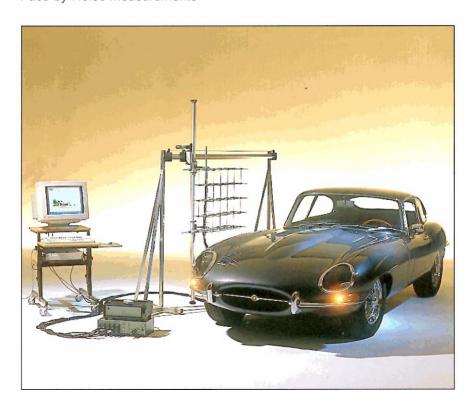
Technical Review

No.1-1995

Spatial Transformation of Sound Fields in the Automotive Industry Pass-by Noise Measurements





Previously issued numbers of Briiel & **Kjær** Technical Review

(Continued from cover page 2)

4-1985 Validity of Intensity Measurements in Partially Diffuse Sound Field Influence of **Tripods** and Microphone Clips on the Frequency Response of Microphones 3-1985 The Modulation Transfer Function in Room Acoustics RASTI: A Tool for Evaluating Auditoria **2–1985** Heat Stress A New Thermal Anemometer Probe for Indoor Air Velocity Measurements 1-1985 Local Thermal Discomfort 4-1984 Methods for the Calculation of Contrast Proper Use of Weighting Functions for Impact Testing Computer Data Acquisition from **Brüel & Kjær** Digital Frequency Analyzers **213112134** Using their Memory as a Buffer 3-1984 The Hilbert Transform Microphone System for Extremely Low Sound Levels Averaging Times of Level Recorder 2317 2-1984 Dual Channel FFT Analysis (Part II) 1-1984 Dual Channel FFT Analysis (Part I) **4-1983** Sound Level Meters — The Atlantic Divide

Special technical literature

3–1983 Fourier Analysis of Surface Roughness

Brüel & Kjær publishes a variety of technical literature which can be obtained from your local Briiel & **Kjær** representative.

Design Principles for Integrating Sound Level Meters

2-1983 System Analysis and Time Delay Spectrometry (Part II) 1-1983 System Analysis and Time Delay Spectrometry (Part I)

The following literature is presently available:

0	Modal Analysis of Large Structures–Multiple Exciter Systems (English)
0	Acoustic Noise Measurements (English),5th. Edition
0	Noise Control (English, French)
О	Frequency Analysis (English), 3rd. Edition
0	Catalogues (several languages)
0	Product Data Sheets (English, German, French, Russian)
Fu	rthermore, back copies of the Technical Review can be supplied as shown in

list above. Older issues may be obtained provided they are still in stock.

the

Technical Review

No. 1 - 1995

Contents

Use of Spatial Transformation of Sound Fields (STSF) Techniques	
in the Automotive Industry	1
by J.Hald	
Pass-by Noise Measurements	25
by Per Rasmussen	

Copyright @1995, Brüel & Kjær A/S

All rights reserved. No part of this publication may be reproduced or distributed in any form, or by any means, without prior written permission of the publishers. For details, contact: Brüel & Kjær MS, DK-2850 Nærum, Denmark.

Editor: K. Zaveri Photographer: Peder Dalmo Layout: Judith Sarup Printed by Nærum Offset

Use of Spatial Transformation of Sound Fields (STSF) Techniques in the Automotive Industry

by J.Hald

Abstract

Ever stricter legislation and competition compel vehicle manufacturers to reduce external vehicle noise. In order to simulate the actual working conditions of vehicles, acoustic testing must be performed not only under stationary but also under non-stationary conditions, e.g. run-up. This article deals with the use of Brüel & Kjær's Spatial Transformation of Sound Fields (STSF) technique as a tool for assessing and understanding the noise radiation phenomena, both qualitatively and quantitatively. Two different implementations of STSF will be described: the well known Cross spectral approach enabling a scan measurement to be applied on a stationary sound source and a new Time domain approach requiring all measurement positions to be acquired simultaneously. While the Cross-spectral STSF is a well established commercially available technique, the Time domain method is still under development.

Résumé

Une legislation de plus en plus sévère, a laquelle s'ajoutent des imperatifs de concurrence, pousse les constructeurs de vehicules automobiles a reduire le bruit que leurs produits imposent a l'environnement. Or, pour pouvoir simuler les conditions réelles de fonctionnement d'un vehicule, des essais acoustiques doivent non seulement être realises dans des conditions stationnaires, mais aussi dans des conditions non-stationnaires, de montee en regime, notamment. La présente communication decrit l'exploitation de la technique STSF Brüel & Kjær de transformation spatiale de champ sonore pour comprendre et evaluer, quantitativement et qualitativement, le phéno-

mène d'émission sonore. Deux différentes approches sont ici traitees. L'une intégrant la technique connue de l'Interspectre pour la mesure par balayage d'une source stationnaire, l'autre une nouvelle technique, dans le Domaine temporel, necessitant l'acquisition simultanée des données a tous les points de mesure. La premiere méthode par interspectre STSF est bien etablie et disponible sur le marché, tandis que la seconde est encore en cours de développement.

Zusammenfassung

Zunehmend strengere Gesetzgebung sowie der Wettbewerb zwingen die Fahrzeughersteller, externe Fahrzeuggerausche zu reduzieren. Um die tatsachlichen Betriebsbedingungen von Fahrzeugen simulieren zu konnen, sind akustische Priifungen nicht nur unter stationaren, sondern auch unter nichtstationaren Bedingungen (z.B. beim Beschleunigen) durchzuführen. Dieser Artikel beschäftigt sich mit der raumlichen Transformation von Schallfeldern (STSF), einer von Brüel & Kjær verwendeten Technik, die Erscheinungen der Schallabstrahlung qualitativ und quantitativ zu beurteilen gestattet. Es werden zwei Varianten der STSF beschrieben: das bekannte Kreuzspektrum-Verfahren, bei dem eine Abtastmessung auf eine stationare Schallquelle angewendet wird, und ein neues Zeitbereich-Verfahren, bei dem alle Meßpositionen gleichzeitig erfaßt werden. Während Kreuzspektrum-STSF eine bewahrte, auf dem Markt erhaltliche Technik darstellt, befindet sich die Zeitbereich-Methode noch in der Entwicklung.

Stationary noise: Cross-spectral STSF

The Cross-spectral STSF technique has been developed in close collaboration with research and development groups in the automotive industry. Basically, this STSF technique involves a two dimensional acoustical scan measurement in a plane close to a noise source and then transformation of the scan plane data to any other plane either closer to or further away from the noise source under test. 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 near field acoustical holography and the Helmholtz integral equation.

Since the sound field representation is based on only Auto- and Crossspectra, the stationarity assumption can be given the following practical formulation. The sound source must be sufficiently stationary during the measurement period such that all the necessary Auto- and Cross-spectral measurements are repeatable to a sufficiently high degree of accuracy.

The Cross-spectral sound field representation includes only parts of the sound field 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 analyzed. 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 proper set of reference signals.

Complete description of the sound field in three dimensions

Development engineers use the STSF technique to provide a far-field radiation pattern from a near-field measurement. This is particularly important when the noise source is in a laboratory or test cell and cannot be moved outdoors or to a very large anechoic chamber. Additionally, STSF can calculate the acoustic field closer to the noise source than the measurement plane. This can be important in those cases where it is not possible to make measurements physically close to the noise source because of rotating parts or very high temperatures. The near-field calculations can provide pressure, intensity or velocity maps which are very useful in locating noise generating regions.

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 [3]. Thus it is 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. Thus, "what if' scenarios can be enacted which enables the design engineer to decide how efforts are best applied in order to lower the far-field levels.

Measurement technique

In theory, an STSF measurement could be performed using a minimum of instrumentation consisting of a reference transducer, a scan microphone, a Cross-spectrum analyzer and a computer. Due to the quantity of data involved, such a measurement would be unacceptably slow. A practical system includes a set of reference transducers, an array of scan microphones,

an automated positioning system (i.e. a robot), a Cross-spectrum analyzer and a computer. A multichannel system is preferable not only because of the increased speed but also because the exclusion of references facility can be employed. This requires simultaneous acquisition of all reference channels.

