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Chronicle

58th Open Seminar on Acoustics joined with 2nd Polish-German Structured Conference on Acoustics

The 58th Open Seminar on Acoustics and 2nd Polish-German Structured Conference on Acoustics are organized by the Gdańsk Division of the Polish Acoustical Society, the latter jointly with the German Acoustical Society DEGA.

These scientific events are going to be held in the Delfin Hotel in Jurata on Hel Peninsula on 13 – 16 September 2011.

The topics of the papers submitted to the 58th Open Seminar on Acoustics cover a large area of theoretical, technical and experimental research in the field of acoustics. The 2nd Polish-German Structured Conference on Acoustics will be held in the following sections: multi-modal and object-oriented approaches to audition, computational acoustics, active noise control, room and building acoustics, and seabed acoustics.

Five invited papers will be delivered by the following speakers:

1. Prof. Otto von Estorff
   Hamburg University of Technology, Hamburg;
2. Prof. Grażyna Grelowska
   Polish Naval University, Gdynia;
3. Prof. Marek Iwaniec
   AGH University of Science and Technology, Kraków;
4. Prof. Andrzej Nowicki
   Institute of Fundamental Technological Research
   Polish Academy of Sciences, Warszawa;
5. Prof. Anna Preis
   Adam Mickiewicz University, Poznań.

The 58th Open Seminar on Acoustics and 2nd Polish-German Structured Conference on Acoustics together have received 115 papers. Some abstracts of these papers are given below.

58th Open Seminar on Acoustics		2nd Polish-German Structured Conference on Acoustics
Organizing Committee	Scientific Committee	
Chairman	*Chairman*	*Coordinator*
Roman Salamon	*Eugeniusz Kozaczka*	*Grażyna Grelowska*

Abstracts

1. What can we learn from psychoacoustics regarding the perception of whole-body vibrations? Tactile descriptors for whole-body vibrations

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In classical psychoacoustics, participants of listening experiments describe the auditory events regarding the interested perceptual descriptors, such as loudness, pitch, timbre, direction of the auditory event, etc. Such kind of descriptors can be categorized into three groups as temporal, spatial and spectral.

Although in the past some of the fundamental psychophysical quantities of whole-body vibrations were investigated, the perceptual descriptors, which are needed to describe, differentiate and scale the perceptual characteristics of whole-body vibrations, are missing. People are exposed to many forms of whole-body vibrations while driving different kinds of vehicles like cars, trucks, ships and helicopters or in other situations, e.g. organ concert in a church, etc. The spectral and temporal content of the vibration signals play an important role on our well-being and product evaluation.

In this study, two sets of experiments were conducted to investigate the tactile descriptors of whole-body vibration events. In the first experiment, representative whole-body vibration signals, such as sinusoidal vibrations at different frequencies, impulsive vibrations and broadband noises, were presented to the subjects. They were asked to describe their all the tactile perceptible impressions. Free verbalization interview resulted in 38 different descriptive terms. In the second experiment, the terms were evaluated according to their suitability in describing tactile perceptible properties of the whole-body vibration signals by all subjects.

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2. Modal analysis and measurements of low-class classical guitars

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The work presented in this paper is concerned with the analysis of the vibrational behavior of low-class classical guitars in relation to their quality.

Measurement methods of determining sound pressure and body vibrational responses are presented. Structural modes of top and back plates have been measured with the aid of a scanning laser vibrometer. Additionally, sound pressure response measurements were performed in the anechoic chamber. The instruments were excited with a modal hammer or a bone vibrator. Issues related to measurement accuracy and precision as well as the influence of excitation point on modal analysis are discussed. Amplitude and frequency relations were investigated in reference to instruments' quality and top plate wood type.

The experiments show that although 85% of energy radiated by the classical guitar come from mode T(1,1), T(1,2) and T(3,1) which appears below 600 Hz, the frequency band above 2 kHz may have crucial impact on the classical guitar sound quality.

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3. Numerical testing of nonlinear generation efficiency in a bubble layer

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The aim of the paper is a theoretical analysis of acoustic waves propagation through a bubble layer. The layers with spherical bubbles of the same sizes and uniformly distributed are considered. The transmitted and reflected waves, their first and second harmonic amplitudes are studied. The mathematical model of the pressure propagation in bubbly liquid layer is constructed by the linear non-dissipative wave and the Rayleigh-Plesset equations. Numerical analysis is carried out for different layer thicknesses, different values of volume fraction and parameters of generated signals, first of all frequency. Some results of numerical investigations are presented.

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4. Educational implementation of a sound level meter in the LabView environment

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As a consequence of implementation of EU Directives related to noise protection there is a growing need for education of students in the respective programs of engineering studies. More and more students of various AGH Faculties are introduced to the basics of acoustic measurements. That population consists of students at various levels of theoretical background in the field of acoustics measurements. During the laboratory activities they are offered devices, which are in most cases quite advanced types of digital sound analyzers, including also a sound level meter. For practical reasons it would be most favorable if each student could have a device at his/her own disposal. Unfortunately such a situation is not possible at the moment because of various reasons.

Having the above problem in mind a dedicated software package has been prepared, implemented in the LabView environment, which allows detailed studies of problems related to the acoustic signal measurements. The software package covers measurements using sound level meters as well as tasks in spectral analysis (1/1 and 1/3 band filters) and narrow band (FFT) analysis. The application includes sound level meters with indicators in both digital and analog versions as well as visual presentations of the measured quantities, depending on the device settings applied by the student.

With such an arrangement during the introductory laboratory classes, carried out using individual PC's, each student is offered a direct individual contact with a device that is properly pre-programmed for realization of a well-constructed learning process. Such a situation definitely facilitates the understanding of the essence of acoustic signal measurements and provides a good basis for further laboratory work, carried out as team-activities.

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5. Estimation of the long-term noise indicators on the basis of the random environment control investigations

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Problems related to the estimation of long-term noise indicators on the basis of random 'momentary' environment control investigations, are presented in the paper. An attention was directed towards deficiencies of the standard estimation procedures of the long-term noise indicators; it means estimations of their average values and standard deviations attributed to their uncertainty assessments. These standard deviations are usually determined on the basis of the random sample of the environmental examination results. Possible ways of looking for the problem solution were shown. The proposed method of modelling the results of a random, 'momentary', sample of control investigations, leading to the determination of the expected value of the long-term noise indicators and to the assessment of their uncertainty, was described. The authors related the selection and realization of the proposed solution to the theory and methods of analysis of time series. Investigations, analyses and verifying procedures accompanying the proposed mathematical formalisation, were given. Exemplifying contents of the presented paper were related to the assessment of the traffic noise at the one of the main arteries of Kraków. They are considered in a context of the practice of the 'momentary' measurements and the estimation of long-term noise indicators performed on their basis.

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6. Determination form of distribution for sound level measurement results

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The article show the way of estimation probability density distribution for the random sample of measurements of sound levels L_A shifted gamma distribution.

Formulates the need of its known in the context of acoustic environmental checking inspections and related assessments of their uncertainty. An analysis of differences and references to a normal distribution, commonly used in the statistical analysis of acoustic measurements.

The intention of the authors of the proposed solution should be a basis for wider discussion on the exploration of new algorithms for the estimation of controlled noise indicators and assessment of their uncertainty and procedures for acoustic identification and verification formalisms adopted in the research model.

