Noise source identification techniques: simple to advanced applications

The number of noise source identification (NSI) techniques available to engineers working on noise, vibration and harshness problems has increased considerably in recent years. The choice of the most appropriate technique depends upon the application and the information required. This paper reviews techniques for noise source identification and quantification ranging from simple hand-held sound intensity systems, hand-held array systems to large ground based microphone arrays. The methods include Beamforming, Spherical Beamforming and Acoustic Holography. Guidelines are given to help the engineer choose a suitable technique based on the frequency range of interest, the distance from the measurement array to the test object and the resolution required. Practical application examples ranging from hearing aids to wind turbines are presented to illustrate the various NSI techniques.

1   Introduction

Noise source identification (NSI) techniques are used to optimise the noise emission from a wide range of products including vehicles, household goods and wind turbines. The goal of NSI is to identify the most important sub-sources on an object in terms of position, frequency content and sound power radiation. Ranking of the sub-sources can then be used to identify where design changes will most effectively improve the overall noise radiation. With a considerable number of techniques available, NVH engineers might have need of an overview to help select the most appropriate solution. This paper gives some guidelines to help the selection together with some practical applications of NSI techniques. Emphasis is placed on the useful frequency range and the resolution of the various methods.

2   NSI techniques

A number of NSI techniques use just one or a few transducers. Examples are sound mapping based on sound pressure, sound intensity and selective intensity.

Array based techniques using a dozen to up to hundreds of microphones are where the greatest developments are taking place. The main methods are:

- Near-field Acoustic Holography related techniques (STSF, Non-Stationary STSF, SONAH)
- Beamforming (Phased Array technique)
- Refined beamforming (deconvolution methods)
- Moving source beamforming
- Spherical beamforming

The technique selected by the user will depend on the required frequency range, the measurement distance, the required resolution and the area covered by the measurement.

3   Sound mapping

The most straightforward method to detect a noise source is to map the Sound Pressure Level (SPL) at various locations around the noisy product. The SPL is measured with a single channel FFT or 1/3 octave analyser at each point of a defined grid. The measured data is then used to produce a contour map that gives a rough idea of the location and characteristics of the source. The method is rarely used nowadays as it has mainly been superseded by sound intensity mapping. The main benefit of sound pressure mapping is the low cost involved. The limitations are that the method is time consuming, is particularly susceptible to the influence of background noise and can only be applied to stationary noise sources.

Since the 1980's the sound intensity technique based on a phase-matched pair of microphones, has been used to measure the acoustic energy flow. This yields not only the amplitude but also the direction of the sound energy. The method has been incorporated into a number of international standards to determine sound power.

Sound intensity mapping involves the measurement of sound intensity spectra at a number of discrete points on a grid close to the object under test. A contour plot is then produced and superimposed on a photograph to enable identification and documentation of the noise sources. To speed up the process a robot is often used to move the sound intensity probe from position to position.

The main benefit of sound intensity compared to sound pressure is that it is a vector quantity. The acoustic field can thus be represented with a magnitude and a direction. Thus it is possible to determine sound power of a source even in the presence of background noise. The technique is now mature so that complete sound intensity mapping systems are available as a 2 channel sound level meter. The spatial resolution of a sound intensity map is limited by the wavelength of sound and the distance between the measurement points in the mapping grid.

Selective intensity calculates that part of the full measured sound intensity that is coherent with a specific reference signal. If, for example, the vibration of a specific component is suspected to be the main cause of the radiated noise, then an accelerometer mounted on that component can be use to provide a reference signal for the selective intensity calculation. If the suspicion is correct then the selective intensity will be close to the full sound intensity observed. The reference signal may be of any nature: acoustic, vibration, force, electrical etc whichever provides the cleanest and least noisy representation of the suspected cause. The benefit of selective intensity compared to traditional sound intensity measurement is that it permits a more precise localization of the sound source.

4 Nearfield Acoustic Holography

Spatial Transformation of Sound Fields (STSF) was one of the first commercially available noise source identification techniques based on nearfield acoustic holography [1]. The technique requires a regular grid of microphones together with a number of references transducers. Measurements of autospectra and cross spectra are made over a plane which completely covers the test object. Using principal component decomposition techniques, a model of the acoustical field is generated from which all acoustical parameters (pressure, particle velocity, intensity, power) can be calculated both closer to and further away from the test object. The calculations are based on 2D FFT and are very fast. However there are limitations due to the fact that unless the test object is measured entirely, artifacts occur in the resulting mapping. For sources which are not stationary in character, a transient method in the time domain was developed known as Non-Stationary STSF.

