An objective measure for the sensitivity of room impulse response and its link to a diffuse sound field

This study is relevant to acoustic measurements in reverberation rooms such as measurements of sound transmission, sound absorption, and sound power levels of noise sources. The study presents a quantitative measure for the diffuseness in a room, which is first introduced theoretically and subsequently examined experimentally. The sensitivity of a room due to changes in the initial conditions is quantified by measuring a pair of impulse responses in a room differing only in the sound source position. Such changes are linked to mixing and the diffuse sound field. The measure is based on the maximum of the absolute value of the cross-correlation between the time windowed sections of the two impulse responses. By integrating this quantity normalized by the energy of the impulse response of the room, a single number rating is obtained. Results based on three sets of experiments indicate that the diffusers and absorbers in the room influence the proposed sensitivity measures systematically.

Conservation of power of the supersonic acoustic intensity

The supersonic intensity is a quantity that represents the net acoustic output that a source couples into the medium; it can be regarded as a spatially low-pass filtered version of the active intensity. This spatial filtering can lead to significant error due to spatial truncation. In this paper, based on a space-domain formulation of the problem, the finite aperture error is analyzed and examined experimentally. The results indicate that the finite aperture error can be mitigated with the appropriate processing and that the supersonic intensity provides a valid quantitative representation of the effective radiation of acoustic sources.
An objective measure for the sensitivity of the room impulse response

This study is relevant for a number of important acoustic measurements in reverberation rooms such as measurement of sound transmission and measurement of sound power levels of noise sources. From a pair of impulse responses measured in a room differing only in the position of the sound source, it might be possible to quantify the sensitivity of the room due to changes in initial conditions. Such changes are linked to mixing. The proposed measure is the maximum of the absolute value of the cross-correlation between the time windowed sections of the two impulse responses. By integrating this quantity normalized by the energy of the impulse response of the room, a single number rating is obtained. The proposed measure is examined experimentally and the results are discussed. The results indicate that the number of absorbers and diffusers in the room influences the proposed measures systematically.

Biomechanical models of damage and healing processes for voice health

In voice-loading occupations employees are required to use their voice for continuous and large periods of time, which might lead to voice problems. In this work anomalous vocal-fold vibrations due to long-time high voice-load are investigated. Laryngeal endoscopic high-speed images within the vocal-fold plane are available. This data is used to improve existing continuum biomechanical models of the vocal-folds by analyzing the injury processes. The project is expected to result in methods that objectively demonstrate the impact of high voice-load on voice. A detailed description of the currently developing work will be presented, including a rigorous analysis of the hypothesized injury processes of the vocal folds. © 2013 Acoustical Society of America.
Combination of acoustical radiosity and the image source method.
A combined model for room acoustic predictions is developed, aiming to treat both diffuse and specular reflections in a unified way. Two established methods are incorporated: acoustical radiosity, accounting for the diffuse part, and the image source method, accounting for the specular part. The model is based on conservation of acoustical energy. Losses are taken into account by the energy absorption coefficient, and the diffuse reflections are controlled via the scattering coefficient, which defines the portion of energy that has been diffusely reflected. The way the model is formulated allows for a dynamic control of the image source production, so that no fixed maximum reflection order is required. The model is optimized for energy impulse response predictions in arbitrary polyhedral rooms. The predictions are validated by comparison with published measured data for a real music studio hall. The proposed model turns out to be promising for acoustic predictions providing a high level of detail and accuracy.

Deconvolution for the localization of sound sources using a circular microphone array
During the last decade, the aeroacoustic community has examined various methods based on deconvolution to improve the visualization of acoustic fields scanned with planar sparse arrays of microphones. These methods assume that the beamforming map in an observation plane can be approximated by a convolution of the distribution of the actual sources and the beamformer's point-spread function, defined as the beamformer's response to a point source. By deconvolving the resulting map, the resolution is improved, and the side-lobes effect is reduced or even eliminated compared to conventional beamforming. Even though these methods were originally designed for planar sparse arrays, in the present study, they are adapted to uniform circular arrays for mapping the sound over 360°. This geometry has the advantage that the beamforming output is practically independent of the focusing direction, meaning that the beamformer's point-spread function is shift-invariant. This makes it possible to apply computationally efficient deconvolution algorithms that consist of spectral procedures in the entire region of interest, such as the deconvolution approach for the mapping of the acoustic sources, the Fourier-based non-negative least squares, and the Richardson-Lucy. This investigation examines the matter with computer simulations and measurements.
Experimental validation of sound field control with a circular double-layer array of loudspeakers.

This paper is concerned with experimental validation of a recently proposed method of controlling sound fields with a circular double-layer array of loudspeakers [Chang and Jacobsen, J. Acoust. Soc. Am. 131(6), 4518-4525 (2012)]. The double-layer of loudspeakers is realized with 20 pairs of closed-box loudspeakers mounted back-to-back. Source strengths are obtained with several solution methods by modeling loudspeakers as a weighted combination of monopoles and dipoles. Sound pressure levels of the controlled sound fields are measured inside and outside the array in an anechoic room, and performance indices are calculated. The experimental results show that a method of combining pure contrast maximization with a pressure matching technique provides only a small error in the listening zone between the desired and the reproduced fields, and at the same time reduces the sound level in the quiet zone as expected in the simulation studies well above the spatial Nyquist frequency except at a few frequencies. It is also shown that errors in the positions of the loudspeakers can be critical to the results at frequencies where the distance between the inner and the outer array is close to half a wavelength.
Extending the frequency range of free-field reciprocity calibration of measurement microphones to frequencies up to 150 kHz

Measurement microphones are typically calibrated in a free field at frequencies up to 50 kHz. This is a sufficiently high frequency for the most sound measurement applications related with noise assessment. However, other applications such as the measurement of noise emitted by ultrasound cleaning machines and failure detection in aeronautic structures require that the sensitivity of the microphone is known at frequencies up to 150 kHz. Another area of particular interest is the investigation of the perception mechanisms of ultrasound. In any of these applications, it is of fundamental importance to establish a well-defined traceability chain to support the measurement results. In order to extend the frequency range of free-field calibration the measurement system and measurement methods must undergo a series of changes and adaptations including the type of excitation signal, techniques for eliminating unwanted reflections from walls, cross-talk, etc. This paper presents the results of an investigation of the calibration of measurement microphones at high frequencies. A strategy for the changes and adaptations to the existing measurement methodologies, and the determination of the microphone parameters is outlined and the results of its implementation are discussed.

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Holographic reconstruction of sound fields based on the acousto-optic effect
Recent studies have shown that it is possible to measure a sound field using acousto-optic tomography. The acousto-optic effect, i.e., the interaction between sound and light, can be used to measure an arbitrary sound field by scanning it with a laser Doppler vibrometer (LDV) over an aperture; This can be described mathematically by means of the Radon transform of the acoustic field. An interesting feature of this measurement technique is that the spatial characteristics of the sound field are captured in the measurement. Therefore, the technique has an inherent holographic potential, implicitly yielding a full characterization of the sound field. In this study, a direct projection of the Radon transform from one plane to another and into the space domain, based on an elementary wave expansion is proposed. The relationship between the Radon and the wavenumber domains is examined, and the reconstruction potential of the method analyzed. The study includes both numerical and experimental results.

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Contributors: Fernandez Grande, E., Torras Rosell, A., Jacobsen, F.
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Linearized versus non-linear inverse methods for seismic localization of underground sources

The problem of localization of underground sources from seismic measurements detected by several geophones located on the ground surface is addressed. Two main approaches to the solution of the problem are considered: a beamforming approach that is derived from the linearized inversion problem, and the Bayes nonlinear inversion method. The travel times used in the beamformer are derived from solving the Eikonal equation. In the linearized inversion method, we assume that the elastic waves are predominantly acoustic waves, and the acoustic approximation is applied. For the nonlinear inverse method, we apply the Bayesian framework where the misfit function is the posterior probability distribution of the model space. The model parameters are the location of the seismic source that we are interested in estimating. The forward problem solver applied for the nonlinear inverse method is a Finite Difference elastic wave-field numerical method. In this paper, the accuracy and performance of the linear beamformer and nonlinear inverse methods to localize an underground seismic source are checked and compared using computer generated synthetic experimental data. © 2013 Acoustical Society of America.

Practical computational aeroacoustics for complex confined scattering geometries in low Mach number flows

The purpose of this paper is to demonstrate that a recently published methodology for predicting flow generated noise by compact surfaces under free-field conditions [1] can be extended to a different and more complex configuration of industrial interest. In the previous paper, the methodology was applied to low Mach number flow past a circular cylinder in free-field, where the Green's function and its derivative were obtained analytically. In this paper, the method will be applied to the case of low Mach number flow past a complex confined scattering geometry where both compact and non-compact surfaces are involved. Here the generation of noise is dominated by the interaction of the flow with a surface whose maximum dimension is shorter than the wavelength of interest. The analysis is based on the surface-source term of the Ffowcs Williams-Hawkins equation. The acoustic source data of the flow are generated by use of a Computational Fluid Dynamics (CFD) simulation. Due to the complexity of the scattering surfaces, the derivative of the Green's function must be obtained numerically through a Computational Acoustics (CA) simulation. The results have been validated through comparison with sound power measurements. © S. Hirzel Verlag · EAA.
Reconstruction methods for sound visualization based on acousto-optic tomography

The visualization of acoustic fields using acousto-optic tomography has recently proved to yield satisfactory results in the audible frequency range. The current implementation of this visualization technique uses a laser Doppler vibrometer (LDV) to measure the acousto-optic effect, that is, the interaction between sound and light, over an aperture where the acoustic field is to be investigated. By identifying the relationship between the apparent velocity of the LDV and the Radon transform of the acoustic field, it is possible to reconstruct the sound pressure distribution of the scanned area using tomographic techniques. The filtered back projection (FBP) method is the most popular reconstruction algorithm used for tomography in many fields of science. The present study takes the performance of the FBP method in sound visualization as a reference and investigates the use of alternative methods commonly used in inverse problems, e.g., the singular value decomposition and the conjugate gradient methods. A generic formulation for describing the acousto-optic measurement as an inverse problem is thus derived, and the performance of the numerical methods is assessed by means of simulations and experimental results.

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Regularised reconstruction of sound fields with a spherical microphone array

Spherical near field acoustic holography with microphones mounted on a rigid spherical surface is used to reconstruct the incident sound field. However, reconstruction outside the sphere is an ill-posed inverse problem, and since this is very sensitive to the measurement noise, straightforward implementation might lead to disastrous reconstructions. A large number of regularisation tools based on singular value decomposition are available, and it has been found that the acoustic holography problem for certain geometries can be formulated in such a way that similarities to singular value decomposition become apparent. Hence, a number of regularisation methods, including truncated singular value decomposition, standard Tikhonov, constrained Tikhonov, iterative Tikhonov, Landweber and Rutishauser, have been adapted for spherical near field acoustic holography. The accuracy of the methods is examined by means of simulations and measurements, which leads to practical recommendations on the use of regularisation techniques regarding space and frequency.

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Room acoustic transition time based on reflection overlap

A transition time is defined based on the temporal overlap of reflected pulses in room impulse responses. Assuming specular reflections only, the temporal distance between adjacent reflections, which is proportional to the volume of a room, is compared with the characteristic width of a pulse at time t, which is mainly controlled by the absorption characteristics of the boundary surfaces of the room. Scattering, diffuse reflections, and diffraction, which facilitate the overlapping process, have not been taken into account. Measured impulse responses show that the transition occurs earlier in a room with nonuniform absorption and furniture than in a room that satisfies the underlying assumptions.

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The versatility of the acousto-optic measuring principle in characterizing sound fields

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A beamforming system based on the acousto-optic effect

Beamforming techniques are usually based on microphone arrays. The present work uses a beam of light as a sensor element, and describes a beamforming system that locates sound sources based on the acousto-optic effect, this is, the interaction between sound and light. The use of light as a sensing element makes this method immune to spatial aliasing. This feature is illustrated by means of simulation and experimental results. For ease of comparison, the study is supplemented with results obtained with a line array of microphones.

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A combination of the acoustic radiosity and the image source method

A combined model for room acoustic predictions is developed, aiming to treat both diffuse and specular reflections in a unified way. Two established methods are incorporated: acoustical radiosity, accounting for the diffuse part, and the image source method, accounting for the specular part. The model is based on conservation of acoustical energy. Losses are taken into account by the energy absorption coefficient, and the diffuse reflections are controlled via the scattering coefficient, which defines the portion of energy that has been diffusely reflected. The way the model is formulated allows for a dynamic control of the image source production, so that no fixed maximum order is required.

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Contributors: Koutsouris, G. I., Brunskog, J., Jeong, C., Jacobsen, F.
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A Monte-Carlo investigation of the uncertainty of acoustic decay measurements

Measurement of acoustic decays can be problematic at low frequencies: short decays cannot be evaluated accurately. Several effects influencing the evaluation will be reviewed in this paper. As new contribution, the measurement uncertainty due to one-third octave band pass filters will be analysed, taking into account the influence of the magnitude response and the phase distortion. It will be shown how the error not only depends on the filter but also on the modal density and the position of the resonances of the system under test within the frequency band. A Monte-Carlo computer simulation has been set up: the model function is a model of the acoustic decays, where the modal density, the resonances of the system, and the amplitude and phase of the normal modes may be considered as random variables. Once the random
input variables and the model function are defined, the uncertainty of acoustic decay measurements can be estimated. Different filters will be analysed: linear phase FIR and IIR filters both in their direct and time-reversed versions. © European Acoustics Association.