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7. LMS algorithm stability for short adaptive filters

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The derivation of the LMS algorithm stability necessary condition for short adaptive filters is described in this paper. It is not, contrary to all currently known conditions, based on statistic properties of input signals, nor it does make use of the small step size assumption. Instead, the derivation is based on the theory of discrete systems and properties of discrete state space matrix. Simulation experiments, where the step size reaches a couple of thousands without the loss of stability, are shown to support the theory. Therefore, the new condition may be useful in determining the maximum allowed step size in cases where, at some stage of adaptation, fast convergence is necessary and the small step size assumptions do not hold.

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8. Method of structural intensity calculation based on frequency response function

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The practical aspect of application of structure surface intensity to the analysis of thin-walled structures was the subject of work. Method based on complex modal parameters derived from frequency response function was used. The exactitude of proposed procedure of calculations was analyzed and compared with method based on modal analysis. The example of numerical analysis was done for model of the lightly damped simply supported plate forced to vibrate with harmonic excitation. The numerical model included the source of vibrations in form of force excitation with known position of application and sink of energy with localized damper. Under special interest was the shape of intensity vector field. There were shown the advantage of applied method of modeling in comparison with method based on modal superposition. The results of the numerical calculations allow assessing the spatial distribution of structural intensity vector values on the surface parts and enabled more precise analysis of vibration energy flow. The energy balance done for particular elements the leads to valuable practical conclusions for confinement if the vibration energy flows and lowering of the sound radiation. The method of intensity calculation was intentional to show the utility of structure surface intensity method in discontinuity and damage diagnostics of mechanical constructions.

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9. Acoustic design of a broadcast studio by finite element method

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In the papers results of design of room acoustics of broadcast studios with complex shape are presented. The analysis was done in low frequency range by finite element method. The influence of shape of the rooms on room's resonances and particular on its distribution in frequency domain is demonstrated. Values of reverberation time obtained by statistical Sabin theory and obtained by wave-based finite element method are compared and discussed. Relation between particular localization of sound absorption materials in a room and its reverberation time in low frequency range is discussed.

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10. A note on application of frequency domain indirect method in time domain sound field analysis of rooms

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In the papers frequency domain indirect method for time domain sound field analysis in rooms is presented. The method is based on the inverse Fourier transform of the frequency domain response computed in frequency domain by finite element method or boundary element method. Method's validation through comparing calculated and measured results of a small rectangular box has been carried out. It was shown the proper estimation of acoustic admittance of room's walls is crucial for correct prediction acoustic parameters which are derived from room's impulse response. It seems the evaluation of Q-factor of individual resonances in spectrum of measured impulse response is proper way to estimate admittance boundary conditions and to calibrate a model.

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11. Active sound control in open space

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In this article, an original system of active control of the parameters of an acoustic field has been presented. Operation of the system is based on ensuring the best parameters in the controlled area, such as: uniformity of sound, impressions of spatial sound, intelligibility of speech and music with minimization of sound level in the protected area.

Within the framework of the realized work, a computer program algorithm was elaborated upon and a sixteen channel system for sound control was designed and executed. The core of the constructed system consists of amplifying systems and digital delay lines implemented on the ADSP BF-537 multi-channel card made by the Analog Devices Company, which was equipped with an expansion module. These systems have been programmed using the VisualDSP++ 4.5 environment. Modeling and simulation studies were performed using the geometric methods implemented in the EASE pack. The final phase of work included tests confirming the effectiveness of operation of the designed and executed system, which were carried out in an open space in a selected area.

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12. Surface acoustic impedance estimation of enclosed space

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The study of sound fields in enclosed spaces can be approached in different ways. One can solve the wave equation with the prescribed boundary conditions either analytically or using numerical methods. Alternatively, the problem can be studied using statistical considerations. The authors have combined the experimental and numerical methods to determine the acoustic impedance of wall materials. Experimental studies performed using a multichannel measuring system in the interior of the prepared factory room model. The measuring system allowed for simultaneous measurements of parameters of the source – the vibration velocity of speaker diaphragm, the sound pressure at the measuring points inside the model and the phase shifts between registered signals. Numerical modeling with using the measured values of sound pressure inside the analyzed area allowed the determination of parameters the boundary area.

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13. Time-frequency analysis of acoustic signals using concentrated spectrogram

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The paper presents improved method of time-frequency (TF) analysis of discrete-time signals. The method involves signal's local group delay (LGD) and channelized instantaneous frequency (CIF) to purposely redistribute all Short-time Fourier transform (STFT) lines. Additionally, the energy concentration index (ECI) and some histogram-like statistics are used to evaluate readability of estimated TF distributions of the energy. Recorded acoustic signals such as sound of the Stradivarius violin, a humpback whale song and a theoretical signals such as impulse response of a rigid and tungsten carbide sphere confined in water are employed to test the novel approach.

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14. The methods of accounting for the influence of sound level meter parametrs on the uncertainty of the measurement of noise describing quantities

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Three different approaches to the estimation of components of sound level measurement uncertainty related to the imperfect metrological characteristics of sound level meters are presented and compared in this paper:

- based on maximum permissible errors specified in the appropriate standard,
- based on statistical analysis of the results of periodic calibration of sound level meters,
- based on the results of periodic calibration of particular sound level meter.

The method of the estimation of uncertainty related to the error of sound level meter adjustment and calibration performed by sound calibrator and the method of accounting for the influence of self-generated noise on the uncertainty of measurement of low level sound pressure are also presented.

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15. Modified numerical algorithm for diffusion equation applied in room acoustics

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The paper shows numerical way of solving an acoustical diffusion equation for 1D, 2D, and 3D using finite difference method. Presented methods include modifications which eliminate additional energy appearing in the model for small absorption coefficient as well as modification which eliminates the need to include additional external points to the model.

Despite of proposed modifications, in 2D and 3D model, some additional energy can be found in model when absorption coefficient is equal to zero. Reverberation times obtained for rooms with small number of calculation points, have better accuracy when are compared to RT60 counted from classical Sabine and Eyring formulas.

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16. Modeling of noise-type distortions of loudspeaker with nonlinear voice-coil inductance

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The method of modeling of nonlinear distortions in loudspeakers with the nonlinear voice-coil inductance is presented in the paper. The broadband noise excitation is used for modeling. This method requires use of digital band-rejection filter at the input of the system and digital band-pass filter at the output. The differential equations set for modeling the nonlinear properties of loudspeaker is included between the input and the output for simulation process. It makes possible to observe the influence of different parameters of loudspeaker on the distortion product.

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17. Comparison of magnetic circuits with permanent magnets and electromagnets for dynamic loudspeakers

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Two solutions of magnetic circuits generating magnetic field in the same air gap are compared and design as well as optimization procedures are given in the paper. Rarely used circuits with electromagnets show positive and interesting properties, among others they allow for adjustment of speaker efficiency and momentary increasing of magnetic induction in the air gap. This is important for the pulse mode work. The calculation examples concerns the high-effective speaker designed for the use in sodars.

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18. Testing of the positioning device for the measuring microphone in the research room

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This paper presents a mechanical positioning system for a measurement microphone designed for acoustic studies in an anechoic and reverberation chambers at the Department of Mechanics and Vibroacoustics, AGH. The results are discussed in the context of mechanical positioning and its impact on the outcome of the execution of individual measurement procedures. Moreover, areas for research were identified and solution concepts shown for further development of the automation of acoustic measurements in different research rooms in order to reduce human involvement in them.

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19. On the creation of acoustic spatial impressions in open areas

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In the article was presented the alternative way of sound reinforcement, which allows to obtained in reinforced area sound spatial impressions as well as good speech intelligibility. The idea of the system is the appropriate distribution of loudspeakers around the audience, which can be obtained by using Inverse Image Source Method (IISM). In the open area it is desirable to get as good sound impressions, as in a closed space of the same size. For this purpose, simulations of sound field in Munich Herkulessaal were carried out. Then was attempted

to transfer sound impressions of this object to open area using sound reinforcement system with additional sound sources of appointed parameters, determination of which was based on the IISM.