Suitable instrumentation can be based on the Multichannel Analysis System Type 3550 or Type 3551 which will support the use of up to 16 channels. The software packages WT 9260 and WT 9263, for 3550 and 3551 respectively, support the use of a single-column array only.

As opposed to that, the software package Type 7688 for the Intelligent Data Acquisition (IDA) system supports the use of a general two-dimensional microphone array. The user just specifies the scan area size and the frequency range, and the computer will then automatically control the positioning of the array to cover the specified area with a sufficiently small spacing between the measurement positions. Fig.1 (see Front Cover) shows the IDA based STSF system and Fig.2 gives a schematic overview of the data acquisition hardware.

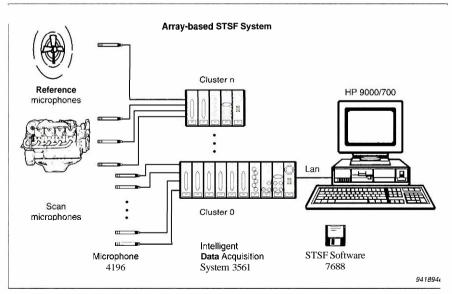


Fig. 2. Hardware overview for STSF system based on Intelligent Data Acquisition System Type 3561

The IDA system consists of a number of 6 channel modules, each with its own digital signal processor and on-board RAM for direct storage of raw time data. The 6 channel input modules are controlled by a Synchronization Module for recording data in up to 3066 channels simultaneously with full frequency range. The IDA system is connected to a workstation via a LAN interface (Ethernet). The STSF software controls the data acquisition, the positioning of the microphone array and the post-processing of the data.

The reference transducers can in principle be any kind of transducers but are most often either microphones or accelerometers. The absolute values of the reference signals are not important, as they only provide phase and coherence information. The reference microphones are used to distinguish between the different mutually uncorrelated partial fields in the sound field. Thus, the number of reference transducers must be at least equal to the number of significant, independent virtual noise sources present in the sound field.

Once a measurement has been completed, various types of calculations can be performed using Near-field Acoustic Holography (NAH) or the Helmholtz Integral Equation (HIE). Based on the measurement result in the scan plane, Fig. 3, the NAH can calculate the sound pressure, particle velocity and sound intensity in any plane parallel to the measurement plane. This also enables the 3-dimensional intensity vectors to be calculated. The acoustic far-field parameters such as SPL along a line or the radiation pattern can be calculated with the HIE.

The STSF system includes functions for Simulation of Source Attenuation. With this technique, the consequences of different noise reduction strategies can be predicted before actual implementation. The simulations are performed by mathematically adding different attenuation functions to the "vibration" velocities (i.e. the particle velocities) of the different surface areas of the noise source. This results in a new set of input data for the NAH and HIE calculations, enabling all the near-field as well as far-field parameters to be calculated for the new sound field. It is thereby possible to calculate, for example, the resulting sound pressure 7.5 m away from the source if the noise radiation of a part of the source is reduced. By comparing the effect of reducing the noise radiation from different parts, the optimum noise reduction strategy can be predicted before making the actual changes to the source.

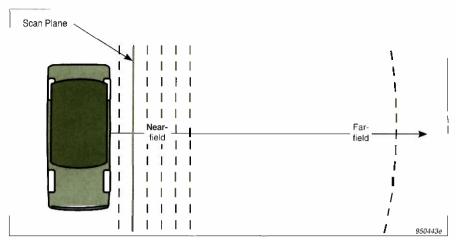


Fig. 3. Calculation of near-field and far-field, based on measurements made in the scan plane

Application examples

The STSF measurement technique has been applied to solve noise problems for a wide range of products. Here we will concentrate on a few application examples from the automotive industry, namely the application of STSF to measurements of whole vehicle exterior noise, measurements in wind tunnels and tyre noise measurements.

In the first example, the objective was to reduce the exterior noise level from a passenger car (described in detail in [4], by Nigel Taylor, Rover Group, UK, et al.). Traditional pass-by noise measuring techniques suggested that the main noise source was the exhaust orifice. Detailed analysis of the sound field using STSF showed that the engine air intake system was dominant. As a result the overall sound pressure level was reduced by minor adjustments to the intake system as opposed to major modifications to the exhaust system.

The noise field on the right side of a Rover 220 GSi was scanned with a microphone array The array of microphones was moved along the side of the car by a computer controlled microphone positioning system. At each array position, the Cross-spectra between all array microphones and a reference microphone were measured with a multichannel analyzer and all the spectra were then transferred to a workstation and stored. After the data

acquisition process, all relevant information about the noise field on the right side of the car was stored in the computer.

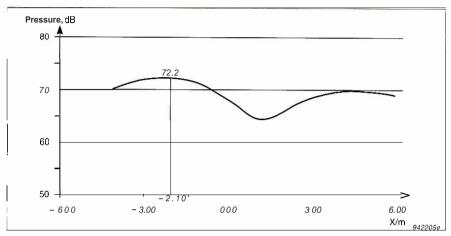


Fig. 4. Sound pressure level (100 Hz A-weighted) along a line at a distance of 7.5 m from the ear, with the Θ ar positioned at x=0

Fig. 4 shows the calculated A-weighted SPL at 100 Hz (bandwidth 20 Hz) along a line 7.5 m from the centre line of the car at a height of 1.2 m. It can be seen that the SPL has two maxima, one around 2.10 m behind the centre of the car and one 4.5 m in front of the car centre line. Intuitively, the highest peak occurring at the rear of the car could be associated with the noise radiated from the exhaust orifice and the smaller peak could be associated with the air intake orifice. Such an explanation, however, does not agree with the sound intensity distribution in the near-field. Fig. 5 shows the radiated intensity right at the surface of the car. Here it can be clearly seen that the dominant noise source is the air intake orifice and not the exhaust orifice. This apparent conflict between the far-field measurement and the nearfield measurement is caused by the fact that the exhaust and the intake are correlated sources and, as such, interfere with each other. This can be seen from a calculation of the intensity radiation away from the car towards the SPL calculation line. Fig.6 shows the intensity vectors away from the exhaust orifice and the intake orifice. Two "beams" can be identified: one away from the exhaust and one away from the intake. The interaction

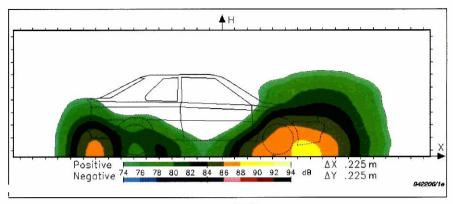


Fig. 5. Sound intensity radiation from side of oar at 100 Hz

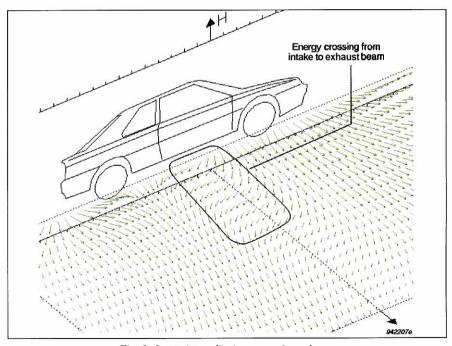


Fig. 6. Intensity radiation away from the car

between the two sources can be seen around $1-2\,\mathrm{m}$ away from the car, where energy from the intake radiation beam crosses over to the exhaust radiation beam and amplifies this.

The overall SPL of the car can be reduced either by reducing the noise from the exhaust orifice or by reducing the noise from the intake orifice. These two cases were simulated with the STSF Simulation of Source Attenuation technique and the SPL along a line was calculated. The reduction of the noise from the exhaust orifice was simulated by introducing a 3dB attenuation function in the mathematical STSF model of the area around the exhaust orifice. From this new acoustical model of the near-field, the resulting SPL along a line was calculated, giving the result shown in Fig. 7. By comparision with the original result in Fig. 5, it can be seen that a reduction of the exhaust noise by 3dB only reduces the overall maximum SPL by 0.9 dB. Similarly, the effect of reducing the noise from the intake was simulated by applying a 3dB attenuation function to the area around the intake orifice. The SPL along a line calculation, Fig. 8, shows that this results in a reduction of the overall maximum SPL by 1.8 dB. It can therefore be concluded that a reduction of the intake noise has a greater effect on the overall SPL than a similar reduction of the exhaust noise. This is also in agreement with the results from the near-field sound intensity calculations, which showed that the major contribution to the far-field SPL was coming from the intake.