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**An acousto-optic beamformer**
There is a great variety of beamforming techniques that can be used for localization of sound sources. The differences among them usually lie in the array layout or in the specific signal processing algorithm used to compute the beamforming output. Any beamforming system consists of a finite number of transducers, which makes beamforming methods vulnerable to spatial aliasing above a certain frequency. The present work uses the acousto-optic effect, i.e., the interaction between sound and light, to localize sound sources in a plane. The use of a beam of light as the sensing element is equivalent to a continuous line aperture with an infinite number of microphones. This makes the proposed acousto-optic beamformer immune to spatial aliasing. This unique feature is illustrated by means of simulations and experimental results within the entire audible frequency range. For ease of comparison, the study is supplemented with measurements carried out with a line array of microphones.

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**A note on the modal kurtosis and the concentration factor in reverberation rooms**
The effect known as "weak Anderson localization," "coherent backscattering," or "enhanced back-scattering" is a physical phenomenon that occurs in random systems, e.g., disordered media and linear wave systems, including reverberation...
rooms: The mean square response is increased at the drive point. In a reverberation room, this means that one can expect an increase of the reverberant sound field at the position of the source that generates the sound field. This affects the sound power output of the source and is therefore of practical concern. The relative increase of reverberant energy is described by the concentration factor, which is usually assumed to be 2. However, because of the stronger direct sound field at the source position, it is obviously very difficult to measure this quantity directly under steady-state conditions. A related parameter of crucial importance for the ensemble statistics of responses in rooms is the modal kurtosis, which is usually assumed to be 3. The modal kurtosis is also very difficult to measure directly. This paper presents the results of an indirect experimental estimation of the two parameters.

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A practical implementation of microphone free-field comparison calibration according to the standard IEC 61094-8
An international standard concerned with the calibration of microphones in a free field by comparison has recently been published. The standard contemplates two main calibration methodologies for determining the sensitivity of a microphone under test when compared against a reference microphone. The two methodologies assume that the two microphones are exposed to the same sound pressure. This can be achieved by measuring the ratio of output voltages either sequentially or simultaneously. The first method requires a stable source to ensure that the sound pressure is approximately the same when the reference and test microphones are measured, whereas the second requires a source with a symmetrical directivity that ensures that the microphones placed at opposite positions are subjected to the same sound pressure. The two methods have been investigated experimentally in an extended frequency range. A third method, consisting of a combination of the sequential and simultaneous methodologies, has also been investigated. Though the application of time selective techniques is not discussed, the experimental results indicate the immunity to unwanted reflections in the sequential and combined approaches while it may be necessary to apply these techniques in the simultaneous approach.

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Control of sound fields with a circular double-layer array of loudspeakers

This investigation is concerned with generating a controlled sound field for listeners inside a circular array of loudspeakers without disturbing people outside the array. Ideally this configuration would have the advantage that reflections from the surroundings would be of no concern. Inspired by the Kirchhoff-Helmholtz integral theorem a double-layer array of loudspeakers is used. Several solution methods are suggested and examined with computer simulations: pure contrast control, pure pressure matching, and a weighted combination. In order to compare the performance of the methods two performance indices are used, i) the ratio of the sound energy in the listening zone to the sound energy in the quiet zone, and ii) a normalised measure of the deviations between the desired and the generated sound field in the listening zone. The best compromise is obtained with the method that combines pure contrast control with a pressure matching technique.

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Direct formulation of the supersonic acoustic intensity in space domain

This paper proposes and examines a direct formulation in space domain of the so-called supersonic acoustic intensity. This quantity differs from the usual (active) intensity by excluding the circulating energy in the near-field of the source, providing a map of the acoustic energy that is radiated into the far field. To date, its calculation has been formulated in the wave number domain, filtering out the evanescent waves outside the radiation circle and reconstructing the acoustic field with only the propagating waves. In this study, the supersonic intensity is calculated directly in space domain by means of a two-dimensional convolution between the acoustic field and a spatial filter mask that corresponds to the space domain representation of the radiation circle. Therefore, the acoustic field that propagates effectively to the far field is calculated via direct filtering in space domain. This paper presents the theory, as well as a numerical example to illustrate some fundamental principles. An experimental study on planar radiators was conducted to verify the validity of the technique. The experimental results are presented, and serve to illustrate the usefulness of the analysis, its strengths and limitations. © 2012 Acoustical Society of America.

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Improving the resolution of three-dimensional acoustic imaging with planar phased arrays

This paper examines and compares two methods of improving the quality of three-dimensional beamforming with phased microphone arrays. The intended application is the detection of aerodynamic noise sources on wind turbines. Both methods employ Fourier based deconvolution. The first method involves a transformation of coordinates that tends to make the response to a point source, the point spread function, more shift invariant. The result is a significant improvement in sound source imaging in the transformed coordinate system. However, the inverse transformation to Cartesian coordinates introduces range dependent resolution limitations because of the irregular distribution of the focal points. The second method combines the transformation of coordinates with an alternative scanning technique. This method can be used in near field three-dimensional acoustic imaging to produce maps free of sidelobes and with constant resolution. The robustness of the proposed methods is validated both with computer simulations and experimentally.

Investigating the use of the acousto-optic effect for acoustic holography

Recent studies have demonstrated that the acousto-optic effect, that is, the interaction between sound and light, can be used as a means to visualize acoustic fields in the audible frequency range. The changes of density caused by sound waves propagating in air induce phase shifts to a laser beam that travels through the acoustic field. This phenomenon can in practice be captured with a laser Doppler vibrometer (LDV), and the pressure distribution of the acoustic field can be reconstructed using tomography. The present work investigates the potential of the acousto-optic effect in acoustic holography. Two different holographic methods are examined for this purpose. One method first reconstructs the hologram plane using acousto-optic tomography and then propagates it using conventional near-field acoustic holography (NAH). The other method exploits the so-called Fourier Slice Theorem and bases all the calculations of the holographic algorithm on the Radon transform of the acoustic field. The validity of the proposed methods is examined in a simple study case by means of simulations and preliminary measurements.
Noise mapping inside a car cabin

The mapping of noise is of considerable interest in the car industry where a good noise mapping can make it much easier to identify the sources that generate the noise and eventually reduce the individual contributions to the noise. The methods used for this purpose include delay-and-sum beamforming and spherical harmonics beamforming. These methods have a poor spatial resolution at low frequencies, and since much noise generated in cars is dominated by low frequencies the methods are not optimal. In the present paper the mapping is done by solving an inverse problem with a transfer matrix between the volume velocities of the sources and the measured sound pressures at the microphone array. This is an illposed problem and therefore regularisation have to be applied when the transfer matrix is inverted in order to give good results.

Sound field control with a circular double-layer array of loudspeakers

This paper describes a method of generating a controlled sound field for listeners inside a circular array of loudspeakers without disturbing people outside the array appreciably. To achieve this objective, a double-layer array of loudspeakers is used. Several solution methods are suggested, and their performance is examined using computer simulations. Two performance indices are used in this work, (a) the level difference between the average sound energy density in the listening zone and that in the quiet zone (sometimes called “the acoustic contrast”), and (b) a normalized measure of the deviations between the desired and the generated sound field in the listening zone. It is concluded that the best compromise is obtained with a method that combines pure contrast maximization with a pressure matching technique.

Sound field reconstruction using acousto-optic tomography

When sound propagates through a medium, it results in pressure fluctuations that change the instantaneous density of the medium. Under such circumstances, the refractive index that characterizes the propagation of light is not constant, but influenced by the acoustic field. This kind of interaction is known as the acousto-optic effect. The formulation of this physical phenomenon into a mathematical problem can be described in terms of the Radon transform, which makes it possible to reconstruct an arbitrary sound field using tomography. The present work derives the fundamental equations governing the acousto-optic effect in air, and demonstrates that it can be measured with a laser Doppler vibrometer in the audible frequency range. The tomographic reconstruction is tested by means of computer simulations and measurements. The main features observed in the simulations are also recognized in the experimental results. The effectiveness of the tomographic reconstruction is further confirmed with representations of the very same sound field measured with a traditional microphone array.

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Sound field separation with sound pressure and particle velocity measurements

In conventional near-field acoustic holography (NAH) it is not possible to distinguish between sound from the two sides of the array, thus, it is a requirement that all the sources are confined to only one side and radiate into a free field. When this requirement cannot be fulfilled, sound field separation techniques make it possible to distinguish between outgoing and incoming waves from the two sides, and thus NAH can be applied. In this paper, a separation method based on the measurement of the particle velocity in two layers and another method based on the measurement of the pressure and the velocity in a single layer are proposed. The two methods use an equivalent source formulation with separate transfer matrices for the outgoing and incoming waves, so that the sound from the two sides of the array can be modeled independently. A weighting scheme is proposed to account for the distance between the equivalent sources and
measurement surfaces and for the difference in magnitude between pressure and velocity. Experimental and numerical studies have been conducted to examine the methods. The double layer velocity method seems to be more robust to noise and flanking sound than the combined pressure-velocity method, although it requires an additional measurement surface. On the whole, the separation methods can be useful when the disturbance of the incoming field is significant. Otherwise the direct reconstruction is more accurate and straightforward.

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The effect of scattering on sound field control with a circular double-layer array of loudspeakers

A recent study has shown that a circular double-layer array of loudspeakers makes it possible to achieve a sound field control that can generate a controlled field inside the array and reduce sound waves propagating outside the array. This is useful if it is desirable not to disturb people outside the array or to prevent the effect of reflections from the room. The study assumed free field condition, however in practice a listener will be located inside the array. The listener scatters sound waves, which propagate outward. Consequently, the scattering effect can be expected to degrade the performance of the system. This paper computationally examines the scattering effect based on the simple assumption that the listener’s head is a rigid sphere. In addition, methods to solve the problem are discussed.

The effect of scattering on sound field control with a circular double-layer array of loudspeakers

A recent study has shown that a circular double-layer array of loudspeakers makes it possible to achieve a sound field control that can generate a controlled field inside the array and reduce sound waves propagating outside the array. This is useful if it is desirable not to disturb people outside the array or to prevent the effect of reflections from the room. The study assumed free field condition, however in practice a listener will be located inside the array. The listener scatters sound waves, which propagate outward. Consequently, the scattering effect can be expected to degrade the performance of the system. This paper computationally examines the scattering effect based on the simple assumption that the listener’s head is a rigid sphere. In addition, methods to solve the problem are discussed.
The influence of the group delay of digital filters on acoustic decay measurements
In this paper the error due to the phase response of digital filters on acoustic decay measurements is analyzed. There are two main sources of errors when an acoustic decay is filtered: the error due to the bandwidth of the filters related to their magnitude response, and the error due to their phase response. In this investigation the two components are separated and the phase error analyzed in terms of the group delay of the filters. Linear phase FIR filters and minimum phase IIR filters fulfilling the class 1 requirements of the IEC 61260 standard have been designed, and their errors compared. This makes it possible to explain the behavior of the phase error and develop recommendations for the use of each filtering technique. The paper is focused on the filtering techniques covered by current versions of the standards for measurement of acoustic decays and in the evaluation of the acoustic decay for narrow filters at low frequencies and low reverberation times (BT

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Thresholds for the slope ratio in determining transition time and quantifying diffuser performance in situ
This study is concerned with an objective measure called the slope ratio that can detect acoustic defects due to unexpected pressure increases such as strong reflections and coincidental constructive interference. The slope ratio is the ratio of the instantaneous slope to the mean slope in a decay curve. The slope ratio was suggested for determining the room acoustic transition time experimentally, but its threshold criteria have not been thoroughly investigated. The thresholds for the slope ratio, particularly for applications such as determining the room acoustic transition time and quantifying in situ diffuseness, are examined for various room impulse responses. For the tested rooms, a slope ratio threshold of 11 gives the most consistent and systematic results.

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A comparison of two strategies for generating sound zones in a room.

For some purposes it may be of interest to generate sound zones with different acoustic properties in a room. This paper compares two strategies for generating such zones. One method is based on 'contrast optimisation': the idea is to maximise the ratio of the potential energy in a 'bright' (esonified) zone to the potential energy in a 'dark' (quiet) zone with a given source configuration. An alternative method based on 'sound field synthesis' has the more ambitious goal to control the sound field in the bright zone in detail, for example, to generate a plane wave that propagates in a certain direction. The two methods are analysed theoretically and examined through simulations and experimentally.

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Contributors: Jacobsen, F., Olsen, M., Møller, M., Agerkvist, F. T.
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Acoustical source mapping based on deconvolution approaches for circular microphone arrays

Recently, the aeroacoustic community has examined various methods based on deconvolution to improve the visualization of acoustic fields scanned with planar arrays of microphones. These methods are based on the assumption that the beamforming map in an observation plane parallel to the array can be approximated by a convolution of the actual sources and the beamformer’s point spread-function, i.e., the beamformer’s response to a point source. By deconvolving the resulting map, the resolution is improved and the side-lobes effect is reduced or even eliminated compared to conventional beamforming. Even though these methods are originally designed for planar sparse arrays, they can be adapted to uniform circular arrays for mapping the sound over 360°. Such geometry has the advantage that the beamforming response has always the same shape around the focusing direction, or in other words, that the beamformer’s point-spread function is shift-invariant, which makes it possible to apply spectral procedures on the entire region of interest so that the deconvolution algorithm becomes computationally more efficient. This investigation examines the matter by means of computer simulations and experimental measurements.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Tiana Roig, E., Jacobsen, F.
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Adaptive Feedback Cancellation With Band-Limited LPC Vocoder in Digital Hearing Aids

Feedback oscillation is one of the major issues with hearing aids. An effective way of feedback suppression is adaptive feedback cancellation, which uses an adaptive filter to estimate the feedback path. However, when the external input signal is correlated with the receiver input signal, the estimate of the feedback path is biased. This so-called “bias problem” results in a large modeling error and a cancellation of the desired signal. This paper proposes a band-limited linear predictive coding based approach to reduce the bias. The idea is to replace the hearing-aid output with a synthesized signal, which sounds perceptually the same as or similar to the original signal but is statistically uncorrelated with the external input signal at high frequencies where feedback oscillation usually occurs. Simulation results show that the proposed algorithm can effectively reduce the bias and the misalignment between the real and the estimated feedback path. When combined with filtered-X adaptation in the feedback canceller, this approach reduces the misalignment even further.

