time hand held acoustic camera based on 12 pu-mini probes, a 24channel data acquisition system and a normal PC.
A beamforming array is a large device consisting on a lot of microphones that is only usable for higher frequencies (f>2kHz) and it has a very small dynamic range (a few dB) [1].

The ‘direct method’, that is simply measuring the sound pressure and particle velocity close to the surface, can be used in diffuse sound fields and with correlated and non-correlated sound sources. So real life measurements in a car or plane, just as source finding on an engine, are very well possible. Both the bandwidth (20Hz-20kHz) and the dynamic range (<40dB) is large.

Background noise causes methods to fail. Sound pressure measurements are far more influenced by background noise than particle velocity measurements [7]. So for this reason alone in all the visualization methods the velocity measurement is advantageous.

### 8.2 Introduction

There are several methods to use an array in the near field of a sound source; the methods are called Nearfield Acoustical Holography (NAH).

The primary objective of NAH is to recover the normal particle velocity and sound pressure on the reconstruction surface, allowing to calculate the active intensity and thus obtain the total power radiated from the source.

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**Fig. 8.3**: An array of 12 pu-mini probes. At the back a CCD optical camera is constructed so that the optic and acoustic picture can be overlayed.
NAH is a methodology that enables one to reconstruct acoustic quantities such as the acoustic pressure, particle velocity, sound intensity, and radiated power from an object based on acoustic measurements in a 2D surface. It is a way to find relative small sized noise sources on structures. An array of sound probes or a robot positioned (row of) microphones is used to scan a certain "scan plane" close to a sound source. The measured 2D sound field data can be "transformed" to other surfaces enabling (in theory) a complete 3D description of the sound field.

Planar Nearfield Acoustical Holography (PNAH) is the oldest method that is developed in the 1980s [9]. It allows calculating the sound field on plane from a 2D measurement plane. So the method works in principle only for flat surfaces.

IBEM (inverse boundary element method) is a two-step method (developed in the late 1980s) that calculates the sound field from a grid of a surface towards sound probes with a boundary element method (BEM). The matrix that contains all transfer functions has to be inverted (this is what the ‘I’ stands for in IBEMS) to be able to reconstruct the sound field at the surface from the measurement points. The method allows for non-planar problems but is extremely labour intensive [6].

A special form of IBEM might be advantageous. The method is called TRIM (Toeplitz Rayleigh Integral Method). If the source is not seen as the structure itself but as an imaginary plane just above it, a more general IBEM solution can be found that does not have to be altered each time the acoustic problem changes. An example of this way of thinking is given in [13]. In this way the disadvantages of the NAH (e.g. the double Fourier transformation with a few measurement points, the regularisation, the limited low frequency bandwidth, etc.) are avoided.

Another visualisation method of acoustic sources is the use of a beamforming array. It is a far field technique so inherent different to the near field techniques. Room properties (reflections, standing waves etc.) are included in far field measurements. The mathematics behind this method is quite simple but the method works only at higher frequencies and requires free field conditions. The dynamic range of this method is low.

The methods mentioned above require quite some calculations and with that, quite some uncertainties. With the use of a Microflown a new method is possible that requires no (ill posed) transformations or sound field predictions. It is simply an array of sound pressure and particle velocity probes that is used to measure the particle velocity and the sound intensity directly near the surface. Due to its ease of use, its high dynamic range and broad band use, the method is shown to be very powerful. The method is called the ‘Direct Method’.

A comparison of three methods is made in [1], see Fig. 8.4. It shows the result of a beamforming array (BFA), planar NAH (SONAH) and the direct method (intensity probe). As can be seen, the beamforming array works only at high frequencies, planar NAH works at somewhat lower frequencies but not higher frequencies and the direct method is performing well at all frequencies.
This example is done without background noise and it shows only how a single source is found. This is an extremely simple example. To find a single source in an anechoic environment, without background noise has no practical meaning. Even in this extremely simple problem it shows that the beamforming and the NAH method give only a limited result.

![Beamforming and NAH Methods Comparison](image)

Fig. 8.4: Resolution of Planar NAH (here called SONAH) compared to a beamforming array and the Direct Method (here called intensity probe), from [1].

### 8.3 What is important in visualization methods?

In this paragraph some features are given in general terms to be able to compare the visualization methods (PNAH, beamforming array, direct method) without going into (mathematic) details. In the following paragraphs the methods are discussed in more detail.

**Complexity.** NAH methods are very complex. It involves many sound sensors and a multichannel analyzer or a robot to scan a surface, the position of the sensors has to be known precisely and proper settings are crucial for a possible good result. To be able to truly understand the results, one needs to understand some heavy mathematics.

The working principle (and therefore the proper use) of beamforming arrays is much easier to understand. However because it is a far field method, ghosts (sources due to reflections) have to be expected.

The direct method is very simple to understand. The positioning of the sensors is not important and extraneous sound sources do not influence the measurement much.

**Ease of use** is an important issue because these methods are normally used to solve acoustic problems in the end of an engineering project (car,
plane etc.) where measurement hours are expensive. NAH requires a lot of effort to be applied properly. The beamforming array is much easier to use but it should be used in a (semi) anechoic environment. Extraneous sound sources and reflections cannot be suppressed. A beamforming array should be used somewhat far from the source. This makes it somewhat difficult to use it inside e.g. a car. At high frequencies beamforming methods can be used inside vehicles but the results are rather vague. One gets an impression to where to look for an acoustic problem. The direct method can be used directly.

NAH requires a coherent sound field measured on an imaginary surface. No sound sources should be present between the measurement plane and the reconstruction plane.

**The time to get results** is depending on the setup time, the processing time and the need to do measurements. NAH requires the most time, the beamforming array and the direct method are ready to go.

**Accuracy based on sound pressure measurements.** For NAH a lot of sound pressure microphones are needed to calculate a particle velocity field. Therefore low cost microphones are used. This of course lowers the quality of the measurement that is not so high anyway [4], [5]. Although low cost microphones are used, due to the number of acquisition channels and the complex software, the total solution cost is high. A beamforming array has a low dynamic range so the accuracy cannot be high. Reflections lower the accuracy even more. The direct method cannot be applied with pressure microphones only.

