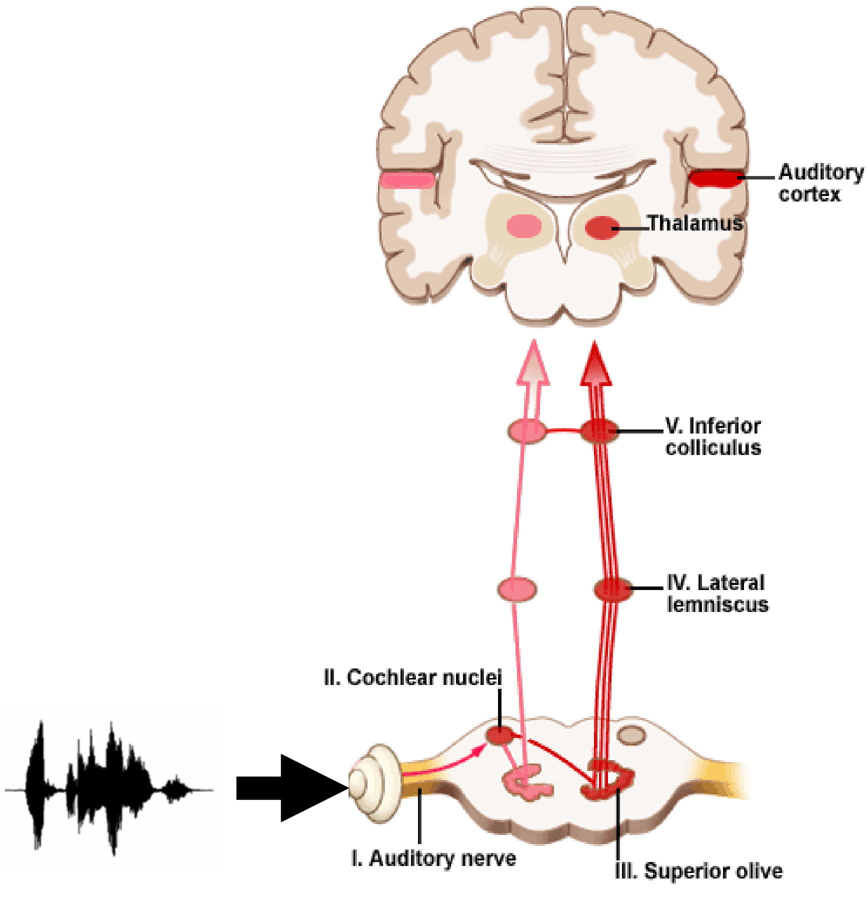
Acoustic Technology, Ørsted•DTU

Technical University of Denmark



Annual Report 2005

ACOUSTIC TECHNOLOGY

Ørsted•DTU

TECHNICAL UNIVERSITY OF DENMARK

ANNUAL REPORT

2005

Edited by Finn Jacobsen in February 2006

Front figure: Spectro-temporal receptive field in the auditory cortex in the brain. Drawing by P. Minary from 'Promenade in the cochlea', www.cochlea.org, EDU website by R. Pujol et al., INSERM and University Montpellier 1. See the description of the project 'Multi-resolution spectro-temporal analysis of complex sounds in the human auditory system' on p. 36.

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CHAIRMAN'S REPORT

The year 2005 again saw increasing activities at Acoustic Technology (AT)—one of the six sections that form the department Ørsted•DTU. The increasing interest in all the areas of acoustics we cover is very welcome. Numerous guest researchers visited us in 2005, and in addition to our Danish students and foreign visiting students on short-term-stay we now 'house' twenty-three foreign master students in our international two-year MSc-programme in Engineering Acoustics. As a result we are supervising and housing an increasing number of MSc-thesis projects for both international and Danish students.

The year saw a further consolidation of the Centre for Applied Hearing Research (CAHR), headed by Torsten Dau. The centre's new research facilities have been used extensively in 2005 by both staff and students. On the staff side there have been some changes; in the late summer Stephan Ewert returned to University of Oldenburg, Germany, and Oliver Fobel left for a position in The Netherlands. To continue their work we welcome Jörg Buchholz (from University of Western Sydney, Australia) and James Harte (from ISVR in Southampton, England).

Because of staff reorganisation in the department Lisbeth Winter has moved to Automation, another section of Ørsted•DTU. We wish Lisbeth all the best in her new 'section environment', and we welcome our new secretary Gunhil Nørhave, and also industrial PhD-student Mikael Odderup Jensen (from the Ørsted•DTU section Electronics and Signal Processing), who will continue his project here at AT supervised by Finn Jacobsen.

At the end of the year Torsten Dau and Thomas Ulrich Christiansen received a 2½-year grant from the Carlsberg Foundation for the project 'Building blocks in spoken Danish—acoustically and perceptually', and Torben Poulsen received DKK 260.000 from Ørsteds Fond and DKK 100.000 from Widex, Oticon and Phonak for a project on 'Recording of natural sounds for hearing aid measurements and fitting'. Other new research projects granted in 2005 include a three-year industrial PhD scholarship to Lars Friis for the project 'Minimisation of vibroacoustic feedback in hearing aids' with Widex A/S as the industrial partner, a PhD scholarship from DTU to Eric Thompson for the project 'Modelling monaural and binaural signal detection in reverberant environments', and a PhD scholarship from the Oticon foundation to Jens Bo Nielsen for a project on enhancement of speech intelligibility. Moreover, Torben Poulsen and Torsten Dau received a grant for a state/company/DTU co-financed doctoral school (SNAK); and in October Olaf Strelcyk began a PhD study on perceptual consequences of impaired auditory signal processing in complex acoustical environments, funded via this SNAK-school with industrial support from Oticon. We also welcome Iris Arweiler who is working with Torben Poulsen and Torsten Dau on the EU-project 'Application to improve auditory communication for the elderly and hearing impaired'.

In the new EU-funded PhD student exchange programme 'European doctorate in sound and vibration studies' (EDSVS 2) we have hosted three visiting PhD students in 2005: Konca Şaher, working on the acoustic quality of living rooms and workspaces in institutions for people with intellectual disabilities, Fabrice Ducret, working on determining properties of micro-perforated plates, and José Escolano-Carrasco, working on the finite difference time domain method for modelling sound fields in enclosures. After her stay formally finished Konca Şaher decided to stay some extra months. We also hosted two PhD students in the (now terminated) first EDSVS project, Dario Paini working on the acoustic influence of arcades on open-air musical performance, and Heli Laitinen, working on musicians' hearing, prevalence of hearing disorders and their effect on their work and hearing protector usage. And a Czech PhD student, Zdenek Havranek, spent four months working with Finn Jacobsen on acoustic holography.

Lily Wang from University of Nebraska, USA, spent one month at AT working with Jens Holger Rindel and Anders Christian Gade on room acoustic models together with her PhD-student Jon Rathsam. In July Xiangyang Zeng returned to Northwestern Polytechnical University, Xi'an, China, after a ten-month sabbatical leave here at AT where he worked on room acoustic auralisation. Guest researcher working with Torsten Dau and colleagues in CAHR include an Australian PhD-student Jess Hartcher-O'Brien investigating the auditory precedence effect, and Steven Greenberg, The Speech Institute, Berkeley, USA, who again spent three months here working with Thomas Ulrich Christiansen on speech research. At the end of the year Xiaojun Shi began a ten-month stay supported by Cirius. He will work on the influence of airflow on sound intensity measurements together with Finn Jacobsen. Other guests in 2005 include three employees from New Jialian Electronic Ltd., China, who spent three month here following two courses and working on loudspeaker development. The collaboration agreement on transducer development between New Jialian Electronic Ltd. and Finn Agerkvist has been extended by yet another year, and also this year Finn Agerkvist gave a series of lectures on electroacoustic systems and transducers at New Jialian Electronic Ltd. in Jiashan, China.

In January 2005 Salvador Barrera-Figueroa, Danish Fundamental Metrology (DFM), began working on a three-year project on diffuse field calibration of microphones, financed by the Danish Research Agency. This project is carried out in collaboration between AT, Danish Primary Laboratory of Acoustics (DPLA) and DFM. At the end of the year DFM took formally over the responsibilities of running DPLA, and the facilities and equipment were moved to DFM's premises in DTU building 307.

In the beginning of the year Jens Holger Rindel received the Rockwool 2005 Prize for his continuing work on improving sound quality in apartment buildings. In November 2005 Torsten Dau was elected Fellow of the Acoustical Society of America.

Our international MSc programme in Engineering Acoustics had 27 applicants in 2005, eleven of whom were accepted. In 2005 the Danish State in cooperation with the companies Oticon and Ødegaard & Danneskiold-Samsøe financially supported two of the MSc-students enrolled in 2003.

Mogens Ohlrich

STAFF

Head of Acoustic Technology
Mogens Ohlrich, BSc, PhD, associate professor

Professor, Head of Centre for Applied Hearing Research
Torsten Dau, MSc, PhD, Dr. Habil

Reader/Professor
Jens Holger Rindel, MSc, PhD

Associate Professors
Finn T. Agerkvist, MSc, PhD
Hans-Heinrich Bothe, MSc, PD, Dr.-Ing. habil.
Anders Christian Gade, MSc, PhD
Finn Jacobsen, MSc, PhD, Dr. Techn.
Torben Poulsen, MSc

Assistant Professors
Jörg Buchholz, MSc, PhD
Stephan D. Ewert, MSc, PhD
Oliver Fobel, MSc, PhD
James Harte, MSc, PhD

Scientists and Research Assistants on External Grants
Iris Arweiler, MSc
Thomas Ulrich Christiansen, MSc, PhD
Salvador Barrera-Figueroa, MSc, PhD (DFM)
Claus Lynge Christensen, MSc (ODEON)
Martin Lisa Nielsen, MSc

PhD Students
Torsten Haaber Leth Elmkjær, MSc (Terma)
Brent C. Kirkwood, BSc, MSc
Allan Larsen, MSc, *Informatics and Mathematical Modelling, DTU*
Jens Bo Nielsen, MSc
Tobias Piechoviak, MSc
Gilles Pigasse, MSc
Olaf Strelcyk, MSc
Eric Thompson, BSc, MSc

Industrial PhD Students
Lars Friis, MSc (Widex)
Mikael Odderup Jensen, MSc (Sonion)
Erik Schmidt, Cand. Mag. (Widex)

Visiting PhD Students
Fabrice Ducret, *Royal Institute of Technology (KTH), Sweden*
José Escolano-Carrasco, *University of Alicante, Spain*
Jess Hartcher-O'Brien, *University of Western Sydney, Australia*
Zdeněk Havránek, *Brno University of Technology, Czech Republic*
Cheol-Ho Jeong, *Korea Advanced Institute of Technology, Korea*
Heli Laitinen, *Helsinki University of Technology, Finland*
Jirí Novácek, *Czech Technical University, Prague, Czech Republic*
Dario Paini, *Politecnico di Milano, Italy*
Jon Rathsam, *University of Nebraska, USA*
Konca Şaher, *Technical University of Delft, The Netherlands*

Visiting Scientists
Steven Greenberg, *The Speech Institute, Berkeley, CA, USA*
Steindor Gudmundsson, *The Icelandic Building Research Institute, Iceland*
Lily M. Wang, *University of Nebraska, USA*
Xiaojun Shi, *China Ship Scientific Research Center, Jiangsu, China*
Xiangyang Zeng, *Northwestern Polytechnical University, Xi'an,China*

Emeritus Professors
Ole Juhl Pedersen, MSc
Knud Rasmussen, MSc

Administrative and Technical Staff
Anne-Dorthe Hansen, Accountant
Gunhil Nørhave, Administrative assistant
Tom A. Petersen, Assistant Engineer
Jørgen Rasmussen, Assistant Engineer
Aage Sonesson, Assistant Engineer
Caroline van Oosterhout, Project secretary
Lisbeth Winter, Trilingual Secretary

1. RESEARCH

Research at Acoustic Technology, Ørsted•DTU, comprises investigations of generation, propagation and effects of sound and vibration, as well as auditory signal processing and perception of sound. The research may involve theoretical analyses, numerical techniques, subjective experiments and advanced measurement techniques. Computer models are used for a variety of purposes. Electro acoustic systems ranging from active hearing protectors to loudspeakers are investigated. The purpose of the research is threefold: i) the development of improved technology for recording and reproduction of sound, and improved acoustic design of rooms; ii) the development of improved solutions for noise abatement indoors and out of doors as well as in vehicles; and iii) the development of state-of-the-art quantitative models of auditory signal processing and perception for advanced technical applications, eg, in audio coding, speech recognition and digital hearing aids. In the following the research projects in 2005 are briefly presented. The projects are divided into three groups according to the main purpose as mentioned above.

1.1 ACOUSTIC COMMUNICATION SYSTEMS

Research in acoustic communication systems deals with the properties of our hearing (psychoacoustics) as well as transducer technology (loudspeakers and microphones). The properties of our hearing and the choice of transducers are fundamental to acoustic communication systems. Furthermore, room acoustics is of crucial importance. Developments in measurement techniques also play an important part in relation to the analysis and design of acoustic communication systems.

Room Acoustics

ODEON, room acoustic calculation model

Claus Lynge Christensen and *Jens Holger Rindel*

The main improvements in 2005 have been focused on the way scattering is handled in the ODEON program. The new methods that have been implemented are the Reflection Based Scattering Method and Oblique Lambert. The benefits of the new methods include better predictions, but they should also make it easier for the user to specify frequency dependent scattering. Other enhancements are support for the Common Loudspeaker Format (www.clfgroup.org), an improved geometry modelling program, and enhanced import facilities for geometries in the DXF format.