A new interpretation of distortion artifacts in sweep measurements

The characterization of acoustical spaces by means of impulse response measurements is often biased by the nonlinear behavior of the loudspeaker used to excite the system under test. In this context the distortion immunity provided by the sweep technique has been investigated. The results show that the sweep method can reject a significant amount of distortion artifacts but, in contrast to what is claimed in the literature, it cannot exclude all distortion artifacts from the causal part of the estimated impulse response.
An investigation of sound fields based on the acousto-optic effect

Various types of transducers are nowadays capable of translating different properties of sound waves into mechanical/electrical quantities, which can afterwards be reinterpreted into acoustical ones. However, in certain applications, for example when using microphone arrays, the presence of bulk transducers can bias the acoustic measurement. Although this influence can often be either neglected at low frequencies or compensated for (typically in the form of a frequency response), the present work alternatively explores the interaction between sound and light as a means to characterize an acoustic field. This non-invasive technique is based on the so-called acousto-optic effect, i.e., the variations of the refractive index of a medium caused by density fluctuations, which follow sound pressure fluctuations. In the current study, this phenomenon is investigated in air, within the audible frequency range, and in two different measurement scenarios where the sound field is well-known: in a rectangular duct and in an anechoic room. Models for predicting the acousto-optic effect in such scenarios are derived and measurements are carried out with a laser Doppler vibrometer. The results show a fairly good agreement between the experimental and simulated data.

Beamforming with a circular array of microphones mounted on a rigid sphere (L)

Beamforming with uniform circular microphone arrays can be used for localizing sound sources over 360. Typically, the array microphones are suspended in free space or they are mounted on a solid cylinder. However, the cylinder is often considered to be infinitely long because the scattering problem has no exact solution for a finite cylinder. Alternatively one can use a solid sphere. This investigation compares the performance of a circular array mounded on a rigid sphere with that of such an array in free space and mounted on an infinite cylinder, using computer simulations. The examined techniques are delay-and-sum and circular harmonics beamforming, and the results are validated experimentally.
Ensemble statistics of active and reactive sound intensity in reverberation rooms
This paper examines fundamental statistical properties of the active and reactive sound intensity in reverberant enclosures driven with pure tones. The existing theory for sound intensity in a diffuse sound field, which is based on Waterhouse's random wave model and therefore limited to the region of high modal overlap, is extended to the region of low modal overlap by taking account of the random fluctuations of the sound power emitted by the source that generates the sound field. The validity of the extended model is confirmed by experimental and numerical results.

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Extracting the invariant model from the feedback paths of digital hearing aids
Feedback whistling is a severe problem with hearing aids. A typical acoustical feedback path represents a wave propagation path from the receiver to the microphone and includes many complicated effects among which some are invariant or nearly invariant for all users and in all acoustical environments given a specific type of hearing aids. Based on this observation, a feedback path model that consists of an invariant model and a variant model is proposed. A common-acoustical-pole and zero model-based approach and an iterative least-square search-based approach are used to extract the invariant model from a set of impulse responses of the feedback paths. A hybrid approach combining the two methods is also proposed. The general properties of the three methods are studied using artificial datasets, and the methods are cross-validated using the measured feedback paths. The results show that the proposed hybrid method gives the best overall performance, and the extracted invariant model is effective in modeling the feedback path.
Free-field calibration of measurement microphones at high frequencies
Measurement microphones are typically calibrated in a free field at frequencies up to 50 kHz. This is a sufficiently high frequency for the most of sound measurement applications related with noise assessment. However, other applications such as assessment of the noise emitted by ultrasound cleaning machines, and fail detection in aeronautic structures require that the sensitivity of the microphone is known at frequencies up to 150 kHz. Such a high frequency can only be reached using small measurement microphones with very low sensitivity. Thus, in order to extend the frequency range of free-field calibration the measurement system, and measurement methods must undergo a series of changes and adaptations including the type of measurement signal, methods for eliminating unwanted reflections from walls, cross-talk, etc. Furthermore, the properties of the measurement microphones used in high frequency calibration are relatively less known, and must be determined either from experimental methods or numerically. This paper presents the results of an initial investigation of the calibration of measurement microphones at high frequencies. A strategy for the changes and adaptations to the existing measurement methodologies, and the determination of the microphone parameters is outlined.

Generation of sound zones in 2.5 dimensions
A method for generating sound zones with different acoustic properties in a room is presented. The method is an extension of the two-dimensional multi-zone sound field synthesis technique recently developed by Wu and Abhayapala; the goal is, for example, to generate a plane wave that propagates in a certain direction within a certain region of a room and at the same time suppress sound in another region. The method is examined through simulations and experiments. For comparison a simpler method based on the idea of maximising the ratio of the potential acoustic energy in an ensonified zone to the potential acoustic energy in a quiet zone is also examined.
Improvements of the smearing technique for cross-stiffened thin rectangular plates

New developments in the simplified smearing technique for modeling vibrations of cross-stiffened, thin rectangular plates are presented. The computationally efficient smearing technique has been known for many years, but so far the accuracy of, say, predicted natural frequencies has been inadequate. The reason is that only the stiffeners at a right angle to the axis of angular motion are taken into account when calculating the bending stiffness, whereas the stiffeners that are parallel to this axis of angular motion are neglected. To improve predictions, the parallel stiffeners are taken into account in this paper. The improved smearing technique results in better accuracy for predicted natural frequencies of flat stiffened plates, as demonstrated for both simply supported and clamped boundary conditions. The improved prediction accuracy is demonstrated by comparing results from a numerical model based on the current development with results from finite element (FE) simulations that include the exact cross-sectional geometries of the stiffened panel. In order to demonstrate applications of the improved smearing technique, the predicted forced response is compared with both experimental and FE results. Another improvement concerns the orientation of the stiffeners. The original smearing technique presupposes that the stiffeners are parallel to the edges of the plate, but simple considerations make it possible to relax this requirement. To test the validity of the resulting technique a series of plates are examined for stiffeners angled relative to the plate edges.
Near field acoustic holography with microphones on a rigid sphere

Spherical near field acoustic holography (spherical NAH) is a technique that makes it possible to reconstruct the sound field inside and just outside a spherical surface on which the sound pressure is measured with an array of microphones. This is potentially very useful for source identification. The sphere can be acoustically transparent or it can be rigid. A rigid sphere is somewhat more practical than an open sphere. However, spherical NAH based on a rigid sphere is only valid if it can be assumed that the sphere has a negligible influence on the incident sound field, and this is not necessarily a good assumption when the sphere is very close to a radiating surface. This Letter examines the matter through simulations and experiments.

Optimization of multiple-layer microperforated panels by simulated annealing

Sound absorption by microperforated panels (MPP) has received increasing attention the past years as an alternative to conventional porous absorbers in applications with special cleanliness and health requirements. The absorption curve of an MPP depends on four parameters: the holes diameter, the panel thickness, the perforation ratio, and the thickness of the air cavity between the panel and an impervious wall. It is possible to find a proper combination of these parameters that provides an MPP absorbing in one octave band or two, within the frequency range of interest for noise control applications. However, when a wider absorption frequency band is required, it is necessary to design multiple-layer MPP (ML-MPP). The design of an N-layers MPP depends on 4N parameters. Consequently, the tuning of an optimal ML-MPP by exhaustive search within a prescribed frequency band becomes impractical. Therefore, simulated annealing is proposed in this paper as a tool to solve the optimization problem of finding the best combination of the constitutive parameters of an ML-MPP providing the maximum average absorption within a prescribed frequency band.
Practical computational aeroacoustics for compact surfaces in low mach number flows

Sound generation has been widely studied using numerical hybrid methods. The aim of this paper is to introduce a flexible procedure where the acoustic source data may be synthesized and stored from commercially available Computational Fluid Dynamics (CFD) codes and later used to predict radiated noise. Different applications will require either analytical or numerical methods for the radiation calculations. Attention is restricted to low Mach number flows where the noise generation is dominated by the interaction of the flow with a surface with at least one characteristic dimension short compared to the wavelength of interest. This makes it possible to focus on the surface source term of the Ffowcs Williams-Hawkins equation. In this paper, in order to illustrate the basic method for storing and utilizing data from the CFD analysis, the flow past a circular cylinder at a Reynolds number of $Re = 1.4 \times 10^5$ will be studied, where the cylinder is compact and therefore the analytical free-space Green's function may be used. © S. Hirzel Verlag.

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Organisations: Acoustic Technology, Department of Electrical Engineering, University of Navarra, Lloyd's Register Group Ltd.
Contributors: Pradera-Mallabiabarrena, A., Keith, G., Jacobsen, F., Rivas, A., Gil-Negrete, N.
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Smearing technique for vibration analysis of simply supported cross-stiffened and doubly curved thin rectangular shells

Plates stiffened with ribs can be modeled as equivalent homogeneous isotropic or orthotropic plates. Modeling such an equivalent smeared plate numerically, say, with the finite element method requires far less computer resources than modeling the complete stiffened plate. This may be important when a number of stiffened plates are combined in a complicated assembly composed of many plate panels. However, whereas the equivalent smeared plate technique is well established and recently improved for flat panels, there is no similar established technique for doubly curved stiffened shells. In this paper the improved smeared plate technique is combined with the equation of motion for a doubly curved thin rectangular shell, and a solution is offered for using the smearing technique for stiffened shell structures. The developed prediction technique is validated by comparing natural frequencies and mode shapes as well as forced responses from simulations based on the smeared theory with results from experiments with a doubly curved cross-stiffened shell. Moreover, natural frequencies of cross-stiffened panels determined by finite element simulations that include the exact cross-sectional geometries of panels with cross-stiffeners are compared with predictions based on the smeared theory for a range of different panel curvatures. Good agreement is found.

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Contributors: Luan, Y., Ohlrich, M., Jacobsen, F.
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Sound field reconstruction based on the acousto-optic effect

Acoustic measurements are usually carried out with transducers that interact mechanically with the sound field under investigation. The goal of this work is to employ a completely different measurement principle, the determination of sound pressure based on the interaction between sound and light, namely the acousto-optic effect. When sound propagates through a medium, it gives rise to pressure fluctuations that change the instantaneous density of the medium. Under such circumstances, the speed of light is not constant, but changed by the acoustic field. This acousto-optic interaction can be measured with a laser Doppler vibrometer; furthermore, it can be exploited to characterize an arbitrary sound field using tomographic techniques. This paper briefly reviews the fundamental principles governing the acousto-optic effect in air, and presents an investigation of the tomographic reconstruction within the audible frequency range by means of simulations and experimental results. The good agreement observed between simulations and measurements is further confirmed with representations of the sound field obtained with traditional microphone array measurements.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Torras Rosell, A., Barrera Figueroa, S., Jacobsen, F.
Publication date: 2011
Sound field separation with a double layer velocity transducer array (L)
In near-field acoustic holography sound field separation techniques make it possible to distinguish between sound coming from the two sides of the array. This is useful in cases where the sources are not confined to only one side of the array, e.g., in the presence of additional sources or reflections from the other side. This paper examines a separation technique based on measurement of the particle velocity in two closely spaced parallel planes. The purpose of the technique is to recover the particle velocity radiated by a source in the presence of disturbing sound from the opposite side of the array. The technique has been examined and compared with direct velocity based reconstruction, as well as with a technique based on the measurement of the sound pressure and particle velocity. The double layer velocity method circumvents some of the drawbacks of the pressure-velocity based reconstruction, and it can successfully recover the normal velocity radiated by the source, even in the presence of strong disturbing sound. © 2011 Acoustical Society of America.

Supersonic acoustic intensity with statistically optimized near-field acoustic holography
The concept of supersonic acoustic intensity was introduced some years ago for estimating the fraction of the flow of energy radiated by a source that propagates to the far field. It differs from the usual (active) intensity by excluding the near-field energy resulting from evanescent waves and circulating energy in the near-field of the source. This quantity is of concern because it makes it possible to identify the regions of a source that contribute to the far field radiation, which is often the ultimate concern in noise control. Therefore, this is a very useful analysis tool complementary to the information provided by the near-field acoustic holography technique. This study proposes a version of the supersonic acoustic intensity applied to statistically optimized near-field acoustic holography (SONAH). The theory, numerical results and an experimental study are presented. The possibility of using particle velocity measurements instead of conventional pressure measurements is examined. The study indicates that the calculation of the supersonic intensity based on measurement of the particle velocity is more accurate than the one based on sound pressure measurements. It also indicates that the method based on SONAH is somewhat more accurate than the existing methodology based on Fourier transform processing.
Weak Anderson localisation in reverberation rooms and its effect on the uncertainty of sound power measurements

The effect known as ‘weak Anderson localisation’, ‘coherent backscatter’ or ‘enhanced backscattering’ is a physical phenomenon that occurs in random systems, e.g., disordered media and linear wave systems, including reverberation rooms: the mean square response is increased at the drive point. In a reverberation room this means that one can expect an increase of the reverberant sound field at the position of the source that generates the sound field. This affects the sound power output of the source and is therefore of practical concern. However, because of the stronger direct sound field at the source position it is obviously very difficult to measure the effect directly. The ‘concentration factor’ is usually assumed to be 2, and the ‘modal kurtosis’ is assumed to be 3. This paper presents the results of an indirect experimental estimation of these two parameters, and discusses implications for the uncertainty of sound power measurements.