**Accuracy based on particle velocity measurements.** Microflows are relative expensive but provide a much better sound field reconstruction in NAH [4], [5], [6]. Less acquisition channels and the processing power is required to get results. For beamforming arrays Microflows do not have real advantages.

The direct method is based on the measurement of the sound pressure and the particle velocity close to a surface. However it is possible of only measure the velocity. Then the method is not influenced by extraneous sources and sound field is measured directly, the accuracy is high.

It might be argued that the velocity based holography (measured at a certain distance) for the determination of the surface velocity provides more information than the direct measurement of the surface velocity with Microflows close by the surface. This is because all velocity measurements are used to determine the surface velocity at that particular point. However the errors that are made with NAH might cancel out this (small) advantage.

**A dynamic range** of 40dB and more is quite normal for the surface velocity of a vibrating object. If the dynamic range of the reconstruction method is less, this means that large errors are made. Only velocity based methods are capable of reconstructing such large dynamic ranges. Pressure based NAH methods have a dynamic range in the order of 20dB and beamforming arrays have an even much lower dynamic range. Only the direct method has such a high dynamic range.
The Bandwidth of the direct method is much higher than the other methods. In principle the bandwidth of the direct method is 2Hz to 20kHz. The practical bandwidth of NAH methods is in the order of 500Hz-3kHz, the practical bandwidth of the beamforming arrays is limited to high frequencies only.

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8.4 Direct method

The direct method is straightforward: measure the particle velocity and sound pressure close to the source. From these quantities the sound intensity can be derived and that quantifies directly the amount of sound that the sources emit. Either the sound intensity or the particle velocity is visualised.

Bandwidth

Compared to planar holography (200Hz-2kHz) and beamforming arrays (2kHz-10kHz) the direct method has a large bandwidth. In theory the bandwidth can be from frequencies far below 20Hz and above 20kHz. At the lower end however the measurement error increases for intensity measurements due to that the relative distance (the measurement distance
compared to the wave length) decreases and therefore the reactivity of the sources increases.

The measurement error is decreased by enlarging the measurement distance. A rule of thumb is that the phase shift should be below 85 degrees to ensure an error below 1dB. This phase shift can be determined easily.

Compared to planar holography (~15dB) and beamforming arrays (~3dB) the direct method has a large dynamic range. The dynamic range is of course determined by the sound field but values of 30dB to 60dB are common.

**Simple mathematics, no errors, fast method**

Compared to planar holography the mathematics and operation conditions are extremely simple. This ensures that no (reconstruction) errors are made. Because of simple mathematics, the measurement results can be visualised fast. This allows real time visualisation and even the display of transient events like explosions or engine runups.

Another important advantage of the direct method over holography and beamforming systems is that data acquisition systems can be made of a much simpler architecture. This is because for intensity measurements only pairs of channels have to be synchronized. For a velocity profile a limited synchronisation is needed. Due to this it is possible to do the required calculations in a distributed manner or make use of soundcards. See further chapter 16: ‘data acquisition’.

![Fig. 8.5: First pictures of the direct method from the experiments at the Otto-von-Guericke-Universität, Magdenburg.](image)
Background noise

All measurement methods that are discussed in this chapter are influenced by background noise. This is strongly reduced by particle velocity measurements (or intensity measurements) as compared to sound pressure measurements because of three reasons [7]:

1. The sound pressure level and particle velocity level are of similar magnitude in the free field. If the sound wave reflects on a rigid surface, the sound pressure doubles and the particle velocity reduces to zero.

2. On the other hand, close to a vibrating (sound emitting) surface, the sound pressure level is reduced as compared to the particle velocity level that coincides with the surface velocity.

3. A sound pressure microphone is omni-directional and thus measures the sound field in all directions. A Microflown measures the particle velocity in one direction. Therefore Microflown measures only one third of the sound field whereas a pressure microphone measures the total sound field.

Direct method can be used in diffuse sound fields

The direct method can be used in diffuse sound fields. The PU intensity method is not affected by a diffuse sound field. This is proven in [12] and more explicit in [11], [14], [15], [16], [17], [18]. The basis of the proof is that the directivity of the Microflown is an exact figure of eight.

If the method is used to determine the velocity field only, the method works also because of the fact that background noise is reduced [7]. Because both the intensity and the velocity are not affected by the background noise the background-noise-free sound pressure can be derived.

Direct method is usable for non correlated noise sources

In contrast to PNAH where each measurement point is needed for the reconstruction (due to the 2D Fourier based method, see below) the direct method uses the intensity measurement determined at each measurement point. Therefore also apart from correlated noise sources also non-correlated noise sources can be measured and visualized.

Measurement errors of the direct intensity method

A p-u sound intensity measurement system combines two fundamentally different transducers. The sound intensity is simply the time average of the instantaneous product of the pressure and particle velocity signal,

\[ I_r = \langle pu_r \rangle_r = \frac{1}{2} \text{Re} \{ pu^*_r \} \] (8.1)
where the latter expression is based on the complex representation of harmonic variables. Quite apart from the particulars of the Microflown transducer it can be shown that any sound intensity measurement system based on the combination of a pressure microphone and a particle velocity transducer is sensitive to reactive sound fields; if the reactivity (the ratio of the reactive to the active intensity in logarithmic form) takes a high value, as for example in the near field of a source, then even a very small phase mismatch error between the two transducers gives rise to a considerable bias error, as can be seen from the expression:

\[
\hat{I}_r = \text{Re}\{S_{pu} e^{j\varphi}\} = I_r \cos \varphi_e - J_r \sin \varphi_e = I_r \left(1 - \varphi_e \frac{J_r}{I_r}\right) = I_r \left(1 - \varphi_e \tan \varphi_{\text{field}}\right) \quad (8.2)
\]

where \(\hat{I}_r\) is the measured intensity, \(S_{pu}\) is the ‘true’ cross spectrum between the sound pressure and the particle velocity, \(\varphi_e\) is a small phase error between the measured and the ‘true’ particle velocity, \(I_r\) is the ‘true’ intensity, and \(J_r\) is the ‘true’ reactive intensity. The phase \(\varphi_{\text{field}}\), is the phase shift of the sound field. The reactive intensity is given by:

\[
J_r = \frac{1}{2} \text{Im}\{pu_r^*\} \quad (8.3)
\]

Whereas the (active) intensity describes the net flow of sound energy the reactive intensity describes the non-propagating part of the energy that is merely flowing back and forth, corresponding to the instantaneous particle velocity being in quadrature with the sound pressure. Many sources have strongly reactive nearfields at low frequencies where they mainly generate evanescent waves. Near such a source the air is essentially moving back and forth as if it were incompressible.