The Reflection Based Scattering method is a method that calculates a scattering coefficient that is unique for each reflection. This coefficient, which includes diffraction phenomena, is based on:

- Scattering coefficient provided for the surface, specifying the roughness of the surface
- Incident path length
- Reflected path length
- Dimensions of the surface
- Distance from reflection point to edge of the surface
- Angle of incidence

Even though only one scattering coefficient is entered for each surface, this coefficient is expanded to cover the whole frequency range, using typical frequency functions for scattering coefficients (increasing with frequency) and diffraction (decreasing with frequency). The diffraction part of scattering being handled by ODEON; it is no longer needed to specify different scattering coefficients for surfaces with the same material in order to take into account the size or location of a surface—indeed ODEON will do a better job.

Oblique Lambert is a technique that makes it possible to include frequency depending scattering in the late reflection method of ODEON. Depending on the scattering coefficient the area radiation provided by late secondary sources is tilted towards the specular direction. If the calculated scattering coefficient is zero the secondary source will point in the direction of specular reflection, and if it is one it will point in the normal direction of the surface as a traditional Lambert source. Because this technique will make a part of the Oblique Lambert source point out of the room, a normalisation factor is applied. The factor depends on the angle of tilting, being 1 for 0 degrees and 2 for 90 degrees.

Finite-difference time-domain modelling of sound fields in enclosures
José Escolano-Carrasco
Supervisor: *Finn Jacobsen*

The finite-difference time-domain (FDTD) method is a common method for simulation of sound fields. It provides a simple and accurate solution of initial-boundary value problems. However, most problems in acoustics require frequency dependent boundary conditions and including them in time-domain solutions is not so easy. Most existing solutions to this problem have high computational costs, and stability is not always ensured. In this work, a solution is proposed based into mixing modelling strategies. This involves separate FDTD mesh and boundary conditions (a digital filter representation of reflecting factors), combined in a global solution. This solution is based in the interaction model of different paradigms by means of wave digital filters.

This work is supported by a Marie Curie fellowship from the 'European Doctorate in Sound and Vibration Studies II' project.

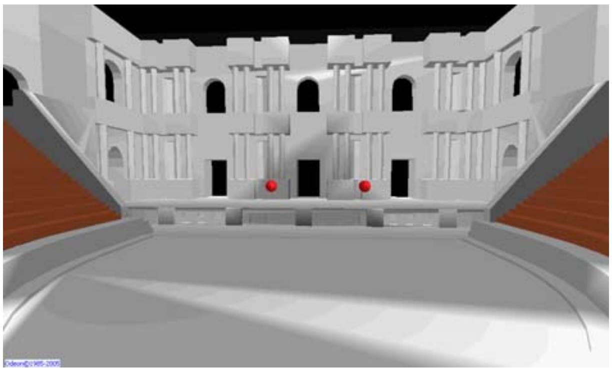
The phased ray tracing method and its applications to room acoustic simulations
Cheol-Ho Jeong
Supervisor: *Jens Holger Rindel*

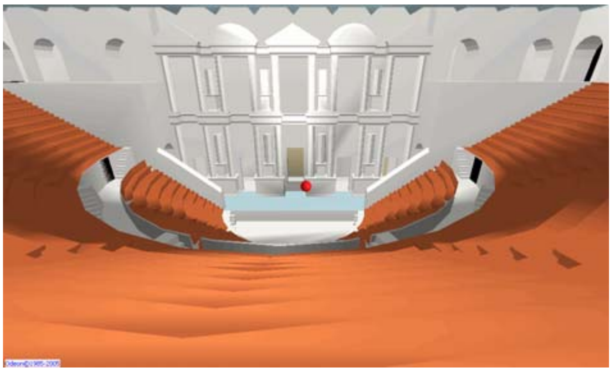
This project is part of a PhD project carried out in Korea. Whereas the classical ray tracing method in room acoustics does not take the phase of the sound into account, this new method improves the accuracy of the computer simulations, especially at low frequencies. Different approximations to the complex surface impedance have been investigated. Although the best results are obtained using the full complex representation, reasonably good approximations can be obtained using a simplified, real impedance model.

Sound propagation in a long, narrow hallway
Jonathan Rathsam, *Lily M. Wang*, *Claus Lynge Christensen* and *Jens Holger Rindel*

The distribution of the sound pressure level down a long, narrow hallway due to a sound source at one end has been measured and simulated. The slope of the spatial decay curve has been satisfactorily simulated using a new scattering method that has been implemented in the ODEON software.

ERATO
Jens Holger Rindel, *Martin Lisa Nielsen*, *Anders Christian Gade* and *Claus Lynge Christensen*





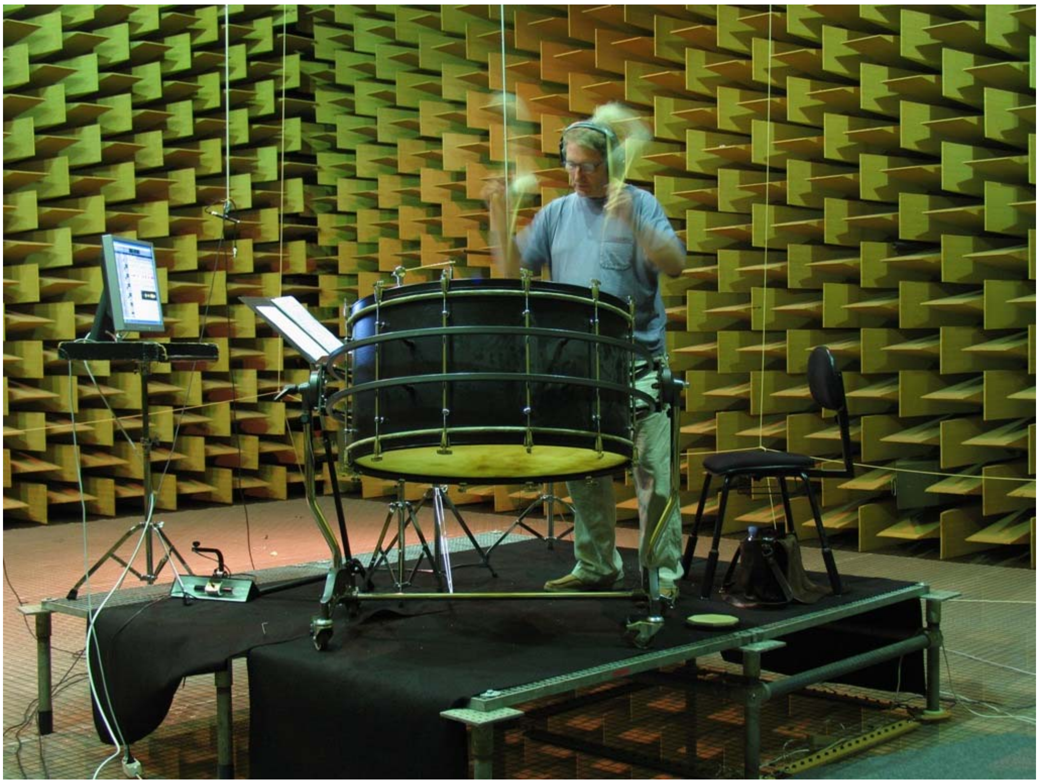
A view into the acoustic computer models of two virtually reconstructed theatres, the South Theatre of Jerash (left) and the Odeon of Aphrodisias (right).

In this project the acoustics of typical performance venues in the Roman and Greek era have been studied in detail using computer simulations. Models have been created of the monuments both as originally built (to the extent that this is known), in their present state (as ruins), and with the temporary stages and scenery often applied when these spaces are used as performance venues today. The models of the theatres have been investigated by changing or removing certain parts of the geometry and observing the impact on the overall acoustics. It has become possible to quantify the acoustic properties of the ancient theatres and odea in each condition, and this has led to valuable information regarding the acoustic consequences and degree of authenticity of permanent restorations and temporary amendments.

Acoustic recordings of music, crowd sounds and actors representing a Greek drama have been carried out in collaboration with Turkish actors and musicians. These recordings have been used in the acoustic computer simulations to produce audible sound demonstrations of virtual performances. The resulting sounds have a high degree of acoustic realism featuring movement of the actors and musicians on the stage, and simulations of an audience of thousands of people. These sound files are being integrated with visual restorations to form the audio in 3D virtual realizations of performances in the selected venues and will result in an interactive DVD.

The project is supported by the EU Commission under the 5th framework with the thematic title 'Preserving and use of cultural heritage'.

Anechoic recordings of the musical instruments in a symphony orchestra
Jens Holger Rindel, *Anders Christian Gade* and *Claus Lynge Christensen*



Recording one of the musical instruments of a symphony orchestra in the large anechoic room at DTU.

In connection with a project carried out by the private company AM3D A/S in Aalborg about 3D sound and the musical instruments of a symphony orchestra, a large number of anechoic recordings have been made. During two weeks in June the members of the Tivoli Symphony Orchestra were recorded one by one in DTU's large anechoic room. In addition to the recordings made by AM3D A/S, AT have also made recordings using a new multi-channel recording technique with four microphones surrounding the source. These recordings may be used with the ODEON program for auralisation of the sound from all the instruments together. This work will continue in 2006.

Methods for modelling sound scattering in a room acoustic computer model
Xiangyang Zeng, *Claus Lynge Christensen*, *Jens Holger Rindel*

A new scattering method for the room acoustics computer model ODEON has been introduced. The scattering effect has been divided into two parts, each of them being frequency dependent: One part is due to the edge diffraction and can be calculated automatically, and the other part is due to surface roughness, and this needs input of information about the surface. The new method and the previous, frequency independent method have been applied in various room models, and the calculated results have been compared with measured values of room acoustic parameters. With the new method the inclusion of geometrical details in the room models tends to improve the accuracy of acoustic predictions.

Psychoacoustics

Multidimensional data analysis
Finn Agerkvist

When analysing multidimensional data such as data from a psychophysical experiment, principal component analysis is often used for data reduction. Principal component analysis gives adequate information on how the data is distributed and in which directions it spreads. However, using the direction with the largest spread does not guarantee the best prediction/detection performance. One of the reasons for this that the direction of the principal axes derived from principal component analysis is very sensitive to how the original data is scaled. The goal of the project is to develop a method that always chooses the best axes for the desired task.

The work is carried out in collaboration with Thomas Sams, Electronics & Signal Processing, Ørsted•DTU.

Evaluation of a binaural speech intelligibility model
Iris Arweiler, *Torben Poulsen* and *Torsten Dau*

This project is part of the European project 'Hearing in the communication society' (HEARCOM). The general focus of HEARCOM is on the identification and characterisation of auditory communication limitations, and on modelling and evaluation of ambient conditions that limit auditory communication in everyday situations. HEARCOM is a consortium of 29 university departments, companies, research institutes, and clinics. The project at DTU investigates binaural speech intelligibility models. So far two models exist: one is based on the speech intelligibility index and the other one on the speech transmission index. To evaluate these models three rooms have been simulated in the 'Odeon' program with different speaker-receiver settings. The Danish sentence test 'Dantale II' will be carried out with normal-hearing and hearing-impaired test subjects in order to measure the speech intelligibility in these different simulated acoustic environments via headphones. Speech shaped noise, multi-talker babble, and a single talker serve as interfering noise. The results of the Dantale II measurements will be compared with the calculated results obtained with the speech intelligibility models. The goal of this topic is to have a unified model available that simulates both the acoustics and expected speech intelligibility for arbitrary rooms.

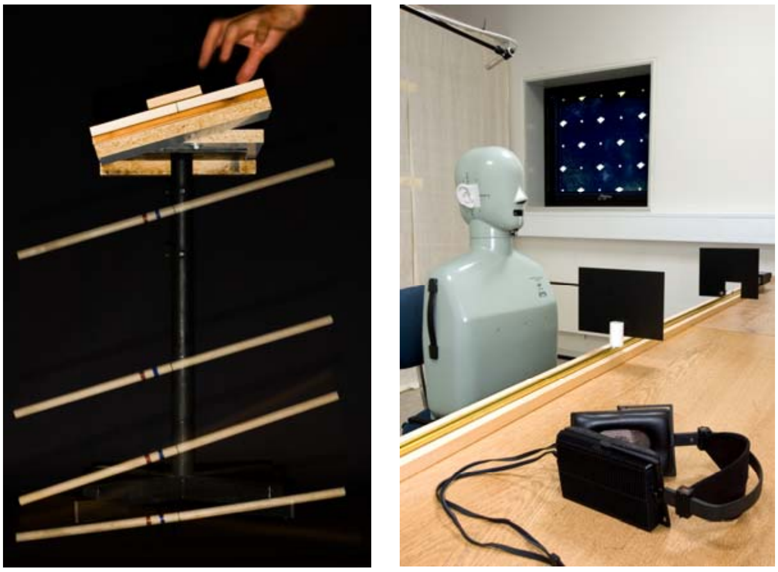
Acoustic description of perceptually salient features in everyday sounds
Brent C. Kirkwood
Supervisor: *Torben Poulsen*

Historically, the study of human hearing has consisted of playing mathematically simple, synthetic sounds to human test subjects who are asked about the sensations they have when hearing the stimuli. This approach has helped us understand parts of the auditory system's physiology and basic functional performance but has done little to provide a description of practical everyday listening. In order to understand how humans use their hearing outside of laboratory environments, it is worthwhile to study hearing using real sounds that listeners would encounter in their daily lives. The goal of this PhD project is to produce new knowledge in this recently emerging field through the execution of ecologically-motivated listening tests. It is hoped to gain an understanding of the human hearing abilities that often exist unnoticed in the presence of visual and haptic sensory systems.