An investigation on methods for free-field comparison calibration of measurement microphones.

A note on measurement of low-frequency noise in rooms
A room acoustic transition time based on the overlap of reflected waves.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jeong, C., Brunskog, J., Jacobsen, F.
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A theoretical approach to room acoustic simulations based on a radiative transfer model
A theoretical approach to room acoustic simulations based on a radiative transfer model is developed by adapting the classical radiative transfer theory from optics to acoustics. The proposed acoustic radiative transfer model expands classical geometrical room acoustic modeling algorithms by incorporating a propagation medium that absorbs and scatters radiation, handling both diffuse and non-diffuse reflections on boundaries and objects in the room. The main scope of this model is to provide a proper foundation for a wide number of room acoustic simulation models, in order to establish and unify their principles. It is shown that this room acoustic modeling technique establishes the basis of two recently proposed algorithms, the acoustic diffusion equation and the room acoustic rendering equation. Both methods are derived in detail using an analytical approximation and a simplified integral equation of the proposed method, respectively, allowing a clear definition of the underlying assumptions, limitations, advantages and disadvantages.

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Beamforming with a circular microphone array for localization of environmental noise sources
It is often enough to localize environmental sources of noise from different directions in a plane. This can be accomplished with a circular microphone array, which can be designed to have practically the same resolution over 360. The microphones can be suspended in free space or they can be mounted on a solid cylinder. This investigation examines and compares two techniques based on such arrays, the classical delay-and-sum beamforming and an alternative method called circular harmonics beamforming. The latter is based on decomposing the sound field into a series of circular harmonics. The performance of the two signal processing techniques is examined using computer simulations, and the results are validated experimentally.
Beamforming with a circular microphone array for localization of environmental sources of noise.

It is often enough to localize environmental sources of noise from different directions in a plane. This can be accomplished with a circular microphone array, which can be designed to have practically the same resolution over 360. The microphones can be suspended in free space or they can be mounted on a solid cylinder. This investigation examines and compares two techniques based on such arrays, the classical delay-and-sum beamforming and an alternative method called circular harmonics beamforming. The latter is based on decomposing the sound field into a series of circular harmonics. The performance of the two signal processing techniques is examined using computer simulations, and the results are validated experimentally.

Effort variation regularization in sound field reproduction

In this paper, active control is used in order to reproduce a given sound field in an extended spatial region. A method is proposed which minimizes the reproduction error at a number of control positions with the reproduction sources holding a certain relation within their complex strengths. Specifically, it is suggested that the phase differential of the source driving signals should be in agreement with the phase differential of the desired sound pressure field. The performance of the suggested method is compared with that of conventional effort regularization, wave field synthesis (WFS), and adaptive wave field synthesis (AWFS), both under free-field conditions and in reverberant rooms. It is shown that effort variation regularization overcomes the problems associated with small spaces and with a low ratio of direct to reverberant energy, improving thus the reproduction accuracy in the listening room.
Improving the resolution of beamforming measurements on wind turbines

The spatial resolution of a beamformer based on a planar microphone array in a measurement plane parallel to the array can be approximated by a two-dimensional convolution of the actual distribution of incoherent sources and the beamformer’s response to a point source. Several methods are available for deconvolving the resulting blurred picture and thus improving the resulting resolution. This investigation is concerned with a similar deconvolution for the three-dimensional case.

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Influence of the turbulent boundary layer pressure fluctuation on the sound intensity measurement in a mean flow

The influence of turbulent boundary layer pressure fluctuation on the sound intensity measurement in a flow is a subject of practical concern, because the sound intensity probe is generally exposed to the airflow and is sensed the turbulent boundary layer (TBL) pressure fluctuation which may even overwhelm the true source pressure in some cases. In this paper, the model of the sound intensity caused by the TBL pressure fluctuation is described firstly. Based upon the developed model, the sound intensity caused by the TBL pressure fluctuation is calculated using the available models of the wave-vector frequency spectra of the TBL pressure fluctuation. In order to validate the model and the numerical results, a serious of measurements were carried out. It is shown that the calculated results of the TBL pressure fluctuation agree fairly well with the measured results which are corrected with the estimated spatial response function of the microphone. Also, the characteristics of the measured sound intensity are consistent with that of the calculated sound intensity.

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Measurement of the sound power incident on the walls of a reverberation room with near field acoustic holography

The conventional method of measuring the insertion loss of a partition relies on an assumption of the sound field in the source room being diffuse combined with the classical relation between the spatial average of the mean square pressure in the source room and the incident sound power per unit area; and it has always been regarded as impossible to measure the sound power that is incident on a wall directly. This paper examines a new method of determining this quantity from sound pressure measurements at positions on the wall using 'statistically optimised near field acoustic holography' (SONAH). The purpose is to examine whether one should use a correction similar to the well-known 'Waterhouse correction' when the incident sound power is deduced from the sound pressure in the source room.

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Measuring long impulse responses with pseudorandom sequences and sweep signals

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Noise injection for feedback cancellation with linear prediction

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Radiation impedance of condenser microphones and their diffuse-field responses.
The relation between the diffuse-field response and the radiation impedance of a microphone has been investigated. Such a relation can be derived from classical theory. The practical measurement of the radiation impedance requires (a) measuring the volume velocity of the membrane of the microphone and (b) measuring the pressure on the membrane of the microphone. The first measurement is carried out by means of laser vibrometry. The second measurement cannot be implemented in practice. However, the pressure on the membrane can be calculated numerically by means of the boundary element method. In this way, a hybrid estimate of the radiation impedance is obtained. The resulting estimate of the diffuse-field response is compared with experimental estimates of the diffuse-field response determined using reciprocity and the random-incidence method. The different estimates are in good agreement at frequencies below the resonance frequency of the microphone. Although the method may not be of great practical utility, it provides a useful validation of the estimates obtained by other means.

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Room acoustic transition time based on reflection overlap
A transition time is defined based on the temporal overlap of reflected pulses in room impulse responses. Assuming specular reflections only, the temporal distance between adjacent reflections, which is proportional to the volume of a room, is compared with the characteristic width of a pulse at time t, which is mainly controlled by the absorption characteristics of the boundary surfaces of the room. Scattering, diffuse reflections, and diffraction, which facilitate the
overlapping process, have not been taken into account. Measured impulse responses show that the transition occurs earlier in a room with nonuniform absorption and furniture than in a room that satisfies the underlying assumptions.

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Contributors: Jeong, C., Brunskog, J., Jacobsen, F.
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**Separation of radiated sound field components from waves scattered by a source under non-anechoic conditions.**
A method of estimating the sound field radiated by a source under non-anechoic conditions has been examined. The method uses near field acoustic holography based on a combination of pressure and particle velocity measurements in a plane near the source for separating outgoing and ingoing wave components. The outgoing part of the sound field is composed of both radiated and scattered waves. The method compensates for the scattered components of the outgoing field on the basis of the boundary condition of the problem, exploiting the fact that the sound field is reconstructed very close to the source. Thus the radiated free-field component is estimated simultaneously with solving the inverse problem of reconstructing the sound field near the source. The method is particularly suited to cases in which the overall contribution of reflected sound in the measurement plane is significant.

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Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Fernandez Grande, E., Jacobsen, F.
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**Statistical properties of kinetic and total energy densities in reverberant spaces**
Many acoustical measurements, e.g., measurement of sound power and transmission loss, rely on determining the total sound energy in a reverberation room. The total energy is usually approximated by measuring the mean-square pressure (i.e., the potential energy density) at a number of discrete positions. The idea of measuring the total energy density instead
of the potential energy density on the assumption that the former quantity varies less with position than the latter goes back to the 1930s. However, the phenomenon was not analyzed until the late 1970s and then only for the region of high modal overlap, and this analysis has never been published. Moreover, until fairly recently, measurement of the total sound energy density required an elaborate experimental arrangement based on finite-difference approximations using at least four amplitude and phase matched pressure microphones. With the advent of a three-dimensional particle velocity transducer, it has become somewhat easier to measure total rather than only potential energy density in a sound field. This paper examines the ensemble statistics of kinetic and total sound energy densities in reverberant enclosures theoretically, experimentally, and numerically.

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**Statistics of sound intensity in reverberation rooms.**

**General information**
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., Molares, A.
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Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2010 › Research › peer-review

**The ensemble variance of pure-tone measurements in reverberation rooms.**
Reverberation rooms are often used for measuring the sound power emitted by sources of sound. At medium and high frequencies, where the modal overlap is high, a fairly simple model based on sums of waves from random directions having random phase relations gives good predictions of the ensemble statistics of measurements in such rooms. Below the Schroeder frequency, the relative variance is much larger, particularly if the source emits a pure-tone. The established theory for this frequency range is based on ensemble statistics of modal sums and requires knowledge of mode shapes and the distribution of modal frequencies. This paper extends the far simpler random wave theory to low frequencies. The two theories are compared, and their predictions are found to compare well with experimental and numerical results.
Using a reflection model for modeling the dynamic feedback path of digital hearing aids

Feedback whistling is one of the severe problems with hearing aids, especially in dynamic situations when the users hug, pick up a telephone, etc. This paper investigates the properties of the dynamic feedback paths of digital hearing aids and proposes a model based on a reflection assumption. The model is compared with two existing models: a direct model and an initialization model, using the measured dynamic feedback paths. The comparison shows that the proposed approach is able to model the dynamic feedback paths more efficiently and accurately in terms of mean-square error and maximum stable gain. The method is also extended to dual-microphone hearing aids to assess the possibility of relating the two dynamic feedback paths through the reflection model. However, it is found that in a complicated acoustic environment, the relation between the two feedback paths can be very intricate and difficult to exploit to yield better modeling of the dynamic feedback paths. (C) 2010 Acoustical Society of America. [DOI: 10.1121/1.3290989]
Vibration of panels with angled stiffeners: A numerical study of the smearing technique.

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Contributors: Luan, Y., Ohlrich, M., Jacobsen, F.
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Analysis of the sources of error in the determination of sound power based on sound intensity measurements

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Santillán, A., Jacobsen, F.
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A new approach for modelling the dynamic feedback path of digital hearing aids

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Ma, G., Gran, F., Jacobsen, F., Agerkvist, F. T.
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DOIs: 10.1109/ICASSP.2009.4959557

A sound field separation technique based on measurements with pressure-velocity probes
It has recently been shown that statistically optimized near field acoustic holography based on measurement with an array of pressure-velocity transducers makes it possible to distinguish between sources on the two sides of the array and thus suppress the influence of a disturbing source [F. Jacobsen and V. Jaud, J. Acoust. Soc. Am. 121, 1550-1558 (2007)].
However, the suggested technique uses a transfer matrix optimized for the source under test and may be expected to perform less well when the disturbing source is not placed symmetrically on the other side of the array, and this will usually be the case. In this letter, a modified method is presented.

Hybrid method for determining the parameters of condenser microphones from measured membrane velocities and numerical calculations

Typically, numerical calculations of the pressure, free-field, and random-incidence response of a condenser microphone are carried out on the basis of an assumed displacement distribution of the diaphragm of the microphone; the conventional assumption is that the displacement follows a Bessel function. This assumption is probably valid at frequencies below the resonance frequency. However, at higher frequencies the movement of the membrane is heavily coupled with the damping of the air film between membrane and backplate and with resonances in the back chamber of the microphone. A solution to this problem is to measure the velocity distribution of the membrane by means of a non-contact method, such as laser vibrometry. The measured velocity distribution can be used together with a numerical formulation such as the boundary element method for estimating the microphone response and other parameters, e.g., the acoustic center. In this work, such a hybrid method is presented and examined. The velocity distributions of a number of condenser microphones have been determined using a laser vibrometer, and these measured velocity distributions have been used for estimating microphone responses and other parameters. The agreement with experimental data is generally good. The method can be used as an alternative for validating the parameters of the microphones determined by classical calibration techniques.
Measurement of incident sound power using near field acoustic holography

The conventional method of measuring the insertion loss of a partition relies on an assumption of the sound field in the source room being diffuse and the classical relation between the spatial average of the mean square pressure in the source room and the incident sound power per unit area; and it has always been regarded as impossible to measure the sound power that is incident on a wall directly. This paper examines a new method of determining this quantity from sound pressure measurements at positions on the wall using 'statistically optimised near field acoustic holography' (SONAH).

The purpose is to examine whether one should use a correction similar to the well-known 'Waterhouse correction' when the incident sound power is deduced from the sound pressure in the source room.