The rightmost approximation of Eq. (8.2) is valid if \(\varphi_e << 1\). If \(J_r > I_r\), as for instance very near a source, then even a fairly small phase error will give rise to a considerable bias error.

![Fig. 8.6: Reactivity as function of the phase shift of sound pressure and particle velocity.](image)
On the other hand it also shows that substantial p-u phase errors can be tolerated if $|J_\phi| < |I_\phi|$. For example, even phase mismatch of 35° gives a bias error of less than 1dB under such conditions. In other words, the phase calibration is critical when measurements are carried out under nearfield conditions, but not at all critical if the measurements are carried out in the far field. The “reactivity” (the ratio of the reactive to the active intensity a tangent function of the phase shift between sound pressure and particle velocity) indicates whether this source of error is of concern or not.

![Graph](image)

**Fig. 8.7: Sound intensity measurement error as function of the phase shift of the sound field and the phase matching error.**

In measurements with p-u probes the acceptable reactivity depends on the accuracy of the phase calibration of the device. A reactivity of more than 5dB is ‘high’, but values of up to 25dB can occur (measured very close to a non radiating source at a very low frequency, see chapter 5: ‘Intensity’; paragraph ‘Measurement of a very reactive sound source’).

Since the phenomenon is associated with near fields of sources the appropriate remedy is to use a measurement surface further away from the source under investigation.

Calibration of p-u sound intensity systems involves exposing the probe to a sound field with a known relation between the pressure and the particle velocity, for example a plane propagating wave (anechoic chamber), a standing wave tube or a simple spherical wave (near field calibration). See further chapter 4: ‘calibration’.

**Relation with source path contribution**

The source path contribution method such as explained in chapter 10 requires near field particle velocity and near field sound intensity. These values are multiplied with a complex number (the path) to reach a sound pressure at a certain listener position.
There is not much practical difference between an acoustic camera and a source path contribution system. One could see a source path contribution system as an acoustic camera of which the results are scaled with a complex factor (the path).

It can be argued that in many cases that one would think that an acoustic camera is required to tackle a certain problem, in fact one need a source path contribution system.

One need an acoustic camera if one only wants to know where sources are located. One needs a source path contribution system if one wants to know how loud the sources are perceived at a certain position.

If one uses a source path contribution system one can always get the acoustic camera results.

8.5 Planar NAH; a Fourier method

Fourier NAH or Planar NAH is the original NAH formulation derived by Williams and Maynard in 1980. Planar NAH is based on the fact that the sound field in one plane can be expressed as the two-dimensional convolution of a ‘propagator’ and the sound field in another plane.

The complex sound pressure is measured in a plane \((z=z_m)\) near the source under examination and a 2D spatial Fourier transform is calculated. The result is the wavenumber spectrum:

\[
P(k_x, k_y) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} p(x, y, z_m) e^{i(k_x x + k_y y)} \, dx \, dy
\]  

(8.4)

The planar NAH theory claims that one can determine the wavenumber spectrum of the sound pressure in another, parallel plane, \((z=z_p)\) simply by multiplying the wavenumber spectrum in the measurement plane with an exponential ‘propagator’:

\[
P(k_x, k_y, z_p) = P(k_x, k_y) e^{-i k(z_p - z_m)}
\]  

(8.5)

The sound pressure in the prediction plane \((z=z_p)\) by the inverse Fourier transform:

\[
p(x, y, z_p) = \frac{1}{(2\pi)^2} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} P(k_x, k_y, z_p) e^{-i(k_x x + k_y y)} \, dk_x \, dk_y
\]  

(8.6)

It is also possible to calculate the particle velocity vector, and thus the sound intensity. The normal component of the particle velocity is given by:

\[
u(x, y, z_p) = \frac{1}{(2\pi)^2} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} P(k_x, k_y, z_p) \frac{k}{\rho c} e^{-i(k_x x + k_y y)} \, dk_x \, dk_y
\]  

(8.7)

It is also possible to measure the normal component of the particle velocity in the plane near the source \((z=z_m)\) transform to the wavenumber domain, multiply with the propagator, and transform back to the spatial
domain. The result is the normal component of the particle velocity in the prediction plane \((z=z_p)\):

\[
U(k_x, k_y) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} u(x, y, z_m) e^{i(k_x x + k_y y)} \, dx \, dy
\]

\[
U(k_x, k_y, z_p) = U(k_x, k_y) e^{-i k_z (z_p - z_m)} \tag{8.8}
\]

\[
u(x, y, z_p) = \frac{1}{(2\pi)^2} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} U(k_x, k_y, z_p) e^{-i(k_x x + k_y y)} \, dk_x \, dk_y
\]

If the wavenumber spectrum is multiplied by \(\rho c k_z / k_z\) before the inverse Fourier transform is carried out one will get the sound pressure in the prediction plane:

\[
p(x, y, z_p) = \frac{1}{(2\pi)^2} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} U(k_x, k_y, z_p) \rho c \frac{k_z}{k_z} e^{-i(k_x x + k_y y)} \, dk_x \, dk_y \tag{8.9}
\]

The above description is grossly oversimplified. In practice the measurement area is finite, the sound field is sampled only at a finite number of discrete positions, and the 2D spatial Fourier transform is approximated by a 2D FFT. To reduce the influence of the finite aperture and the spatial sampling, zero padding and a tapered spatial window should applied before the spatial Fourier transform is calculated. The spatial sampling must satisfy the sampling theorem, one cannot sample very close to the source (where there may be high spatial frequency components), and one should not sample too far away, because then the evanescent modes will be buried in noise. Moreover, whereas forward prediction (in which \(z_p > z_m\)) is numerically stable, backward prediction, which involves predicting the sound field closer to the source than the measurement plane, is an unstable ‘illposed’ inverse problem that requires regularisation because of the fact that the evanescent waves are amplified exponentially with the distance. The standard regularisation technique involves spatial low-pass filtering (multiplying with a window in the wavenumber domain).