The second example of applications of the STSF measurement technique for automotive noise reduction concerns wind-tunnel measurements (described in detail in [5], by M. Nakamura, Honda R& D, Japan, et al.). As other noise sources like engine noise and tyre noise are being reduced, the relative contribution from aerodynamic generated noise under high speed driving conditions is increasing. This noise is difficult to measure because of disturbances such as self-generated wind noise and wind tunnel background noise. The cross spectral measurement technique applied in STSF can be used to reduce the influence of these disturbing noise sources and the holographic calculations of STSF can be used for precise localization of the turbulence noise sources from measurements taken outside the turbulence region.

The measurements were performed in the Honda low-noise wind tunnel, with the set-up shown in Fig. 9. The vertical microphone array is scanned along the side of the car at a distance of 250 mm from the centre of the side window. In this example a single reference microphone was placed inside the car, at the drivers head position. As the STSF uses the Cross-spectra

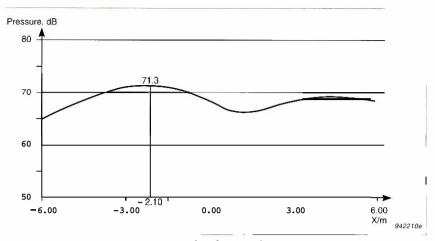


Fig. 7. As Fig. 4, but with simulated 3 dB reduction of exhaust noise

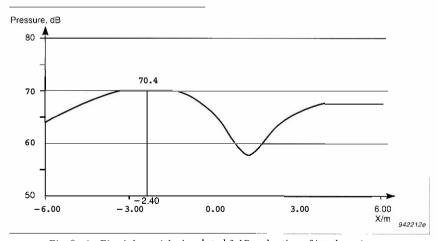


Fig. 8. As Fig. 4, but with simulated 3dB reduction of intake noise

between the scan microphones and the reference microphones for the calculations, only the part of the noise field which correlates with the signal picked up by the reference transducers will be taken into account in the cal-

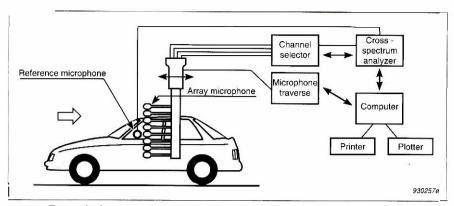


Fig. 9. STSF set-up for measurement in the Honda low-noise wind-tunnel

culations. This means that the STSF measurement outside the car will identify the noise sources which are coherent with the sound pressure at the position of the driver's head, while disturbances like wind induced noise on the individual scan microphones will be excluded from the calculations. Fig. 10 shows the sound pressure level on the left side of three different cars with a wind speed of 100 km/hour.

The last example is from a measurement of tyre noise, the third of the major noise sources contributing to the radiation from passenger cars. The tyre under test was mounted on the front axle without the car on a rolling road in a semi-anechoic test chamber. To simulate cornering driving conditions, the front wheels were at an angle of 1° from the vertical plane through the longitudinal axis of the car. The STSF measurement was concentrated on the area around the tyre and by suitable placement of the reference transducers it was ensured that the background noise from the auxiliary equipment did not disturb the measurement. The use of four reference microphones around the tyre/road contact region was shown to provide a good complete model of the sound field. The measurement was performed with a line array of 12 microphones, which was moved to 20 measurement positions along the side of the tyre by a microphone positioning system, giving a total of 240 scan positions. The distance between the scan positions (grid spacing) was 75 mm, giving a total measurement area of $825 \times 1425 \,\mathrm{mm}^2$. Fig. 11 shows the intensity radiation from the tyre at 740 Hz as an example of the results from this measurement. As ususal, the tyre/road contact area is the principal noise generating region relative to

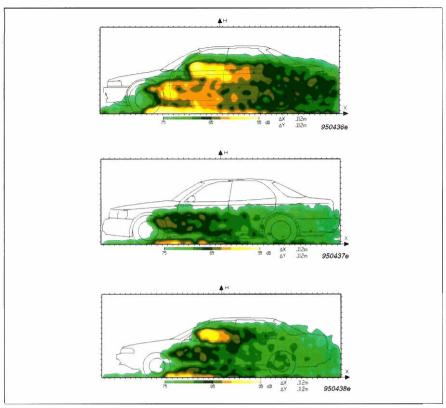


Fig. 10. Outside sound pressure distribution on the side surface of three different cars at a windspeed of 100 km/hour. Frequency range is 194 Hz to 729 Hz A-weighted

the other parts of the tyre. However, the cornering of the tyre excites a vibration pattern on the sidewall, causing relatively high levels of acoustical radiation from the upper part of the tyre.

Non-stationary noise: Time domain STSF

During the past 10 years, Cross spectral STSF has been recognized as an efficient tool for detailed study of acoustical radiation phenomena, for sound sources which are stationary stochastic. There has been no system

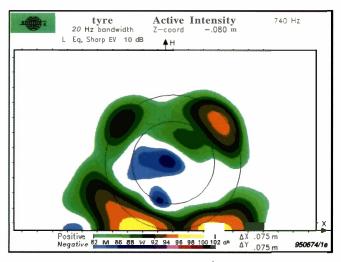


Fig. 11. Sound intensity radiation from tyre at 740 Hz

for performing acoustical holography on general non-stationary sound sources, such as engines under run-up conditions. Ever stricter legislation is compelling vehicle manufacturers to reduce external vehicle noise. In order to simulate the actual working conditions of vehicles, acoustical testing must be performed not only under stationary but also under non-stationary conditions.

A European collaboration project, named PIANO, has been started under the auspices of the CEC BRITE/EURAM program. The participants include heavy-vehicle manufacturers, specialized companies and universities. The task of this project from 1993 to 1995, is to investigate new methods that enable faster, cheaper and more accurate ISO 362 pass-by noise reduction measures for heavy road vehicles. These vehicles are invariably operating under non-stationay conditions. In this connection, a prototype time domain STSF system is used to assess the radiation from components such as engines and gear-boxes. The prototype system is based upon a fixed array of microphones coupled to a multichannel, simultaneous, time-history acquisition unit and a supporting software package.

Instrumentation for Time domain STSF

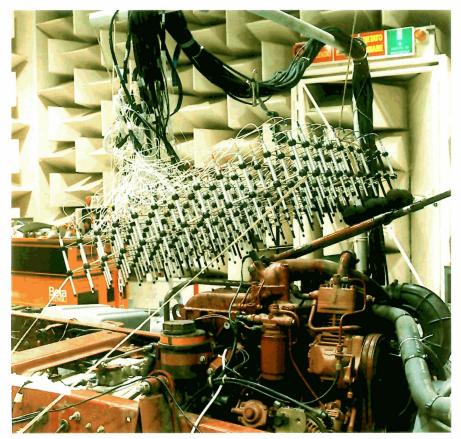


Fig. 12. SARAH microphone array

A fixed microphone array system, named SARAH (Simple ARray for Acoustical Holography) is being developed by Brüel & Kjær for such time-history acquisitions. The array consists of a large number of specially designed microphones which can be clipped into position in an open grid network. The microphones are grouped in sixes into a single D-plug which can then

be connected to a module in the Intelligent Data Acquisition front-end system (IDA) Type 3561. The first IDA unit contains a power supply, an interface and a synchronization module. It can be equipped with a maximum of 7 microphone input modules, each capable of supporting 6 microphones, i.e. 42 channels in all. Subsequent IDA units do not require synchronization modules and can therefore be equipped with up to 8 microphone input modules, i.e. 48 channels. The time-history data is stored in RAM in the frontend until it is transferred to the host UNIX computer, which in the measurements described here was a HP712/80. As an indication of the capabilities of the system; for 6 to 3066 microphone channels, the maximum time record length is 20.8 seconds with a 6.4 kHz frequency range.

As the real-time memory is in RAM in the front-end, not in the host computer, the limitation of the system is not in the number of channels but in the time to transfer the data from IDA to the host computer. As an example, for the measurements described below, for 120 channels and a bandwidth of 3.2 kHz, and a time record of 2.0 seconds, the transfer of 4 Mbytes of data took less than 20 seconds through an **Ethernet** interface connection.