Near field acoustic holography based on the equivalent source method and pressure-velocity transducers

The advantage of using the normal component of the particle velocity rather than the sound pressure in the hologram plane as the input of conventional spatial Fourier transform based near field acoustic holography (NAH) and also as the input of the statistically optimized variant of NAH has recently been demonstrated. This paper examines whether there might be a similar advantage in using the particle velocity as the input of NAH based on the equivalent source method (ESM). Error sensitivity considerations indicate that ESM-based NAH is less sensitive to measurement errors when it is based on particle velocity input data than when it is based on measurements of sound pressure data, and this is confirmed by a simulation study and by experimental results. A method that combines pressure- and particle velocity-based reconstructions in order to distinguish between contributions to the sound field generated by sources on the two sides of the hologram plane is also examined.
On the applicability of the spherical wave expansion with a single origin for near-field acoustical holography

The spherical wave expansion with a single origin is sometimes used in connection with near-field acoustical holography to determine the sound field on the surface of a source. The radiated field is approximated by a truncated expansion, and the expansion coefficients are determined by matching the sound field model to the measured pressure close to the source. This problem is ill posed, and therefore regularization is required. The present paper investigates the consequence of using only the expansion truncation as regularization approach and compares it with results obtained when additional regularization (the truncated singular value decomposition) is introduced. Important differences between applying the method when using a microphone array surrounding the source completely and an array covering only a part of the source are described. Another relevant issue is the scaling of the wave functions. It is shown that it is important for the additional regularization to work properly that the wave functions are scaled in such a way that their magnitude on the measurement surface decreases with the order. Finally, the method is applied on nonspherical sources using a vibrating plate in both simulations and an experiment, and the performance is compared with the equivalent source method.
Patch near field acoustic holography based on particle velocity measurements

Patch near field acoustic holography (PNAH) based on sound pressure measurements makes it possible to reconstruct the source field near a source by measuring the sound pressure at positions on a surface that is comparable in size to the source region of concern. Particle velocity is an alternative input quantity for NAH, and the advantage of using the normal component of the particle velocity rather than the sound pressure as the input of conventional spatial Fourier transform based NAH and as the input of the statistically optimized variant of NAH has recently been demonstrated. This paper examines the use of particle velocity as the input of PNAH. Because the particle velocity decays faster toward the edges of the measurement aperture than the pressure does and because the wave number ratio that enters into the inverse propagator from pressure to velocity amplifies high spatial frequencies, PNAH based on particle velocity measurements can give better results than the pressure-based PNAH with a reduced number of iterations. A simulation study, as well as an experiment carried out with a pressure-velocity sound intensity probe, demonstrates these findings.

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Patch near-field acoustic holography: The influence of acoustic contributions from outside the source

It is a requirement of conventional Near-field Acoustic Holography that the measurement area covers the entire surface of the source. In the case of Patch Near-field Acoustic Holography (patch NAH), the measurement area can be reduced to cover only a specific area of the source which is of particular interest (known as the “patch” or “source patch”). The area of the source beyond this patch is not of interest in the analysis. However, its acoustic output may nevertheless contribute to the total sound field in the measurement plane, and influence the reconstruction of the field close to the patch. The purpose of this paper is to investigate how the acoustic radiation from outside the patch area influences the reconstruction of the sound field close to the source. The reconstruction is based on simulated measurements of sound pressure and particle velocity. The methods used in this paper are the Statistically Optimized NAH (SONAH) and the Equivalent source Method (ESM), also known as the Wave Superposition Method. Particular attention is drawn to how the equivalent sources in the ESM could be distributed in order to achieve an acceptable reconstruction of the sound field. It has been shown that an acceptable reconstruction of the normal velocity can be achieved if the contributions from beyond the patch area are accounted for.

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Contributors: Fernandez Grande, E., Jacobsen, F., Zhang, Y.
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Pressure comparison calibration of measurement microphones in a free field

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Contributors: Barrera Figueroa, S., Ben, M. D., Jacobsen, F.
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Regularization in global sound equalization based on effort variation

Sound equalization in closed spaces can be significantly improved by generating propagating waves that are naturally associated with the geometry, as, for example, plane waves in rectangular enclosures. This paper presents a control approach termed effort variation regularization based on this idea. Effort variation equalization involves modifying the conventional cost function in sound equalization, which is based on minimizing least-squares reproduction errors, by adding a term that is proportional to the squared deviations between complex source strengths, calculated independently for the sources at each of the two walls perpendicular to the direction of propagation. Simulation results in a two-dimensional room of irregular shape and in a rectangular room with sources randomly distributed on two opposite walls demonstrate that the proposed technique leads to smaller global reproduction errors and better equalization performance at listening positions outside of the control region compared to effort regularization and compared to a simple technique that involves driving groups of sources identically.

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Sound power emitted by a pure-tone source in a reverberation room

Energy considerations are of enormous practical importance in acoustics. In "energy acoustics," sources of noise are described in terms of the sound power they emit, the underlying assumption being that this property is independent of the
particular environment where the sources are placed. However, it is well known that the sound power output of a source emitting a pure tone or a narrow band of noise actually varies significantly with its position in a reverberation room at low frequencies, and even larger variations occur between different rooms. The resulting substantial uncertainty in measurements of sound power as well as in predictions based on knowledge of sound power is one of the fundamental limitations of energy acoustics. The existing theory for this phenomenon is fairly complicated and has only been validated rather indirectly. This paper describes a far simpler theory and demonstrates that it gives predictions in excellent agreement with the established theory. The results are confirmed by experimental results as well as finite element calculations.
A comparison of statistically optimized near field acoustic holography using single layer pressure velocity measurements and using double layer pressure measurements

Statistically optimized near field acoustic holography (SONAH) is usually based on the assumption that all sources are on one side of the measurement plane whereas the other side is source free. An extension of the SONAH procedure based on measurement with an array of pressure-velocity probes has recently been suggested. An alternative method uses a double layer array of pressure transducers. Both methods make it possible to distinguish between sources on the two sides of the array and thus suppress the influence of extraneous noise and reflections coming from the “wrong” side. This letter compares the two methods.


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A comparison of statistically optimized near field acoustic holography using single layer pressure velocity measurements and using double layer pressure measurements

Statistically optimized near field acoustic holography (SONAH) is usually based on the assumption that all sources are on one side of the measurement plane whereas the other side is source free. An extension of the SONAH procedure based on measurement with an array of pressure-velocity probes has recently been suggested. An alternative method uses a double layer array of pressure transducers. Both methods make it possible to distinguish between sources on the two sides of the array and thus suppress the influence of extraneous noise and reflections coming from the “wrong” side. This letter compares the two methods.


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Acoustic Signals and Systems
The Handbook of Signal Processing in Acoustics will compile the techniques and applications of signal processing as they are used in the many varied areas of Acoustics. The Handbook will emphasize the interdisciplinary nature of signal processing in acoustics. Each Section of the Handbook will present topics on signal processing which are important in a specific area of acoustics. These will be of interest to specialists in these areas because they will be presented from their technical perspective, rather than a generic engineering approach to signal processing. Non-specialists, or specialists from different areas, will find the self-contained chapters accessible and will be interested in the similarities and differences between the approaches and techniques used in different areas of acoustics.

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A method of measuring the Green's function in an enclosure
The acoustic Green's function can be measured using a device with two matched microphones mounted in a tube driven by a loudspeaker combined with another microphone that represents the observation point. Good agreement is obtained between the measured and theoretical Green's function in a rectangular room below 320 Hz. At higher frequencies the agreement is less good because of the imperfect geometry of the room.

General information
Analysis of room transfer function and reverberant signal statistics

General information
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Contributors: Luan, Y., Jacobsen, F.
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Analysis of room transfer function and reverberant signal statistics: Abstract of paper
For some time now, statistical analysis has been a valuable tool in analyzing room transfer functions (RTFs). This work examines existing statistical time-frequency models and techniques for RTF analysis (e.g., Schroeder’s stochastic model and the standard deviation over frequency bands for the RTF magnitude and phase). RTF fractional octave smoothing, as with 1-slash 3 octave analysis, may lead to RTF simplifications that can be useful for several audio applications, like room compensation, room modeling, auralisation purposes. The aim of this work is to identify the relationship of optimal response smoothing (e.g., as in complex smoothing) with respect to the original RTF statistics. More specifically, the RTF statistics, derived after the complex smoothing calculation, are compared to the original statistics across space inside typical rooms, by varying the source, the receiver position and the corresponding ratio of the direct and reverberant signal. In addition, this work examines the statistical quantities for speech and audio signals prior to their reproduction within rooms and when recorded in rooms. Histograms and other statistical distributions are used to compare RTF minima of typical “anechoic” and “reverberant” audio speech signals, in order to model the alterations due to room acoustics. The above results are obtained from both in-situ room response measurements and controlled acoustical response simulations.

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Organisations: Department of Electrical Engineering, Acoustic Technology, University of Patras
Contributors: Georganti, E., Mourjopoulous, J., Jacobsen, F.
An efficient realization of frequency dependent boundary conditions in an acoustic finite-difference time-domain model.

The finite-difference time-domain (FDTD) method provides a simple and accurate way of solving initial boundary value problems. However, most acoustic problems involve frequency dependent boundary conditions, and it is not easy to include such boundary conditions in an FDTD model. Although solutions to this problem exist, most of them have high computational costs, and stability cannot always be ensured. In this work, a solution is proposed based on "mixing modelling strategies"; this involves separating the FDTD mesh and the boundary conditions (a digital filter representation of the impedance) and combining them into a global solution. This solution is based on an interaction model that involves wave digital filters. The proposed method is validated with several test cases.

An iterative method for determining the surface impedance of acoustic materials in situ

General information
A note on determination of the diffuse-field sensitivity of microphones using the reciprocity technique

The diffuse-field response of a microphone is usually obtained by adding a random-incidence correction to the pressure response of the microphone. However, the random-incidence correction is determined from a relative measurement, and its accuracy depends not only on the relative response at all angles of incidence but also on the accuracy of the frequency response at normal incidence. By contrast, this paper is concerned with determining the absolute diffuse-field response of a microphone using the reciprocity technique. To examine this possibility, a reciprocity calibration setup is used for measuring the electrical transfer impedance between a pair of microphones placed in a miniature (2 m$^3$) reverberation room. The transfer function between the microphones is measured using fast Fourier transform analysis and pseudorandom noise. The calculation of the diffuse-field sensitivity involves (a) separation of the reverberant response from the total response, (b) determination of the reverberation time, and (c) averaging over space and frequency. The resulting diffuse-field correction is compared with an estimate of the random-incidence correction determined in an anechoic room and with a numerical prediction.
**Instrument de mesure en acoustic**
The function of measuring microphones and sound intensity measuring devices is presented for scientific personnel.

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Contributors: Tarnow, V., Jacobsen, F.
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**Instruments de mesure en acoustique**
La mesure en acoustique fait appel à des instruments qui mesurent, d'une part la pression, d'autre part l'intensité du signal. Ce dossier présentera, notamment, deux appareils spécialisés : les sonomètres et les microphones. Il abordera l'ensemble des informations majeures à connaître sur les microphones telles que leur étalonnage et les diverses méthodes de mesure d'intensité acoustique auxquelles ils participent.

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**Intensity techniques**
The Handbook of Signal Processing in Acoustics will compile the techniques and applications of signal processing as they are used in the many varied areas of Acoustics. The Handbook will emphasize the interdisciplinary nature of signal processing in acoustics. Each Section of the Handbook will present topics on signal processing which are important in a specific area of acoustics. These will be of interest to specialists in these areas because they will be presented from their technical perspective, rather than a generic engineering approach to signal processing. Non-specialists, or specialists from different areas, will find the self-contained chapters accessible and will be interested in the similarities and differences between the approaches and techniques used in different areas of acoustics.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F.
Many acoustic measurements rely on determining the total sound energy in an enclosure; and this quantity is usually estimated by measuring the mean square pressure at a number of discrete positions. Almost 30 years ago it was shown theoretically that the normalised spatial variance of the total sound energy density (potential and kinetic) is one third of the normalised spatial variance of the potential energy density (the mean square pressure) in a reverberant sound field above the Schroeder frequency. About ten years later this prediction was confirmed experimentally. However, until recently measurement of the total sound energy density (in air) has required an elaborate arrangement based on finite difference approximations using at least four matched pressure microphones; therefore the method has never come into use. However, with the advent of a three-dimensional particle velocity transducer it has become somewhat easier to measure total rather than only potential energy density in a sound field. This paper examines the spatial uniformity of potential, kinetic and total sound energy density in enclosures theoretically and experimentally with particular emphasis on the frequency range below the Schroeder frequency.
Near field acoustic holography with microphones mounted on a rigid sphere

Spherical near field acoustic holography (spherical NAH) is a technique that makes it possible to reconstruct the sound field inside and just outside an acoustically transparent spherical surface on which the sound pressure is measured with an array of microphones with negligible scattering. This is potentially very useful for source identification. On the other hand, a rigid sphere is somewhat more practical than an open sphere, and it is possible to modify the existing spherical NAH theory so that a similar sound field reconstruction can be made with an array of microphones flush-mounted on a rigid sphere. Rigid spheres with flush-mounted microphones are also used for beamforming, and it is known that they are advantageous compared with open spheres for this application. However, whereas beamforming is a far field technique, NAH is a near field technique, and spherical NAH based on a rigid sphere is only valid if it can be assumed that the sphere has a negligible influence on the incident sound field, and this is not necessarily a good assumption when the sphere is very close to a radiating surface. This paper describes the modified spherical NAH theory and examines the matter through simulations and experiments.

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Contributors: Jacobsen, F., Moreno, G., Fernandez Grande, E., Hald, J.
Publication date: 2008

Near field acoustic holography with microphones mounted on a rigid sphere

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On determination of microphone response and other parameters by a hybrid experimental and numerical method

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Contributors: Barrera Figueroa, S., Jacobsen, F., Rasmussen, K.
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Publication date: 2008
On determination of microphone response and other parameters by a hybrid experimental and numerical method

Typically, numerical calculations of the pressure, free-field and random-incidence response of a condenser microphone are carried out on the basis of an assumed displacement distribution of the diaphragm of the microphone; the conventional assumption is that the displacement follows a Bessel function. This assumption is probably valid at frequencies below the resonance frequency. However, at higher frequencies the movement of the membrane is heavily coupled with the damping of the air film between membrane and back plate, and with resonances in the back chamber of the microphone. A solution to this problem is to measure the velocity distribution of the membrane by means of a non-contact method, such as laser vibrometry. The measured velocity distributions can be used together with a numerical formulation such as the Boundary Element Method for estimating the microphone response and other parameters such as the acoustic centres. In this work, a hybrid method is presented. The velocity distributions of condenser Laboratory Standard microphones were measured using a laser vibrometer. This measured velocity distribution was used for estimating the microphone responses and parameters. The agreement with experimental data is good. This method can be used as an alternative for validating the parameters of the microphones determined by classical calibration techniques.