Planar NAH is valid for free space only. In practice, we cannot have an infinite hologram plane, so a compromise must be made. In general, the larger the size of a hologram plane is, the more accurate the reconstruction is, but the more cumbersome the process becomes. A rule of thumb is to have the edges of a hologram plane to be four times those of the source surface.

A diffuse sound field or the influence of background noise can be reduced by correlating techniques.

Since planar NAH relies on spatial sampling of acoustic pressure, the edge of the hologram plane must be larger than one wavelength of the low frequency limit.

Meanwhile, the microphone spacing must be smaller than one half wavelength of the high frequency limit, and the standoff distance must not
be smaller than the microphone spacing. This means that one cannot measure acoustic pressure as close to a source surface as possible, which may lower the accuracy of reconstruction for the high-frequency components.

A low signal to noise (S/N) ratio is another reason that NAH technologies are limited in their ability to provide accurate reconstruction in high frequencies. This is because a typical spectrum produced by a sound source has relatively high amplitudes in the low frequencies and relatively low amplitudes in the high frequencies; whereas the amplitude of ambient (background) sound pressure is more or less constant. Therefore the differences in the amplitudes of the signal and background sound at low frequencies are large, but those in the high frequencies are small. If the amplitude of the signal is less than 100 times that of the background (an S/N ratio of 40dB) it is difficult to get a satisfactory reconstruction of acoustic quantities by any NAH methods.

In the PNAH method the 2D spatial Fourier transform is used. To get a feeling of what is going on here: a measurement with 256 microphones can be compared with a 16 point 1D Fourier transform. The 1D Fourier transform is well known in transferring time data to frequency data and a 16 point Fourier transform will give an (almost) useless result. This (almost) useless result is used to make a ‘backward’ prediction. Mathematically speaking this represents an ill-posed inverse problem due to the fact that small perturbations in the measurement can lead to large perturbations in the reconstructed quantities. The reconstruction is stabilized by using a regularization procedure. The velocity reconstruction is a more ill-posed problem than the pressure reconstruction and thus requires more regularization. It is very unlikely that such a method result in reliable information.

Another consideration is the following example. To determine the sound field of a point source with spectrum of 50Hz-5kHz a would need an array size of $\lambda = c/f = 340/50 = 6.8$ meters (!). It can be questioned what the contribution is of the microphones 3 meters away of the source. On the other hand at least at each half wavelength there must be a microphone so the upper frequency limit of 5kHz dictates that the microphone spacing must be at least 3.4cm. This would mean that 40,000 measuring points are required to cover just two decades! (A bandwidth of 20Hz-20kHz would mean a 17x17 meters array and 8.5mm sensor spacing and 4,000,000 measuring points…).

Some manufactures of PNAH systems have 3D positioning systems so that a relative small array can be moved and the pressure data stored. In such way a large virtual array is constructed (over a 1000 measuring points). This increases the lower frequency limit and the accuracy.

**Microflows and Planar NAH**

There are two main reasons why Microflows should be used for Planar NAH. Cancellation of background noise and a far better sound field
prediction. The topic “cancellation of background noise” can be found previous in the text.

In [4], [5] simulations and measurements show that the particle velocity based reconstruction is far better than the sound pressure based reconstruction.

The measurements that show this are presented in [4], [5] and a brief summary is given below. The sound source was a 3-mm steel plate with dimensions of 39×63cm, see Fig. 8.8.

Fig. 8.8: The Microflown array in front of the steel panel in the baffle.

The spatial transform was an 18×56 point transform. A robot moved the single transducer or the array over the two measurement planes, one typically about $z_m=12$cm from the source plane and one $z_p=4$cm from the source plane.

From the sound pressure and particle velocity measurements in the measurement plane ($z_m$) the sound pressure and particle velocity is predicted in the prediction plane. Since the sound pressure and particle velocity is also measured in the prediction plane, the prediction results can be verified. Fig. 8.9 shows a comparison of the directly measured sound pressure in the prediction plane and predictions based on measurements of sound pressure and the normal component of the particle velocity in the
plane further away. It can be seen that velocity-based prediction is acceptable and somewhat more accurate than the pressure-based prediction.

![Sound pressure predictions](image1)

**Fig. 8.9:** Sound pressure predictions and true results based on sound pressure measurements and particle velocity measurements, from [4], [5].

Fig. 8.10 shows the ‘true’ particle velocity in the prediction plane predictions based on measurements of the pressure and the particle velocity in the measurement plane. It is clear that the velocity-based prediction is by far the best.

![Particle velocity predictions](image2)

**Fig. 8.10:** Particle velocity predictions and true results based on sound pressure measurements and particle velocity measurements, from [4], [5].
Fig. 8.11 shows the ‘true’ sound intensity in the prediction plane and a prediction based solely on pressure measurements in the measurement plane, a prediction based particle velocity measurements prediction plane, and a prediction based on both pressure and particle velocity measurements in the measurement plane. Again the velocity-based prediction performs much better than the prediction based on measurements of the pressure, and similar to a prediction based on measurements of both quantities.