As is clear from simulations, it is possible to obtain in open areas similar sound spatial impressions to sound impressions in closed spaces, and it is also possible to improve sound quality in terms of speech intelligibility.

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20. Influence of the medium flow on directivity characteristics

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The aim of the paper is to provide a review of different acoustic far field radiation measures with special emphasis put on the presence of the medium flow, the phenomenon commonly occurring in industrial systems where the theory of acoustic ducts founds its application, such as jet engines or heating and ventilation plants. Graphical representation of the far field radiation in polar and azimuth angles, called directivity characteristics, may include depiction of the sound pressure, the sound intensity, or the sound power-gain function, either in linear or decibel scale. In case of the acoustic pressure alone, the menagerie of acoustic nomenclature (according to "Formulas of acoustics" by Mechel *et al.*) includes such quantities as: the directivity factor, the directivity coefficient, the directivity value, the directivity index, and the directivity as such, of which the last two are measured in decibel scale. In the absence of the medium flow, the directivity coefficient defined as the square of the directivity factor represents at the same time the intensity directivity up to a constant factor; the same relation exists between the directivity index and the directivity. In the presence of flow, however, the far field relation between the sound pressure and the sound intensity gets complicated and substantial differences between the pressure and intensity directivity characteristics appear, so derivation of one of them does not lead automatically to the knowledge of the other one. The sound pressure and intensity characteristics depend also on the direction of flow expressed usually by means of positive or negative Mach number.

The paper presents also experimental results of directivity characteristics depicting radiation from a cylindrical duct with a uniform flow in both directions compared to directivity characteristics without flow. The duct was excited by a fan, the results were presented in one third octave bands from 500 Hz to 16 kH.

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21. Active noise reduction in emergency vehicles

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A sound warning signal, emitted by emergency vehicles, makes a very bothersome noise for the crew. A technical solution, depending on integration of a system of a warning signal generator with a communication system and system of active noise reduction in one unit, is presented in this paper. Basic legal and standard-setting regulations, concerning warning signals applied in emergency vehicles, are discussed and a new warning signal is proposed, as it enables

more effective operation of the system of active noise reduction. Structure of the system of active noise reduction and the control algorithm, based on an adaptive filter type NOTCH, is presented. Verifying tests concerning effectiveness of a system of active reduction of noise and improvement in speech intelligibility are presented as well.

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22. Ultrasonic method of monitoring of environmental risks associated with precipitation

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Monitoring of environmental risks associated with precipitation is especially important as it allows early warnings about possible results of exceeded critical values of e.g. thickness of the layer of snow on flat roofs or water level in rivers and water basins.

The work presents an ultrasonic method of monitoring the snow layer thickness or water level based on measurement from air. It describes the principle of operation of a measurement device using three methods of compensating for external factors affecting the appliance's precision. It also presents a general block diagram of a device providing temperature and parametric compensation as well as the structure of an ultrasound measurement probe. In order to verify the proposed solutions the research team tested the measurement device in laboratory and operating conditions. The results that were obtained make it possible to select specific configuration of device operation depending on the required measurement precision and limitations associated with installing the system for actual operation.

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23. Selected methods of vibroacoustical energy flow in modified structures

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In general under the term structural modification, the process of prediction and identification of changes in structure properties aiming at assumed alteration of structure dynamic behaviour is understood. The process of structural modification frequently requires taking into account the energy balance. In case of local modification, such a situation takes place when additional energy is supplied or received from the system being modified. In case of a junction of subsystems, such a situation takes place when a subsystem being adjoined has non-zero energy, or an additional energy resulting from the modification process is released (eg. during collisions, residual processes, appearance of loads changing system structure, cracks propagation, etc).

In this study the selected methods of structural modification are presented and systemized in respect to energy balance. Those methods are considered as an effective tool for energy flow control in simple and complex structures.

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24. Influence of measurement conditions in reverberation chamber on the result of acoustic research

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The correctness of the results of acoustic tests carried out in the reverberation chamber depends on proper tuning of the reverberation time and on determinating of the sound absorption parameters in acoustic chamber. During the process of starting a new test bench, it is necessary to carry out the first measurements using an object with known acoustic parameters. This is similar to the calibration process and allows us to confirm the accuracy of the results. This paper presents these research, conducted in the sistem of reverberation chambers for measurements of insulation, acoustic power and sound absorption in the laboratory at the Ship Design and Research Centre.

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25. Test signal selection for determining the sound scattering coefficient in a reverberation chamber

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The paper focuses on the problem of test signal selection while determining sound scattering coefficient according to ISO 17497-1. Our research shows that the use of MLS signal is preferred in this procedure. Sine sweep signal, despite its advantages, presents certain limitations if the sample is moving. An attempt has been made to develop a method that enables error minimization, taking advantage of the dependence of the obtained values of sound scattering coefficient s on rotational speed of the turntable and the type of the test signal. The conditions have been specified in which sine sweep test signal can be applied at continuous and discreet measurement.

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26. Sound reflection from reflecting screens depending on ceiling adaptation

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Reflecting structure placed over the stage in concert halls and auditoria should provide sound reflection in a certain way, which will assist the emission from the scene and will not

cause defects in the interior acoustics. Model studies were used to determine the relative level of reflections by the reflecting structure as a function of frequency for a number of geometric and material configurations. Analysis of the results allows to define the influence of ceiling adaptation on the quality of the reflected sound. It was shown that adaptation of the ceiling over reflecting panels using strongly absorbing materials, significantly improve rebound characteristics. It is also necessary to avoid some arrangements of elements since they could cause unfavorable acoustic effects.

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27. Studying diffusive acoustic systems with textile covers

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Textile covers for acoustic systems are willingly used by interior designers, because they make it possible to easily mask certain items. However, when put on acoustic diffusers that alter the phase of the reflected wave, these covers lower the system's effectiveness by increasing its sound absorption. The authors have measured the sound absorption and diffusion coefficient of the system in several configurations of the textile cover. The measurements have been conducted according to standard procedures. The results show a significant impact of the cover on the acoustic parameters of the system.

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28. Usage of participatory GIS-type tools in managing acoustic climate of urban area

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The paper presents the problems of the new challenges generated by the need to draw up strategic acoustic maps and action plans against noise pollution by cities' administrations by the end of 2013. The possibilities and advantages of the use of tools such as Participatory GIS (PGIS) to manage the acoustic climate of urban areas are described. The objective of such activities is to adjust the planned solutions for noise reduction to social perception and the actual needs of residents in this area. The conclusions of earlier studies conducted on the use of elements of the PGIS methodology, (supplementing the classic GIS methodology) to map noise and setting priorities for protective action were quoted. The paper is a contribution to the development project "Network system for counseling and consultation in the process of creation and use of acoustic maps" that is conducted at the Institute of Production Engineering of the Silesian University of Technology.