On the relation between the radiation impedance and the diffuse-field response of measurement microphones

Bibliographical note
Copyright (2008) Acoustical Society of America. This article may be downloaded for personal use only. Any other use requires prior permission of the author and the Acoustical Society of America.
Parameter survey of a rib stiffened wooden floor using sinus modes model

General information
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Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Sjökvist, L., Brunskog, J., Jacobsen, F.
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Publisher: Société Française d'Acoustique
Source: orbit
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Research output: Chapter in Book/Report/Conference proceeding – Annual report year: 2008

Parameter survey of a rib stiffened wooden floor using sinus modes model: Abstract of paper
In buildings built with new techniques there exists a need for better understanding of their acoustical performance. The development of large wooden houses slows down by the uncertainty and costly testing that have to be performed many times before gaining good results. A greater understanding of the sound insulation for lightweight buildings have the possibility to speed up the development of new techniques and in the end give tenants better quality of life. This study uses Fourier sinus series to calculate the vibrations on a rib stiffened plate. The beams are modelled as line forces and moments that reacts onto the plate vibrations. A parameter study is performed with the aim to identify the most important parameters and their behaviour. The preliminary results show that the attenuation of the system is by far most evident in the direction across the beams. The influence from the basic input parameters on the attenuation is then studied. And it is preliminary shown that the placement of the excitation force within a bay actually is irrelevant for the calculated attenuation.

Power-output regularization in global sound equalization
The purpose of equalization in room acoustics is to compensate for the undesired modification that an enclosure introduces to signals such as audio or speech. In this work, equalization in a large part of the volume of a room is addressed. The multiple point method is employed with an acoustic power-output penalty term instead of the traditional
quadratic source effort penalty term. Simulation results demonstrate that this technique gives a smoother decline of the reproduction performance away from the control points.

**General information**
Publication status: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Stefanakis, N., Sarris, J., Cambourakis, G., Jacobsen, F.
Pages: 33-36
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Web of Science (2008): Indexed yes
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Source: orbit
Source ID: 208788
Research output: Contribution to journal › Journal article – Annual report year: 2008 › Research › peer-review

**Sound propagation in forests: A comparison of experimental results and values predicted by the Nord 2000 model.**
The purpose of the work described in this paper is twofold: (i) to present the results of an experimental investigation of the sound attenuation in different types of forest, and (ii) to validate a part of the Nord 2000 model. A number of measurements have been carried out in regular and irregular forests with trees with deciduous and evergreen leaves, different tree density, different trunk diameter, etc. The experimental results indicate that trees have a noticeable effect on sound propagation at medium and high frequencies at distances longer than 40m. The Nord 2000 model uses a simple algorithm to predict scattering effects when sound propagates in outdoor spaces with obstacles. The comparison of experimental results and predictions shows that the Nord 2000 model predicts the ground effect dip in forests with acceptable accuracy in about 60% of the cases if the flow resistivity of the ground is allowed to vary with distance. The high frequency effect of trees is in general not predicted very well.

**General information**
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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Tarrero, A., Martín, M., González, J., Machimbarrena, M., Jacobsen, F.
Pages: 662-671
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Peer-reviewed: Yes

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Scopus rating (2008): SJR 0.665 SNIP 1.796
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Original language: English
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10.1016/j.apacoust.2007.01.007
Spherical near field acoustic holography with microphones on a rigid sphere

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering, Acoustic Technology, Brüel and Kjær Sound and Vibration Measurement A/S
Contributors: Jacobsen, F., Hald, J., Fernandez Grande, E., Moreno, G.
Pages: 2869-2873
Publication date: 2008

Host publication information
Title of host publication: Proceedings of the European Conference on Noise Control
Source: orbit
Source ID: 221797

Spherical near field acoustic holography with microphones on a rigid sphere: Abstract of paper
Spherical near field acoustic holography (SNAH) is a recently developed technique that makes it possible to reconstruct the sound field inside and just outside an acoustically transparent spherical surface on which the sound pressure is measured with an array of microphones with negligible scattering. Because of the versatile geometry of a sphere SNAH is potentially extremely useful for source identification. On the other hand a rigid sphere is somewhat more practical than an open sphere, and it is possible to modify the SNAH theory so that a similar sound field reconstruction can be made with an array of microphones flush-mounted on a rigid sphere. However, this approach is only valid if it can be assumed that the sphere has a negligible influence on the incident sound field, in other words if multiple scattering can be ignored, and this is not necessarily a good assumption when the sphere is close to a radiating surface. This paper describes the modified SNAH theory and examines the matter through simulations and experimentally.

General information
Publication status: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, Brüel and Kjær Sound and Vibration Measurement A/S
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Peer-reviewed: Yes

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BFI (2008): BFI-level 2
Scopus rating (2008): SJR 0.83 SNIP 1.636
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Research output: Contribution to journal › Conference abstract in journal – Annual report year: 2008 – Research – peer-review
The incident sound power in a diffuse sound field

General information
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Organisations: Acoustic Technology, Department of Electrical Engineering, Oticon A/S
Contributors: Jacobsen, F., Chen, X.
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Publication date: 2008

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Title of host publication: Proceedings of Fifteenth International Congress on Sound and Vibration
Publisher: International Institute of Acoustics and Vibration
Source: orbit
Source ID: 221801

The Microflown particle velocity sensor
The Handbook of Signal Processing in Acoustics will compile the techniques and applications of signal processing as they are used in the many varied areas of Acoustics. The Handbook will emphasize the interdisciplinary nature of signal processing in acoustics. Each Section of the Handbook will present topics on signal processing which are important in a specific area of acoustics. These will be of interest to specialists in these areas because they will be presented from their technical perspective, rather than a generic engineering approach to signal processing. Non-specialists, or specialists from different areas, will find the self-contained chapters accessible and will be interested in the similarities and differences between the approaches and techniques used in different areas of acoustics.

General information
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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., de Bree, H.
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Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jónsson, G., Jacobsen, F.
Number of pages: 6
Publication date: 2007

Host publication information
Title of host publication: Proceedings of 19th International Congress on Acoustics
Source: orbit
Source ID: 202567
Research output: Chapter in Book/Report/Conference proceeding – Article in proceedings – Annual report year: 2007 – Research
An equivalent roughness model for seabed backscattering at very high frequencies using a band-matrix approach
This work concerns modeling of very high frequency (>100 kHz) sonar images obtained from a sandy seabed. The seabed is divided into a discrete number of 1D height profiles. For each height profile the backscattered pressure is computed by an integral equation method for interface scattering between two homogeneous media as formulated by Chan (IEEE Trans. Antennas Propag. 46, 142-149 (1998)). However, the seabed is inhomogeneous, and volume scattering is a major contributor to backscattering. The SAX99 experiments revealed that the density in the unconsolidated sediment within the first 5 mm exhibits a high spatial variation. For that reason, additional roughness is introduced: For each surface point a stochastic realization of the density along the vertical is generated, and the sediment depth at which the density has its maximum value will constitute the new height field value. The matrix of the full integral equation is reduced to a band matrix as the interaction between the point sources on the seabed is neglected from a certain range; this allows computations on long height profiles with lengths up to approximately 25 m (at 300 kHz). The equivalent roughness approach, combined with the band-matrix approach, agrees with SAX99 data at 300 kHz.

Publications & citations
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Research output:
Contribution to journal › Journal article – Annual report year: 2007 › Research › peer-review

An investigation of microphone calibration in a diffuse sound field

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Barrera Figueroa, S., Rasmussen, K., Jacobsen, F.
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Host publication information
Title of host publication: Proceedings of Inter-Noise 2007
Source: orbit
Source ID: 202564
Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2007 › Research

A note on the physical interpretation of frequency dependent boundary conditions in a digital waveguide mesh
Digital waveguide mesh (DWM) is a popular method for time domain modelling of sound fields. DWM consists of a recursive digital filter structure where a D'Alembert solution of the wave equation is propagated. One of the attractive characteristics of this method is related to the simplicity of incorporating frequency dependent boundary conditions. The impedance of the boundaries can be simulated by means of digital filtering. So far such digital filters have been designed to provide reflection factors corresponding to the impedance of the boundaries for normal sound incidence. However, the...
resulting model of the boundary does not agree with the behaviour of a locally reacting surface, and this can give rise to contradictions in the physical interpretation of the reflected sound field. This paper analyses the behaviour of frequency dependent boundary conditions in DWM in order to obtain a physical interpretation of the simulated impedance surfaces. The interpretation is validated by several examples.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Escolano-Carrasco, J., Jacobsen, F.
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Issue number: 3
ISSN (Print): 1610-1928
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Web of Science (2007): Indexed yes
Original language: English
Source: orbit
Source ID: 199166
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A numerical study of fluid flow past a circular cylinder at RE = 3900 and a practical approach to noise prediction.

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Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Pradera, A., Keith, G., Jacobsen, F., Gil-Negrete, N., Rivas, A.
Number of pages: 8
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Source: orbit
Source ID: 202537
Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2007 › Research › peer-review

A survey on simultaneous measurement of the free-field and diffuse-field sensitivity of microphones

**General information**
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Barrera Figueroa, S., Rasmussen, K., Jacobsen, F.
Number of pages: 6
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**Host publication information**
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Source ID: 202566
Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2007 › Research

In-situ measurements of the complex acoustic impedance of porous materials

**General information**
Publication status: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Alvarez, J., Jacobsen, F.
On experimental determination of the free-field correction of laboratory standard microphones at normal incidence

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Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Barrera Figueroa, S., Rasmussen, K., Jacobsen, F.
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Source ID: 194022
Research output: Contribution to journal › Journal article – Annual report year: 2007 › Research › peer-review

On experimental determination of the random-incidence response of microphones

The random-incidence sensitivity of a microphone is defined as the ratio of the output voltage to the sound pressure that would exist at the position of the acoustic center of the microphone in the absence of the microphone in a sound field with incident plane waves coming from all directions. The random-incidence correction of a number of laboratory standard microphones has been determined experimentally. Although the measurement procedure seems to be straightforward, some practical and fundamental problems arise: i Reflections from the mounting rig contaminate the measured frequency response, and whereas some of these reflections can be removed using a time-selective technique, others coincide with the direct impulse response and consequently cannot be removed in the time domain and thus affect the accuracy of the estimate; ii the accuracy of the estimate is relying on the rotational symmetry of the microphone and depends on the angular resolution. The effect of the angular resolution has been compared with the analytical solution of the scattering and diffraction around a solid sphere. Numerical calculations supplement the experimental results. Although the procedure has only been applied to laboratory standard microphones, it is not restricted to such microphones and may be applied to other types of measurement microphones.

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Contributors: Barrera Figueroa, S., Rasmussen, K., Jacobsen, F.
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Publication date: 2007
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Journal: Journal of the Acoustical Society of America
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Ratings:
On the uncertainty in measurement of sound power using sound intensity

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F.
Pages: 20-28
Publication date: 2007
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Original language: English
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Sound intensity

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F.
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Publisher: Springer Verlag
Edition: 1st
ISBN (Print): 0-387-30446-0
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Sound intensity measurements: chapter 45

General information
Publication status: Published
Statistically optimised near field acoustic holography and the Helmholtz equation least squares method: a comparison

General information
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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Gomes, J., Jacobsen, F., Bach-Andersen, M.
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Source ID: 210357

Statistically optimised near field acoustic holography with pressure-velocity probes and with a double-layer array

General information
Publication status: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Jacobsen, F., Chen, X., Jaud, V.
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Source: orbit
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Research output: Chapter in Book/Report/Conference proceeding – Article in proceedings – Annual report year: 2007 – Research

Statistically optimized near field acoustic holography using an array of pressure-velocity probes

Statistically optimized near field acoustic holography (SONAH) differs from conventional near field acoustic holography (NAH) by avoiding spatial Fourier transforms; the processing is done directly in the spatial domain. The main advantage of SONAH compared with NAH is that the usual requirement of a measurement aperture that extends well beyond the source can be relaxed. Both NAH and SONAH are based on the assumption that all sources are on one side of the measurement plane whereas the other side is source free. An extension of the SONAH procedure based on measurement with a double layer array of pressure microphones has been suggested. The double layer technique makes it possible to distinguish between sources on the two sides of the array and thus suppress the influence of extraneous noise coming from the “wrong” side. It has also recently been demonstrated that there are significant advantages in NAH based on an array of acoustic particle velocity transducers (in a single layer) compared with NAH based on an array of pressure microphones. This investigation combines the two ideas and examines SONAH based on an array of pressure-velocity intensity probes through computer simulations as well as experimentally.