Typical examples of human hearing in the everyday world include the ability to hear the size of objects interacting in a person's environment, the gender or identity of a person conveyed through the sound of their footsteps, and the amount of liquid in a glass that is being filled. A systematic study of these kinds of listening tasks could provide an improved understanding of this mostly unexplored territory, which could then in turn be used to enhance man-machine interfaces, improve simulations (virtual reality, video games, etc), and improve treatment of hearing disorders.

The project is funded by DTU and co-supervised by Graham Naylor, director of research for Oticon A/S.

The influence of presentation method on auditory perception of dropped-rod length
Brent C. Kirkwood



Left-hand picture: dropping apparatus. Right-hand picture: listening station

When designing a listening test, it is an option to present the acoustic stimuli to test subjects via recordings of the stimuli, via synthesised versions of the stimuli, or live. Live presentation of stimuli has the great advantage of bypassing any imperfections in a recording and playback chain and can thereby increase the ecological validity of a listening task (also enhanced in other ways in a live task). However, this ecological validity comes at a cost in the form of a significant reduction in the choices of test method and in the types of stimuli that can be presented.

It has previously been shown that humans are reasonably good at hearing the lengths of wooden dowels dropped onto hard surfaces in the confines of a live listening test. The purpose of this experiment was to investigate the influence of presentation method on test subject performance for this typical everyday listening task. Such information is useful for designing future listening tests, and a comparison of the results can also reveal some of the underlying acoustic structure important for the auditory perception of object size.

Test subjects performed the same rod-length estimation task by listening to stimuli presented in three different ways: (i) live, (ii) headphone (diotic) playback of recordings made from a single microphone, and (iii) headphone playback of recordings made using an acoustic head and torso simulator. Results indicate that normal-hearing listeners perform worse when listening to monaurally recorded stimuli than when listening to stimuli presented live. Inconclusively, subject performance with the binaural-recording presentation method was not statistically different from the live or monophonic presentation methods. It is at least clear, however, that monaurally recorded stimuli should be used with caution.

Hearing Aid Research

Hearing aid processing of loud speech and noise signals—Consequences for loudness perception and listening comfort
Erik Schmidt
Supervisor: *Torben Poulsen*

Hearing aid processing of loud speech and noise signals is the focus of this industrial PhD-project. The project is carried out in co-operation with the Danish hearing aid manufacturer Widex A/S with Carl Ludvigsen as the industrial supervisor.

Modern hearing aids use digital sound processing in several channels, compression, noise reduction, etc., to optimise the reproduction of the signal. The goal is to amplify speech and environmental sounds to the residual audible area of the hearing impaired person such that speech intelligibility and listening comfort is as good as possible.

Many principles for gain setting in hearing aids are based on empirical research, investigating loudness perception and gain-preference of the hearing impaired. Most research has been focusing on speech at average vocal-effort. It is well documented that for speech at this level, hearing aid-users prefer gain to be approximately half the size of the hearing loss. This positions the amplified speech signal in the middle of the user's audible area, at a comfortable listening level. There has been little research on the optimum gain-prescription for loud and soft speech-levels. For the present, such prescriptions are based mainly on logic, as there is limited evidence on eg what compression characteristics are best to be used for these levels.

This project looks into the perceptual effect of different sound processing-strategies for presenting loud speech and sounds to the hearing aid user. From a sound quality-point of view, it make sense to provide the hearing aid user with a sensation of the sound level-variations in the acoustical environment—that is, there should be some degree of level-variation in the output that reflects the overall level variations in the given environment.

In a recent experiment, the perception of level variations in loud speech and noise signals were investigated, using a simulated nonlinear hearing aid. Four different signals with a built-in level-variation of 20 dB were compressed with seven different compression ratios, in the range from 1:1 to 10:1. Test-signals were presented under free field conditions to eight hearing impaired listeners wearing binaural behind-the-ear hearing aids fitted linearly for the individual loss according to the procedure developed by National Acoustic Laboratories in Sydney, Australia. The subjects then rated their perception of the signals on 10-point interval scales in regard to the degree of level-variation, loudness, and acceptance of the reproduced level-difference. Subsequent analysis has shown significant differences in the ratings among signals, and this is assumed to be caused by spectral and temporal differences among signals. This test-method may be used with other types of input-signals (different spectra) and other types of compression systems. Signals with level variation may also be used in the validation of hearing aid-fittings in the clinic.

The project is due to end in May 2006.

Transducers

Compensation of nonlinearities in micro-loudspeakers
Allan Larsen
Supervisor: *Finn Agerkvist*

Micro-loudspeakers are used in mobile phones and headsets. Because of their limited size they are often driven beyond the limit of linear operation. This is especially true for mobile phones which now also serve as mp3 players. Therefore it is more relevant to use compensation techniques on these loudspeakers than on normal hi-fi speaker systems which are rarely driven as hard. The project aims to develop suitable compensation algorithms for microspeakers.

Most of the work is carried out at Informatics and Mathematical Modelling, DTU, with Jan Larsen as the main supervisor.

Error correction of loudspeakers
Finn Agerkvist

The aim of this project is to improve loudspeaker performance. Currently under investigation is the idea of improving the power efficiency and/or reducing the required power by deliberately introducing nonlinearities in the loudspeaker. This idea assumes that the introduced distortion can be removed by applying a suitable compensation algorithm.

Most of the work is carried out by Bo Petersen, PhD student at Esbjerg Institute of Technology, Aalborg University, with Jens Arnsbjerg as the main supervisor.

PUBLICATIONS

Journal Papers

J. Xu, *J.M. Buchholz* and F.R. Fricke: Flat-walled multi-layered anechoic linings: optimization and application. *Journal of the Acoustical Society of America* **118**, 3104-3109, 2005.

F. Otondo and *J.H. Rindel*: A new method for the radiation representation of musical instruments in auralizations. *Acta Acustica united with Acustica* **91**, 902-906, 2005.

Chapters in Books

B.C. Kirkwood: Maintaining realism in auditory length-perception experiments. PhD papers in Technology and Science—Fall 2005: Selected papers from the PhD course 'Writing and Reviewing Scientific Papers', pp. 46-48, Aalborg University, 2005.

J.H. Rindel: Dansk akustik i musikkens tjeneste—kunst eller videnskab? In *Dansk Akustik 1955 - 2005, Jubilæumsskrift* (ISBN 87-990425-0-9), pp. 41-54, ed. O.J. Pedersen, Dansk Akustisk Selskab, 2005.

Conference Papers

J. Xu, *J.M. Buchholz* and F.R. Fricke: Multi-layered polyurethane foams for flat-walled anechoic linings, *VSTech 2005 Symposium*, Hiroshima, Japan, 2005.

C.L. Christensen and *J.H. Rindel*: Predicting acoustics in class rooms. *Proceedings of Inter-Noise 2005*, Rio de Janeiro, Brazil, 2005.

C.L. Christensen and *J.H. Rindel*: A new scattering method that combines roughness and diffraction effects. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

A.C. Gade, *C.L. Christensen*, M.L. Nielsen and *J.H. Rindel*: Matching simulations with measured acoustic data from Roman theatres using the ODEON programme. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

B.C. Kirkwood: 'The influence of presentation method on auditory length perception' *in Studies in Perception and Action VIII: Thirteenth International Conference on Perception and Action*, eds. H. Heft and K.L. Marsh (Lawrence Erlbaum Associates, Mahwah, NJ, 2005).

B.C. Kirkwood: As good as the real thing? Performance differences for live versus recorded stimuli in an everyday listening task. *Proceedings of 21st Danavox Symposium 'Hearing Aid Fitting'* (eds. A.N. Rasmussen and *T. Poulsen*), Kolding, Denmark, 2005.

M.L. Nielsen, *J.H. Rindel*, *C.L. Christensen* and *A.C. Gade*: Matching simulations with measured acoustic data from Roman theatres using the ODEON programme. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

M.L. Nielsen, *J.H. Rindel*, *C.L. Christensen*, *A.C. Gade*: How did the ancient Roman theatres sound? *Forum Acusticum 2005*, Budapest, Hungary, 2005, pp. 2179-2184.

D. Paini, *A.C. Gade* and *J.H. Rindel*: Agorá Acoustics—Effects of arcades on the acoustics of public squares. *Forum Acusticum 2005*, Budapest, Hungary, 2005

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J.H. Rindel and *F. Otondo*: The interaction between room and musical instruments studied by multi-channel auralization. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

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E. Schmidt: Perception of level variations in loud speech and noise signals, processed by a simulated non-linear hearing aid. *Proceedings of 21st Danavox Symposium 'Hearing Aid Fitting'* (eds. A.N. Rasmussen and *T. Poulsen*), Kolding, Denmark, 2005.

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L.M. Wang, *J. Rathsam*, *C.L. Christensen* and *J.H. Rindel*: Sound pressure distribution in a long, narrow hallway: Measurements versus results from a computer model with scattering from surface roughness and diffraction. *150th Meeting of the Acoustical Society of America*, Minneapolis (paper 4aAA3), *Journal of the Acoustical Society of America* **118**, p. 1998, 2005.

1.2 ENVIRONMENTAL ACOUSTICS

Environmental acoustics deals with the attenuation of sound and vibration at the source (by means of damping of vibrations, active noise control or screening), along the propagation path (noise barriers, for example), or at the receiver (attenuation at the façade, hearing protectors, etc.). Computer simulations and acoustic measurement techniques play an important role in noise abatement and vibration control.

Effects of Noise

Musicians' evaluation of their working environment, self reported ear symptoms, and the use of hearing protectors.
Heli Laitinen
Supervisor: *Torben Poulsen*

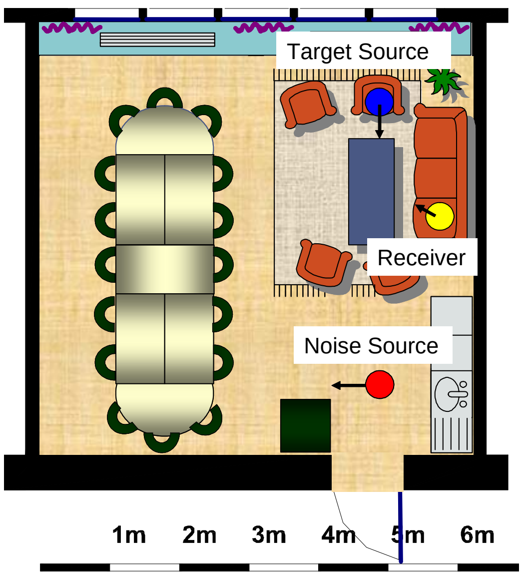
The hearing problems and the hearing protector usage of classical musicians in three Danish symphony orchestras have been studied. The results shows that the use of only one hearing protector is quite common among musicians. This has to be taken into account when the sound exposure of a musician is estimated. The study has investigated the annoyance caused by the occlusion effect for hearing protector users. A paper for publication in *International Journal of Audiology* is under preparation.

Heli Laitinen's research stay, which ended in February 2005, was financed by a Marie Curie fellowship from the 'European Doctorate in Sound and Vibration Studies' project.

Noise from human activity
Torben Poulsen and *Jens Holger Rindel*

AMI, The National Institute of Occupational Health (Copenhagen) is doing a review of 'Noise from human activity', ie non-industrial noise. Typical examples are noise in kindergartens and noise in schools. The noise may or may not be harmful to the hearing. The review shall give an overview of the available literature and the state of the art, and shall point to important research needs. The review is made by a number of researchers from different institutions. Torben Poulsen has a contribution about noise and hearing problems for musicians. Jens Holger Rindel is writing about the problems in open-office environments. The review should be finished in 2006.

Impacts of reverberation time, absorption location and background noise on listening conditions in a multi source environment
Konca Şaher
Supervisor: *Jens Holger Rindel*



The ground floor plan of the room. The floor area is approximately 38 m^2, and the height of the room is 3 m.

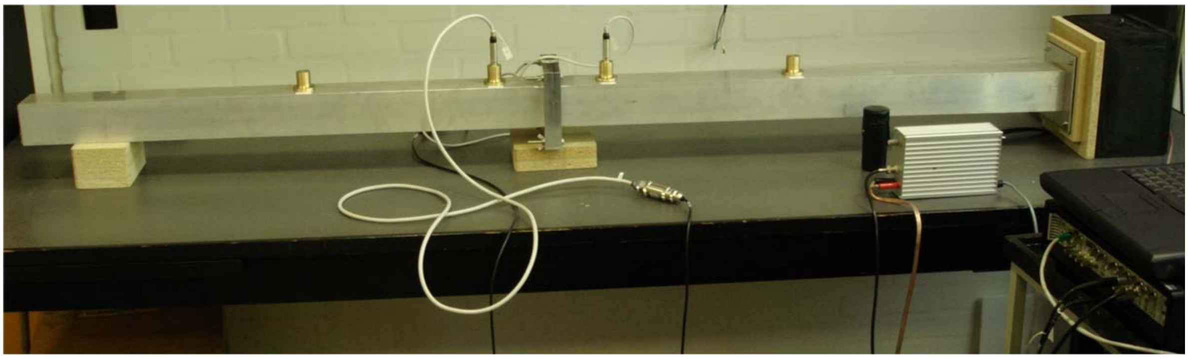
The PhD project at Delft University of Technology focuses on the prediction of the acoustical quality in living rooms and workspaces in institutions for people with intellectual disabilities. A part of this project is carried out at Acoustic Technology, Ørsted•DTU.