![Graph showing sound intensity predictions and true results based on sound pressure measurements and particle velocity measurements, from [4], [5].](image)

Conclusion from [5]: "... velocity-to-pressure backward predictions are far better than pressure-to-velocity backward predictions because of the fact that the wave number ratio that enters into such cross predictions reduces high spatial frequencies in the former case but amplifies them in the latter case. For the same reason amplitude and phase mismatch, which is likely to occur in measurements with arrays of transducers, has a far more serious influence on pressure-to-velocity predictions than on velocity-to-pressure predictions: such transducer mismatch introduces high spatial frequencies, and the resulting errors are amplified exponentially with the distance if the prediction plane is closer to the source than the measurement plane.

The superiority of the method based on measurement of the particle velocity has been confirmed by an experimental study in which the sound pressure and the normal component of the particle velocity were measured at some distance from a vibrating, baffled steel panel with a Microflown p-u sound intensity probe and used to predict the pressure, the normal component of the particle velocity, and the normal component of the sound intensity in a plane closer to the panel.
Velocity-based predictions were consistently found to be better than pressure-based predictions. Thus if only one type of transducer is available one should choose to measure the particle velocity. However, slightly better predictions of the sound intensity may be obtained if the pressure that enters into this quantity is predicted from the pressure and the particle velocity is predicted from the particle velocity."

From: Jacobsen [4], [5]:

The particle velocity decays faster towards the edges of the measurement region than the sound pressure and has a larger dynamic range; therefore spatial windowing has less serious consequences on velocity-based holography than on conventional pressure-based holography.

The quality of pressure-to-pressure predictions is nevertheless similar to the quality of velocity-to-velocity predictions.

Velocity-to-pressure backward predictions are far better than pressure-to-velocity backward predictions because the wavenumber ratio in the propagator reduces high spatial frequencies in the former case but amplifies them in the latter case.

Amplitude and phase mismatch has a far more serious influence on pressure-to-velocity predictions than on velocity-to-pressure predictions.

The superiority of the method based on measurement of the particle velocity has been confirmed by an experimental study.

My personal opinion of holography is that the methods look so mathematical elegant (once you understand it) that one almost forgets to check if the reality looks just as good. The JASA paper [4] shows that even when measurements are done under an ideal situation (true planar source, a lot of measurement points, no background noise), large errors are made. The results of the holography method are very difficult to understand and (therefore) it is difficult to set up the system in practice.

If we take one step back and realize what is done: we would like to know the sound pressure and particle velocity at a surface. Because of the method (it is impossible to measure too close and too far from the surface) we have to measure at a specific distance the sound pressure. The measurement is very susceptible for background noise and because it is a scalar values the pressure distribution has a low dynamic range. With this garbled signal with a low dynamic range the particle velocity (with a high dynamic) range has to be calculated with a mathematical very unstable method that needs all the measurement points to calculate the velocity in one point...

Why not simply measure sound pressure and particle velocity?

8.6 HELS and SONAH

Statistically optimized near field acoustic holography SONAH differs from conventional near field acoustic holography NAH by avoiding spatial Fourier transforms; the processing is done directly in the spatial domain. The main
advantage of SONAH compared with NAH is that the usual requirement of a measurement aperture that extends well beyond the source can be relaxed [21].

Conventional planar NAH is based on discrete spatial Fourier transforms of sound pressure data measured with a microphone array. However, to avoid serious truncation errors caused by the finite two-dimensional spatial transform “leakage” in the wave number domain the array must extend well beyond the source so that the sound pressure has decayed to an insignificant level near the edges of the array.

Statistically optimized near field acoustic holography SONAH (and HELS) is an variant of NAH that has the great advantage of avoiding spatial transforms and thus the mentioned truncation effects; therefore the measurement array can be smaller than the source. HELS and SONAH are almost similar [20].

The demand on number of measurement points, the sensor spacing and the measurement distance is not very practical. In a study by GM is showed that even for very simple geometries the measurement distance must much smaller than recommended by the given theory and the number of measurement point had to be larger than theory suggested [19].

To be able to reliable measure a mode shape in a 42cm×23cm 3mm thick plate below 1kHz, one needs 240 pressure measurements, 8mm measurement distance and 2cm sensor spacing.

**Microflows in HELS and SONAH**

The p-u method performs better than any other method when the signal-to-noise ratio is poor; otherwise the best solution is to predict the pressure from the pressure and the particle velocity from the particle velocity [21].

The tendencies observed with SONAH predictions are that velocity-to-pressure predictions perform better than pressure-to-velocity predictions, in particular if the transducers are not perfectly matched. The explanation is undoubtedly the same as with NAH.

A simulation study has shown and experimental results have confirmed that SONAH with advantage can be based on measurement of both quantities (pressure and velocity), in particular if the transducers are less than perfectly matched, since it in general is better to predict the sound pressure from the sound pressure and the particle velocity from the particle velocity than to predict either quantity from the other quantity. Measuring both sound pressure and particle velocity has the additional advantage that it makes it possible—within limits—to reduce the influence of sound coming from the “wrong” side of the measurement array. In this case both the pressure and the particle velocity should be determined from both quantities [21].

But again (just as the previous paragraph), why calculate the particle velocity and sound pressure if you can measure them directly? The
calculation is bandwidth limited, the requirements are extreme and the result is not very accurate.

8.7 **IBEMS NAH; a Helmholtz method**

The IBEM method is very labour intensive and it has no practical use found in industry yet. One application might be in the verification boundary element simulations. Therefore, the method is only discussed briefly here.

The first numerical example of using IBEM to reconstruct an acoustic field inside an enclosure was shown by Gardner and Bernhard in 1988. The general IBEM codes for arbitrary geometry in an exterior region were developed in the early 90's.

To reconstruct acoustic quantities on arbitrary source geometry, one can utilize the Helmholtz integral theory that correlates any field acoustic pressure to the acoustic pressure and normal component of velocity on a source surface.

The most popular numerical implementation is via boundary element method (BEM) that discretizes an arbitrary source surface into small elements and then solves acoustic quantities on the discrete nodes defined on these elements.

Because in this case source information (cause) is sought based on measurement of acoustic pressure or velocity (effect), it is an inverse acoustic problem and its numerical implementation is known as inverse boundary element method or IBEM for short.