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Contributors: Jacobsen, F., Jaud, V.
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The acoustic center of laboratory standard microphones (vol 120, pg 2668, 2006)

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Towards estimating the uncertainty of intensity-based sound power measurements

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Contributors: Jacobsen, F., Ambrosini, M.
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Research

Weighted statistically optimised near field acoustic holography with pressure-velocity probes

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., Chen, X., Jaud, V.
Number of pages: 8
Publication date: 2007

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Source: orbit
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Research output: Chapter in Book/Report/Conference proceeding → Article in proceedings – Annual report year: 2007
Research → peer-review

A note on the calibration of pressure-velocity sound intensity probes
A pressure-velocity sound intensity probe is a device that combines a pressure microphone with a particle velocity transducer. Various methods of calibrating such sound intensity probes are examined: a far field method that requires an anechoic room, a near field method that involves sound emitted from a small hole in a plane baffle, a near field method where the sound is emitted from a hole in a spherical baffle, and a method that involves an impedance tube. The performance of the two near field methods is examined both in an anechoic room and in various ordinary rooms. It is shown that whereas reflections from the edges from a plane baffle disturb the calibration, the method based on a spherical baffle gives acceptable results in a wide frequency range even when the calibration is carried out in a small office, provided that the distance between the hole and the device under test is about 5 cm.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., Jaud, V.
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Peer-reviewed: Yes

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Web of Science (2006): Indexed yes
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URLs:
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Source: orbit
Source ID: 190599
Research output: Contribution to journal → Journal article – Annual report year: 2006 → Research → peer-review
A numerical investigation of the influence of windscreens on measurement of sound intensity

Sound intensity probes are often used with windscreens to minimize the effect of noise caused by airflow. A theoretical and experimental study of the effect of windscreens on p-p intensity probes published ten years ago concluded that windscreens give rise to underestimation of the sound intensity at low frequencies in strongly reactive sound fields. The theoretical part of this study was based on the assumption of a windscreen of infinite extent. In this paper windscreens of realistic size and shape are dealt with by means of a coupled boundary element model for the windscreen and the surrounding air. The error of the estimated intensity caused by the windscreen is calculated under a number of sound field conditions of varying reactivity. It is shown that the resulting error can be much larger than the intensity itself in a very reactive sound field. It is also shown that the shape and size of the windscreen has a significant influence on the error.

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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Juhl, P., Jacobsen, F.
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Publication Information
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A numerically accurate and robust expression for bistatic scattering from a plane triangular facet

This work is related to modeling of synthetic sonar images of naval mines or other objects. Considered here is the computation of high frequency scattering from the surface of a rigid 3D-object numerically represented by plane triangular facets. The far field scattered pressure from each facet is found by application of the Kirchhoff approximation. Fawcett [J. Acoust. Soc. Am. 109, 1319–1320 (2001)] derived a time domain expression for the backscattered pressure from a triangular facet, but the expression encountered numerical problems at certain angles, and therefore, the effective ensonified area was applied instead. The effective ensonified area solution is exact at normal incidence, but at other angles, where singularities also exist, the scattered pressure will be incorrect. This paper presents a frequency domain expression generalized to bistatic scattering written in terms of sinc functions; it is shown that the expression improves the computational accuracy without loss of robustness.

General information
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Organisations: Electronics & Signal Processing, Department of Electrical Engineering, Acoustic Technology
Contributors: Wendelboe, G., Jacobsen, F., Bell, J.
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A numerical study of the random-incidence and diffuse-field sensitivity of laboratory standard microphones using BEM

The difference between the random-incidence sensitivity of a microphone and the diffuse-field sensitivity is that according to the definition of the former, plane waves coming from different angles of incidence impinge successively onto the microphone under free-field conditions, whereas according to the definition of the latter, a number of plane waves coming from random directions and having random phases impinge simultaneously upon the microphone. The random-incidence sensitivity can be estimated using measurements made in an anechoic chamber, while the diffuse-field sensitivity requires a reverberation room. It is widely accepted that the two definitions are equivalent. The purpose of this paper is to examine this equivalence using numerical simulations. A laboratory standard microphone can be considered rotationally symmetrical around the axis, thus, an axi-symmetric formulation of the Boundary Element Method has been used to study different aspects of the two realizations, such as the influence of the number of plane waves impinging on the microphone, the required amount of spatial averaging, etc.

A physical interpretation of frequency dependent boundary conditions in a digital waveguide mesh

A physical interpretation of frequency dependent boundary conditions in a digital waveguide mesh

Calibration of p-u intensity probes
Near field acoustic holography with double layer array processing

General information
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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Havránek, Z., Jacobsen, F.
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Statistically optimised near field acoustic holography based on particle velocity measurements

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Contributors: Jacobsen, F., Jaud, V.
Publication date: 2006

Statistically optimised near field acoustic holography with an array of pressure-velocity intensity probes

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., Jaud, V.
Publication date: 2006
The acoustic center of laboratory standard microphones
An experimental procedure is described for obtaining the effective acoustic distance between pairs of microphones coupled by a free field, leading to the determination of the position of the acoustic center of the microphones. The procedure, which is based on measuring the modulus of the electrical transfer impedance, has been applied to a large number of microphones. In all cases effects due to reflections from the walls of the anechoic chamber and the interference between the microphones have been removed using a time-selective technique. The procedure of determining the position of the acoustic center from the inverse distance law is analyzed. Experimental values of the acoustic center of laboratory standard microphones are presented, and numerical results obtained using the boundary element method supplement the experimental data. Estimated uncertainties are also presented. The results reported confirm values previously defined in an international standard and extend the frequency range.

The random incidence sensitivity of measurement microphones

The use of microperforated plates to attenuate cavity resonances
The use of microperforated plates to introduce damping in a closed cavity is examined. By placing a microperforated plate well inside the cavity instead of near a wall as traditionally done in room acoustics, high attenuation can be obtained for specific acoustic modes, compared with the lower attenuation that can be obtained in a broad frequency range with the conventional position of the plate. An analytical method for predicting the attenuation is presented. The method involves finding complex eigenvalues and eigenfunctions for the modified cavity and makes it possible to predict Green's functions. The results, which are validated experimentally, show that a microperforated plate can provide substantial attenuation of
modes in a cavity. One possible application of these findings is the treatment of boiler tones in heat-exchanger cavities.

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Source: orbit
Source ID: 191899
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A comparison of two different sound intensity measurement principles
The dominating method of measuring sound intensity in air is based on the combination of two pressure microphones. However, a sound intensity probe that combines an acoustic particle velocity transducer with a pressure microphone has recently become available. This paper examines, discusses, and compares the two measurement principles with particular regard to the sources of error in sound power determination. It is shown that the phase calibration of intensity probes that combine different transducers is very critical below 500 Hz if the measurement surface is very close to the source under test. The problem is reduced if the measurement surface is moved further away from the source. The calibration can be carried out in an anechoic room.

**General information**
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Jacobsen, F., de Bree, H.
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Peer-reviewed: Yes

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Electronic versions:
Jacobsen.pdf
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A note on the uncertainty in measurement of sound power using sound intensity

**General information**
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Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F.
Publication date: 2005

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Development of micro-perforated acoustic elements for vehicle applications

**General information**
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Ducret, F., Jacobsen, F., Åbom, M.
Publication date: 2005

**Host publication information**
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Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2005 › Research › peer-review

Intensity-based sound power determination under adverse sound field conditions: p-p probes versus p-u probes

**General information**
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., de Bree, H.
Publication date: 2005

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Localization of seabed domains ensonified by object-reflected sound at very high frequencies

**General information**
Publication status: Published
Organisations: Electronics & Signal Processing, Department of Electrical Engineering, Acoustic Technology
Measurement of sound intensity: p-u probes versus p-p probes

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., de Bree, H.
Publication date: 2005

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Source: orbit
Source ID: 182182
Near field acoustic holography based on an array of particle velocity sensors

Near field acoustic holography is usually based on measurement of the pressure. This paper describes an investigation of an alternative technique that involves measuring the normal component of the acoustic particle velocity. A simulation study shows that there is no appreciable difference between the quality of predictions of the pressure based on knowledge of the pressure in the measurement plane and predictions of the particle velocity based on knowledge of the particle velocity in the measurement plane. However, when the particle velocity is predicted close to the source on the basis of the pressure measured in a plane further away, high spatial frequency components corresponding to evanescent modes are not only amplified by the distance but also by the wave number ratio (kz/k). By contrast, when the pressure is predicted close to the source on the basis of the particle velocity measured in a plane further away, high spatial frequency components are reduced by the reciprocal wave number ratio (k/kz). For the same reason holography based on the particle velocity is less sensitive to transducer mismatch than the conventional technique based on the pressure. These findings are confirmed by an experimental investigation made with a p-u sound intensity probe produced by Microflown.

Predicting the eigenmodes of a cavity containing an array of circular pipes

An array of pipes inside a cavity, as found, for example, in a shell-and-tube heat exchanger, changes the eigenfrequencies of the cavity. It can be tedious to determine the shifted eigenfrequencies with a finite-element model. Based on previous
work by Meyer and Neumann, Parker proposed a simple relationship for predicting the shifted eigenfrequencies. In this paper, results obtained from this relationship are compared with eigenfrequencies obtained from very accurate finite element simulations. From the results it can be concluded that Parker's relationship gives fairly good predictions of the eigenfrequencies for the first few modes in a cavity with pipes arranged in a rectangular configuration. The predictions are not so accurate for pipes arranged in a diamond configuration, and a modified version of the relationship is suggested for this configuration. If the number of pipes in the cavity is small, the simple relationship is no longer valid. ©2005 Acoustical Society of America.
A note on measurement of sound pressure with intensity probes

The effect of scattering and diffraction on measurement of sound pressure with "two-microphone" sound intensity probes is examined using an axisymmetric boundary element model of the probe. Whereas it has been shown a few years ago that the sound intensity estimated with a two-microphone probe is reliable up to 10 kHz when using 0.5 in. microphones in the usual face-to-face arrangement separated by a 12 mm spacer, the sound pressure measured with the same instrument will typically be underestimated at high frequencies. It is shown in this paper that the estimate of the sound pressure can be improved under a variety of realistic sound field conditions by applying a different weighting of the two pressure signals from the probe. The improved intensity probe can measure the sound pressure more accurately at high frequencies than an ordinary sound intensity probe or an ordinary sound level meter. ©2004 Acoustical Society of America.

A note on the concept of acoustic center

The acoustic center of a reciprocal transducer is defined as the point from which spherical waves seem to be diverging when the transducer is acting as a source. This paper examines various ways of determining the acoustic center of a source, including methods based on deviations from the inverse distance law and methods based on the phase response. The considerations are illustrated by experimental results for condenser microphones.
A numerical investigation of the influence of windscreens on sound intensity measurements

Interferencia de ondas acústicas en la calibración primaria de micrófonos por reciprocidad en campo libre

Interferencia de ondas acústicas en la calibración primaria de micrófonos por reciprocidad en campo libre
On the interference between the two microphones in free-field reciprocity calibration

One of the fundamental assumptions in free-field reciprocity calibration of microphones is that the microphones can be substituted by point sources at the positions where the acoustic centers are located. However, in practice the microphones have finite dimensions and, at the distance and in the frequency range where the measurements are made, the direct wave and the subsequent reflections from the microphones interfere with each other, creating a "standing wave." This interference effect gives rise to deviations from the inverse distance law, indicating that the free-field assumption is not strictly valid. The interference has been thought to be caused by specular reflection between the parallel diaphragms of the microphones, and a solution based on tilting the axis of one of the microphones a few degrees has been proposed, but never examined in practice. In this paper a time-selective technique is applied for analyzing the interference and for removing it in the time domain. It is shown that the phenomenon is due to multiple backscattering rather than specular reflection. Thus tilting one of the microphones does not alleviate the problem, as also demonstrated experimentally. However, the time-selective technique is quite effective in removing the interference effect and other disturbances from the direct wave between the microphones. ©2004 Acoustical Society of America.

Perception of modal distribution in critical listening spaces

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Contributors: Fazenda, B., Avis, M. R., Davies, W. J., Jacobsen, F.
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Peer-reviewed: Yes

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Title of host publication: Proceedings of Eleventh International Congress on Sound and Vibration, St. Petersburg, Russia, 2004
Publisher: International Institute of Acoustics and Vibration
URLs:
Source: orbit
Source ID: 60931
Sound equalization in a large region of a rectangular enclosure
The work presented by Santillán [J. Acoust. Soc. Am. 110, 1989–1997 (2001)] about equalization at low frequencies in rectangular enclosures is extended, and topics that remained unaddressed in the original study are treated in this paper. A modification is introduced to the original cost function that leads to a least-squares problem formulation that demonstrates increased robustness, and the multiple-error LMS (Least Mean Square) adaptive algorithm is employed to approximate the coefficients of the equalization filters to their optimum values. Other issues studied in this paper include the dependence of the limits of the zone of equalization on factors such as the damping constant of the modes of the enclosure, the number of sources, and the driving frequency. ©2004 Acoustical Society of America.