Basic theory of Time domain STSF

Presently, only the use of Near-field Acoustical Holography for near-field calculations has been implemented and tested in the protoype Time domain STSF software. Far-field calculations using Helmholtz' integral equation have not been implemented yet, but that is relatively simple. The present chapter gives a short introduction to the basic theory and computer implementation of Time Domain Holography (TDH).

Fig.13 illustrates the geometry of the measurement problem to be treated. A sound field is measured over a plane $z=z_0>0$ in the near-field region of a sound source. All parts of the source are assumed to be in the half space z<0, and the half space $z\ge0$ is assumed to be source-free and homogeneous.

The sound pressure field $p(\bar{r}, t) = p(x, y, z, t)$, which represents a particular event in time domain, fulfils the homogeneous wave equation in the half space $z \ge 0$:

$$\nabla^2 p - \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2} = 0, \qquad z \ge 0 \tag{1}$$

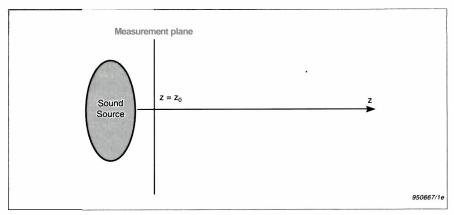


Fig.13. Planar measurement geometry

To be able to use the well-known holography calculation method, we need to transform the event to the frequency domain in time and in the spatial coordinates x and y. We need the following Fourier transform pair for the sound pressure in any plane with constant z-coordinate, $z \ge 0$:

$$p(\mathbf{x}, y, z, t) = \frac{1}{(2\pi)^3} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} P(k_x, k_y, z, \omega) e^{-j(k_x x + k_y y - at)} dk_x dk_y d\omega$$
(2)

$$P(k_x, k_y, z, \omega) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} p(x, y, z, t) e^{j(k_x x + k_y y - at)} dx dy dt$$
 (3)

If we insert the Fourier transform expression (2) for $p(\bar{r}, t)$ in the wave equation (1) and take the Fourier transform, we obtain the following **one**-dimensional differential equation in z:

$$\left[\frac{\partial^2}{\partial z^2} + \gamma^2\right] P(k_x, k_y, z, \omega) = 0, \qquad \gamma^2 \equiv k^2 - k_x^2 - k_y^2, \qquad z \ge 0$$
(4)

where

$$k \equiv \frac{\omega}{c} \tag{5}$$

It is shown for example in reference [7] that provided all sources of the sound field are at z < 0, then the complete solution to (4) can be written as:

$$P\left(k_{x},k_{y},z,\omega\right) = P\left(k_{x},k_{y},z_{0},\omega\right)e^{-j\,\gamma(z-z_{0})},\quad z_{0}\geq0,\qquad z\geq0 \tag{6}$$

where γ is a function of k_x , k_y and ω (through k):

$$\gamma = \begin{cases}
\sqrt{k^2 - k_x^2 - k_y^2} & \text{for } k_x^2 + k_y^2 \le k^2 \\
-j\sqrt{k_x^2 + k_y^2 - k^2} & \text{for } k_x^2 + k_y^2 > k^2
\end{cases}$$
(7)

Since the sound pressure field is measured in the plane $z = z_0$, the plane wave spectrum P must be obtained from equation (3) with z equal to z_0 . Formulae (6) and (2) then allow the sound pressure fieldp for any $z \ge 0$ to be calculated.

Using the well-known relations between the sound pressure and the particle velocity, the particle velocity vector $\overline{u}(\overline{r}, t)$ can be shown to be (see e.g. [1]):

$$\begin{split} \bar{u}\left(x,y,z,t\right) &= \frac{1}{\left(2\pi\right)^{3}\rho c} \int\limits_{-\infty-\infty-\infty}^{\infty} \int\limits_{-\infty}^{\infty} P\left(k_{x},k_{y},z,\omega\right) \frac{1}{k} \left\{k_{x},k_{y},\gamma\right\}^{T} \\ &\times e^{-j\left(k_{x}x+k_{y}y-\omega t\right)} dk_{x} dk_{y} d\omega \end{split} \tag{8}$$

where $\{\}^T$ represents a column matrix (a vector). From the time domain pressure and the time domain particle velocity, the instantaneous intensity can be obtained, and any type of filtering and averaging can be performed.

Reference [6] describes a holography implementation which has some important similarities with the present TDH system. The application in [6] is measurement on broadband, stationary sources with controlled excitation. In the present application, the ability to measure source characteristics at different stages of a non-stationary event is essential.

For the case of calculation of the sound field in a plane further away from the source than the measurement plane, i.e. $z>z_0$, a direct time-domain expression can be derived from Helmholtz' Integral Equation [7]:

$$p(\bar{r},t) = \frac{z-z_0}{2\pi c} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} \frac{1}{R^2} \left[\frac{\partial p}{\partial t} (\bar{r}',t-\frac{R}{c}) + \frac{c}{R} p(\bar{r}',t-\frac{R}{c}) \right] dx' dy', \quad z > z_0 \quad (9)$$

Here *R* is the distance between the integration point $\bar{r}' = (x', y', z_0)$ and the calculation point $\bar{r} = (x, y, z)$:

$$R = \sqrt{(x-x')^2 + (y-y')^2 + (z-z_0)^2}$$

The direct time-domain formula (9) is helpful for understanding some of the effects of a truncation of the event in time: A sharp cut off in time will introduce a rather large impulsive error due to the time derivative in (9). This error term will appear with a delay due to the propagation time Rlc from the measurement plane to the calculation position.

The time domain formula (9), however, is efficient for the numerical calculation only when the sound field at a single or a few points is needed, and it does not apply for calculation closer to the source than the measurement plane. The holographic formulation given in formulae (1) to (8) is efficient because it allows the use of FFTs for the Fourier transform calculations and, further, it allows prediction of the sound field closer to the source than the measurement plane.

Computer implementation

Fig. 14 shows the data flow for the prototype computer implementation. A rectangular box represents data — with an encircled f at the bottom it is frequency domain data, otherwise it is time domain data. Note that the Fourier transforms (FFTs) in time and space are performed at different stages, as opposed to the above mathematical formulation. The spatial

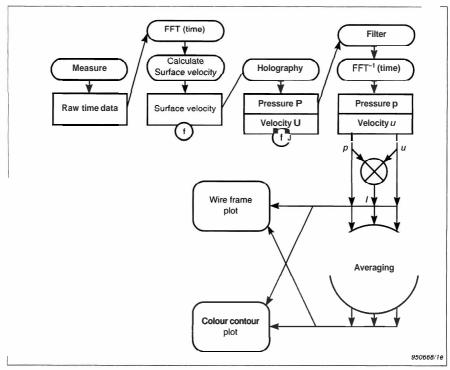


Fig.14. Data flow for prototype TDH software

transforms are performed as parts of the operations Calculate Surface velocity and Holography.

Calculate Surface velocity computes that frequency domain particle velocity distribution in a plane close to the surface of the sound source which will produce the measured sound pressure field. Thereby, the assumption of zero sound pressure outside the measurement window in the measurement plane is avoided, significantly reducing the effects of using a rather small measurement window, which is often necessary because a full sized array is required. An iterative holographic procedure is used for the calculation [8].

Holography provides the frequency domain sound pressure P and particle velocity (*z*-component) U in any plane z > 0 from frequency domain particle velocity or sound pressure data in a parallel plane.

Having the frequency domain data makes it easy to impose a filter, for example a 113-octave filter, before the data are inverse Fourier transformed (FFT⁻¹) to the time domain. The instantaneous intensity I is calculated as a product of the time domain pressurep and particle velocity u.

Measurements on the engine of a small truck

A fixed (full size) array system is more suited for measurements on engines or components installed in a test cell rather than for complete vehicles due to the large number of channels involved. Some adjustment of the test bed is usually required in order to position the array as closely as possible to the object under test. An advantage of the fixed array system relative to a scanned array system is that feed-pipes to the engine can be led away through the array plane.

To test the prototype system, measurements were made on a small IVECO EUROCARGO truck on a dynamometer at Centro Ricerche Fiat in Torino. In order to have access to measure the engine, the passenger cabin was tilted forward, see Fig.12. The engine was **a** 4 litre 8040.25 diesel engine with 4 cylinders.