The main advantages of IBEM are the reduction of problem dimensions by one and its flexibility in selecting measurement locations. However, IBEM has several inherent difficulties.

A major drawback of IBEM is its requirement of a source surface mesh, which may not be available in engineering applications.

Since IBEM relies on a spatial discretization of an acoustic quantity, it is necessary to have at least six nodes per wavelength in order to ensure a satisfactory reconstruction of an acoustic quantity. Consequently, the upper frequency dictates the number of discrete nodes necessary in a source surface mesh.

To preserve the integrity of a reconstructed image, one must take a comparable number of measurements as the discrete nodes on the surface mesh. This can be problematic in practice because the number of discrete nodes for an arbitrary source running at low-to-mid frequencies may be quite large, which can make the reconstruction process complex and time consuming.

First the physical position of the microphones and the structure has to be put into a model. Then at the surface a certain number possible noise sources are defined. For all the microphones the total effect of these possible noise sources is calculated.
Fig. 8.12: If the sound sources of a hairdryer have to be determined, first a model of the hairdryer is made up and a measurement grid is defined. From every point of the model of the hairdryer the acoustic transfer function to every measurement point of the grid determined from [6].

The result of all these transfer functions is of the following form:

\[ P = A \cdot S \]  \hspace{1cm} (8.10)

With \( A \) the matrix of transfer functions from every point of the object under test (\( S \)) to the measurement grid (\( P \)). Once the measurements are performed one is able to identify the noise sources by calculating the inverse of the matrix \( A \):

\[ S = A^{-1} \cdot P \]  \hspace{1cm} (13.11)
As the distance from the noise sources becomes closer it becomes more easy to invert the matrix A. Therefore one has to measure in the near field of the noise sources. Traditional methods use pressure sensors but with the Microfown one is also able to measure the velocity field. This has a number of benefits.

**Microfowns in IBEM**

In the Ph.D. thesis of Visser [6], the use of Microfowns result in a better reconstruction. IBEM has no low frequency limit. It allows for reconstruction at any value of low frequencies and can yield satisfactory results with relatively few measurements.

### 8.8 Direct method Measurements

![Image](image-url)

**Fig. 8.13:** The pu based acoustic camera is used as trouble shooting tool in the automotive.

**Acoustic problem finding of a micropump**

With a PU match acoustic camera the noise sources of a good and faulty pump are measured. As can be seen in Fig. 8.14, when the bearing is damaged the pump starts to radiate from the front and the back. Further information on this measurement can be found in chapter 17: “end of line control: Gears & motors”.
Near Field Acoustics

Fig. 8.14: left an acoustic photo of a good pump, right a photo of a faulty pump.

**Helicopters interior noise testing** [8]

Noise levels recorded in the helicopter cabin are severely affected by the strength and vicinity of noise sources. Jet engines, the gearbox and the rotors can be considered as separated sources - whose spectral content is strongly tonal and rpm dependant - exciting simultaneously the cabin acoustic cavity.

Measuring acoustic data on helicopters is a difficult matter because of the multiplicity of source, their high correlation (all sources are dependant on the rotational speed) and the high reflectivity of the acoustics field in cabin. Traditional testing techniques as Acoustic Intensity or Near Field Acoustic Holography fail correctly addressing the problem.

The Direct Method proves able to deal with harsh environmental testing conditions while improving measurement accuracy and paves the way for a better understanding on how to reduce noise level in helicopter cabins.

Fig. 8.15: Left: EUROCOPTER EC-135, right: PU intensity map of the interior [25].

Fig. 8.16 shows two of the typical noise maps obtained with the new testing technique. The extremely valuable feature of the Microflown array is
Near Field Acoustics

that the acquisition can be carried out in the wide spectra frequency range. There is no need to modify the probe, as it is the case for intensity probe using microphones (spacers dependency form the frequency content of the acquired signal). In the case under analysis only one test run was sufficient to make data all available for processing in the range 0Hz to 25kHz. The standard intensity probe would have required at least 3 different runs corresponding to the 3 different spacers. Furthermore, fee Microflown array works as a wide frequency intensity that allows showing at the same time both low frequency components (e. the main rotor BPF) and the high frequency components such as the turbines.

![Fig. 8.16: Interior noise maps.](image)

**Sound field visualisation of a thin plate**

This is a summary from the application note ‘Surface velocity versus sound intensity visualization’.

For an experiment the differences between surface velocity and sound intensity visualisation in a 33x43cm, 0.8mm thin plate are measured.

The measurement is done with a PU probe on 3.5 centimetre distance of the plate. For every square on the panel, one measurement is done (180 measurements). The plate was excited with white noise and the input signal is used as reference. All recorded data is processed with MATLAB software. Results are given below.

To be able to visualize the phase of the velocity measurements, the input signal of the loudspeaker is used as reference. This is because the measurements are done point by point. If the measurements are done all at the same time (with an array), this reference signal is not required. For the intensity measurements no reference signal is required because the intensity itself has a sign indicating if the measured values have an inward an outward energy flow.

Dynamic range of the intensity measurements is in the order of 65dB, the dynamic range of the velocity measurements is in the order of 48dB.
See also chapter 7.12: ‘Simple method for finding modes in a thin plate’.

Fig. 8.17: Measurement set up.

Fig. 8.18: Visualisation of the velocity (left) and intensity (right) sound field at 53Hz.
Fig. 8.19: Visualisation of the velocity (left) and intensity (right) sound field at 141Hz.

Fig. 8.20: Visualisation of the velocity (left) and intensity (right) sound field at 240Hz.

Fig. 8.21: Visualisation of the velocity (left) and intensity (right) sound field at 240Hz.

Fig. 8.22: Visualisation of the velocity (left) and intensity (right) sound field at 486Hz.
8.9 Acoustic eyes (near field)

The ‘acoustic eyes’ is a name that is given to a way of source localization that makes use of two (or more) 3D intensity probes.

The idea is that an intensity vector is aiming at the loudest sound source that is measured. If two intensity vectors are measured on separated locations, two vectors are pointing towards one point: the loudest sound source, see also Fig. 11.1. In this way the source direction and the source distance can be determined.