5 SONAH

The STSF and Non-Stationary STSF methods have been further developed to overcome some of their practical limitations. A resultant technique is known as Statistically Optimised Nearfield Acoustical Holography (SONAH) [2, 3]. The idea consists in fitting a plane wave model to the measurements using a linear decomposition. In case of a double layer of sensors, it is also possible to differentiate between sources in front and behind the microphone array.

The main benefits of SONAH, compared to standard NAH, are the possibility to use a non-uniform grid of microphones, a low sensitivity to sources outside of the calculation plane and better performance at low frequencies.

The low sensitivity to sources outside of the calculation plane makes it possible to perform local measurements; therefore with SONAH, there is no need to use an array which entirely covers the object under test, as is required for traditional NAH. Furthermore, the applicability of SONAH to an irregular grid of microphones provides more flexibility. In particular, different microphone array techniques can be combined in the same system. For example, combining SONAH with beamforming processing with the same microphone array system provides an extended frequency range.

5.1 Conformal calculations

Local measurements using SONAH open up the possibility to perform conformal calculations. This means that quantities such as sound pressure, intensity or velocity can be calculated on the surface of the test object.

For even more accurate conformal mapping, following the surface details of the object, it is necessary to acquire a model (importing or digitizing) of the test object on one hand, and on the other a precise location of the microphone array.
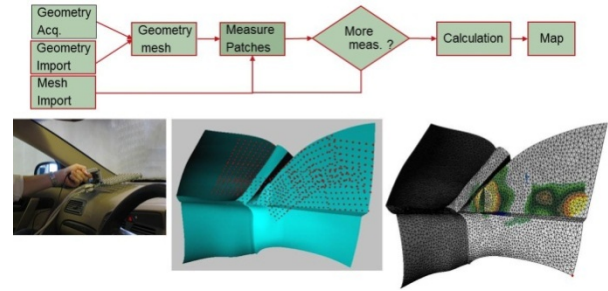

Fig. 1: Conformal mapping workflow: Import or acquisition of the geometry, mesh generation, patch measurements, calculations and displays

In order to reduce the preparation time and to simplify the measurement process, a tracking system based on infrared sensors is attached to the microphone array handle, to obtain real-time positions and the orientations of the array during the measurements. The workflow is summarized in figure 1. In this process, it is necessary to assume that the noise source characteristics are stationary [3].

Using conformal maps reduces the risk of misinterpretation of the data. Figure 2 shows two examples of applications.
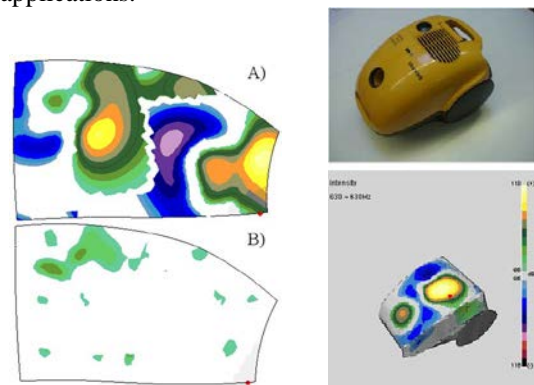

Fig. 2: Left: Measurements on a car window during wind tunnel tests. A: with mirror, B without mirror, intensity at 125 Hz 1/3rd octave. Right: Vacuum cleaner, intensity 630 Hz 1/3rd octave.

5.2 Panel contribution

Noise source identification techniques, can be used beyond their traditional use, namely localizing acoustic sources or estimating the distribution of acoustic quantities on extended areas. One example is panel contribution based on SONAH.