One of the main activities in rooms in institutions for people with intellectual disabilities is simultaneous conversations. The speech intelligibility index (STI) has been investigated as an acoustic quality indicator under single and multi-source conditions. The influence on STI of the reverberation time, the distribution of the absorptive materials, and the introduction of screens has been examined. However, these objective parameters should also be assessed through subjective experiments. Auralisations of single and multi source environments have been made with different reverberation time and distribution of absorption with and without screens. Listening tests with test subjects were accomplished to determine just noticeable differences in STI and to investigate STI in a broader range and under multi-source conditions. The results indicate that just noticeable differences in STI obtained from computer calculations by changing the location of the absorptive surface are verified by the subjects. When investigating STI the judgment of reverberation, the loudness and type of noise source and its impact on speech intelligibility, listening effort and noise distraction, are central issues. A lower reverberation time gives better speech intelligibility, and the introduction of a screen is an effective measure to obtain a lower noise level. However, the correlation between objective and subjective parameters shows some variance. The type and content of the noise source have an important impact on the subjective perception of a noise source. For all calculations the room acoustic ODEON model has been used.

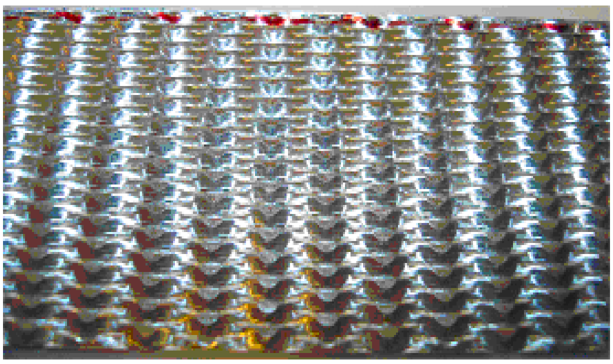
This research has been carried out in cooperation with Lau Nijs, Delft University of Technology. The project is funded by a Marie Curie grant from 'European Doctorate in Sound and Vibration Studies II'.

Noise Control

Development of micro-perforated acoustic elements for vehicle applications
Fabrice Ducret
Supervisor: *Finn Jacobsen*



Test rig for measurement of the flow impedance.



Micro-perforated sample produced by the Swedish company Sontech.

There is a growing interest in the automotive industry in micro-perforated acoustic elements for reducing noise radiation from engine systems. The traditional means of acoustic attenuation through porous materials has some serious drawbacks, including poor efficiency at low frequencies, and low capability to withstand flow and the thermal conditions commonly encountered in engine systems. Novel elements consisting of plates with orifices of submillimetre dimensions are currently investigated. An indicator of paramount importance when designing such elements is the acoustic flow impedance (the ratio of the pressure drop across the sample to the acoustic volume velocity through the holes). A detailed description of the impedance of these elements is necessary for evaluating the efficiency of reflection of acoustic waves. This is required for reducing the propagation of noise further downstream. A concise fundamental theoretical investigation for optimal design is not possible because of the highly complex geometries often found in micro-perforated patterns. Moreover, mean flow, thermal and nonlinear effects come into play for vehicle applications.

The work is funded by a Marie Curie grant from the project 'European Doctorate in Sound and Vibration Studies II'.

Active noise cancellation headsets

Torsten H. Leth Elmkjær

Supervisor: *Finn Jacobsen*

Terma A/S, Acoustic Technology, Ørsted•DTU, and the Engineering College of Aarhus have continued a joint research project on active noise reduction (ANR). Terma is involved in the design of the next generation of pilots' helmets. By wearing a helmet with embedded ANR the pilot will expose himself to less low frequency noise.

In 2005 the work has concentrated on developing algorithms capable of handling non-Gaussian 'heavy-tailed' impulsive signals. In particular the symmetric-stable distribution has been used to model heavy-tailed phenomena encountered in communication, underwater acoustics and radar. Most previous work in the field of active control uses some of the numerous variants of least mean square (LMS) algorithm extended with the filtered-*x* scheme. However, the heavy tailed signals have no finite second order moments. Hence, the filter should be based on *lp*-norm optimisation, and thus a new adaptive algorithm, the regularised normalised least mean *p*-norm algorithm, has been developed. The filter performance was found to be only marginally degraded compared with the standard LMS algorithm for ordinary time-invariant Gaussian noise signals. A part of this work has been carried out in cooperation with Thayer's School of Engineering, Dartmouth College, NH, USA.

Co-supervisor at the Engineering College of Aarhus: Svend-Olof Sjöström. The project is due to end in May 2006.

Estimation of airborne sound insulation between rooms

Jirí Novácek

Supervisor: *Jens Holger Rindel*

This project is part of a PhD project carried out in the Czech Republic. The calculation model described in EN 12354-1 has been compared with a number of other theoretical prediction models for the airborne sound insulation using material data from a database of Czech building constructions. Both single and double constructions have been studied.

ISO/TC 43/SC 1/WG17

Torben Poulsen

Torben Poulsen is convener of Working Group 17, 'Methods of measurement of sound attenuation of hearing protectors'. The working group met in connection with the ISO plenary meeting in Toronto, Canada, in June 2005. The main topic of the WG meeting was to include a section about the uncertainty of the measurements in the draft standards (CD 4869-3 and DTS 4869-5). The titles of these documents are: 'Acoustics—Hearing protectors—Part 3: Simplified method for the measurement of insertion loss of ear-muff type protectors for quality inspection purposes' and 'Acoustics—Hearing protectors—Part 5: Method for estimation of noise reduction using fitting by inexperienced test subjects'.

Measurement Techniques

Higher order structure tensor and symmetries

Finn Agerkvist

In image analysis the structure tensor is often used to detect lines and corners. It operates by analysing the gradient field of the image, exploiting that the line is symmetric, ie, invariant to a rotation of 180 degrees. In this project the complex valued generalised structure tensor is used for detecting higher order objects such as triangles and rectangles or textures composed of these elements. The advantage of this model compared with a Fourier method is that the structure tensor based methods also can handle aperiodic textures.

The work has been carried out in collaboration with Thomas Sams, Electronics & Signal Processing, Ørsted•DTU.

Noise reduction in images by anisotropic diffusion

Finn Agerkvist

Random noise in images can be reduced by local averaging, ie, by lowpass filtering. However this simple method also removes the finer details of the image. If instead the averaging is done over pixels that have the same intensity, ie, if the averaging is done along a line perpendicular to the local gradient, the structure of the image is preserved. In this project such diffusion filtering algorithms are investigated with focus on computational speed and stability.

The work has been carried out in collaboration with Thomas Sams, Danish Defence Research Establishment.

Near field acoustic holography with double layer array processing

Zdeněk Havránek

Supervisor: *Finn Jacobsen*

Near field acoustic holography (NAH) is an experimental technique for analysing sound fields near sources. A simulation study has been carried out and the investigation compares i) classical planar NAH based on spatial transforms (single layer NAH), ii) a similar technique based on measurements in two parallel planes (double layer NAH), and iii) a method that avoids spatial transforms (and thus leakage) by using statistical considerations (statistically optimal NAH, SONAH). The source is a point driven panel that generates a complicated sound field. The influence of reflections and background noise from other sources is also examined. Preliminary results seem to indicate better accuracy in localisation and characterisation of a sound source with double layer array processing in presence of other disturbing sources. The simulation of the influence of transducer errors on the global accuracy of the NAH calculation indicates very similar effects (additional errors) in single and double layer arrays.

Near field acoustic holography based on particle velocity measurements

Finn Jacobsen

Near field acoustic holography is usually based on measurement of the sound pressure. This investigation has examined an alternative technique that involves measuring the normal component of the acoustic particle velocity. An initial simulation study showed 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 wavenumber ratio (k_z/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 wavenumber ratio (k/k_z). 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 have been confirmed by an experimental investigation made with a *p-u* sound intensity probe produced by Microflown.

Most of the practical part of this work was carried out by Yang Liu in his MSc project; see section 2.3.

Sound power determination with two intensity techniques under adverse sound field conditions
Finn Jacobsen

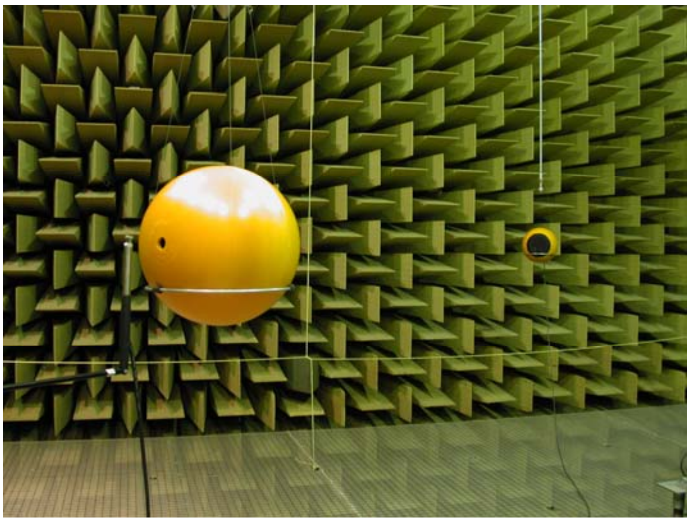
There are two methods of measuring sound intensity. It is well known that the accuracy of sound power measurements with the established method based on two closely spaced pressure microphones (the '*p-p* method') deteriorates when the pressure-intensity index takes high values, which occurs when there is strong background noise from sources outside the measurement surface. It is also known that the accuracy of sound power measurements with the alternative measurement principle based on combining a pressure microphone with a particle velocity transducer (the '*p-u* method') does not depend directly on the pressure-intensity index, but instead on the ratio of the reactive to the active intensity. This might perhaps make the *p-u* measurement principle more accurate under such difficult conditions, because extraneous noise does *not* increase the reactivity, although this potential advantage of *p-u* intensity measurement systems never has been demonstrated experimentally. This investigation has examined the matter. A better performance of the *p-u* measurement system was *not* observed, but the results indicated that the performance of a well-calibrated *p-u* sound intensity measurement system is similar to the performance of a *p-p* sound intensity measurement system of class 1.

The work was carried out in cooperation with Hans-Elias de Bree, Microflown Technologies, The Netherlands.

Calibration of pressure-velocity sound intensity probes
Finn Jacobsen

Various methods of calibrating sound intensity probes based on the combination of pressure and particle velocity transducers have been 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 has been examined both in an anechoic room and in various ordinary rooms. The results showed that whereas reflections from the edges from a plane baffle disturb the calibration, the method based on a spherical baffle gives acceptable results from 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 between 5 and 10 cm.

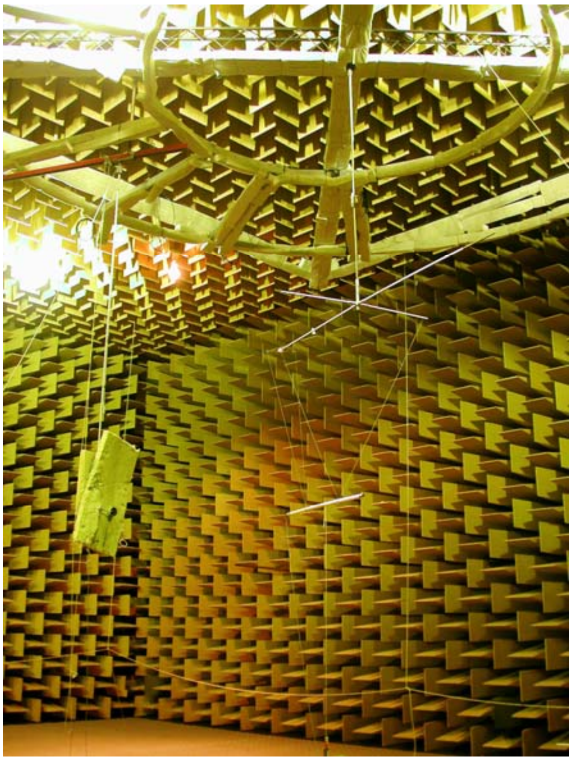
Most of the practical part of this work has been carried out by MSc student Virginie Jaud in a special course.



A Microflown p-u intensity probe 10 cm from a 'monopole on a sphere' in the anechoic room. In the background a KEF 'coincident source' loudspeaker mounted in a sphere can be seen.

Microphone Calibration

Diffuse-field calibration of microphones
Salvador Barrera-Figueroa, *Knud Rasmussen* and *Finn Jacobsen*



Setup for measuring the free-field correction as a function of angle of incidence in the anechoic room.

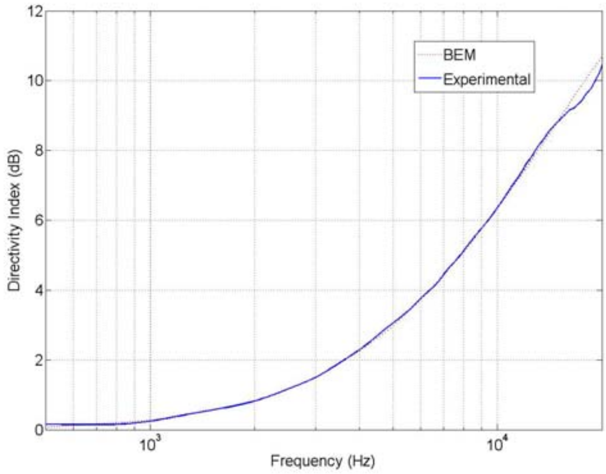
The sensitivity of a microphone depends significantly on the type of the sound field in which it is immersed. The diffuse-field sensitivity is in practice calculated from the measured pressure sensitivity using empirical corrections that depend on the type and geometry of the microphone. The purpose of this project is to develop a method for reciprocity calibration of condenser microphones under diffuse-field conditions, and to demonstrate that such calibrations are possible with high precision.

During the first stage of the project alternative methods have been implemented in order to obtain a reference that can be compared with the diffuse-field sensitivity obtained by reciprocity. These alternative methods are diffuse-field corrections determined experimentally from free-field measurements at different angles of sound incidence, also known as random-incidence sensitivity, and similar numerical corrections calculated using the boundary element method (BEM).