The major measurement condition to be considered was a fast run-up (pass-by simulation) in 3rd gear, taking about 1.6 seconds. To be able to compare stationary and non-stationary conditions, a series of measurements were taken at constant RPM. In order that the stationary conditions could simulate the run-up conditions as closely as possible, the engine was operated at full load in 3rd gear.

Measurements were taken over the top right-hand side of the engine, with the normal of the array plane pointing approximately **45** degrees upwards. The measurement plane could be as close as 12 cm from the nearest point on the intake manifold and 8 cm from the nearest point on the compressor. The array area covered the engine side down to the chassis.

Results

Fig. 15 to Fig. 20 show intensity contour plots over the right side of the engine for the 1600 Hz third-octave band. Fig. 15 to Fig. 19 represent different stages during the fast run-up – at 2150, 2600, 2750, 2900 and 3000 RPM – while Fig. 20 represents a measurement with stationary engine operation at 2750 RPM. All plots have the same threshold, and each colour represents a 1 dB level range. Also, all maps show the intensity in a plane 10 cm closer to the source than the measurement plane, which is a little bit further than the first intersection with the source. For the run-up

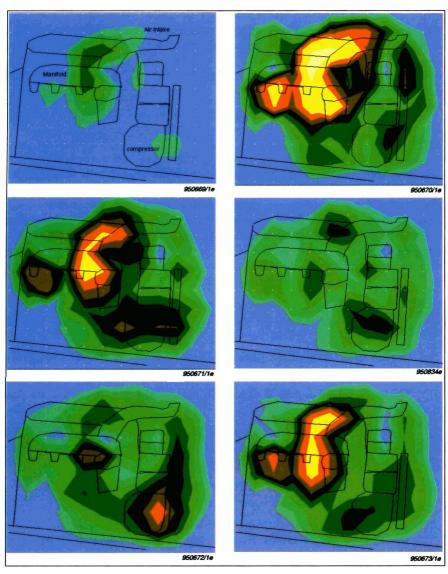


Fig. 15. Run-up, 2150 RPM Fig. 17. Run-up, 2750 RPM Fig. 19. Run-up, 3000 RPM

Fig. 16. Run-up, 2600 RPM Fig. 18. Run-up, 2900 RPM Fig. 20. Stationary, 2750 RPM

maps, averaging has been performed over a 50 RPM interval, while for the stationary condition an averaging time equal to 488 msec has been used.

Fig. 17 and Fig. 20 can be compared because they represent approximately the same RPM condition of the engine – Fig. 17 during the run-up and Fig. 20 during stationary operation. The two plots look very similar around the air intake and the manifold.

Conclusions

Acoustical array techniques are becoming more common in the automotive industry. The development has been from a single microphone moved from position to position manually, to one or more sound intensity probes moved by means of a robot, to linear arrays of microphones used to scan stationary noise sources, to large fixed arrays of microphones used in time-history acquisition systems for non-stationary noise. This article described two array systems; the Cross spectral STSF system, already widely used in the automotive industry for stationary noise analysis, and the Time domain STSF system, still under development, for non-stationary noise analysis.

References

- [1] MAYNARD, J.D., WILLIAMS, E.G. & LEE, Y.: J. Acoust. Soc. Am., 78 (4), 1395 (1985)
- [2] VERONESI, W.A., & MAYNARD, J.D.: *J.Acoust. Soc. Am. 81 (5), 1307* (1987)
- [3] HALD, J..: "STSF A Unique Technique for scan based Near-Field Acoustic Holography without restrictions on coherence", Briiel & Kjær Technical Review No. 1, (1989)
- [4] TAYLOR, N. & RASMUSSEN, P.: "Exterior Noise Measurements on a Rover 220 GSi", Brüel & Kjær Application Note, BO 0430
- [5] NAKAMURA, A., KOMINE, T, TSUCHIYA, M. & HALD, J.: "Measurement of Aerodynamic Noise using STSF", Brüel&Kjær Application Note, BO 0392
- [6] MAYNARD. J.D.: *Proc. NOISE-CON* 87,635 (1987)

- [7] HANSEN, T.B. & YAGHJIAN, A.D.: Rome Lab., AFMC Griffiss Airforce Base, NewYork, Inhouse Report, RL-TR-93-210 (1993)
- [8] HALD, J.: Proc. Inter-Noise 94, 1887 (1994)
- [9] GINN, K.B. & HALD. J.: Proc. Euro-Noise 95,921 (1995)

Pass-by Noise Measurements

by Per Rasmussen

Abstract

Exterior noise measurements on vehicles can be performed according to various national and international standards using a specially developed measuring system. Mathematical simulations of the noise field radiated from a vehicle shows that by using non-standard measurement procedures, it is possible to obtain more information about the noise radiation from the vehicle, and make it possible to identify and quantify individual sound sources.

Résumé

La mesure du bruit au passage de véhicules est regentee par diverses normes, nationales ou internationales, et s'effectue au moyen de systèmes specialement mis au point dans ce but. Or, des simulations mathematiques du champ sonore émis par le véhicule en deplacement montrent qu'il est possible, par le biais de procedures de mesure non normalisées, d'obtenir un plus grand nombre d'informations sur l'émission de bruit provoquee et d'en localiser et quantifier les différentes sources.

Zusammenfassung

Außengeräuschmessungen an Fahrzeugen nach verschiedenen nationalen und internationalen Normen lassen sich mit einem Meßsystem durchführen, das speziell für diesen Zweck entwickelt wurde. Durch mathematische Simulation des vom Fahrzeug abgestrahlten Schallfeldes wird gezeigt, daß sich durch nichtstandardisierte Meljverfahren weitere Informationen über die Schallabstrahlung des Fahrzeugs gewinnen lassen, so daß individuelle Schallquellen identifiziert und quantitativ bestimmt werden können.

Introduction

The acceptable exterior noise level from passenger cars is constantly decreasing through national and international regulations. Most recently, the acceptable level within the European Union has been lowered to 74dB(A). To fulfil these demands, still more advanced measuring techniques have to be used in the development process and for troubleshooting. Exterior vehicle noise is normally measured following the procedure in ISO standard 362 "Acoustics – Measurement of noise emitted by accelerating road vehicles – Engineering method" [1] According to this standard, the noise is measured with a microphone placed 7.5 m from the centre of the test road. The vehicle is accelerated past the microphone at full throttle and the registered noise level is the maximum level measured during the pass-by.

Most standards for measuring the exterior noise from a vehicle intend to quantify the noise impact on the environment rather than to describe the vehicle as a noise source. The noise impact will depend on a number of factors such as the distance to the listener, type of road, typical vehicle speed, driving style, etc. These factors are all considered in the standard and the result of the measuring procedure is a single number rating of the noise emission from the car. By carefully controlling all parameters during the measurements, it is possible to obtain reliable and accurate results.

However, in order to reduce the exterior noise in the most efficient way, it is essential to have detailed information about the vehicle's individual noise sources. This includes information about the locations of the sound sources. the strength of each source and the interaction between the different sources. Pass-by noise measurement standards like ISO 362 are not suited for, and not intended for, such detailed investigations of the different noise sources. As the microphone is 7.5 m away from the centre of the test road, it is essentially a farfield measurement, which gives very little information about the acoustical near-field. The distance from the different parts of the vehicle to the microphones is almost the same, so seen from the microphone positions the car is one big source with a complicated directivity pattern. At higher frequencies, where the wavelength of the sound is much smaller than the dimensions of the car and the distance from the car to the microphones, the individual noise sources can be considered as simple, omni-directional monopoles radiating equally in all directions. However, the shape of the enclosures (e.g. engine bay) and acoustic trim can result in directional noise sources. At lower frequencies, especially around the dominating engine noise frequencies, where the wavelengths are comparable to the dimensions of the car, correlated noise sources will interact and can result in complicated directivity patterns.

Measurement procedure

The pass-by noise test according to ISO362 is performed by accelerating the car past the microphones and measuring the maximum overall A-weighted level. The car approaches the measurement area with a constant speed of 50km/hr. As the front of the car reaches the starting line A-A, see Fig.1, the vehicle is accelerated at full throttle until the rear-end of the car reaches the line B-B. The A-weighted sound pressure level is measured with a type 1 sound level meter on both sides of the vehicle, using an exponential averaging time of 250 ms. The result of the test is the maximum level obtained during the pass-by.