The purpose of this technique [4] is to estimate the contributions of some areas to the acoustic pressure at a chosen point in a cabin. A practical case is for example the acoustic contribution of the different inner parts of a car cabin to the sound pressure level at the ear of the driver.
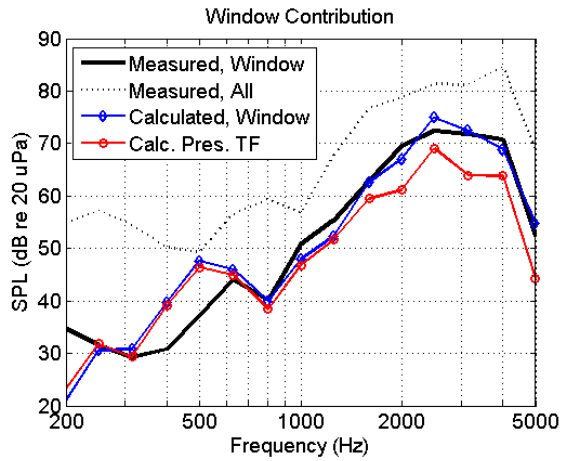
Fig. 3: Pressure contribution at the driver's ear position of a flat loudspeaker attached on a window of a car cabin. The measured true contribution corresponds to solid black curve, the calculated contribution correspond to solid blue curve. For the red curve only the operational particle velocity is used.

Panel contribution based on SONAH estimates the contribution of a panel ΔS to the acoustic pressure at a target point using pressure and particle velocity [4].

Two steps are required. Firstly, the transfer functions between a source at a target position and the panel ΔS using SONAH are measured. Secondly, the operational pressure and sound velocity on the surface ΔS are obtained using SONAH again.

To illustrate this technique, figure 3 shows the contribution of a flat loudspeaker attached to a window of a car cabin, in presence of background noise. Agreement between the true and calculated contributions is good, except below 630 Hz, where the signal level provided by the flat loudspeaker is too low.

5.3   Entering intensity

A different example of application using SONAH is the determination of the entering intensity in a cabin configuration [5]. The purpose is to distinguish between the different intensity components close to walls. In particular, it is of interest to estimate what is actually entering into the cabin from outside the vehicle. Another interesting aspect is the estimation of the panel absorption.

It is assumed that the sound fields due to radiation (entering intensity) on one hand and due to absorption on the other hand are mutually incoherent. This incoherence assumption can be reasonable in case of very large number of incoherent excitations, such as, for example, those created by a turbulent flow around an aircraft fuselage.

A sound intensity probe cannot distinguish these different components, but will measure instead the total intensity. Using SONAH with a double layer microphone array (DLA), it is possible to separate these different contributions [5].

To evaluate the Entering Intensity technique, tests were performed in a transmission suite facility, emulating the noise conditions around an aircraft window.

In the figure 4 (same scale, for a single octave band), the middle display shows a high level of absorbed power compared to the radiated power. The Right display shows that the result is close to the reference case when applying the Entering Intensity technique.
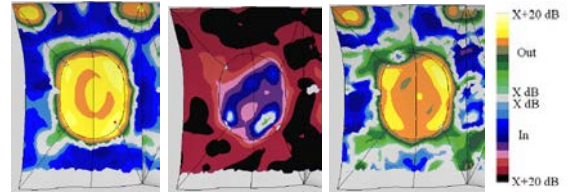

Fig. 4: SONAH calculations. Left: Transmission loss noise without emission from loudspeakers (reference). Middle: Transmission loss with emission from loudspeakers. Right: Transmission loss with emission from loudspeakers, but with Entering Intensity processing applied.

6   Beamforming

Beamforming has become a popular technique for noise source identification for exterior vehicle noise (cars, trains and aeroplanes) and also in wind tunnels on both models and full sized vehicles.

Beamforming is somewhat similar to a camera in that an array of microphones in combination with beamforming calculations behaves as a lens. The technique is based on a delay and sum principle, where an array of microphones is used to capture the sound field [6]. The array itself is usually plane although a funnel shaped array may be used in order to suppress background noise arriving from the rear of the array. The signals are then connected to a signal processor to determine the directional characteristics of the incoming sound. The beamforming array is most sensitive to sound arriving from a focus direction. This region is known as the main lobe. Sound from other directions will also be detected to a certain degree; these regions are known as the side lobes. A good array design is characterized by having a large difference in the sound levels measured at the mainlobe and at the highest side lobes. The greater this difference, (known as the Maximum Sidelobe Level) the better the array is at reducing spurious peaks known as ghost images. Arrays with a regular grid of microphones are notorious in producing ghost images when the spacing between the microphones is larger than half a wavelength. Ideally, the array should have the microphones randomly distributed but this is not feasible for a practical system. A useful compromise is to build an array with identical segments, each segment containing a random distribution of microphones in order to optimize the performance of the beamformer over a wide frequency range.