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29. Recognition of selected sources of transport noise in the acoustic climate monitoring

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Noise is one of the major environmental problems in the inhabited areas of the world. In the European Union, Directive 2002/49/EC establishes uniform requirements for the assessment and management of environmental noise. In Poland, the provisions of the said Directive were implemented into the national Environmental Protection Law Act of 27 April 2001 (as amended). The purpose of these acts to the legal system is the protection of health, quality of life and well-being of the inhabitants of the globe. While exploiting the environment by emission of significant quantities of acoustic energy, the management of a road, a railway, a tram-line, an airport or a port is required to perform continuous measurement of such emissions. There is also important to monitor the environmental impact of industrial plant and installations. Such studies are designed to collect information about the prevailing acoustic climate and to produce conclusions, reports and maps of the areas most threatened with limits being exceeded. Carrying out continuous monitoring of a particular area, involves problems of large quantities of the recorded data, often representing the information unrelated to the study source. Manual verification of data is time-consuming and costly. Therefore, to develop effective methods for automatic identification of transport and industrial noise sources becomes an important task for the proper determination of noise levels. One of possible solution is to use pattern recognition techniques related to acoustic signal recorded by the monitoring station. The paper presents usefulness of special directed measurement techniques, acoustic signal processing (based on human hearing perception) and classification methods using artificial intelligence in the recognition of transport noise (air traffic noise, railway noise) in the acoustic environmental monitoring.

The paper has been performed within the project R03 0030 06/2009.

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30. Interaction between speech intelligibility task and simultaneous distracting task

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A lot of research on various aspects of speech perception has been conducted. However, the unsolved problem is the ability of human mind to perform multiple tasks simultaneously. Perceptual load theory states that people decide which of their activities is the most important, then available perceptual resources are allocated. Less important activities are conducted effectively only if remaining resources are sufficient. It should be emphasized that researchers studying divided attention focused mainly on activities requiring vision. This study focuses on the influence of non auditory tasks, called distracters, on speech intelligibility. Two experiments examining the interactions between distracters and speech intelligibility were conducted. In none of the experiments speech intelligibility decreased due to distracter, however, the percentage of correct responses to distracter when subjects were performing speech intelligibility tests decreased compared to situation where they focused only on distracter.

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31. Efficiency of induction loops in speech intelligibility improvement in hearing aid users

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Although the idea of induction loops was introduced a few decades ago, yet it is still undervalued and relatively seldom used, especially in public areas. However, it appears that the use of such a solution can significantly improve speech intelligibility in hearing aid users. It is worth noting that generally all currently produced hearing aids are equipped with a coupling system, which makes their user not to be exposed to any costs associated with the purchase of additional equipment, such as infrared or FM systems. This article presents the results of the analysis of speech intelligibility improvement in individuals with hearing loss who are users of hearing aids. The study was conducted *in situ* in three rooms, of different purposes and different acoustic conditions. The speech intelligibility test results indicate a very high efficiency of induction loops.

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32. Attenuation of instrumental sounds by chair mounted screen for musicians

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In this study, attenuation of musical instrument sounds by chair mounted screen for musicians was determined. The attenuation was determined as the insertion loss calculated as a difference in the A-weighted sound pressure levels measured by a microphone placed above a chair, in the place of musicians head, with the screen present and removed. Tests were performed in an anechoic chamber and in a chamber music concert hall. Results showed that the screen can be an effective barrier to sound only for instruments of high pitch for which the attenuation determined in an anechoic chamber was up to 18 dB. In concert hall reverberant conditions, the screen decreased sound level by about 2 to 10 dB.

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33. Draft of the revised german regulation DIN 45680 – measurement and assessment of low-frequency noise immissions

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Low-frequency noise (LFN) is a growing problem in Germany. In this situation many reasonable complaints regarding LFN were rejected by the environment agencies, because the data measured and calculated accordingly to the regulation DIN 45680 do not exceed the limits given there. Often, only small details determine the decision, if a complaint is successful or not. For example, the actual version of the DIN 45680 distinguishes between a broadband and tonal LFN and the limits for a tonal LFN are lower and easier to meet. However, the difference in the characteristic between broadband and tonal LFN can be very small by definition. This makes the decision arbitrary up to a certain extent. To overcome such problems und to reduce the inconsistency between reasonable but not successful complaints a revision of the DIN 45680 from 1997 was initiated in 2005 and a draft of the DIN-working-group NA 001-01-02-11 AK "Revision of DIN 45680" is available now, which will be discussed in the next time. This paper reports about the main issues of this draft.

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34. Nonlinear active control of narrowband noise in a duct with a vibrating plate

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Feedforward active noise control system with vibrating plate in the primary path is discussed in this paper. Applied control algorithm is based on the MLP neural network utilized for realization of the NARMAX model. Modified Error Backpropagation algorithm is used for learning the neural network. Parametric system identification using Hammerstein-Wiener models is performed for determination of limitations of the applied active noise control algorithm. Laboratory experiments with pure tone signals are presented.

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35. Modernization of large music studio at Polish Radio Szczecin

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The paper describes the modernization of the music studio with a volume of 2244 m^3. The previous acoustics of the studio was one of the best in Polish broadcast, so the main task of studio modernization was to keep its acoustics. Therefore, large part of the existing design has been preserved to which additional elements were acoustically matched. Modernization covered an interior refurbishment and installation of steel trusses which expanded the function of the studio with TV recording. Roller blinds were also installed which enabled regulation of reverberation time in the range of 0.55–0.95 s. Acoustical measurements and positive subjective assessment of the studio acoustics confirm the validity of the used way of upgrading of the studio function.

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36. Laboratory tests of sound insulation for "sandwich" systems with mineral wool

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Article compares the results of sound insulation for airborne sound, carried out in accordance with international standards, in the reverberation chambers in Acoustic Laboratory of Ship Design and Research Centre. The tested samples were prepared as the "sandwich" type systems, with panels of mineral wool of various densities and thicknesses, as well as with air space between components. The results are presented in the form of mathematical relations for Rw – index of acoustic insulation, and also in the form of diagrams for frequency distribution.

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37. Transmission loss of structured sheet metal

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In this paper measurements of the transmission loss of structured sheet metal and flat, unstructured sheet metal as reference are presented and differences are explained using the mode shapes of the sheet metal gained out of numerical calculation results. The measurements are performed in a window test chamber and the numerical results are obtained using the commercial vibro-acoustic solver FFT Actran. The measured transmission loss of the structured sheet metal shows a reduction within the range of about 5 to 12 kHz in comparison to the flat sheet metal. This behavior is discussed with regard to the modal density of the structured sheet metal. They vibrate in similar global modes as the flat sheet metals do, but for higher frequencies. Furthermore the structured sheet metals show unique local mode shapes due to the bump structure in comparison to flat sheet metals. These mode shapes and the higher stiffness give an explanation for the reduced measured transmission loss in the mentioned frequency range.

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38. An adaptive vibration control procedure based on symbolic solution of diophantine equation

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In this paper, the adaptive control based on symbolic solution of Diophantine equation is used to suppress circular plate vibrations. It is assumed that the system to be regulated is unknown. The plate is excited by a uniform force over the bottom surface generated by a loudspeaker. The axially-symmetrical vibrations of the plate are measured by the application of the strain sensors located along the plate radius, and two centrally placed piezoceramic discs are used to cancel the plate vibrations. The adaptive control scheme presented in this work has the ability to calculate the error sensor signals, to compute the control effort and to apply it to the actuator within one sampling period. For precise identification of system model the regularized RLS algorithm has been applied. Self-tuning controller of RST type, derived

for the assumed system model of the 4th order is used to suppress the plate vibration. Some numerical examples illustrating the improvement gained by incorporating adaptive control are demonstrated.

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39. Formation of acoustic diffusion in rooms

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Paper describes known from literature method of sound diffusity index assesment, which utilizes impulse response of the room. Process of forming diffused sound field using this method was investigated. Increase of sound diffusity index in time was noted in rooms differing in shape, volume and reverberation time. Exponential growth of sound diffusion index was noted which correlates with other acoustical rooms' measures.