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Sarris, J. C., Jacobsen, F., Cambourakis, G. E.
Pages: 3271-3274
Publication date: 2004
Peer-reviewed: Yes

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Research output: Contribution to journal › Journal article – Annual report year: 2004 › Research › peer-review

Sound power measurements using sound intensity at high frequencies

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Jacobsen, F., Juhl, P.
Publication date: 2004

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Title of host publication: Proceedings of Inter-Noise 2004, Prague, Czech Republic
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Source: orbit
Source ID: 60965
Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2004 › Research › peer-review

Sound pressure measurements with sound intensity probes

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Juhl, P., Jacobsen, F.
Pages: 2209-2212
Publication date: 2004
A BEM approach to validate a model for predicting sound propagation over non-flat terrain

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Quirósy Alpera, S., Jacobsen, F., Juhl, P., Henríquez, V. C.
Pages: 781-791
Publication date: 2003
Peer-reviewed: Yes

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Journal: Applied Acoustics
Volume: 64
Issue number: 8
ISSN (Print): 0003-682X
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Web of Science (2003): Indexed yes
Original language: English
Keywords: boundary element method, Sound propagation, non-flat terrain
URLs:
Source: orbit
Source ID: 60709
Research output: Contribution to journal › Journal article – Annual report year: 2003 › Research › peer-review

Analysis and modelling of room responses by time-frequency methods

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Sarris, J., Cambourakis, G., Jacobsen, F.
Pages: 4151-4158
Publication date: 2003

Host publication information
Title of host publication: Proceedings of Tenth International Congress on Sound and Vibration, Stockholm, Sweden
Source: orbit
Source ID: 60789
Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2003 › Research › peer-review

Analysis of nonlinear behavior of loudspeakers using the instantaneous frequency: Abstracts of papers
It is well known that the weakest link in a sound reproduction chain is the loudspeaker. The most significant effect on the sound quality is nonlinear distortion of loudspeakers. Many methods are applied to analyze the nonlinear distortion of loudspeakers. Almost all of the methods are based on the Fourier transform. In this work, a new method using the instantaneous frequency is introduced for describing and characterizing loudspeaker nonlinearities. First, numerical integration is applied to simulate the nonlinearities of loudspeakers caused by two nonlinear parameters, force factor and stiffness, separately. Then, a practical loudspeaker is used in an experiment and its nonlinear characteristics are analyzed with the instantaneous frequency. The results provide a clear physical interpretation of the nonlinearities of loudspeakers and will be useful for understanding the nonlinear behavior of loudspeakers. It will also be helpful for compensating for the nonlinearities and for improving the quality of sound reproduction. [Work supported by Sino-Danish International Co-operative Project, No. AM13: 66 and DANIDA (Denmark).]
An approximate method of modelling scattering by composite bodies

A note on the acoustic centre of condenser microphones

A note on the acoustic centre of condenser microphones

Host publication information
Title of host publication: Proceedings of Tenth International Congress on Sound and Vibration, Stockholm, Sweden, 7-10 July
URLs:
A time-selective technique for free-field reciprocity calibration of condenser microphones

In normal practice, microphones are calibrated in a closed coupler where the sound pressure is uniformly distributed over the diaphragm. Alternatively, microphones can be placed in a free field, although in that case the distribution of sound pressure over the diaphragm will change as a result of the diffraction of the body of the microphone, and thus, its sensitivity will change. In the two cases, a technique based on the reciprocity theorem can be applied for obtaining the absolute sensitivity either under uniform pressure or free-field conditions. In this paper, signal-processing techniques are considered that improve the accuracy of the free-field calibration method. In particular, a fast Fourier transform (FFT)-based time-selective technique for removing undesired reflections from the walls of the measurement chamber has been developed and applied to the electric transfer impedance function between two microphones. The acoustic centers of the microphones have been determined from the "cleaned" transfer impedance values. Then, the complex free-field sensitivities of the microphones have been calculated. The resulting complex sensitivities and acoustic centers have proved to be in good agreement with previously published data, and this confirms the reliability of the time-selective technique, even in nonanechoic environments. Furthermore, the obtained results give a new reference for further comparisons, because they cover a frequency range with an accuracy that has not been obtained by previously published data.

Measurement of the residual pressure-intensity index with extreme precision

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Optimization of low frequency sound fields in small enclosures

General information
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Organisations: Department of Electrical Engineering
Contributors: Conway, C., Jacobsen, F., Corcoran, D.
Pages: 411-418
Publication date: 2003

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URLs:
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Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2003 › Research › peer-review

Sound intensity and its measurement and applications

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Jacobsen, F.
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Publication date: 2003
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Journal: Current Topics in Acoustical Research
Volume: 3
Original language: English
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Source ID: 60759
Research output: Contribution to journal › Journal article – Annual report year: 2003 › Research › peer-review

Active control of loudspeakers: An investigation of practical applications

General information
Publication status: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, Department of Acoustic Technology
Contributors: Bright, A. P., Jacobsen, F., Polack, J., Rasmussen, K. B.
Publication date: Nov 2002

Publication information
Original language: English
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60678_Orsted2002_Andrew Bright-Active control of loudspeakers-An investigation of practical applications.pdf
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Source ID: 60678

An experimental study of free-field reciprocity calibration of condenser microphones

General information
A note on finite difference estimation of acoustic particle velocity

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Jacobsen, F.
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Publication date: 2002
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Original language: English
DOI:
10.1006/jsvi.2002.5023
Source: orbit
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Research output: Contribution to journal › Journal article – Annual report year: 2002 › Research › peer-review

On the non-uniqueness problem in a 2D halfspace BEM formulation

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Juhl, P., Jacobsen, F., Henriquez, V. C., Alpera, S. Q.
Publication date: 2002

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Title of host publication: Proceedings of Ninth International Congress on Sound and Vibration, Orlando, Fl., USA
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Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 2002 › Research › peer-review

Measurement of sound absorption using sound intensity

General information
Publication status: Published
Organisations: Department of Electrical Engineering
Contributors: Jacobsen, F., Zamboni, L.
Pages: 2415-2422
Publication date: 2001
On the modeling of narrow gaps using the standard BEM

Numerical methods based on the Helmholtz integral equation are well suited for solving acoustic scattering and diffraction problems at relatively low frequencies. However, it is well known that the standard method becomes degenerate if the objects that disturb the sound field are very thin. This paper makes use of a standard axisymmetric Helmholtz integral equation formulation and its boundary element method (BEM) implementation to study the behavior of the method on two test cases: a thin rigid disk of variable thickness and two rigid cylinders separated by a gap of variable width. Both problems give rise to the same kind of degeneracy in the method, and modified formulations have been proposed to overcome this difficulty. However, such techniques are better suited for the so-called thin-body problem than for the reciprocal narrow-gap problem, and only the first is usually dealt with in the literature. A simple integration technique that can extend the range of thicknesses/widths tractable by the otherwise unmodified standard formulation is presented and tested. This technique is valid for both cases. The modeling of acoustic transducers like sound intensity probes and condenser microphones has motivated this work, although the proposed technique has a wider range of applications.
Validation of an Efficient Outdoor Sound Propagation Model Using BEM

An approximate, simple and practical model for prediction of outdoor sound propagation exists based on ray theory, diffraction theory and Fresnel-zone considerations [1]. This model, which can predict sound propagation over non-flat terrain, has been validated for combinations of flat ground, hills and barriers, but it still needs to be validated for configurations that involve combinations of valleys and barriers. In order to do this a boundary element model has been implemented in MATLAB to serve as a reliable reference.

An elementary introduction to applied signal analysis

Prediction of random errors in sound intensity measurement
The coherence of reverberant sound fields

A new method of measuring spatial correlation functions in reverberation rooms is presented. It is shown that the coherence functions determined with appropriate spectral resolution contain the same information as the corresponding correlation functions, and that measuring such coherence functions is a far more efficient way of obtaining this information.

The technique is then used to verify theoretical predictions of the spatial correlation between various components of the particle velocity in a diffuse sound field. Other possible applications of the technique are discussed and illustrated with experimental results obtained in an ordinary room.
An MLS coherence function and its performance in measurements on time-varying systems

General information
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Organisations: Department of Acoustic Technology
Contributors: Liu, J., Jacobsen, F.
Pages: 2851-2858
Publication date: 1999

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Place of publication: Lyngby
Publisher: Department of Acoustic Technology, Technical University of Denmark
Source: orbit
Source ID: 172279
Research output: Chapter in Book/Report/Conference proceeding – Article in proceedings – Annual report year: 1999

Attenuation and damping of structure-borne sound: Applied signal analysis

General information
Publication status: Published
Organisations: Department of Acoustic Technology
Contributors: Jacobsen, F.
Number of pages: 4
Publication date: 1999

Publication information
Original language: English
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Is there a systematic disagreement between intensity-based and pressure-based sound transmission loss measurements?

General information
Publication status: Published
Organisations: Department of Acoustic Technology
Contributors: Machimbarrena, M., Jacobsen, F.
Pages: 101-111
A numerical and experimental investigation of the performance of sound intensity probes at high frequencies

The high-frequency performance of a p-p intensity probe with a solid spacer between the two microphones is examined. It is shown theoretically and verified experimentally that with a spacer length that equals the diameter of the microphones, the finite difference error is almost perfectly cancelled by a cavity resonance. It is concluded that the usable frequency range of sound intensity probes can be doubled with this configuration.

General information
Publication status: Published
Organisations: Department of Acoustic Technology
Contributors: Jacobsen, F., Cutanda, V., Juhl, P. M.
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Publication date: 1998
Peer-reviewed: Yes

Publication information
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Issue number: 2
Sound intensity

General information
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Organisations: Department of Acoustic Technology, Auburn University
Contributors: Crocker, M. J., Jacobsen, F.
Pages: 1327-1340
Publication date: 1998

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Title of host publication: Handbook of Acoustics
Place of publication: New York
Publisher: John Wiley & Sons
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Source ID: 170462
Research output: Chapter in Book/Report/Conference proceeding – Annual report year: 1998

The sound power emitted by a source of low acoustic impedance

General information
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Organisations: Department of Acoustic Technology, Technical University of Denmark
Contributors: Jacobsen, F., Verholt, L. M.
Pages: 1361-1364
Publication date: 1998

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Place of publication: Christchurch
Publisher: New Zealand Acoustical Society
Source: orbit
Source ID: 170463
Research output: Chapter in Book/Report/Conference proceeding – Article in proceedings – Annual report year: 1998

An elementary introduction to applied signal analysis

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Publication status: Published
Organisations: Department of Acoustic Technology
Contributors: Jacobsen, F.
Number of pages: 40
Publication date: 1997

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Original language: English
Source: orbit
Source ID: 167820
A note on the influence of the averaging time and the frequency resolution on the accuracy of acoustic measurements

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Publication status: Published
Organisations: Department of Acoustic Technology
Contributors: Jacobsen, F.
Number of pages: 13
Publication date: 1997

Publication information
Original language: English
Source: orbit
Source ID: 167825
Research output: Book/Report › Book – Annual report year: 1997 › Research › peer-review

An overview of the sources of error in sound power determination using the intensity technique
An overview of the most important sources of error in sound power determination with the sound intensity technique is presented. It is concluded that the method is convenient, accurate and reliable provided that a few simple rules are observed. (C) 1997 Elsevier Science Ltd.

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Organisations: Department of Acoustic Technology
Contributors: Jacobsen, F.
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Peer-reviewed: Yes

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Journal: Applied Acoustics
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Original language: English
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Observations on the systematic deviations between two methods of measuring sound transmission loss

General information
Publication status: Published
Organisations: Department of Acoustic Technology, Beijing Municipal Institute of Labour Protection
Contributors: Jacobsen, F., Ding, H.
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Publication information
Journal: Building Acoustics
Volume: 3
Issue number: 1
Original language: English
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Source ID: 167811
Research output: Contribution to journal › Journal article – Annual report year: 1997 › Research › peer-review

Sound Intensity

General information
Sound intensity and its measurement

A critical examination of some of the field indicators that have been proposed in connection with sound power determination using the intensity method

A numerical investigation of the performance of sound intensity probes at high frequencies
A sound intensity probe for measuring from 50 Hz to 10 kHz

General information
Publication status: Published
Organisations: Department of Acoustic Technology, National Autonomous University of Mexico
Contributors: Jacobsen, F., Cutanda, V., Juhl, P. M.
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Publisher: Institute of Acoustics
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Source ID: 164898
Research output: Chapter in Book/Report/Conference proceeding – Annual report year: 1996 – Research › peer-review

A sound intensity probe for measuring from 50 Hz to 10 kHz

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Organisations: Department of Acoustic Technology, National Autonomous University of Mexico
Contributors: Jacobsen, F., Cutanda, V., Juhl, P. M.
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Peer-reviewed: No

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Observations on the systematic deviations between the results of the conventional method and the sound intensity method of measuring transmission loss

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Contributors: Jacobsen, F., Ding, H.
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Publication date: 1996

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Place of publication: St. Albans
Publisher: Institute of Acoustics
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Research output: Chapter in Book/Report/Conference proceeding – Article in proceedings – Annual report year: 1996 – Research › peer-review
Støj fra mindre vindmøller: Vejledning i dæmpning af støj fra vindmøller: [Noise from smaller wind turbine generators: Guidelines for reducing noise from wind turbine generators]

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Ohlrich, M., Jacobsen, F., Andersen, B.
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Publication date: 1987

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Original language: English
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Source: orbit
Source ID: 240641
Research output: Book/Report › Report – Annual report year: 1987 › Research

Vibrational power transmission from multipoint mounted machinery to supporting structure

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., Ohlrich, M.
Number of pages: 136
Publication date: 1986

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Place of publication: DK-2800 Kgs. Lyngby
Publisher: The Acoustics Laboratory, Technical University of Denmark
Original language: English
(No series; No. Report no 35).
Source: orbit
Source ID: 240637
Research output: Book/Report › Report – Annual report year: 1986 › Research

High-frequency vibration isolation of a steel ring element

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Ohlrich, M., Jacobsen, F.
Pages: 183-186
Publication date: 1983

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Title of host publication: Proceedings of 11th ICA, the International Conference on Acoustics
Source: orbit
Source ID: 240525
Research output: Chapter in Book/Report/Conference proceeding › Article in proceedings – Annual report year: 1983 › Research

Isolation of structural vibration from machinery

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Ohlrich, M., Jacobsen, F.
Pages: 309-312
Publication date: 1982
Power transmission from vibrating machine to complicated structure

General information
Publication status: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Jacobsen, F., Ohlrich, M.
Pages: 281-284
Publication date: 1982