7   Refined beamforming

Delay and sum beamforming has a spatial resolution of about a wavelength. However if the source under test can be modelled by a finite number of non-correlated point sources, a refined beamforming technique based on deconvolution algorithms such as NNLS (Non-negative least squares) and DAMAS (Deconvolution Approach for Mapping Acoustic Sources) [7] can be used to improve the spatial resolution by a factor of typically three to ten.
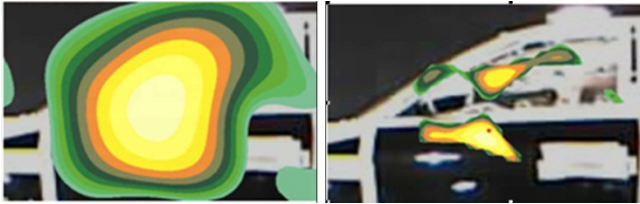
Fig. 5: Sound Intensity maps based on beamforming measurements in a wind tunnel. Left: delay and sum method. Right: Refined beamforming method.

8   Moving source beamforming

The moving source beamforming method can be employed when the source under test is in motion, such as a vehicle or train pass-by, an aircraft fly-over or a wind turbine [8]. In these situations, it is necessary to take into account the Doppler effect, turbulence effects and industry specific representation of the results. Where large ground based areas are employed, for example for aircraft fly-over, the correlation length between the microphones needs to be considered; a frequency dependent shading is useful in these cases to utilize the entire array area at low frequencies and a reduced central area at high frequencies.

For open sources such as lorries and trucks, special line displays and sliced cubes have been employed to locate sources on the vehicles. Moving source beamforming measurements have been successfully added to standardised pass-by measurements on test tracks, thus enabling type testing and research and development work to be run in parallel.
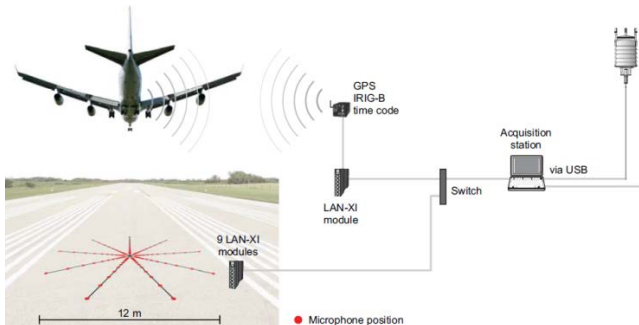

Fig. 6: Typical system for measurement and data acquisition for Noise Source Identification during flyover of passenger aircraft using Moving Source Beamforming.

9   Spherical beamforming

Spherical beamforming is the extension of planar beamforming to spherical arrays. Thanks to the array shape, no preferential directions are considered; therefore we are able to look at all directions around the sphere. This makes this type of array eminently suitable for enclosures, such as rooms or cabins. Obviously, it can be applied to free-field conditions as well.

Different types of techniques are applied to process the signals on the microphones. One typical and robust technique is based on spherical harmonics functions; usually referred to as called Spherical Harmonics Beamforming [9]. The sound field is sampled by the microphones on the sphere, and decomposed into spherical harmonics functions of different orders. Based on this decomposition, a directional function can be derived, to estimate the contribution from a specific direction. Applying this process to all directions around the sphere provides the acoustic map.
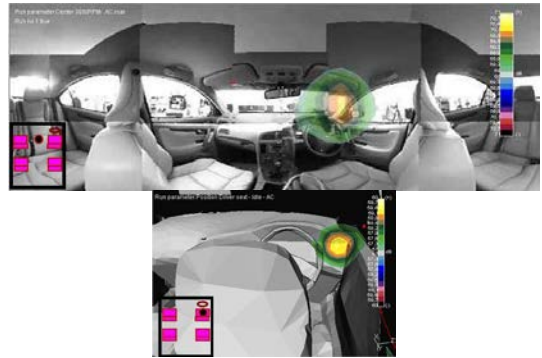

Fig. 7: Pressure contribution based on Spherical Harmonics Beamforming measurements in a car cabin. Air conditioning noise Top: Engine at 3000 RPM, alignment based on pictures. Bottom:idle condition, conformal Spherical Beamforming based on cabin model.