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40. Validation of the Polish Matrix Sentence test in various groups of cochlear implant users

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The paper focuses on the application of the Polish Matrix Sentence Test (PMST) for measuring speech intelligibility for a group of 22 cochlear implant (CI) users. The tested patients were post-lingual CI users who had been using CI devices for 5–6 years. The PMST was presented with a background of masking noise at different signal-to-noise ratios (SNRs). The SNRs were modified adaptively (1-up/1-down decision rule) to determine the speech reception threshold (SRT), i.e. the SNR yielding 50% correct responses. A four-channel experimental setup was used for presentation of the stimuli: the target speech, i.e. the PMST, was presented from a loudspeaker placed at the azimuth of 0 deg, whereas the uncorrelated 'speech-like' noise (babble) was presented via loudspeakers placed at the azimuths of 90, 180 and 270 deg. Differences between patients with unilaterally and bilaterally fitted CI devices, as well as with and without additional acoustic stimulation, were analysed and compared against data from previous studies. In general, the poorest performance was observed for unilaterally fitted CI users, while the best speech recognition was shown for patients equipped with bilateral CIs (with and without additional acoustic stimulation). Apart from that no correlation between individual mean pure tone thresholds and speech intelligibility data was found.

Supported by a grant from Norway through the Norwegian Financial Mechanism (contract no. 7/2009).

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41. Acoustic short helicoidal resonator – computational and experimental investigations

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The paper concerns with acoustic properties of a short helicoidal resonator inside a cylindrical pipe which exhibits the role of the band stop acoustic filter, or the band rejection filter. The similar behavior is observed with Helmholtz resonator incorporated at one side of the pipe. Using three dimensional acoustic simulations verified by the experiment, it was shown that acoustic band stop properties mostly depend on the number or fraction of turns of a helicoidal shape inside the pipe. This means that by a proper choice of this main parameter we can set the filtering properties of this acoustic element, mainly the width and the magnitude of sound or noise rejection. This gives us the possibility to attenuate the noise propagation inside the ventilation systems or chimneys.

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42. Multibeam sonar data processing for seafloor characterisation

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The approach to seafloor characterisation was investigated. It relies on calculation of several descriptors (parameters) related to seabed type using three types of multibeam sonar data obtained during seafloor sensing: 1) the grey-level sonar images of seabed, 2) the 3^D model of the seabed surface which consist of (x, y, z) points, 3) the set of time domain echo envelopes corresponding to several beams. The proposed method has been tested using field data records acquired from several bottom types in the Gulf of Gdańsk region and the promising results have been reported. This paper presents and discusses several techniques of processing the calculated descriptors in the context of their use in seabed classification. Namely, the calculating the standard deviation of the descriptors for small subsets of data, as well as the application of the Principal Component Analysis procedure (PCA) was investigated.

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43. Structure-born-sound of an UIC60 rail and the radiated sound field

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This paper gives a mechanical model of the railway track in full 3-D FEM. Furthermore the infinite character of the track is considered. Because of the 3-D elements the operations to

find a wave base lead to numerical problems. These difficulties occur just if discrete damping-elements are considered to model the damping character of pad and ballast. The reason for the ill conditioning of the matrices will be explained and a solution will be presented. Afterwards the radiated sound field will be calculated using a Boundary element method and discussed.

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44. Active measures to reduce the transmission of noise and vibrations in engine mounts – concepts and application examples

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Active measures to reduce the transmission of noise and vibrations in engine mounts are usually based on the concept of generating additional forces at appropriate locations (e.g., mounts, bearings. The additional forces are controlled with respect to frequency, phase, and amplitude in such a way that they counteract the unwanted excitation and, therefore, significantly reduce the overall vibration or the radiated sound.

Some of them are based on innovative mechatronic add-on concepts; others are directly integrated into the mechanical load path and thus are based on adaptronic (i.e., smart structure) concepts. Both approaches allow using smart materials or mechatronic devices as sensors and/or actuators. Both, the potential level of noise and vibration reduction and the resulting cost (e.g., development, component, and system costs) depend, among other factors, strongly on the particular vibration source (kind of the engine) and its installation conditions.

This paper describes most promising active approaches in practical application examples, classified with respect to the physical principles.

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45. Adaptive equalization of sound radiation from a plate using acceleration sensors and virtual-microphone control

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Vibrating plates are potentially interesting sound sources for industrial applications. They are more resistant to harsh environmental conditions than loudspeakers. However, they have much worse sound generation properties. In this paper a method for improving those properties by using an adaptive feed-forward control system of sound radiation from a plate is investigated. FIR filters are used for each actuator to generate control signals based on the reference signal. The filters are updated with the FXLMS algorithm in order to minimize the instantaneous

squared value of the difference between the sound pressure at the specified point in the acoustic field, and the reference signal filtered by the desired secondary path transfer function. In contrary to previous publications of the authors, the sound pressure is not directly measured, but a virtual microphone approach employing accelerometers mounted on the plate is used. Performance of the system is experimentally verified and compared to the system based on microphone measurements, and obtained results are reported.

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46. The influence of source location on the distribution of steady-state sound field inside two coupled rooms

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In the paper, the computer modelling application based on the modal expansion method is developed to study an influence of a sound source location on a steady-state response of coupled rooms. In the research, an eigenvalue problem is solved numerically for a room system consisting of two rectangular spaces connected to one another. A numerical procedure enables the computation of shape and frequency of eigenmodes, and allows one to predict the potential and kinetic energy densities in a steady-state. In the first stage, a frequency room response for several source positions is investigated, demonstrating large deformations of this response for strong and weak modal excitations. Next, particular attention is given to studying how the changes in a source position influence the room response when a source frequency is tuned to a resonant frequency of a strongly localized mode.

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47. Feeling the sound: audio-tactile intensity perception

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In this study, an experiment was conducted to determine the influence of vibration on loudness perception. Vertical whole-body vibrations were produced using an individually calibrated electro-dynamic shaker. A pair of closed dynamic headphones was used for audio reproduction. A loudness matching experiment was carried out with 28 participants. In this experiment, sinusoidal sounds were presented in the absence and presence of simultaneous vibrations. Participants had to match the level of the acoustic stimuli while ignoring the tactile stimulation. The loudness matching experiment was carried out for four sinusoidal tones, at the frequencies of 10, 20, 63 and 200 Hz. Three different vibration levels (4, 8 and 12 dB above the vibration threshold for each participant) were used to study the influence of the vibration amplitude. Results indicate that whole-body vibration had a significant influence on loudness perception. When an acoustic stimulus was accompanied by vibration, the level of the acoustic stimulus was perceived one decibel higher on average.

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48. Parameterization of adaptive control algorithms for multi-channel active noise control system

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The paper summarizes an initial research on parameterization of adaptive control algorithms applied for three-channel active noise control (ANC) system, used to create three dimensional local zones of quiet in a reverberant enclosure by reduction of tonal disturbance. Two different ANC system structures are concerned: classical ‘filtered-x’ and modified ‘filtered-x’ structures, both for control algorithms: least mean squares three (LMS), recursive least squares (RLS) and affine projection (AP) algorithms. Apart of classical simultaneous adaptation method, a round robin adaptation method is also applied and verified. The problem of parameterization of chosen adaptive control algorithms is evaluated in computer simulations – an applicable range of parameters is determined and next the best parameters set is chosen for each control algorithm to compare their performance.