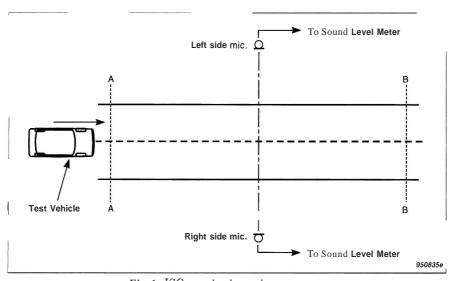


Fig. 1. ISO standard pass-by test set-up

The noise is transmitted from the different sources like tyres, exhaust orifice, air-intake, etc., to the microphone placed 7.5 m away from the centre of the test road. The microphone converts the acoustical signal into an electrical signal which in turn passes through an RMS-detector with an equivalent averaging time of $^{1}/_{4}s$ (Fast).

The measurement is started when the car is at the starting line $10\,\text{m}$ before the microphones. The speed of the car at the starting line is $50\,\text{km/hr}$, corresponding to approx. $14\,\text{m/s}$. The full measurement is finished when the rear end of the car is $10\,\text{m}$ past the microphones, so the total measuring time is approximately $1.5\,\text{s}$. This is comparable to the averaging time of $^{1}/_{4}\,\text{s}$, so when interpreting the measurement data this should be considered.

Also the geometry of the measurement set-up affects the measurement result. For a normal passenger car of length 3 to 5m, the distance between individual noise sources like engine and exhaust orifice or front and rear tyre, is much smaller than the distance from the source to the microphones. Furthermore, at low frequencies the distance between the sources is small compared to the wavelength of the sound and it is therefore necessary to consider the interaction of the different noise sources.

Pass-by measurement instrumentation

Fig. 2 shows a full set-up for pass-by noise measurements. The system consists of three functional parts: the position measurement part, the rpm/telemetry part, and the frequency analyzer. The position measurement part consists of a radar and two sets of photocells with reflectors. The telemetry part consists of a "Transmitter Unit" that transmits the measured rpm and the throttle position via an RS-232 radio modem to the "Receiver Unit". All the functions of the system are controlled by the frequency analyzer. The frequency analyzer used in this system is a specially developed Order Tracking Analyzer, including both dual-channel frequency and order analysis, tacho input processing in two channels and logging of four DC-channels.

The radar is a standard police-radar used in more than 25 countries around the world. It works on the Doppler principle at a frequency of approximately 22 GHz. At this frequency, the wavelength of the radio waves is around 12 mm. If an object such as a car is in the radar beam, it will reflect the transmitted wave back to the radar antenna. If the car now moves relative to the radar antenna, the reflected wave will be Doppler shifted relative to the transmitted wave causing a phase shift. So every time the car moves approximately 6 mm or half a wavelength, the radar will detect a 180 degree phase shift and produce a pulse on the output. If the car is traveling at 50 km/h, this will correspond to around 2300 pulses every second.

This tacho signal is processed by the order tracking analyzer which calculates the velocity and position of the vehicle.

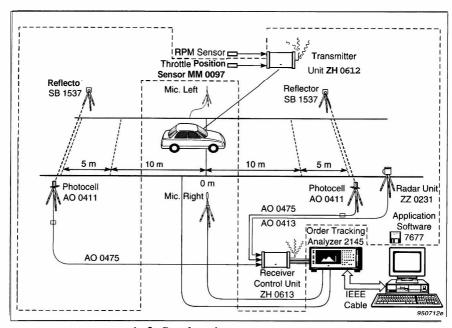


Fig.2. Pass-by noise measurement set-up

In order to calculate the absolute position of the car, the analyzer needs information about an initial position. This is provided by two light-barriers, with photocells and reflectors, one at each end of the test section. The analyzer is set up so that measurements are started by an external trigger signal from the photocells so that the position calculation is initiated simultaneously with this trigger signal. The analyzer also detects which of the two photocells was triggered first, in order to keep track of the direction of the vehicle.

The noise data are acquired by the analyzer as a function of vehicle position. This is controlled by the radar signal, so that for a typical measurement the analyzer will store noise spectra at constant intervals. A typical set-up would be to start at $-15\,\mathrm{m}$ and end at $+15\,\mathrm{m}$ with an interval of 0.25 m. This means that the analyzer will store 121 spectra at the positions $-15\,\mathrm{m}$, $-14.75\,\mathrm{m}$, $-14.50\,\mathrm{m}$ and so on until $+15\,\mathrm{m}$.

The ISO standard prescribes that the initial velocity of the vehicle has to be 50km/hr and that the driver has to apply full throttle at the starting line 10m before the microphone line. As the result of the pass-by test depends on the ini-

tial vehicle velocity and the exact position where the driver applied full throttle, it is important to record these data during the measurement. The pass-by system measures the engine rpm and the throttle position in the car and transmits the results to the stationary analyzer. This is done via a two-way digital radio-modem, which ensures very reliable data transmission. The Transmitter Unit in the car measures the rpm value and the result is a number which is transmitted via an RS-232 radio-modem. As the radio-modem establishes two-way communication, information cannot only be sent from the Transmitter Unit in the car to the stationary Receiver Unit near the analyzer, but also from the analyzer to the Transmitter Unit in the car. The two-way communication is also used for sending start information from the analyzer to the Transmitter Unit in the car. When the analyzer is ready to measure, a Start indication lamp on the Transmitter Unit lights up and the driver can start the pass-by drive.

The Transmitter Unit also measures the throttle position with a Throttle Position Sensor. This is basically a force transducer which is mounted on the accelerator. When the car approaches the test section, the test driver will use a moderate pressure on the accelerator to maintain a steady speed. At the line 10m before the microphone line (position –10 m), the driver should apply full acceleration by pressing the accelerator down fully. This will be detected by the force transducer and transmitted back to the analyzer.

Measurement results

The full measurement is controlled by Windows™ software which is also able to display the measured data in different data formats. Fig.3 shows a typical example of results obtained from a pass-by noise measurement. This gives the overall A-weighted SPL from both sides of the vehicle as a function of the vehicle position. At the same time the 400 Hz $^{1}/_{3}$ -octave bands and the engine rpm are displayed.

It can be seen that the overall maximum SPL occurs when the vehicle is 2.25m past the microphone and that this is highest on the left side of the car. The corresponding $^1/_3$ -octave spectrum is seen in Fig. 4. This shows two different noise phenomena: the discrete frequency components around $125\,Hz$ and $400\,Hz$, coming from the engine and exhaust system, and the broadband noise from $500\,Hz$ to $2.5\,kHz$ coming from the road/tyre interaction. When interpreting the pass-by measurement results, it is important to consider the difference in the origin of these two different groups of signals. The engine/exhaust noise is generated by the same source (the engine) and will therefore be correlated,

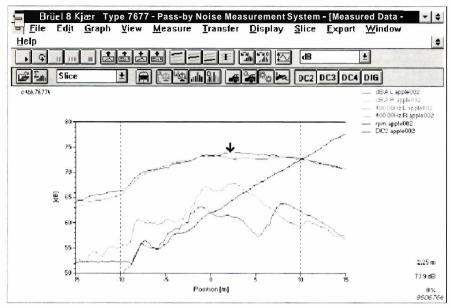


Fig. 3. Overall noise, 400 Hz and engine rpm as a function of vehicle position

while the **road/tyre** interaction noise is generated by **uncorrelated** sources. The consequences of this will be exemplified later on by simulating the different situations with a simple mathematical model.