As for (planar) beamforming technique, limitations occur in terms of resolution and dynamic range. The resolution is mainly governed by the sphere's radius: the larger the sphere, the better is the resolution for a given frequency. To increase the resolution for a selected radius, a hard (closed) sphere is preferable. In the context of Spherical Harmonics Beamforming, a hard sphere makes the processing more stable. The dynamic range, related to the highest sidelobe level, is determined by the number of microphones and their distribution on the sphere. Examples are presented in figure 7.

10   Sound Quality Metrics Mapping

Mapping sound intensity permits the localization of noise sources. However, to understand the effect of the noise source on a person, it is usually more appropriate to map sound quality metrics such as loudness, sharpness or impulsiveness [10]. The arrays required for such mapping are the same as for SONAH or beamforming, only extra processing needs to be added.

11   Selection of an NSI technique

Beamforming techniques assume that the sound field can be modeled by typically plane waves or point sources, whereas NAH using an array and sound mapping using a single probe make no such assumptions. Therefore, broadly speaking, at measurement distances far from the source (several wavelengths), beamforming techniques can be used, whilst close to the source (fractions of a wavelength) NAH techniques are applicable.

The lower and upper limiting frequency of interest gives an indication of the techniques which can be used (Fig.8).

Technique	Lower limit, f_{min}, Hz	Upper limit, f_{max}, Hz
NAH	c/D	c/(2dx)
SONAH	c/(8D)	c/(2dx)
Beamforming	Lc/(DR)	See note
Spherical Beamforming	> c/(aR$_\theta$)	See note

c = speed of sound; D = diameter of array;
dx = average spacing between measurement points
R = resolution; R$_\theta$ = resolution in radians
L = distance of array from source; a= radius of array
Note: upper limit can be improved by processing. Optimisation of microphone positions can improve the Maximum Side Lobe level which determines the fmax.

Fig. 8: Lower & upper frequency limits of arrays

Roughly speaking, the lower the frequency of interest, the larger the array must be. The upper frequency limit is set by the average spacing between the microphones. However, the upper limit for Beamforming and Spherical Beamforming can be increased by using different algorithms and by an optimization of the microphone positions to adjust the Maximum Side Lobe pattern.

The resolution obtainable for NAH/SONAH is typically equal to the average spacing between the microphones. For Beamforming, the resolution depends on the wavelength, the nature of the sound source (coherent or non-coherent), on the algorithm employed and also on the size of the array (Fig.9). For the classical delay and sum algorithm, resolution is about one wavelength. This can however be improved by employing a deconvolution technique such as refined beamforming [7].

Another factor to be considered is the measurement area covered by the array (Fig. 10). For NAH, the array must be large enough so that virtually all acoustical energy passes through the array area. In practice this means that the array must often be of greater dimensions than the object under test. For SONAH, even though the array only covers part of the source, good representation of the sound field can be obtained inside the covered area. For spherical beamforming with omnidirectional directivity, provided that the measurement distance is more than about twice the radius of the spherical array (to avoid near field effects), there is no limitation to the area covered.

Technique	Planar Combo Array Spatial resolution, R, metres	Handheld Array Spatial resolution, R, metres	Spherical Array Angular resolution, R$_\theta$ degrees
NAH/SONAH	Average spacing between micro's	Average spacing between micro's	Average spacing between micro's
Beamforming Coherent sources	= max(λ; 1,4 Lλ/D)	= max(λ; 1,4 Lλ/D)	180/ka
Beamforming Non-coherent sources	= max(0,7λ, Lλ/D)	= max(0,7λ, Lλ/D)	180/(1,4ka)
Refined Beamforming Non-coherent sources	= λ/3 to λ/10	= λ/3 to λ/10	To be investigated

k = wavenumber; a = radius of array; λ = wavelength
L = distance to source; D = diameter of array