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49. Results of measuring the speech transmission index (STI) in rooms with a method employing a mouth simulator and an omnidirectional source

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The quality of verbal content reception in a speaker-listener system (e.g. teacher-pupil) in rooms is affected by the speaker’s speech characteristics, the speaker’s and listener’s locations, background noise level and acoustic properties of rooms. The quality may be characterised by the speech transmission index. It is being determined on the basis of the reduction of the speaker’s voice modulation index (or of a phonic signal emitted by a loudspeaker – transmitting system) with reference to the modulation index of the signal measured at the listener’s location (the phonic signal recorded in the measuring system). The measurement standard for the index has been determined in PN-EN 60268-16 standard with the aim to minimise the effect of measurement method on the value of the index. One of the important elements of the measurement method is the usage of the artificial mouth source employed as the acoustic source in the transmitting channel. The authors made measurements of the speech transmission index (STI) employing the method defined in the above mentioned standard and compared results obtained when using the artificial mouth source (reference measurements) with the ones obtained by means of an omnidirectional source. Measurements were conducted in four rooms having different acoustic properties. The authors also made a comparison between how the measure values of the speech transmission index (STI) are influenced by different orientation of the main axis of the directional source (artificial mouth) in a horizontal plane.

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50. A lumped-circuit model for radial vibration mode of circular piston transducer

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The paper presents network equivalent circuit of piezoceramic circular disc transducer that takes into account thickness and radial mode of vibrations. The novel model uses Cauer LC ladder of the second form for implementation of radial mode of vibrations. Its values are obtained by expanding the quotient of zero and first order Bessel functions of the first kind into continued fraction representation and can be expressed directly by geometry of the piezoelectric disc (its radius and thickness) and its material properties (stiffness coefficient and density). As the ladder contains only lumped elements, the model is portable across different implementations of SPICE simulators. It has clear interpretation considering mechanical properties of vibrating disc and comparing to thickness vibration model. Presented equivalent circuit of whole transducer is especially useful for simulating complete electronic circuit of multi-frequency echo-sounders based on single circular disc transducer in time and frequency domains.

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51. Integration of acoustics simulation into a development environment for adaptronic systems

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This paper proposes an approach for integrating the simulation of acoustics problems into an existing data management tool. As a motivation, differences between and challenges in the fields of adaptronics (i.e., smart structure technology) and mechatronics are discussed. The aim of the research at the LOEWE-Zentrum AdRIA is presented and the reasons for the need for acoustics simulations are shown. An existing data management tool is presented and it is shown how this data management tool needs to be altered and adapted so that it can be used for the development of adaptronic products as well.

Then, the procedure of acoustics simulations in general and the specifics of the acoustics simulation of adaptronic structures are described considering principles of automation. Both the schematic workflow and the actual implementation of such a simulation are shown. The paper concludes with a summary and suggestions for further integration of acoustics simulations.

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52. Statistical verification of models reverberation time in small rooms

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In this article presents the results of measurements and theoretical calculations for small spaces. With the aim of analysis carried out 5 models of the rooms, which have been placed in non reverberant chamber. Measurements of the reverberation time in these rooms were made in two variants, the first in isotropic acoustic field, the second one in the field disturbed by elements of mineral wool. Then, for all models were made theoretical calculations in two variants, using three models: Neubeauera, Sabine, Eyring. All results were analyzed in STATISTICA program and on this basis, were made statistical inference of the theoretical model usefulness according to small room's results.

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53. Acoustic emission and the PLC effect in compressed CuZn30 monocrystals

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The instability of plastic deformation of the type Portevin-Le Chatelier (PLC) of monocrystals of single-phase brass CuZn30 with crystallographical orientation $[\bar{1}\ 3\ 9]$ was investigated applying the method of free compression at a constant strain rate and a temperature within the range from 200°C to 400°C, simultaneously recording this phenomenon by means of acoustic emission. It has been found that the values of the amplitude of stress oscillation and the energy of acoustic emission of the investigated alloy, recorded in the respective stage of strain-hardening curves, displaying the PLC effect, depend mainly on the temperature of compression. Moreover it was proved that in the range of the occurrence of the PLC effect the acoustic signal displays the characteristics of an impulse of cyclic repeatability, distinctly correlated qualitatively with the stress oscillations on the curves $\sigma - \varepsilon$. The analysis of the results indicates that the PLC effect in the tested monocrystals results from dynamical strain ageing (DSA).

Keywords: plastic strain, Portevin–Le Chatelier effect (PLC), monocrystals, copper alloys, compression test, acoustic emission.

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54. Method of examining of sets multiple sources of sound in a working environment with the use of modeling methods

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The article presents method of examining of sets of multiple sources of sound for the needs of supporting design of technical systems and workplaces. The objective of this topic is elaboration of efficient method of supporting identification of sets of multiple sources occurring in the working environment. Knowledge about identification of source has a significant meaning in a close field, in direct proximity of a technical object, due to its non-homogenous character. In the research there were used various methods of modelling representation of sound sources taking into account location, acoustic features and direction. There was formulated a proposal to introduce a model of substitute sound source representing a set of multiple sources.

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55. Analysis of the uncertainty in railway noise modelling by the RMR SRM II method with the interval arithmetic application

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The interval arithmetic application in the uncertainty assessment in the railway noise modelling by means of the Dutch RMR SRM II method is presented in the paper. This method is recommended by the European Union Directive 2002/49/WE. The uncertainty analysis in modelling was performed in dependence on the uncertainty of correction coefficients occurring in the presented emission model.

The basic notions were presented and the relations in between interval numbers described. These relations were used for the determination of the variability range of the correction coefficients occurring in the model as well as for the assessment of the uncertainty of the modelling results.

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56. Creating acoustic parameters of sound diffusing structures

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Sound diffusing structures, such as reflection phase grating diffusers (Schroeder's diffusers) are nowadays very popular in acoustics as a effective method of canceling defects and improvement acoustical parameters of interiors. Paper presents the possibilities of creating acoustical properties of diffusers and main faults made in adapting them in rooms. Paper deals also with diffusing structures sound absorption, usually neglected in acoustical treatments. Presented results of laboratory measurements show, that stiffness of the construction, and geometry of diffusers, has a great impact on sound absorption for low frequencies. Influence of elements composition on diffusion coefficient d was analyzed for two types of QRD diffusers based on primary number N=7.

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57. The dual surface method for acoustic exterior problems

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This paper proposes the application of the Dual Surface method to overcome the problem of nonuniqueness of the solution that arises when acoustic exterior problem are solved using integral equations. The method adds interior integral equations, on an appropriately located interior surface, to the surface integral equations multiplied by a purely imaginary factor to

ensure the uniqueness of the solution. This coupling factor and the distance of the interior from the original surface are two free parameters within the method. Applications of the method are given for the radiation as well as for the scattering from different structures. The results show the effectiveness of the method at the critical frequencies where the untreated integral equations cannot be solved uniquely.

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58. Tinnitus therapy based on high-frequency linearization

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The aim of this work was to present problems related to tinnitus symptoms, its pathogenesis, hypotheses on tinnitus causes, and therapy treatments to reduce or mask the phantom noise. In addition, the hypothesis on the existence of parasitic quantization that accompanies hearing loss was recalled. The paper contains a description of experiments carried out with the application of high-frequency dither having specially formed spectral characteristics. Report, discussion on results obtained and conclusions are also included.

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59. Investigation of N-wave propagation parameters of free field projectiles moving

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Basing on an experiences of local military conflicts in the 90th of last century, in the technologically developed countries it were began researches on development of acoustic detection systems and location of the shot. Performing a series of simulation and experimental researches of the distribution of space-time characteristics of pressure disturbances caused by a movement of fired projectile are necessary to develop technical requirements of designed devices.

Experimental distributions of the values of selected parameters of the shock wave propagation resulted from flight 7.62 mm caliber projectile are determined in this paper. Experimental results were compared with the theoretical values.