In the first case, the measurement jig was designed and mounted inside the large anechoic room, and the sensitivity of a number of LS1 and LS2 microphones was measured using different angular steps. Because of unwanted reflections from the measurement setup, a time selective technique was applied in order to eliminate these unwanted perturbations. The result is a smooth estimate of the directivity index which can be used to determine the random incidence sensitivity of the microphones.

The numerical estimate has been obtained using an axisymmetrical BEM formulation. The use of this formulation simplifies the problem of integrating the sound pressure over the diaphragm of the microphone. However, it does not take account of the complex interactions of the air film between the diaphragm and the back plate and back cavity within the microphone. Nevertheless there seems to be a good agreement between experimental and numerical results.

The figure below shows the directivity index of LS1 microphone estimated experimentally and numerically.



Comparison of experimental and calculated microphone directivity index of microphones of type LS1.

Danish Primary Laboratory of Acoustics

Knud Rasmussen

Danish Primary Laboratory of Acoustics (DPLA) was established in 1989 by the Agency for Development of Trade and Industry as a cooperation between Brüel & Kjær Sound and Vibration and Acoustic Technology, DTU. DPLA is responsible for maintaining and disseminating the basic unit of sound pressure (the pascal) and acceleration (m/s^2) in Denmark. The associated research and international cooperation is mainly performed by Acoustic Technology. International cooperation is accomplished through meetings within EUROMET, the International Electrotechnical Commission (IEC) and Consultative Committee for Acoustics, Ultrasound and Vibration (CCAUV), and by participating in common projects organised by these bodies.

At the end of 2005 DPLA's responsibilities and activities were transferred from Acoustic Technology to Danish Fundamental Metrology, located in building 307 on the DTU Campus.

Free-field reciprocity calibration of LS1/LS2 microphones

Knud Rasmussen and *Salvador Barrera-Figueroa*

Calibration of a number of microphones using the time-selective free-field reciprocity calibration technique described in Salvador Barrera-Figueroa's PhD thesis from 2003 revealed a systematic difference in the mid-frequency range relative to numerical calculations of the free-field sensitivities. By definition the free-field sensitivity refers to a plane progressive wave. However, the practical implementation of a reciprocity calibration can only be performed in a spherical sound field, which leads to a systematic difference from the 'true' free-field sensitivity. A paper describing this problem is under preparation, and a paper was presented at the Twelfth International Congress on Sound and Vibration in Lisbon, Portugal.

CCAUV.A-K3 Key Comparison

Knud Rasmussen

DPLA has undertaken the role as pilot laboratory for an international key comparison on pressure calibration of type LS2 microphones organised under the auspices of CCAUV.[1] Such Key Comparisons are arranged in order to demonstrate and verify traceability on metrological units, in this case the unit for sound pressure, pascal. The calibrations are performed in the frequency range from 20 Hz to 25 kHz. National metrology institutes from fifteen countries all over the world took part in the project. The calibrations were finished by the end of October 2003 and a draft report was presented to the Plenary Meeting of CCAUV in September 2004.

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[1] Consultative Committee for Acoustics, Ultrasound and Vibration under BIPM (Bureau International des Poids et Mesures) in Paris.

During 2005 all participants have recalculated their results using a common set of microphone parameters in order to eliminate this part of the uncertainty. A final report will be issued in 2006. The analysis and the reporting are mainly performed by Vicente Cutanda, now at the University of Vigo, Spain.

CCAUV.A-K2 Key Comparison
Knud Rasmussen and *Salvador Barrera-Figueroa*

An international Key Comparison on low frequency calibration of type LS1 microphones in the frequency range from 2 Hz to 1 kHz has been conducted in 2005. Ten countries (Austria, Denmark, Germany, Great Britain, Japan, Korea, Russia, Spain, Turkey and USA) took part in this comparison. The very first calibrations were performed by DPLA in January 2005.

The Austrian Bundesamt für Eich- und Vermessungswesen (BEV) acts as pilot laboratory and a first draft report is expected in early 2006.

EUROMET.A-K3 Key Comparison, Project 674
Knud Rasmussen

The number of participants from each geographical region is limited in the above CCAUV.A-K3 Key Comparison. An additional nine countries in Europe wanted to participate, and as a consequence a regional Key Comparison was arranged within Europe following the international CCAUV Key Comparison. DPLA also acted as pilot laboratory for this work, and the microphones circulated belong to one of the sets used for the CCAUV comparison. The calibrations were finished by the end of May 2004. The analysis of the results and the final report is prepared by IEN, Italy. A draft report has been prepared but cannot be finalised until the report from the above CCAUV.A-K3 Key Comparison has been approved. However, preliminary results indicate (not surprisingly) larger deviations than obtained by the major laboratories that took part in the CCAUV Key Comparison.

COOMET.A-K3 Key Comparison
Knud Rasmussen and *Salvador Barrera-Figueroa*

The number of participants from each geographical region is limited in the above CCAUV.A-K3 Key Comparison. An additional four countries in the Euro-Asian region wanted to participate, and as a consequence a regional Key Comparison has been arranged between four countries (Poland, Romania, Russia and Ukraine) following the international CCAUV Key Comparison.

DPLA has undertaken the role as pilot laboratory for this regional key comparison on pressure calibration of type LS2 microphones organised under the auspices of COOMET.[2] The calibrations will be finished in the first quarter of 2006.

Structureborne Sound

Minimisation of vibroacoustic feedback in hearing aids
Lars Friis
Supervisors: *Mogens Ohlrich* and *Finn Jacobsen*

There are many requirements on modern hearing aids regarding aesthetics and structural design. This has resulted in hearing aid constructions where the loudspeaker and the microphones are very close. As a consequence of this, problems with the vibro-acoustic transmission from the loudspeaker to the microphones often arise. Vibrations and sound pressure is picked up by the microphones and an unwanted electric loop is formed. This phenomenon, which at certain critical gain levels in the hearing aid causes an uncomfortable howling sound, is called *feedback.* So far it has been attempted to minimise the feedback by use of knowledge based on practical experience and simple modelling of the vibro-acoustic transmission paths. However, this has not led to any major structural or scientific advances.

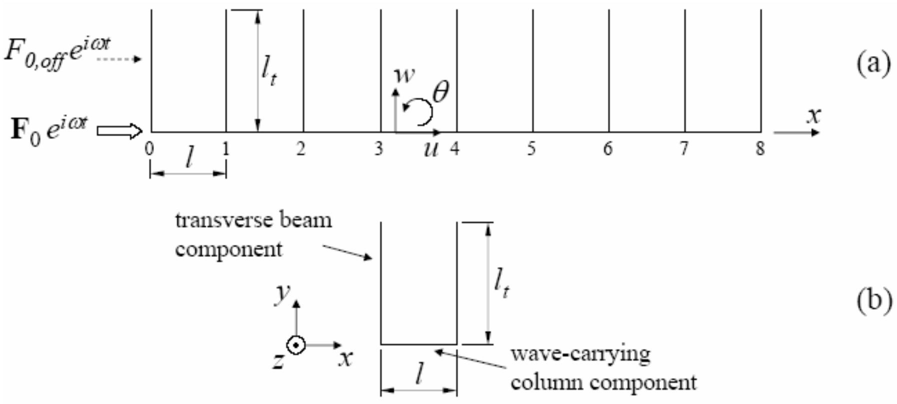
[2] Coopération Métrologique (Euro-Asian Cooperation of National Metrological Institutions)

The objective of this PhD project is to describe the vibro-acoustic transmission paths from loud-speaker to microphones in order to minimise the feedback. As a starting point a specific troublesome hearing aid design is investigated. The goal of the investigation is to obtain knowledge and simulation tools that can improve the design during the development phase. In order to examine the vibro-acoustic transmission paths of the hearing aid, classic and new modelling techniques, such as 'Mobility analysis' and 'Fuzzy Structure Theory' will be applied. Furthermore, combined electrical and vibro-acoustic measurements will be performed in order to investigate the feedback phenomenon and to determine the dynamic properties of the individual hearing aid components.

The project is an industrial PhD project carried out in cooperation with the Danish hearing aid manufacturer Widex A/S, with Lars Baekgaard Jensen as the industrial supervisor.

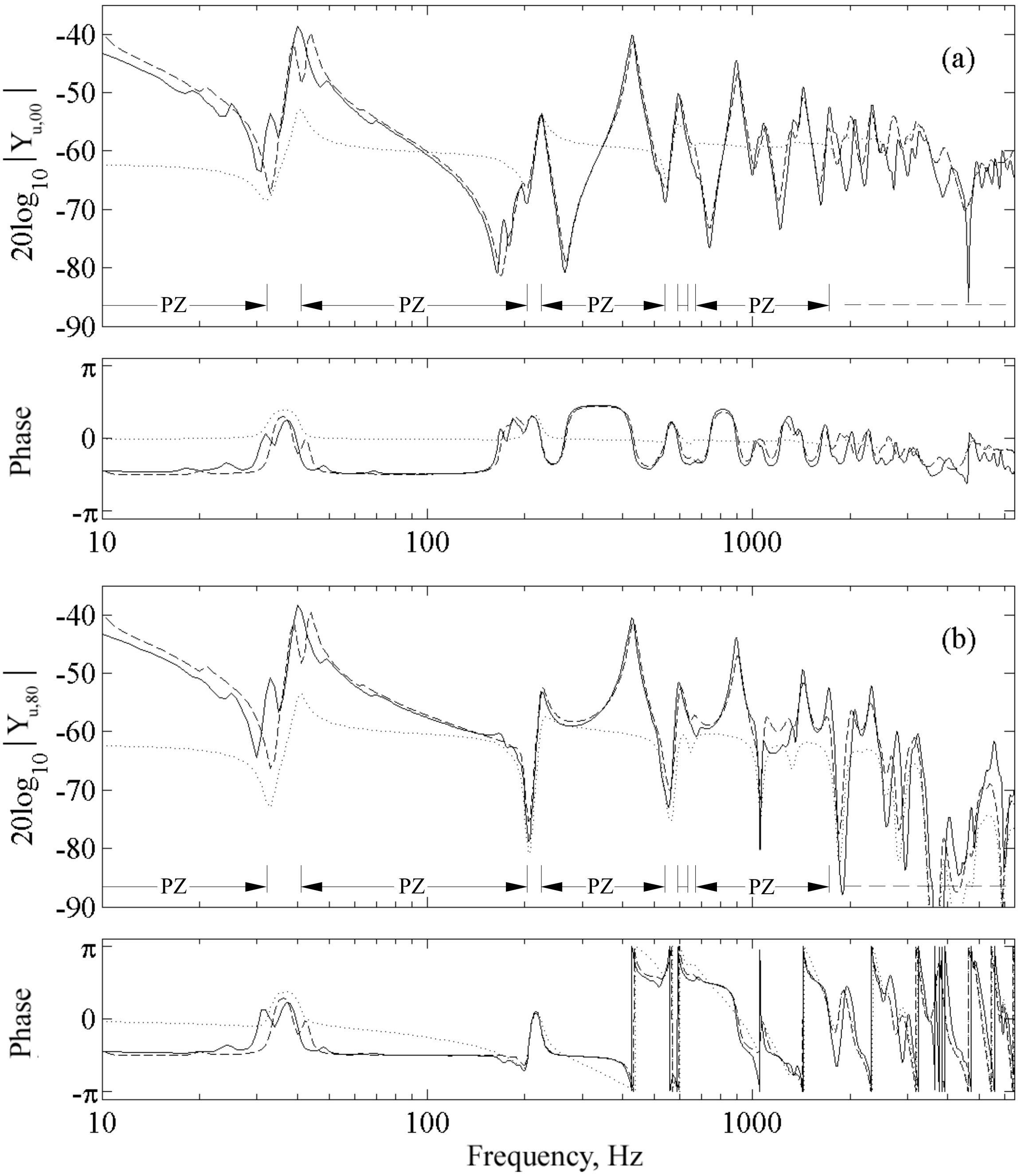
Coupled longitudinal-flexural wave motion in finite periodic wave-guides
Lars Friis and *Mogens Ohlrich*

This project investigates further the coupling of flexural and longitudinal wave motions in structural wave-guides with side-branches attached *asymmetrically* at regular intervals. Wave transmission is considered in a tri-coupled, periodic assembly of beam-type elements (or plane wave transmission for normal incidence in a similar plate assembly). An understanding of wave coupling in such spatially periodic transmission paths is of considerable practical interest, for example in the prediction of vibration levels and structure-borne sound transmission in web-stiffened panels, in ship hulls that have deck structures to one side only, and in outer supporting column structures in building skeletons.



Periodic structure with asymmetrical side-branches in the form of transverse beam elements attached to a long beam or column.

The effect of structural damping on wave coupling and vibration responses in such a semi-infinite wave-guide was first examined, and the damping-dependent occurrence of decoupled wave motions was revealed for a structure with multi-resonant side-branches. In contrast to an equivalent but uncoupled symmetrical system it was found that the flexural-longitudinal wave coupling in a system with resonant side-branches results in a highly enhanced wave transmission with very little attenuation from element to element. Next, the simplifying semi-infinite assumption was relaxed, and general expressions for the junction responses of finite and multi-coupled periodic systems were derived as a generalisation of the governing expressions for finite, mono-coupled periodic systems [Ohlrich, *J. Sound Vib.* **107**, 411-434 (1986)]. The new derivation of the general frequency response of a finite system utilises the eigenvectors of displacement responses and wave forces associated with the characteristic wave-types that can exist in a multi-coupled periodic system. A freely suspended, specific test-structure with eight periodic elements and with structural terminations at the extreme ends was also considered. Audio-frequency vibration responses of this tri-coupled periodic structure were predicted numerically over a broad range of frequencies, and excellent agreement was found with results obtained from an experimental investigation with a nominally identical, periodic test-structure.