An example of the influence of the types of noise sources on the interpretation of pass-by data is described in the Brüel & Kjær Application Note "Exterior Noise Measurements on a Rover 220 GSi" by Nigel Taylor, Rover Group, et al. [2]. The Rover 220 GSi was tested at the Rover Gaydon test site, on the ISO 10844 [3] Standard noise test road surface. The vehicle was tested in both 2nd and 3rd gear, according to the procedure in ISO 362. The pass-by noise level in 2nd gear was successfully reduced using, among other things, a finite element model of the exhaust system. The result of a test in third gear is shown in Fig. 5. This gives the overall A-weighted sound pressure level for the right side as a function of the vehicle position relative to the microphones. The maximum SPL is $75\,\mathrm{dB}(A)$ and a further frequency analysis shows that the overall SPL is dominated by noise in the $100\,\mathrm{Hz}^{-1}/_3$ -octave band, which corresponds to the engine firing frequency or second harmonic of the engine revolution. It can be seen that the SPL for the $100\,\mathrm{Hz}^{-1}/_3$ -octave band has two

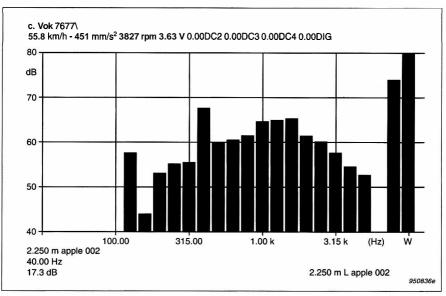


Fig. 4. 113-octave spectrum of pass-by noise in position with maximum SPL

maxima, one when the car is around **0.5 m** before the microphone line and one when the car is 7 m past the microphone line. Intuitively, the highest peak occurring at the rear of the car could be associated with the noise radiated from the exhaust orifice and the smaller peak could be associated with the **air** intake orifice. This interpretation can, however, be shown to be incorrect both by mathematically **modeling** the noise radiation from a car, as below, and by other measurement techniques such as STSF.

Noise model of car

The sound field from a car can, as a first approximation, be simplified to two monopoles. At low frequencies, the noise radiation is dominated by the noise from the exhaust orifice and from the engine (air intake), while at higher frequencies tyre noise is dominant. Considering first the low-frequency enginegenerated noise, the car can be modelled as two correlated monopoles placed $l_{\rm d}$ apart (usually in the range of 2 to 3 m), where the front monopole A1 represents the noise from the engine and the rear monopole A2 represents the

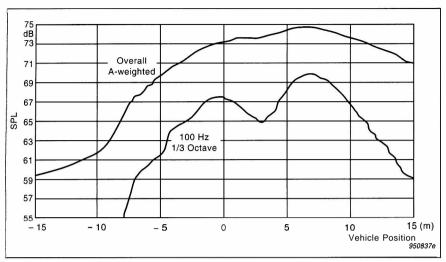


Fig. 5. Measurement of pass-by noise for right side of vehicle

noise from the exhaust orifice, see Fig. 6. The two monopoles are moving past the microphone at a distance of $l_1 = 7.5 \,\mathrm{m}$ with speed v_0 . In a real pass-by test, the vehicle is accelerating past the microphone, but for simplicity the speed will be considered constant at $14 \,\mathrm{m/s}$. The position of source A1 relative to the microphone is given by x, and the position of A2 is therefore $x - l_d$.

The sound field from a monopole is a simple function of the distance r from the monopole, so the sound radiated from A1 is given by:

$$p_{1}(r_{1}) = \frac{C_{1} \cdot k \cdot \rho \cdot c}{4 \cdot \pi \cdot r_{1}} \cdot e^{ik \cdot (c \cdot t - r_{1})}$$

$$\tag{1}$$

where C_1 is the source strength of source A1, k the wavenumber, p the density of air and c the speed of sound. The distance r_1 from the source to the microphone is given by:

$$r_1(x) = \sqrt{l_m^2 + x^2} (2)$$

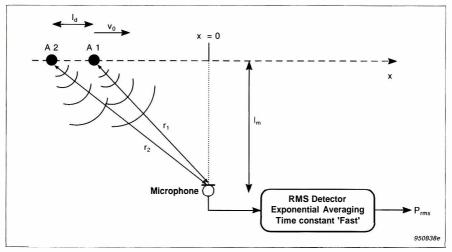


Fig.6. Model of pass-by measurement with two point sources A1 and A2

Similarly, the sound pressure from source **A2** is given by:

$$p_{2}(r_{2}) = \frac{C_{2} \cdot k \cdot \rho \cdot c}{4 \cdot \pi \cdot r_{2}} \cdot e^{ik \cdot (c \cdot t - r_{2}) + i\psi}$$

$$(3)$$

where ψ is the phase between the direct signal from the **air** intake and the signal from the exhaust orifice and r_2 is the distance from the source to the microphone position given by:

$$r_2(x) = \sqrt{l_m^2 + (x - l_d)^2} \tag{4}$$

These two pressures are then added together at the microphone position to give the resulting pressure p_{τ} . This is then passed through an RMS detector with time constant τ , giving the resulting output P_{τ}

$$P_{rms} = \sqrt{\frac{1}{\tau} \int_{-\infty}^{t} p_s(\theta)^2 \cdot d\theta}$$
 (5)

At higher frequencies, the different noise sources can be treated as uncorrelated and in calculating the sound pressure level at the microphone position it is not necessary to use the instantaneous sound pressures from the individual sources. The equations for the sound pressure level at the microphone position can then be simplified to:

$$p(r_1) = \frac{C_1 \cdot k \cdot \rho \cdot c}{4 \cdot \pi \cdot r_1} \tag{6}$$

and similarly for the other source.

Simulations with uncorrelated sources

The results of a pass-by measurement can be simulated by numerically integrating equation (5) with the appropriate functions for the sound pressure level at the microphone position. The result is given as the sound pressure level as a function of the vehicle position as it passes the microphone. The vehicle starts at a position $x = -15 \,\mathrm{m}$, passes the microphone at $x = 0 \,\mathrm{m}$ and continues to $x = +15 \,\mathrm{m}$.

In the first example, Fig. 7, the pass-by noise has been calculated for two uncorrelated sources with a source separation of 2.5 m, for the frequency range around 1 kHz, typical for tyre noise. The two sources are assumed to be of equal strength. As the distance between the two sources is much smaller than the distance from the sources to the microphone, it is not possible to distinguish the contributions from the individual sources.

In order to separate the contribution of the front source from the rear source, it is necessary to move the microphone much closer to the vehicle, so that the distance between the sources is large compared to the distance to the microphone. Fig. 8 shows a simulation of a pass-by where the microphone is only 0.5 m from the car. In this case it is possible to distinguish the two sources, although it is not possible to evaluate the magnitude of the individual contributions. This is because of the relatively long averaging time used in the RMS calculation. The averaging time is $\frac{1}{4}$ s and during this time the car will move

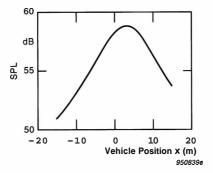


Fig. 7. Simulation of pass-by: two uncorrelated noise sources, with microphone 7.5 m away

almost 4 m. Therefore the contributions from the front and rear sources will be smeared out by the averaging process. By using a shorter averaging time, the contributions from the two sources can be separated. Using, for example, an averaging time of $^{1}/_{16}$ s, as simulated in Fig. 9, it is possible to separate the contributions from the two sources.

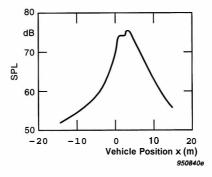


Fig. 8. Simulation of pass-by: two uncorrelated noise sources, with microphone 0.5 m away

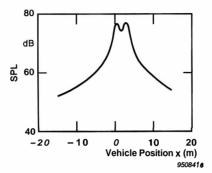


Fig. 9. Simulation of pass-by: two uncorrelated mise sources, with microphone 0.5 m away and 1/16s averaging time

As we have seen, it is possible to obtain more information from a pass-by measurement by moving the microphone closer to the vehicle than the 7.5m prescribed in the ISO standard and by using a shorter averaging time. Such measurements will, of course, not be in accordance with the ISO standard and cannot be used for homologation purposes. On the other hand, they may give more information about the noise radiation from the vehicle, and make it possible to identify and quantify individual sound sources. In practice there are, however, limits to how close the microphone can be placed, and how short an averaging time can be used. As the distance from the microphone to the vehicle is decreased it also becomes more critical. In order to make repeatable measurements, the distance from the car to the microphone must be the same for each measurement. Ib maintain an accuracy off 1 dB the distance has to be kept within ±10%. For a microphone distance of 7.5 m this is easily achieved, but for microphone distances lower than 0.5 m it becomes very difficult. Also, the use of shorter averaging time reduces the accuracy of the measurement result as the statistical uncertainty is increased. For a ¹/₃-octave frequency analysis, an averaging time of $\frac{1}{32}$ s will only give a result to within ± 1 dB for frequencies above 1.25kHz. For lower frequencies, the BT product will be smaller **and the** statistical uncertainty will be larger.