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60. Measurements of sound absorption coefficient for auditorium seating for various geometry of sample

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The paper presents the results of measurements of sound absorption coefficient for auditorium seating using two methods. According to the first one, seats sound absorption were

measured in reverberation chamber for various configurations of small amount of chairs. Extrapolations of the results were used to predict the absorption coefficients of infinite samples. In second method, samples were covered around with barriers. Comparison of results obtained from both methods, makes it possible to estimate the sound absorption coefficient for any dimensions of audience in auditoria. Moreover, sound absorption of side surfaces could be included in results. The proposed method will simplify the previously known measuring procedures.

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61.Signal selection for silent sonar with mathed filtration

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Secretiveness is a desirable characteristic of sonar in certain military applications. It can be obtained through the issuance of wideband signals with low power and long time of repetition. The main problem in such silent sonar is negative impact of the Doppler effect in terms of detection and accuracy of determining the distance of detected targets. In the paper three types of probing signals namely signals with linear and hyperbolic frequency modulation and linear modulation period are compared in above respect. Presented results of numerical simulations showed that the signals with the hyperbolic frequency modulation are the least sensitive to the Doppler effect from the point of view of detection. For all the compared signals to determine the distance errors are virtually identical.

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62. Experimental analysis of acoustic image of a vibrating three layered cantilever beam with MR fluid

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The paper summarises the results of laboratory testing of a vibrating three-layered cantilever beam with MR fluid. The aim of the tests was to find out how the magnetic field should affect the acoustic image. The end of a beam is fixed to an electrodynamic shaker TIRA S514 with a rigid fixture. The magnetic field acting upon the beam is generated by an electromagnet. The slot width in the electromagnet is 20 mm. The laboratory set-up incorporates two measurement circuits. The first circuit records the displacements of the beam's attachment point and of a selected point at the beam's end. The other circuit registers the acoustic signals accompanying the beam's vibrations. The shaker is controlled by the module LMS SCADAS III via the computer PC supported by the TestLab program. The beam vibrations were induced by kinematic excitations. The applied signals were sine signals of fixed frequency f (51, 52, 53 Hz) and of variable frequency in the range (51, 70 Hz), corresponding to the magnetic field strength in the slot of $H = 74$, 101, 115, 128 kA/m. In each experiments the authors registered time patterns of the beam's displacement and of acoustic pressure. The analysis of beam displacement data and noise levels was done in the frequency domain, in 1/12 octave bands, in

the frequency range (11 Hz to 10 kHz). The experiments indicate that in the acoustic image of the vibrating beam the amplitudes of harmonic frequencies of the spectrum should change, depending on the magnetic field strength.

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63. Analysis of the inluence of some parameters of multitone on sensory perception of consonans in the model based on functioning as autoassociative neuron network

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This article concerns the overall possibility of using auto associative neural networks for measuring the characteristics of auditory sensations of a quantitative nature, for which the classical scaling methods do not give satisfactory results, or performing the measurement using such methods is not possible at all, due to the complex and often ambiguous – from the standpoint of current knowledge – process of perception of these traits. The paper focuses on the proposed concept of description of the dynamics of human auditory system, using this type of network and proposes a methodology of measuring ratio of the total volume of the multi-tone sensation to the volume of its spectral components, perceived simultaneously. This procedure uses the method of statistical correlation of psychoacoustic observables measurement.

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64. Evaluating kinesthetic feedback and its audio-tactile extension for virtual shape and object identification

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The identification of virtual shapes and objects is a particularly important task in the virtual world. To investigate to what extent it is possible to explore and identify virtual shapes and objects with a haptic device providing only force-feedback, suitable experiments have to be conducted. A formalization of such experiments with a basic schematic representation of the actively exploring subject is introduced in this work. The formalization is described with the help of a haptic identification experiment revealing that subjects experience various difficulties during the exploration and recognition process. Different approaches are imaginable to overcome these difficulties. A possibility to classify the approaches is described by introducing the terms "Being There" and "Beyond Being There". This categorization is explained by considering the auditory and tactile extension of kinesthetic feedback to overcome one of the revealed difficulties, namely, the insufficient haptic curvature discrimination. The usefulness of such approaches can be illustrated with the help of the basic schematic representation of the actively exploring subject.

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65. The acoustic vector sensor, Acoustic Scoring and Locating System for Rockets, Artillery & Mortars (RAM-LOC)

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Acoustic sensors can be used to detect, classify and locate battlefield threats such as mortars, rifles and various vehicles. Sound pressure microphones are commonly used for this purpose, but this article focuses on Acoustic Vector Sensors (AVS's). These sensors consist of three orthogonal particle velocity sensors in combination with a sound pressure microphone. These sensors make it possible to measure the direction-of-arrival of a sound wave instantaneously. The use of multiple sensors leads to very robust source localization and classification. This paper presents a system which consists of multiple Unattended Ground Sensors (UGS's). Applications, with an emphasis on Acoustic Scoring and Locating System for Rockets, Artillery & Mortars (RAM-LOC), are discussed.

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66. Adaptation of the stage in an opera house for concert

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This paper adresses the issue of adapting the opera stage to the needs of the concert. The introduction of the orchestra shell for both hypothetical and already existing stages was analysed. Computer analyses of acoustic parameters at the audience seats for Chamber Opera in Kalisz were conducted. In Teatr Wielki in Poznań experimental research was conducted for the stage prepared for the needs of the concert and for the one that was empty. In a ray method – based computer analysis, the distribution of the sound pressure level, RT, EDT, C_{80}, and T_S were measured. In turn, in the experimental research RT and the decrease of the sound level ΔL_P were measured. The conducted research enabled to define what requirements the orchestra shell should meet to obtain the improvement of acoustic conditions.

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67. Identification of noise sources at nonpublic airports

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The report presents the results of acoustic measurements carried out for nonpublic airports. Object of study were propeller-powered aircrafts whose weight were under 8.618 kg. Flight operations performed by the propeller-powered aircrafts may adversely affect the acoustic climate around airports.

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68. An application of Gammatone filter to MPEG-7 sound indexing

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No modeling of filtering in human auditory system has been so far implemented in the design of audio descriptors in MPEG-7 standard. In this study, an application of Gammatone filter to MPEG-7 descriptors is proposed. The aim is to examine if applying model of human auditory filter improves the quality of sound feature extraction, which is relevant for the sound indexing process. The Gammatone auditory filter is implemented as a low-level audio descriptor consistent with MPEG-7 standard, which replaces analysis by the short term Fourier transform (STFT). Tests of audio indexing by high-level audio descriptors using proposed descriptor employing the Gammatone auditory filter showed improved identification of musical instruments as compared to indexing with the use of STFT based original low-level MPEG-7 descriptors.

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69. Optimal aperture in MSTA method for medical ultrasound imaging applications

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The paper presents the optimization problem for the multi-element synthetic transmit aperture method (MSTA) in ultrasound imaging applications. The optimal choice of the transmit aperture size is performed as a trade-off between the lateral resolution, penetration depth and the frame rate. Results of the analysis obtained by a developed optimization algorithm are presented. Maximum penetration depth and the best lateral resolution at given depths are chosen as the optimization criteria. The results of numerical experiments carried out in MATLAB® using synthetic aperture data of point reflectors simulated by Filed II for the case of 5 MHz 128-element linear transducer array with 0.48 mm pitch are presented. The visualization of experimentally obtained synthetic aperture data of a tissue mimicking phantom and in vitro measurements of the beef liver are also shown. The data were obtained using the SonixTOUCH Research system equipped with a linear 4 MHz 128 element transducer with 0.3 mm element pitch, 0.28 mm element width and 70% fractional bandwidth was excited by one sine cycle pulse burst of transducer's center frequency.