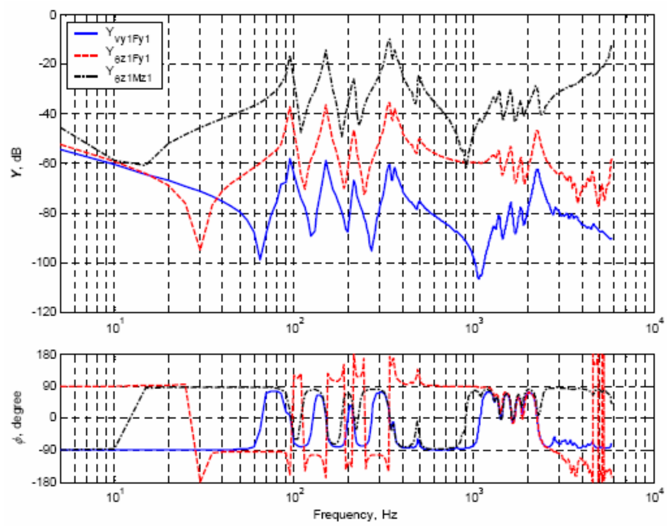


Longitudinal junction mobility of finite periodic structure with eight elements; the system is driven at junction no. 0 by an axial (longitudinal) point force F_0. ——, Measurements; – – –, numerical prediction; ⋯⋯, numerical prediction for a corresponding semi-infinite periodic structure. (a) Direct mobility $Y_{u,\,00} = \dot{u}_0 / F_0$, and (b) transfer mobility $Y_{u,\,80} = \dot{u}_8 / F_0$. (Note: frequency bands denoted by 'PZ' identify wave propagation zones.)

Vibratory strength of machinery sources and structural power transmission

Mogens Ohlrich

This ongoing work focuses on a practical characterisation of vibrating machinery sources and on suitably simple techniques for estimating the vibratory power that such sources inject into connected receiving structures, eg foundations. Because of the high complexity of most machinery structures and their inherent excitation mechanisms, the vibrational source strengths generally have to be determined from measurements of source feet vibration velocities and corresponding feet mobilities. However, in certain cases estimates of source mobilities in the audible frequency range may be obtained by using available finite element software, eg ANSYS. An example of this is illustrated in the figure below, which show calculations of the local point and transfer mobilities at the foot of a two-dimensional model of a machine; these results are from a Master project by Waseim Alfred, see Section 2.3. The effect of cross coupling on power transmission in terminal coordinates was also examined in this project, although for a simpler beam source-receiver case. Practical receiving structures are also very complex, and their input mobilities—and especially the damping controlled real parts—often have to be measured if reliable estimates of the power transmission are to be made. The developed source characterisation, which specifies the 'terminal power' of a vibrating source, is incorporated in the project 'VibPower' as the chosen technique to be used for estimating vibratory power transmission from machinery. This project is described in the following.



Modulus and phase of the calculated point and cross mobilities at the first coupling point of a two-dimensional machine source model; see Section 2.3.

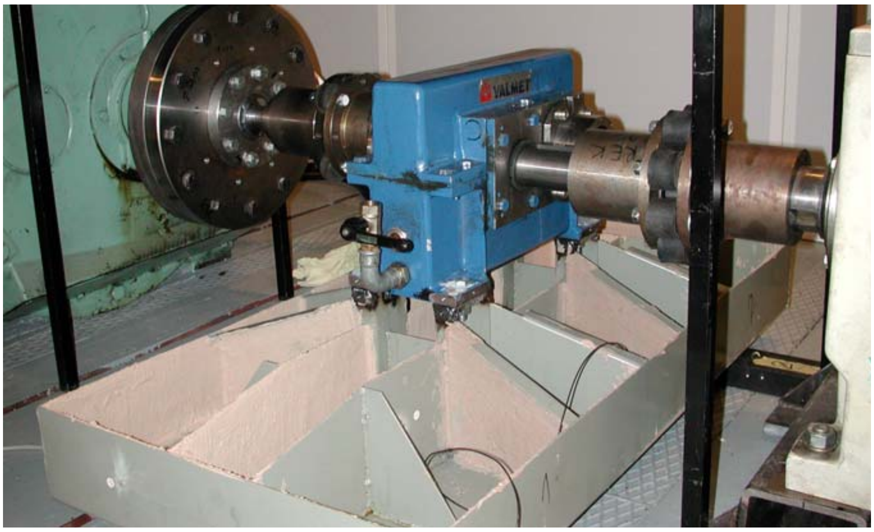
Determination of structureborne sound power and reduction of vibrational power transmission (VibPower): Round Robin Test

Mogens Ohlrich and *Lars Friis*

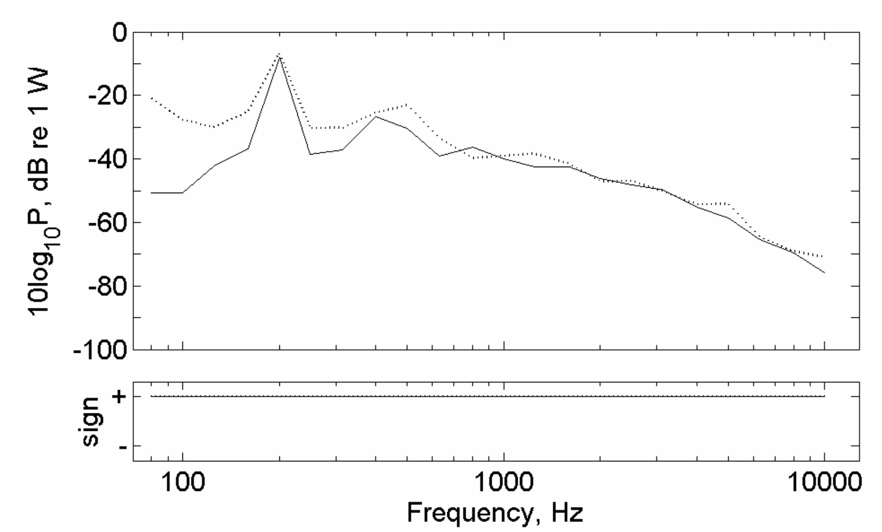
A cooperative research-project initiated by VTT Industrial Systems, Finland, was started in 2004 with the purpose of developing practical design guides for reducing transmission of structureborne sound from large machinery to foundation structures. The two main objectives of this research project are to develop prediction and mobility measurement techniques to be applied to machines and foundation structures, and to develop experimental methods for determining audio-frequency transmission of structureborne sound power to the machine environment.

Based upon previous research at Acoustic Technology (AT), test specifications were developed for source/receiver test arrangements, measurements and post processing required in a Round Robin Test of AT's source-strength measurement techniques. These specifications and associated 'recipes' for data processing methods were applied successfully in March 2005 in a Round Robin Test of an industrial gearbox source and foundation arrangement at the laboratory of Machine Design, Tampere University of Technology (TUT), Finland. AT's test results show that the structureborne sound power transmitted from the relatively compact test gearbox to a receiving foundation structure can be predicted (from the measured data) with a reasonably good accuracy in a broad frequency range that stretches from about 300 Hz to 13 kHz. Comparison of measurement results from the tests by TUT and VTT will be done in 2006.

The research institutes participating in the project are VTT, TUT, AT; and the industrial partners are ABB Industry OY, Wärtsilä Marine Finland OY, and Moventas OY, all from Finland. The National Technology Agency of Finland finances the research with contributions from the three participating companies.



Mounting and test arrangement for rigidly connected gearbox on stiffened foundation plate-structure; low speed drive shaft is on the right hand side and high speed shaft is on the left.



Structureborne sound power transmitted from operating gearbox to foundation structure for a test condition of 10.26 kW shaft mechanical power at 700 rpm. ——, Calibrated far-field reference for transmitted power; ········, predicted power transmission from measured source strengths and system mobilities.

Determination of structureborne sound power and reduction of vibrational power transmission (VibPower): Cross-coupling in foundation structure

Mogens Ohlrich and *Lars Friis*

In this work an analytical and numerical model of a finite, line-type foundation structure is developed with the purpose of examining the effect of cross-coupling on structureborne sound power injected to a receiving structure. For this purpose periodic structure theory described above is used and developed for determining the transmission properties and wave motion in a comb-like receiving structure. Allowing for three motion-coordinates this tri-coupled structure experiences flexural-longitudinal wave coupling. Results have been computed for transfer mobilities of a finite periodic receiving structure with variable boundary conditions. These results are furthermore used for examining the effect of cross-coupling on the power injection for cases of 'distributed' excitation by three point forces with variable phases.

PUBLICATIONS

Journal Papers

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B. Fenech and *F. Jacobsen*: Predicting the eigenmodes of a cavity containing an array of circular pipes. *Journal of the Acoustical Society of America* **117**, 63-67, 2005.

F. Jacobsen and H.-E. de Bree: A comparison of two different sound intensity measurement principles. *Journal of the Acoustical Society of America* **118**, 1510-1517, 2005.

Y. Liu and *F. Jacobsen*: Measurement of absorption with a *p-u* probe in an impedance tube. *Journal of the Acoustical Society of America* **118**, 2117-2120, 2005.

F. Jacobsen and Y. Liu: Near field acoustic holography with particle velocity transducers. *Journal of the Acoustical Society of America* **118**, 3139-3144, 2005.

Chapters in Books

T. Poulsen: Høreværn—fra vattotter til aktive høreværn. In *Dansk Akustik 1955 - 2005, Jubilæumsskrift* (ISBN 87-990425-0-9), pp. 116-126, ed. O.J. Pedersen, Dansk Akustisk Selskab, 2005.

Conference Papers

S. Barrera-Figueroa, *K. Rasmussen* and *F. Jacobsen*: The free-field correction of laboratory standard microphones. *Proceedings of Twelfth International Congress on Sound and Vibration*, Lisbon, Portugal, 2005.

F. Ducret, *F. Jacobsen* and M. Åbom: Development of micro-perforated acoustic elements for vehicle applications. *Proceedings of Twelfth International Congress on Sound and Vibration*, Lisbon, Portugal, 2005.

F. Jacobsen and H.-E. De Bree: Measurement of sound intensity: *p-u* probes versus *p-p* probes. *Proceedings of Noise and Vibration Emerging Methods 2005*, Saint Raphaël, France, 2005.

F. Jacobsen: A note on the uncertainty in measurement of sound power using sound intensity. *Symposium on Managing Uncertainty in Noise Measurement and Prediction*, Le Mans, France, 2005.

F. Jacobsen and H.-E. De Bree: Intensity-based sound power determination under adverse sound field conditions: *p-p* probes versus *p-u* probes. *Proceedings of Twelfth International Congress on Sound and Vibration*, Lisbon, Portugal, 2005.

F. Jacobsen and Y. Liu: Near field acoustic holography based on an array of particle velocity sensors. *Proceedings of Inter-Noise 2005*, Rio de Janeiro, Brazil, 2005.

G. Wendelboe, *F. Jacobsen* and J.M. Bell: Localization of seabed domains ensonified by object-reflected sound at very high frequencies. *Symposium on Boundary Influences in High Frequency, Shallow Water Acoustics*, Bath, UK, 2005.

B. Rasmussen and *J.H. Rindel*: Concepts for evaluation of sound insulation of dwellings. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

K. Şaher, *J.H. Rindel*, L. Nijs, L. and M van der Voorden: Impacts of reverberation time, absorption location and background noise on listening conditions in multi source environment. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

Technical Reports

M. Ohlrich: Specification of Round Robin Test of structureborne sound source strength. AT VibPower Report no. 1-05, 6 pp. (Acoustic Technology, Ørsted•DTU, 2005.)

M. Ohlrich: Final specification of Round Robin Test of structureborne sound source strength. AT VibPower Report no. 2-05, 7 pp. (Acoustic Technology, Ørsted•DTU, 2005.)

M. Ohlrich: Post processing of data for Round Robin Test of structureborne sound source strength. AT VibPower Report no. 3-05, 7 pp. (Acoustic Technology, Ørsted•DTU, 2005.)

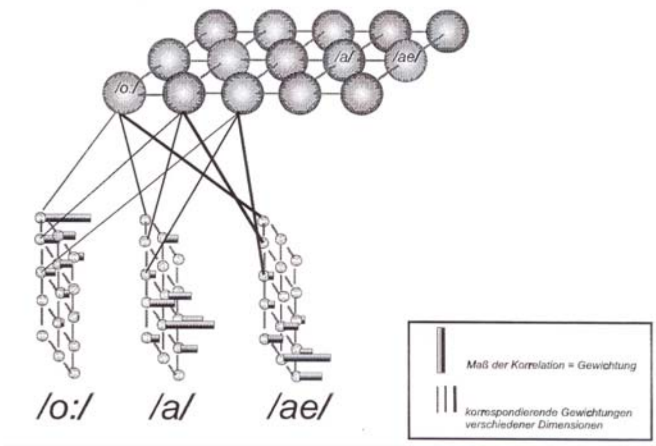
M. Ohlrich and *L. Friis*: Structureborne sound source strength and prediction of power transmission: Results from Round Robin Test. AT VibPower Report no. 4-05, 65 pp. (Acoustic Technology, Ørsted•DTU, 2005.)

1.3 CENTRE FOR APPLIED HEARING RESEARCH

The Centre for Applied Hearing Research has been established in order to strengthen DTU's teaching and research in psychoacoustics, physiological acoustics, and audiology. The purpose of the centre is to promote research and education in the field of acoustic communication with emphasis on the signal analysis and processing principles in the human auditory system, the perceptual consequences of hearing impairment, functional models of auditory processing and perception, applications of auditory models in hearing instruments, measurement and diagnosis of auditory function in clinical and technical audiology, and speech perception.