Simulations with correlated sources

At low frequencies, the noise **from** the car will in general be dominated by noise from the engine (air intake) and the exhaust orifice. At the typical driving conditions for a pass-by test, the engine firing **frequency** will be around **100 Hz**, and this **will** dominate the low-frequency part of the spectrum. The wavelength of the sound at this frequency is around **3.4m**, which is relatively large compared to both the dimensions of a typical car and to the distance to the microphone of 7.5 m. As the noise from the engine and the noise from the exhaust both originate from the engine, the noise from the two sources is correlated and the phase between them must be taken into account when calculating the resulting sound pressure level at the microphone position.

Assuming again that the vehicle can be modelled as two simple sound sources placed 2.5 m apart with equal strength, and that the phase between the signals is 0°, the resulting sound pressure level at the microphone position as a function of the vehicle position will be as shown in Fig. 10.

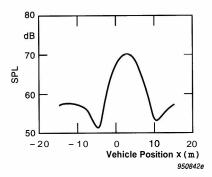


Fig. 10. Simulation of pass-by: two 100 Hz correlated noise sources, with 0° phase shift and with microphone 7.5 m away

This shows a maximum when the car is approximately 2m past the microphone, when the signals from the two sources arrive at the microphone position in phase and the distance from the sources to the microphone is smallest. Also the **curve** shows two minima, one when the vehicle is **5m** before the microphone and the other when the microphone is **11**m past the microphone.

This corresponds to the positions where the noise from the two sources arrives at the microphone position out of phase and therefore cancels each other.

Normally the noise from the exhaust orifice will be phase shifted when passing through the exhaust system and this will change the positions of constructive and destructive interference. For a phase **shift** of 90° of the exhaust signal, but otherwise as in the previous example, the maximum level position will occur when the car is approximately 5m past the microphone, see Fig. 11. In this **case** the minimum, where the noise from the two sources will arrive out of phase, occurs when the car is in front of the microphone, at x = 0 m.

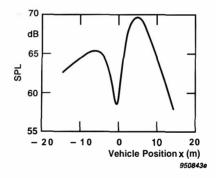


Fig.11. Simulation of pass-by: two 100 Hz correlated noise sources, with 90° phase shift and with microphone 7.5 m away

Intuitively, the result in **Fig.11** could be interpreted as being caused by a weak source in the front part of the car (i.e. engine) and a stronger source at the rear of the car (i.e. exhaust orifice), since the maximum noise level occurs when the car has passed the microphone. The simulation shows, however, that the maximum at the rear of the car is caused by constructive interference between the fi-ont and rear sources, since the two sources used in the calculations were of equal strength. Even if the **front** source is higher than the rear source, the maximum level will still occur when the car has passed the microphone. Fig.12 shows the result of a simulation where the **front** source is twice as strong as the rear source and the maximum level still occurs when the car is approximately 5 m past the microphone.

The position where the noise **from** the **front** and rear arrives at the microphone will therefore to a certain extent **determine** the maximum level obtained

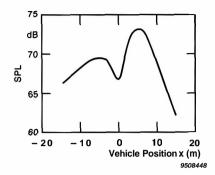


Fig. 12. Simulation of pass-by: two 100Hz correlated noise sources, front source twice as strong as the rear source, with 90° phase shift and with microphone 7.5 m away

during the pass-by test, and this position can be changed by changing the phase between the signals. Fig.13 shows a curve for the calculated reduction in the pass-by test for different phases between the two sources. This curve is for two sources of equal strength and gives the reduction in the maximum noise level relative to the noise level when the noise sources are in phase. It can be seen that when the noise sources are out of phase, the noise level is reduced by approximately 1.6 dB. Similar curves can be made for sources which are not of equal strength.

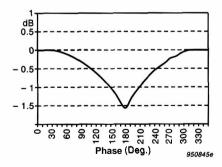


Fig. 13. Reduction in pass-by noise for two correlated sources for different phases

Conclusion

Exterior noise measurements on vehicles is described in detail in various national and international standards. These standards intend to quantify the noise impact on the environment rather than to describe the vehicle as a noise source. The noise impact will depend on a number of factors such as the distance to the listener, type of road, typical vehicle speed, driving style, etc. These factors are all considered in the standards and the result of the measuring procedure is a single number rating of the noise emission from the car. In order to reduce the exterior noise in the most efficient way, it is essential to have detailed information about the vehicle's individual noise sources. This includes information about the locations of the sound sources, the strength of each source and the interaction between the different sources. By using nonstandard measurement procedures, it is possible to obtain more information from a pass-by measurement. Such measurements will, of course, not be in accordance with the standards and cannot be used for homologation purposes. On the other hand, they may give more information about the noise radiation from the vehicle, and make it possible to identify and quantify individual sound sources.

References

- [1] International Standards Organization 362-1982(E): Acoustics Measurement of noise emitted by accelerating road vehicles Engineering method
- [2] TAYLOR, N. & RASMUSSEN, P.: "Exterior Noise Measurements on a Rover 220 GSi", Brüel & Kjær Application Note, BO0430
- [3] International Standards Organization 10844: Acoustics Test surface for road vehicles noise measurements

Previously issued numbers of Briiel & Kjær Technical Review

- 2-1994 The use of Impulse Response Function for Modal Parameter Estimation Complex Modulus and Damping Measurements using Resonant and Non-resonant Methods
 1-1994 Digital Filter Techniques vs. FFT Techniques for Damping
- Measurements (Damping Part I)
- 2-1990 Optical Filters and their Use with the Type 1302 & Type 1306 Photoacoustic Gas Monitors
- 1–1990 The Brüel & Kjær Photoacoustic Transducer System and its Physical Properties
- 2–1989 STŜF Practical **instrumentation** and application Digital Filter Analysis: Real-time and Non Red-time Performance
- 1-1989 STSF A Unique Technique for scan based Near-Field Acoustic Holography without restrictions on coherence
- 2-1988 Quantifying Draught Risk
- 1-1988 Using Experimental Modal Analysis to Simulate Structural Dynamic Modifications
 Use of Operational Deflection Shapes for Noise Control of Discrete Tones
- 4-1987 Windows to FFT Analysis (Part II)
 Acoustic Calibrator for Intensity Measurement Systems
- 3-1987 Windows to FFT Analysis (Part I)
- 2-1987 Recent Developments in Accelerometer Design Trends in Accelerometer Calibration
- 1 1987 Vibration Monitoring of Machines
- 4-1986 Field Measurements of Sound Insulation with a Battery-Operated Intensity Analyzer
 Pressure Microphones for Intensity Measurements with Significantly Improved Phase Properties
 Measurement of Acoustical Distance between Intensity Probe
 Microphones

Wind and **Turbulence** Noise of **Turbulence** Screen, Nose Cone and Sound Intensity Probe with Wind Screen

- 3-1986 A Method of Determining the Modal Frequencies of Structures with Coupled Modes
 Improvement to Monoreference Modal Data by Adding an Oblique Deeree of Freedom for the Reference
- 2-1986 Quality in Spectral Match of Photometric **Transducers** Guide to Lighting of Urban **Areas**
- 1-1986 Environmental Noise Measurements



WORLD HEADQUARTERS: DK-2850 Nærum . Denmark

Telephone: +45 42800500 . Telex: 37316 bruka dk . Fax: +45 42801405

Australia (02) 450-2066 . Austria 0043-1-8167400 . Belgium 0161449225 . Brazil (011) 246-8166

Canada: East (514)695-8225 West (604)591-9300. China 10-8419625 . Czech Republic 02-67021100

Finland (90)1481577 · France (1) 6457 2010 · Germany 0615118149-0. Great Britain (0181) 954-2366

Holland 03402-39994 · Hong Kong 2548 7486 . Hungary (1) 215830512158929 . Italy (02) 57 604141

Japan 03-5420-7302 . Republic of Korea (02) 554-0605 . Norway 66 904410 . Poland (0-22) 40 93 92

Portugal (1) 4711453 . Singapore 7358933 . Slovak Republic 07-37 6181 · Spain (91) 3681000

Sweden (08) 7112730 · Switzerland 011940 09 09 · Taiwan (02) 713 9303 · Tunisia (01) 750 400 · USA (800) 332-2040

Local representatives and service organisations worldwide