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70. Golay coded sequences in synthetic aperture imaging systems

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The paper presents the theoretical and experimental study of synthetic transmit aperture (STA) method combined with Golay coded transmission for medical ultrasound imaging applications. The transmission of long waveforms characterized by a particular autocorrelation

function allows to increase the total energy of the transmitted signal without increasing the peak pressure. It can also improve signal-to-noise ratio and increase the visualization depth maintaining the ultrasound image resolution.

In the work the 128-element linear transducer array with 0.3 mm pitch excited by the 8 and 16-bits Golay coded sequences as well as a one cycle at nominal frequencies 4 MHz were used. The comparison of 2D ultrasound images of the tissue mimicking phantoms is presented to demonstrate the benefits of coded transmission. The image reconstruction was performed using synthetic STA algorithm with transmit and receive signals correction based on a single element directivity function.

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71. Metamodeling and sensitivity analysis of smart structures for active noise reduction

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Smart structures are characterized by structure-integrated sensors and actuators with associated controller systems and power electronics. Particularly, shell structures (such as plates, car bodies, or machine housings) are suitable for smart structure applications that aim at reducing sound emission. As an additional benefit smart structure systems tend to require less additional mass than conventional passive systems for noise and vibration control. The development of smart structure systems requires specific know-how about the system parameters and their interactions. It is also advantageous to perform optimization procedures to develop an efficient system. This paper describes an adequate approach of systems engineering for smart structures considering optimization, design exploration, and sensitivity analysis. After a short introduction the importance of the coupling factor between the active and the passive structure is described. It is shown how to identify important system parameters and their influence using surrogates and sensitivity analysis.

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72. Genesis and development of research on temporary and frequency patterns in analysis of speech signal disturbances

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The examination methods of the vocal organ base usually on comparison of patterns of healthy persons' speech with speech samples of persons with different disorders of this organ. The majority of models of speech perception assume a form of hierarchical processing of the input signal by a series of transformations. It is logical that there exist simple temporal or frequency patterns included in a speech signal. They can affect unitarily on sensorial impulses and they can play a key role in speech perception. Examination of extracted patterns as well

as their changes can give a relatively full description of the vocal organ functions. Presented problems concern research on a non-invasive method of the larynx diagnostics. Sound samples of patients with Reinke's Edema or larynx polyps were examined by spectral analysis and using Elman neural network conditioned by time.

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73. Feedforward adaptive active noise control system with time-varyingnumber of zones of quiet

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Adaptive active noise control system with time-varying number of zones of quiet is considered in a situation when users of the system might either enter or leave a room in which the system operates. The system aims at minimizing noise level at a number of locations which are referred to as zones of quiet.

It is assumed that the system should createa zone of quiet for each of the persons inside the room. This leads to the requirement of activating and deactivating the zones of quiet as the number of users changes over time. Further, it involves changing duties of operating secondary sources. It is also assumed that an instant change of the noise level experienced by the user is highly undesirable because it makes unpleasant and annoying acoustic effects.Whenever such change might occur, an active noise control system should permit a gradual change of the noise reduction level instead of a rapid change.

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74. The influence of synchrony in audio-visual stimuli on annoyance

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Within this study, the influence of synchrony on the simultaneous perception of auditory and visual stimuli is being evaluated. A total of 25 subjects assessed different multimodal stimuli regarding their annoyance, which consisted of visual and auditory files presented either synchronously or asynchronously. While it was not yet possible to find a definitive clear answer, in which way this affected the probands, some interesting approaches are shown and some important improvements for future studies are given.

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75. Measurement method of leveling of sound intensity between the strings of violin

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The leveling of sound intensity between the strings of violin is one of the most important criteria which determine the utility of violin. The objective measurement of this parameter is very difficult. The greatest problem is an inability of the direct measurement of energy which

comes from a violinist. This paper presents the method which solves this problem by using the low-energy violin modes as the reference to calculate the differences between sound intensity of violin strings.

The recordings of two octaves of the chromatic scales played separately on each string of violin are used in the presented method. The energy of modes, which were found in the band from 101 Hz to 190 Hz, was used as the reference to calculate the leveling of sound intensity. An algorithm based on a correlation function was used for searching for the violin modes.

The measurement method of leveling of sound intensity between the strings of violin was tested on the violin sound recordings from the AMATI multimedia database. This database contains the recordings of violins from 10th International Henryk Wieniawski Violin Making Competition.

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76. Evaluation of speech quality in Polish for patients after total laryngectomy

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Tasks regarding the analysis and recognition of pathological speech signal, which has been deformed by various types of vocal tract pathology, are extremely difficult. The difficulties result from the fact that the pathologies of speech organs generate various forms of speech signal deformation, which are often very difficult to predict and also very difficult to recognize in a real speech signal registered by a given patient. The correlations between the phonetic and acoustic phenomena, which are observed both in frequency and time domain representation of speech sounds, are in general poorly attributed to the morphological or pathophysiological features of the deformed speech signal generator.

The paper presents selected results of studies regarding the speech quality evaluation for persons after larynx removal (total laryngectomy), who have been trained to use the esophageal speech. The creation mechanism of the oesophageal speech is the formation of vibration generator in the upper part of oesophagus. The substitute air container is formed by the oesophagus itself and the air vibrations in the upper part of oesophagus, generated during the antiperistaltic motion of the air removed from oesophagus, create the primary pitch. The modulation of the primary pitch is obtained by the unchanged articulation organs and resonance cavities of the vocal tract.

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77. Automatic detection of noise long-term indicators in continuous monitoring of audible noise from corona in power lines

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To noise control in the environment are increasingly used continuous monitoring station for short and long-term measurement. One of the most important tasks of the station is au-

tomatic extraction of parameters of the test signal to determine noise indicators necessary for noise impact assessment. In case of audible noise (AN) generated from power lines, due to large fluctuations in both the interference and the useful part of the registered signal, it is necessary correct choice of monitored parameters so you can automatically calculate the indicators. Analysis of previous work shows that it is difficult to apply a classic approach, such as the use of statistic levels for filtration some interference, it is necessary to simultaneously use some distinctive features of corona AN and separate them from the interference. Continuous measurement of the spectrum is associated with registering a large number of data and obstacles with their gathering and processing, so it is advisable to find an optimal approach.

Paper contains some selected measurement results of corona AN from 400 kV power line giving particular attention to definition of gathered parameters for an automatic estimation of basic long-term indicators for corona AN assessment. Introduced some new parameters of acoustic signal – spectral moments, spectral coefficients of the presence of tonal components and power factors of selected bands, allowed more effective detection of the signal sample containing corona acoustic components. For the final selection of the samples, based on the above parameters, an artificial neural networks techniques has been applied. Selected and filtered samples were the basis for calculating of long-term indicators. The effectiveness of the method was tested on sections of recorded sound with listening selection for samples with and without corona.

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78. Combination tones as a phenomenon of the central level in auditory system

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This work addresses the problem of difficulties in classical interpretation of combination tones as nonlinear distortions. One of the basic problems of such an interpretation is to point out the sources of these distortions. Besides, these kinds of distortions have numerous “anomalies” which are difficult to explain on the grounds of physics or physiology. The aim of the model presented in this paper is to show that combination tones phenomenon can be explained as an effect of central mechanisms. Most of existing theories of pitch perception focus mainly on virtual pitch perception and do not take into account combination tones as an element of the same mechanism. In this paper only the simplified model was presented because of the short length of conference paper. This model in the scope of virtual pitch belongs to the class of spectral models.

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