Speech Processing and Perception

Self-organising maps for measuring similarities in audio-visual speech percepts
Hans-Heinrich Bothe

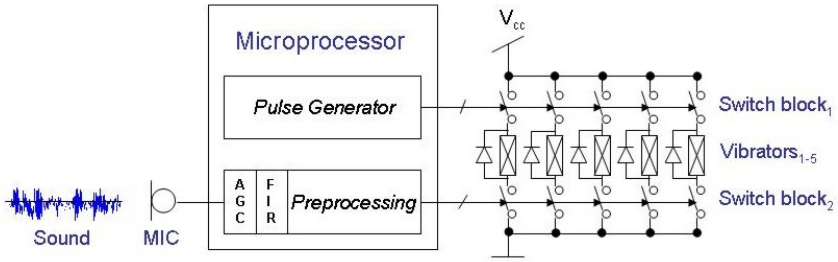


Self-organising map

Hearing-impaired people use audio-visual speech perception to better understand their partners during conversation. In noisy environments, also normal hearing people intuitively make use of visual cues to improve speech perception.

The goal of this project is to find similarity measures for audio-visual speech percepts so that a classification can be realised. For this purpose, phoneme-related self-organising maps (SOM) are trained with data material from a (labelled) video film with speaking faces. A combination of acoustic speech features and corresponding visual lip features is used for the training of the SOM. The process results in highly selective receptive fields on the SOM basis. Overlapping main slopes indicate similarity of respective units; distortion or extra peaks originate from the influence of other, possibly neighbouring units. The data material for the training process consists of 44 fully labelled sentences with a nearly balanced phoneme repertoire. As a result it can be stated that i) SOMs can be trained to map auditory and visual features in a topology-preserving way, and ii) they show strain due to the influence of other audio-visual units. The SOMs may be used to measure co-articulatory effects or similarities amongst audio-visual speech percepts.

Hand-hold vibro-tactile fingertip stimulator supporting the process of speech-reading
Hans-Heinrich Bothe



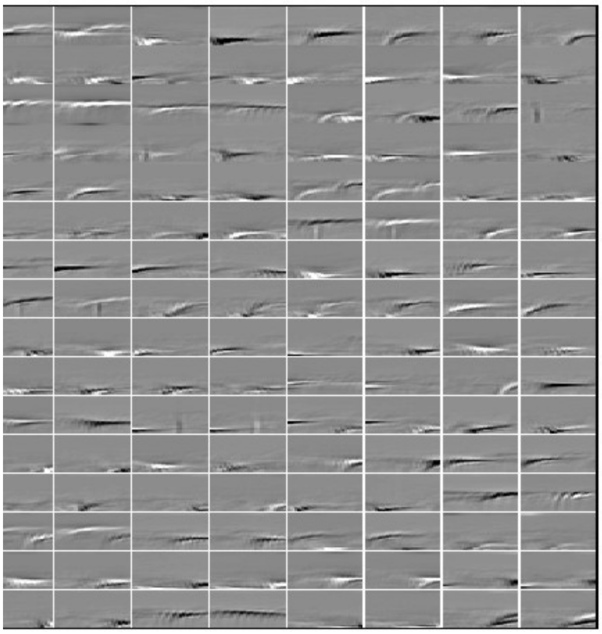
Schematic diagram.

'FLIPPS' is a hand-hold tactile stimulation device developed for the improvement of speech-reading conversation with or amongst hearing-impaired persons. It makes additional speech information available through multiple fingertip stimulation using power-balanced frequency bands for the vibrator stimulation. FLIPPS is battery-powered and microprocessor-controlled; it may be used in daily life conversation. The small size allows for pocket use. First field tests indicate the applicability of the device. At the moment, a prototype hardware realisation that allows for parameter optimisation and for further field tests is at hand. The prototype has been extended and improved by integrating a printed board with the complete electronics into the handle.

Most of the practical work in this project has been carried out by Golam Sadeghnia in a special course.

Evolution of artificial auditory spatio-temporal receptive field induced by long speech signals
Hans-Heinrich Bothe

Neurons in the primary auditory cortex utilise multidimensional multi-scale receptive activation fields associated with specific frequency bands to specific timing patterns of the input signal. Such an auditory 'spatio-temporal receptive field' (STRF) can be determined with the help of reverse correlation methods if the activation or output signal pattern of the neuron is known. Since this seems impossible for neurons in the human auditory cortex, only simulations can be preformed. Such simulations show that the evolution of respective banks of STRF is possible and also useful from the perspective of an 'optimal' information processing. A method is developed to emulate the evolution of monaurally-based sets of STRF utilising normally articulated speech as input and modified forms of independent component analysis plus a sparse matrix criterion for the development of optimal banks of STRF. The acoustic speech signal is preprocessed in a biologically plausible form and arrives in the auditory cortex in modified spectrogram representation. The bank of STRF acts on this spectrogram, producing temporal courses of 'micro-features' that can be used for further processing as, for example, recognition, identification, object binding, localisation, etc., as they are expected in higher cortical areas.



Example of 80 prototype spatio-temporal receptive field.

Most of the practical work in this project has been carried out by Ondrej Lassak in a special course.

Frequency selective filtering of the modulation spectrum and its impact on consonant identification

Thomas U. Christiansen and *Steven Greenberg*

The spectro-temporal coding of Danish consonants has been investigated using an information-theoretic approach. Listeners were asked to identify eleven different consonants spoken in a CV[l] syllable context (where C refers to the initial consonant, V refers to one of three vowels [i, a, u], and [l] refers to the syllable-final liquid segment). Each syllable was processed so that only a portion of the original audio spectrum was present. Narrow (three-quarter octave) bands of speech, with centre frequencies of 750 Hz, 1500 Hz and 3000 Hz, were presented individually and in combination with each other. The modulation spectrum of each band was low-pass filtered at 24, 12, 6 and 3 Hz. Confusion matrices of the consonant-identification data were computed, and from these the amount of information transmitted for each of three phonetic feature dimensions—voicing, manner and place of articulation—was calculated for each condition. This form of analysis provides a simple means of determining whether information associated with each phonetic feature dimension combines linearly across the audio spectrum, and, if not, delineates a method for characterising the (nonlinear) nature of information integration. In addition, the analysis provides a means to associate specific portions of the modulation spectrum with phonetic feature properties. Such analyses indicate that i) accurate, robust decoding of place-of-articulation information requires broadband cross-spectral integration; ii) place-of-articulation information is associated most closely with the modulation spectrum above 6 Hz, with the most significant contribution coming from the region above 12 Hz; iii) place-of-articulation information is crucial for accurate consonant recognition; hence, consonant decoding requires cross-spectral integration of the modulation spectrum above 8 Hz; iv) voicing is mainly associated with the modulation spectrum between 3 and 6 Hz (with a smaller contribution made by the region above 12 Hz); v) manner of articulation is most closely associated with the portion of the modulation spectrum above 12 Hz.

This form of information-theoretic analysis can be used to delineate those parts of the speech signal of greatest importance for encoding phonetic features associated with intelligibility and speech understanding.

Enhancement of speech intelligibility using novel synthesis methods to delineate the fundamental building blocks of spoken language

Jens Bo Nielsen

Supervisors: *Thomas U. Christiansen* and *Torsten Dau*

Far more is involved in understanding spoken language than the decoding of the acoustic spectrum; also the temporal modulations of speech are known to be of great importance. Not only does the auditory system perform a frequency analysis, it also performs a temporal analysis crucial for parsing the speech stream into such linguistic units as the syllable, word and phrase. Anything that reduces a listener's sensitivity to amplitude modulations is likely to lead to problems in speech understanding.

The temporal analysis of the auditory system can be assessed by determining the temporal modulation transfer function (TMTF), a diagram with the just-detectable modulation depth plotted as a function of the modulation frequency. The acoustic carrier when measuring TMTFs has traditionally been broadband noise; only in a few cases narrow-band noise has been used. The concept of TMTF can, however, be extended by using acoustic stimuli not only modulated in time, but also across the frequency spectrum; these two-dimensional modulations in the signal are often referred to as ripples. All complex sounds contain modulations in this two-dimensional domain, and modulation manipulation of acoustic stimuli is an effective approach to mapping the sound analysis of the auditory system.

In the first phase of the PhD project the spectro-temporal properties of a speech signal (nonsense syllables) will be systematically degraded in order to associate specific components of the acoustic waveform with linguistically meaningful dimensions such as phonetic-segment identity and syllable structure, as well as production-based dimensions such as voicing, place and manner of articulation.

The results of the experiments are expected to enhance our knowledge about which parts of the acoustic signal that are most important for speech intelligibility in complex environments. The spectro-temporal elements in the acoustic signal and their internal representation in the auditory system will be investigated, and the obtained knowledge will be used to devise methods for enhancing intelligibility in quiet and noisy environments, both for normal-hearing and hearing-impaired individuals.

Construction of a Danish hearing-in-noise test

Jens Bo Nielsen

A hearing-in-noise test (HINT) is used for determining the speech recognition threshold (SRT). It is applicable eg for assessing hearing impairments or testing the efficiency of hearing aids. Although available in several languages, a Danish HINT version has never been designed. For this reason Centre for Applied Hearing Research, Ørsted•DTU, started a project in 2004 to create such a test. In the first part of the project a sentence corpus with 400 sentences was designed and recorded with a male speaker. A spectrally matched noise was produced by superimposing the recorded sentences. In the second part of the project intelligibility tests of the recorded sentences will be conducted with normal-hearing subjects in order to obtain a corpus of sentences with equal intelligibility. Software for running these tests has been constructed.

It is planned to publish the final HINT with matching noise on a CD. The test will consist of three practice lists and 25 test lists, allowing up to 25 SRT determinations for each test subject.

Perception and Processing of Complex Sounds

Simultaneous reflection masking: Monaural and binaural processes

Jörg Buchholz

Masked thresholds (MT) for a single test reflection masked by the direct sound (200 ms long broad band noise) were measured dependent on the time delay of the reflection for diotic as well as dichotic stimulus presentation. In the diotic case, the direct sound and the test reflection were presented equally to both ears via headphones. In the dichotic case, an inter-aural time difference of 0.5 ms was added to the test reflection. In order to focus on simultaneous masking effects, the reflection was truncated in such a way that it formed a common offset with the direct sound. For the diotic stimulus condition, the resulting MT increased with increasing reflection delay and for the dichotic condition, the MT decreased with increasing reflection delay, producing an intercept between the two curves at a reflection delay of about 7-10 ms. Hence, negative binaural masking level differences (BMLDs) up to about -8 dB were found for very early reflections, and positive BMLDs up to about +8 dB were found for later reflections, suggesting binaural mechanisms that suppress very early reflections and enhance later reflections. Evidence is provided that the critical delay of 7-10 ms refers to the duration of a temporal window that is employed by the auditory system to analyse the considered stimuli and might be related to the temporal resolution of the auditory system at low signal frequencies (ie, 1-2 kHz). In order to describe the monaural signal processing underlying the experimental data, a spectral modulation filterbank concept is proposed that analyses periodicities inherent in the power spectrum produced by a gammatone filterbank. Moreover, the data is discussed in the background of different binaural model approaches taken from the literature. In this way it is shown that either an equalisation-cancellation (EC) model or a combination of an EC model and an inverse filtering model could describe the involved binaural processes.

Sound source localisation: Microphone array design and evolutionary estimation
Jörg Buchholz

The use of speech in human-robot interaction is an indispensable feature from the points of view of both the human and the robot. In order to realise such capability, sound source localisation becomes a mandatory or a prerequisite requirement. This project approaches the localisation of a speaker by using a microphone array. A time-delay-of-arrival based technique is employed to obtain the angular direction of the sound source relative to the array. Geometric array construction that facilitates sound source localisation is taken into consideration. Since sound reception is corrupted by noise, an evolutionary computation method based on genetic algorithms is also used to provide an improved estimation of the sound source location. A simplification of the genetic algorithm is carried out, such that the complexity of the calculation is reduced and the estimator made practically tractable. By using the proposed method, the need for an accurate measurement model and a speaker motion model is removed, and thus the estimation robustness is increased. Simulations are included to demonstrate the effectiveness of the proposed technique.

This work was carried out in collaboration with Raymond Kwok, Gu Fang, and John Gal, University of Western Sydney, Australia. The study was essentially conducted at University of Western Sydney.

Forward masking: Neural adaptation versus temporal integration
Thomas U. Christiansen, *Stephan D. Ewert*, and *Torsten Dau*

As a possible explanation for forward masking, two different mechanisms have been discussed in the literature: persistence (or temporal integration) and adaptation. In this study, two well established models of temporal processing in the auditory system are compared in a unified modelling framework: (a) the temporal-window model representing a temporal-integration mechanism, and (b) the adaptation-loop model as the representative for the adaptation mechanism. The unified modelling framework shares a compressive, nonlinear auditory filter stage and a template-based (optimal detector) decision stage. In a first set of experiments, forward and simultaneous masking was investigated with a tone-pulse signal at 1 or 4 kHz and a 200-ms broadband-noise masker. In the second set of experiments, forward-masking functions were measured for maskers of different durations, whereby the masker level was adjusted such as to produce the same amount of forward masking for a masker-signal delay of 2 ms. Mathematically, the temporal-window model predicts the same forward masking functions for these conditions, which is consistent with the experimental results. The two models were found to be essentially equivalent in predicting forward masking: the combination of integration and the signal-to-noise-ratio based detection criterion in the temporal-window model acts effectively as adaptation. However, the adaptation model might be the more general approach since it shows the effect of adaptation in the internal representation of the stimuli similar to that observed in neural responses and can be applied successfully to a broader class of experimental masking conditions than the temporal window model.