The idea is tested on a two way loudspeaker with a positive result: for lower frequencies the bass loudspeaker was found and for higher frequencies the tweeter was found.
Fig. 11.2: The intensity field of a two way loudspeaker. For low frequencies the bass loudspeaker is emitting (left). Right: at higher frequencies the tweeter emits (from Budapest University, Vibroacoustics Lab.)

One issue that always pops up in discussions is the problem of two spatial separate but identical sound sources. If two identical sources emit sound, the intensity vector will aim in between the two sources if the probe is located exact between the sources. And with that the method fails.

Fig. 11.3: The intensity field of the bass loudspeaker as shown in Fig. 11.2, arrow representation. As can be seen: all arrows aim towards the source. (From Budapest University, Vibroacoustics Lab.)

Fig. 11.4: The intensity field of the tweeter as shown in Fig. 11.2, arrow representation. As can be seen: all arrows aim towards the source. (From Budapest University, Vibroacoustics Lab.)

First of all it is a very theoretical problem that solves itself if the method is exercised by hand: only a small distance variation will cause one of the
identical sources to be dominant. This is because the intensity field normally decays with the distance squared.

One can think of other solutions to get around this problem.

1) One can make a phased array (with only two pressure transducers). This solution has a very poor directivity but it should be enough to find the difference between two identical separated sources or one source.

2) If the sources are far field, it is possible to make a microphone with a variable polar pattern (it is also possible near field but then it is more difficult to explain). The pressure microphone has an omnidirectional polar pattern and the Microflown has a figure of eight polar pattern.

The axis of zero sensitivity varies when a summation is made with the sound pressure \( p \) and particle velocity \( u \).

If the line of zero sensitivity is defined as 90° for a pure velocity signal, the line of zero sensitivity is 0° for \( p+u \) and 180° for \( p-u \). So the line of zero sensitivity is ‘steered’ with the ratio of \( p \) and \( u \) in the summation, see Fig. 11.5.

This line of zero sensitivity is very sharp so it is possible to find out if there are one, two or even more identical sources.

Fig. 11.5: If the pressure signal and velocity signal are summed, the line of zero sensitivity can be ‘steered’.

**8.10 Acoustic noise source finder (based on [24])**

An apparatus is developed that is able to find acoustic sound sources in the near field of a radiating object which is operating in a noisy environment. It is based on two orthogonally placed Microflowns. The complete system is working with battery powered analogue circuitry, it is therefore very small and handheld. One Microflown is used to listen to a source whilst rejecting the background noise and another Microflown is used to create a stereo sense.
In ‘difficult’ sound fields, e.g. diffuse sound fields or a situation with a lot of sound sources, it is a difficult task to find specific sound sources. In the near field of sources the use of a Microflown is advantageous over a pressure microphone because of three reasons:

4. The sound pressure level and particle velocity level are of similar magnitude in the free field. If the sound wave reflects on a rigid surface, the sound pressure doubles and the particle velocity reduces to zero. Therefore the Microflown will not pick up much of the background noise. The sound pressure microphone will pick up this (doubled) noise.

5. Vice versa, close to a vibrating (sound emitting) surface, the sound pressure level is reduced compared to the particle velocity level perpendicular to the surface. Therefore the sound pressure microphone will pick up less signal level from the source than the Microflown.

6. A sound pressure microphone is omni-directional and thus measures the sound field in all directions. A Microflown measures the particle velocity in one direction. Therefore when measuring in a diffuse sound field a Microflown measures only one third of the total sound field whereas a pressure microphone measures the total sound field [3].

The Acoustic Noise Source Finder (ANSF) consists of two Microflows placed orthogonally to each other. Microflown one aims in the forward direction and Microflown two is placed orthogonally so that the sensitivity is zero in the forward direction, see Fig. 11.6. On top a Microflown can be seen, the sensitivity direction is in line with the printed circuit board (S1). The other element is placed on the other side and has sensitivity directions of S2.

The signal of Microflown one (S1) is amplified and connected to headphones. Because of the three effects that are mentioned before, Microflown one has the feature that it rejects the background noise and ‘amplifies’ the source signal.

So with this Microflown alone, one is able to find sound sources. The source is located at the position where the sound is at its loudest level. With the use of the second Microflown however it is possible to generate a signal that indicates direction of particle velocity. With some additional electronics there is now a possibility to send directivity information to the ears of a user. Once the two Microflows are moved over the sound source, the source signal is amplified of attenuated and distributed between the headphones left and right channel so that the location of the source can be found easier. How this signal is created is the subject of this paragraph.

The frequency filters are adjusted in such a way that the annoying sound is heard trough the headphone in an as small as possible bandwidth. When the source of the noise is on the left side of the sensors the voltmeter showing the average multiplied signal shows a negative voltage (meter is turned to the left), and vice versa when the noise source is on the right side. When the source is exactly in front of the sensor the meter will show “zero”, or “straight ahead”.

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Fig. 11.6: Photo of the sensor device.

Together with this voltmeter two variable gain amplifiers are steered in such a way that the user experiences the noise going from left to right when moving the noise source finder from right to left. This assists the process of finding the noise source. The source finding technique is similar to that used with intensity probes.

Fig. 11.7: Photo of a lab-prototype of the sensor device. The source is a Microflown that emits sound due to thermal effects, the 2D Microflown probe that is used here is shown in detail in chapter 3, in the paragraph on experimental Microflows.
Schematically the electronics used can be described as shown in Fig. 11.8.

The signal of Microflown one is amplified, band pass filtered and fed to a headphone section with averaging voltmeter. Band pass filters are needed because of two reasons. Firstly they enable the user to focus on a specific noise source, and secondly they are needed to get a useful signal from multiplying the signals.

The signal of Microflown two is also band pass filtered and then multiplied with the band pass filtered signal of Microflown one. The resulting signal is averaged. This results in a signal that is zero if the sensors are located exactly in front of the hot spot, a negative DC signal when the sensor is on the right of the hot spot and a positive DC signal when the sensor is on the right of the hot spot. This signal is fed into a voltmeter which can be used for indicating where to move the ANSF in order to locate the hot spot.