Fig. 9: Resolution of various acoustical arrays

Technique	Measurement area covered by the array	Measurement distance, L
NAH	≤ 0,9 x Area of array (see note)	= dx
SONAH	≈ area of the array	= dx
Beamforming	≈ +/- 30 degrees from array axis	≥ 0,7D
Spherical Beamforming	No limitation	≥ 2a

D = diameter of array; a = radius of array
dx = average spacing between measurement points
Note: when the NAH array is larger than the source under test

Fig. 10: Measurement area & distance

12   Selection of an array design

Once the engineer has decided upon an NSI technique, there is still the question of selecting the best array design. If NSI needs to be performed inside a vehicle cabin, then it could be advantageous to use a double layered array for SONAH in order to minimize the effects of the reverberant field or a spherical array for Beamforming. In engine test cells and other nearly free field situations, a planar array would be adequate; if such an array is designed to be used with both SONAH and Beamforming, it is usually termed a Combo array. Outdoor measurements, such as measurements on wind turbines and trains, are prone to interference from extraneous sources. In such cases a foldable array in the form of a funnel, can be used to reduce the influence of sound arriving from the rear of the array. For some outdoor measurements such as large wind turbines and fly-over beamforming, the necessary array is so large that it has to be mounted on the ground. Each microphone should be situated on a hard reflective surface (e.g. the runway tarmac or large hard plate) to avoid the effects of ground impedance as a function of the angle of incidence of the sound. For measurements on small objects such as hearing aids and dental drills, the array could consist of no more than two microphones: a stationary reference microphone and a probe microphone moved from position to position in a very fine mesh by means of a robot.

Conclusion

The list of techniques available for the NVH engineer to perform noise source identification is continually expanding. This article reviewed and compared the most common techniques together with guidelines as to how and when they can be employed. For the engineer to optimally exploit the techniques the implementation should be simple; superfluous parameters should be hidden from view and engineering expertise should be embodied in the solution so that the engineer can concentrate on the task in hand: noise source identification.
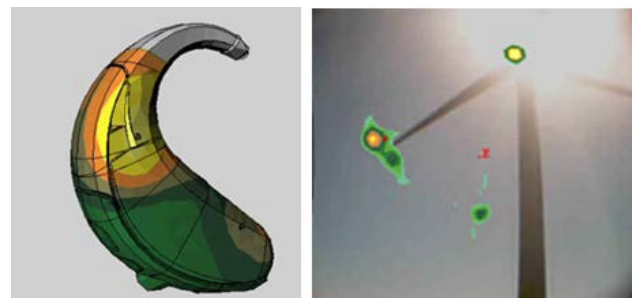
Fig.11. Left: Conformal sound intensity map on a hearing aid at 2400Hz. Right: refined beamforming sound intensity map on wind turbine at 4000Hz

References

[1] J.Hald: "STSF-a unique technique for scan-based Near-field Acoustic Holography without restrictions on coherences", Bruel & Kjaer Technical Review, nr.1 1989.

[2] J. Hald: "Patch Near-field acoustical holography using a new stastically optimal method", Proceedings of InterNoise 2003 (2003).

[3] J. Hald: "Patch holography in cabin environments using a two-layer handheld array with extended SONAH algorithm", Proceedings of Euronoise 2006 (2006).

[4] J. Hald, J. Mørkholt: "Panel Contribution Analysis in a vehicle cabin using a dual layer handheld array with integrated position measurement", Proceedings of SAE Noise and Vibration Conference, 2009.

[5] J. Hald et al: "Array based measurement of radiated and absorbed sound intensity components", Proceedings of Acoustics'08 Conference, 2008.

[6] J. Hald: "Beamforming", Bruel & Kjaer Technical Review, nr.1 2004.

[7] S. Gade, J. Hald and B. Ginn: "Noise source Identification with increased spatial resolution used in automotive industry", Proceedings of Acoustics 2012 Hong Kong, 2012.

[8] K.B.Ginn: "Contribution analysis in pass-by testing using moving source beamforming", Proceedings of SAE Noise and Vibration Conference, 2011.

[9] K. Haddad, J. Hald: "3D localization of acoustic sources with a spherical array", Proceedings of Acoustics'08 Conference, 2008.

[10] W. Song et al.: "Improved noise source identification using sound-quality metrics mapping in vehicle noise measurements", Proceedings of SAE Noise and Vibration Conference, 2011.