The multiplied and integrated signal is also used to operate two voltage controlled amplifiers feeding the audio signal to the headphones. The balance of the audio is changed by the DC signal. When the hot spot is on the left side of the ANSF the left headphone signal is louder and vice versa. Only when the hot spot is in front of the ANSF the signal is equal on both headphone channels.

There is a Microflown correction circuit; this is implemented to correct the Microflown frequency response in terms of a high frequency amplification to enhance audio fidelity.

**Measurements**

First the ANSF is tested in a quiet environment with only one source nearby the sensor. Next a comparing measurement between the use of pressure sensors and particle velocity sensors when measuring near a sound source is done. Measurements are repeated, but then with background noise present.
The two Microflown sensors are placed at a distance of 2.5cm from a plate with a sound source in it. Then a scan is made in a plane parallel to the plane with the source, Fig. 11.6, the plate can be seen fuzzy in the back. The source consists of a loudspeaker behind a hole with diameter of 2cm and is excited with white noise. In the case of these measurements a band pass filter is chosen with a pass band from 100Hz to 1kHz, which is a very wide frequency band and a more difficult test for the system than when a small band is used. A scan with 15 points in the horizontal direction and 9 points in the vertical direction is made, each point has distance of one centimetre between neighbouring points.

![Fig. 11.9: Measurement results of scanning a point source and plotting the averaged multiplied signal source. Left: with a horizontal sensitivity direction; Right: with a vertical sensitivity direction](image)

In a first measurement the sensitivity axis of sensor two (S2 in figure 1) is in the horizontal direction and after that in the vertical-direction. The result of the first measurement is depicted in Fig. 11.9 (left). As can be seen the amplitude of the signal is maximal when approaching the source, and changing sign when crossing the source. The similar is seen for the measurement second measurement result Fig. 11.9 (right).

![Fig. 11.10: Left: vector plot of measurements at 2.5 cm distance. Right: the values plotted are taken as (1/log(data)) for readability.](image)
Since measurements are done in the whole plane at a fixed distance both in the horizontal and vertical direction, this creates the possibility to display a vector plot of the measurements. This vector plot can be seen in Fig. 11.10 left. Clearly the arrows point towards the point source; however at larger distances the arrows are too small to recognize a direction. Therefore the same graph is printed in Fig. 11.10 right, but with a logarithmic scale of the values. At the edges the values do not point towards the source anymore, so there the noise level is too high. The sensor pointing to the plane with the source has barely signal strength because the component in this direction is very small; this is the explanation why the averaged multiplied signal does not point in the right direction.

At 6 cm distance the vector plot is much smoother and values all over the whole grid point toward the source, the signal is now much more constant over the scanning surface. See Fig. 11.11 for the 2-D vector plot of this data.

**Comparison between pressure and velocity signals**

A measurement scan over the source with one particle velocity sensor pointed towards the plane of the source and one pressure sensor is done to illustrate the difference in performance for finding a source.

The measured auto spectrum of both sensors (microphone and Microflown) is taken for each place in the scan and in a bandwidth between 100Hz and 200Hz. The results of the measurements are shown in Fig. 11.12. As can be seen in Fig. 11.12 left the pressure sensor signal shows a clear peak in the response when near to the source. The particle velocity sensor has a more sharp peak, which originates to the directive nature of the particle velocity sensor. Both methods are clearly useable. However in much cases background noise is present. So the scanning test is done again, but now with an external sound source generating background noise. Clearly the pressure signal now gives a blurred image in which the origin of
the noise source can hardly be determined anymore. The particle velocity sensor however gives a comparable signal as without background noise. See Fig. 11.12 right.

Another scan with the two sensor system is performed, this time with 5cm distance to the plane with the source and background noise present. The level of background noise is equal to the level used with the test described above. Together with direction signal (the averaged multiplied signal) the amplitude of the signal from sensor one, pointing towards the plane, is monitored. This time the filters are adjusted to a smaller bandwidth. In Fig. 11.13 both values are plotted against the distance. Values are scaled to a good fit, since absolute values are not of importance here.

The source can be found at the maximum of the amplitude of the signal from sensor one and the changing of the sign of the direction signal. Here the use of the direction signal comes to its right; with help of the output signal of sensor one the source can be found globally, but the direction signal gives the exact point.

Fig. 11.12: Surface scan without background noise, surface scan with background noise.

Fig. 11.13: Measurement at 5cm distance and background noise present.
**Noise source finder, Mid-side technique**

Alan Dower Blumlein was born in England, 1903. He has become well known in the audio community for the creative nature of his stereo contributions. Among the many theories of stereo production proposed by Alan Blumlein perhaps his most innovative was a hybrid form of stereo microphone pickup: the mid/side technique. It was not until the 1950s, however, that this approach was realised in a practical way by the Danish State Radio [1].

The concept of this ingenious mid/side technique is simple. The Mid (or M) signal is the discrete monophonic pickup provided by a microphone of any kind, aimed with its most sensitive direction to the centre of the sound source. The Side (S) information is provided by a figure-of-eight microphone positioned so that the axis of minimum pickup is aimed to the centre of the sound source, see Fig. 11.14.

Fig. 11.14: A traditional MS assembly consisting of an omni-directional microphone providing the M (mid) signal and a figure-of-eight microphone providing the S (side) signal. Right the result of the MS technique.

Fig. 11.15: A MS assembly consisting of two figure-of-eight microphones (Microflown). Right the result of the MS technique.
8.11 References


[17] Arnaud Duval et al., Vehicle Acoustic Synthesis Method 2nd Generation: an effective hybrid simulation tool to implement acoustic
lightweight strategies, Journée SFA / Renault / SNCF, November the 30th 2005.


[22] T. Basten, HE de Bree, and E. Tijs, Localization and tracking of aircraft with ground based 3D sound probes, ERF33, Kazan, Russia, 2007