Comodulation masking release: Spectro-temporal pattern analysis in hearing
Torsten Dau and *Stephan D. Ewert*

Across-frequency comparisons of temporal envelopes are likely to be a general feature of auditory pattern analysis and play an important role in extracting signals from noisy backgrounds, or in separating competing sources of sound. Comodulation of different frequency bands in background noise facilitates the detection of tones in noise, a phenomenon known as comodulation masking release (CMR). There has been recent interest in elucidating the physiological basis for CMR, and it has been suggested that CMR is generated at a very early (peripheral) stage of auditory signal processing. However, the results of the present project clearly indicate that CMR interacts with auditory object formation and grouping associated with higher-level processes in the auditory system. The present project specifically investigates effects of sequential and concurrent auditory streaming in CMR. The results place strong constraints on physiological models of CMR and on auditory perception models.

The project has been carried out in cooperation with Andrew J. Oxenham, MIT, Boston, USA.

Estimates of auditory filter phase response at and below the characteristic frequency
Stephan D. Ewert

Animal studies have shown that the basilar-membrane phase curvature (or rate of change of group delay with frequency) is negative around the characteristic frequency (CF), but near zero well below CF. In this project it has been tested whether psychophysical masking experiments in humans show the same difference between on- and off-CF phase curvatures. Masked thresholds were measured for a 2-kHz signal in the presence of harmonic tone complex maskers with a fundamental frequency of 100 Hz, band-limited between 200 and 1400 Hz (off-frequency masker) or between 1400 and 2600 Hz (on-frequency masker). The results from four normal-hearing listeners are consistent with predictions from animal physiological data: a negative phase curvature is found for the on-frequency masker, whereas the phase curvature for the off-frequency masker is near zero. The method and results provide a strong test for the temporal response of computational models of human cochlear filtering.

This project has been carried out in cooperation with Andrew. J. Oxenham, MIT, Boston, USA.

Interaction between pitch and modulation perception
Stephan D. Ewert

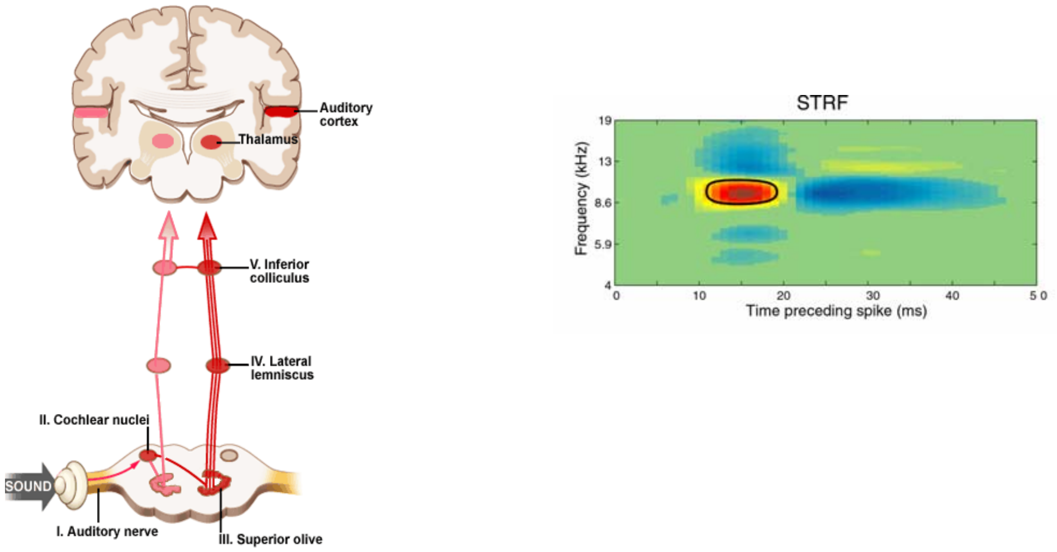
Recent temporal models of pitch and modulation perception converge on a relatively realistic implementation of cochlear processing followed by a temporal analysis of periodicity. There is large overlap in the existence region of pitch and modulation perception. This project has investigated the interaction between carrier periodicity and envelope periodicity using broadband stimuli derived from Gaussian noise. The current results indicate that in the presence of carrier periodicity, detection of amplitude modulation is impaired throughout the tested range (16 to 1000 Hz). In contrast, detection of carrier periodicity in the presence of amplitude modulation is impaired only for very low fundamental frequencies below the pitch range. The goal of the project has been to develop a model framework that allows a combined analysis of pitch and amplitude modulation which have so far been described separately in complementary models. For modulation perception, a modulation filterbank has been applied to the cochlear output, whereas for pitch perception, autocorrelation has been applied.

This work has been carried out in cooperation with Lutz Wiegrebe from the University of Munich, Germany.

Investigating the precedence effect: A comparison of masked and echo thresholds
Jess Hartcher-O'Brien
Supervisor: *Jörg Buchholz*

The clarification of the mechanisms underlying the auditory precedence effect (PE), whose upper limit is the echo threshold (ET), has involved discussions of auditory mechanisms from preprocessing to cognition. An alternative approach to investigating the auditory processing of reverberant sounds is based on the reflection masked threshold (RMT), which as shown in previous studies can be successfully explained by mechanism associated with auditory pre-processing [Buchholz, 2001]. Although the ET may not be solely explained by pre-processing mechanisms, a number of RMT and ET parameter dependencies show similar behaviour, and thus indicate involvement of similar mechanisms. In order to further clarify the processes involved in the PE, RMT and ET measurements were conducted for dependence of level, duration, delay, and stimulus type under comparable conditions. The results, and in particular, the comparison of the two thresholds, contribute to the clarification of the auditory processes addressed by the ET, and thus underlying the PE.

Multi-resolution spectro-temporal analysis of complex sounds in the human auditory system
Tobias Piechowiak
Supervisors: *Torsten Dau* and *Jörg Buchholz*



Spectro-temporal receptive field in the auditory cortex in the brain. Excitation-inhibition pattern of cell populations to a so-called 'ripple', a sinusoidally amplitude modulated stimulus in the temporal and the spectral domain. The aim of the Tobias Piechowiak's project is to investigate the perceptual consequences of this spectro-temporal transformation by using specifically designed psychoacoustical masking experiments. The results will be important for the developing a new model of auditory signal processing and perception.

When sound enters the ear the auditory system performs a frequency analysis, ie the information contained in the signal is separated into different frequency filters. In the next stage the signal is rectified and low-pass filtered such that the auditory system follows only the amplitude modulation (the *envelope*) of the sound. Across-filter comparisons of temporal envelopes are very common in natural sounds and are supposed to play an important role in extracting signals from noisy backgrounds. An example where a comparison of the envelopes in different frequency regions leads to a substantial facilitation of detecting a signal in a noisy background is comodulation masking release (CMR). Comodulation means that the sounds in the different frequency regions have coherent envelope fluctuations. However, it has been shown that CMR is sensitive to the acoustic context. For example, presenting sounds with different onsets as offsets in the various frequency regions can destroy the benefit usually obtained by the comodulation. Since it is assumed that monaural across-channel processing and binaural across-ear processing underlie similar processing principles this PhD project investigates if binaural listening is also sensitive to the acoustic context. In addition, the project focuses on a model that quantitatively accounts for CMR in various stimuli configurations. Since Durlach suggested an equalisation-cancellation based circuit as an effective strategy of 'noise reduction' in the binaural system (1963) it could also be an effective way of accounting for CMR. In this case the activity in the different frequency bands would be averaged (equalisation) and then subtracted from the target containing signal (cancellation). The project will later focus on the investigation of so-called spectro-temporal modulations (also referred to as ripples) and their role for the auditory system in the understanding of complex sounds like speech and music.

Perceptual consequences of impaired auditory signal processing in complex acoustic environments

Olaf Strelcyk

Supervisor: *Torsten Dau*

One of the most common complaints of people with hearing loss concerns their difficulty in understanding speech. In particular, hearing-impaired people often have great difficulty with speech communication when background noise is present. Also, they are typically less able than normally hearing people to take advantage of spatial separation of the target speech and the interfering sounds. Recent advances in hearing aid technology have addressed this problem by focusing on improving the signal-to-noise ratio delivered to the listener. However, despite enormous technological progress in this area, the benefit varies strongly among the individual listeners. While some listeners show good performance in speech communication, others continue to experience difficulties.

There has been considerable controversy in the literature about the reasons for this difficulty. Some researchers suggest that the difficulty arises primarily from reduced audibility; for a given speech level, the proportion of the speech spectrum which is above hearing threshold is less than for normal listeners. Other researchers argue that the difficulty arises, at least partly, from changes in the perception of sound which are *well above* the absolute threshold.

The hypothesis of the current PhD project is that speech communication problems in the individual hearing-impaired listener result from one of—or a combination of—the following effects: (i) desynchronised neural activity in the auditory system, (ii) reduced frequency selectivity and frequency discrimination, (iii) central processing deficits in perceptually segregating sound objects. In order to choose the right compensation strategy (eg, in a hearing aid) for the individual hearing-impaired person, the hearing impairment needs to be characterised as precisely as possible in terms besides just audibility. Further questions are: Where are the limitations 'located' in the brain? How can the performance of the listeners be predicted using models of (impaired) auditory signal processing? The approach to be taken will combine perceptual listening experiments in complex acoustic environments, non-invasive acoustically evoked brain responses, and computational neural modelling in order to allow an integrative characterisation of auditory function.

The project is co-supervised by Graham Naylor, Oticon

Modelling monaural and binaural signal detection in reverberant listening environments

Eric Thompson

Supervisor: *Torsten Dau*

This PhD project attempts to model and understand sound perception in realistic acoustic environments. A state-of-the art model of monaural processing is available at the Centre for Applied Hearing Research. The key processing stage of the detection model is a modulation band-pass filterbank that analyses the dynamic properties of the sound. The model combines concepts from linear system theory and signal detection theory (Dau *et al*., 1997) and accounts for various experimental monaural detection conditions in normally-hearing and hearing-impaired listeners. However, only anechoic stimuli have been considered, and no directional information has been evaluated in the case of different positions for the signal and the interferer (masker). Using a similar modelling approach, a binaural detection model has been developed in parallel at Eindhoven University and Philips research laboratories (Breebaart *et al*., 2001). The model assumes a network that computes the capabilities of the auditory system to process, eg, interaural time and level differences. The model has been very successful in classical conditions of binaural detection. However, this processing model does not contain a modulation filterbank, and this is needed to account for the analysis of the auditory system of more complex sounds like speech and music. The model does also not include effects of hearing impairment in its preprocessing. The idea of the current project is to combine these two processing models in order to develop a generalised framework for the processing of complex sounds. In order to simulate the acoustic input to the two ears (and to the model) in terms of the room-acoustic parameters, source location and head diffraction effects, the software package Odeon at Acoustic Technology in combination with a head model of the simulated listener will be used. The current project thus attempts to combine the expertise from three different research groups that have recently published state-of-the-art models of sound processing in their respective areas. A combination of knowledge from room acoustics and hearing is necessary for a better understanding of speech perception of normal-hearing and hearing-impaired listeners in realistic environments.

Objective Measures of the Auditory Function

Influence of cochlear travelling wave and neural adaptation on auditory brainstem responses

Torsten Dau

The present study has investigated the relationship between evoked responses to transient broadband chirps and responses to the same chirps when embedded in longer-duration stimuli. It has been examined to what extent the responses to the composite stimuli can be explained by a linear superposition of the responses to the single components. For stimulation levels up to approximately 70 dB SPL, the responses to the embedded chirp corresponded to the responses to the single chirp. At high stimulus levels (80-100 dB SPL), disparities occurred between the responses, reflecting a nonlinearity in the processing when neural activity is integrated across frequency. The results further demonstrate the importance of cochlear processing for the formation of brainstem potentials. The data may provide constraints on future models of peripheral processing in the human auditory system. The findings might also be useful for the development of effective stimulation paradigms in clinical applications.

The study was carried out in cooperation with Dirk Junius, University of Oldenburg, Germany. Most of the work was done in Oldenburg.

The effects of neural synchronisation and peripheral compression on the acoustic-reflex threshold

Torsten Dau

This study investigated the acoustic reflex threshold (ART) dependence on stimulus phase utilising low-level reflex audiometry. The goal has been to obtain optimal broadband stimuli for elicitation of the acoustic reflex and to obtain objective determinations of cochlear hearing loss. Three types of tone complexes with different phase characteristics were investigated: A stimulus that compensates for basilar-membrane dispersion thus causing a large overall neural synchrony (basilar membrane tone complex, BMTC), the temporally inversed stimulus (iBMTC), and random-phase tone complexes (rTC). The ARTs were measured in eight normal-hearing and seven hearing-impaired subjects. Five different conditions of peak amplitude and stimulus repetition rate were used for each stimulus type. The results suggest that the ART is influenced by at least two different factors: (a) the degree of synchrony of neural activity across frequency, and (b) the fast-acting compression mechanism in the cochlea that is reduced in the case of a sensorineural hearing loss. The results allow a clear distinction of the two subject groups based on the different ART for the utilised types and conditions of the stimuli. These differences might be useful for objective recruitment detection in clinical diagnostics.

This study has been carried out in collaboration with Matthias Müller-Wehlau, Manfred Mauermann, and Birger Kollmeier, University of Oldenburg, Germany. Most of the study took place in Oldenburg.

Recording neural adaptation from different stages in the auditory pathway—A technical feasibility study

James Harte

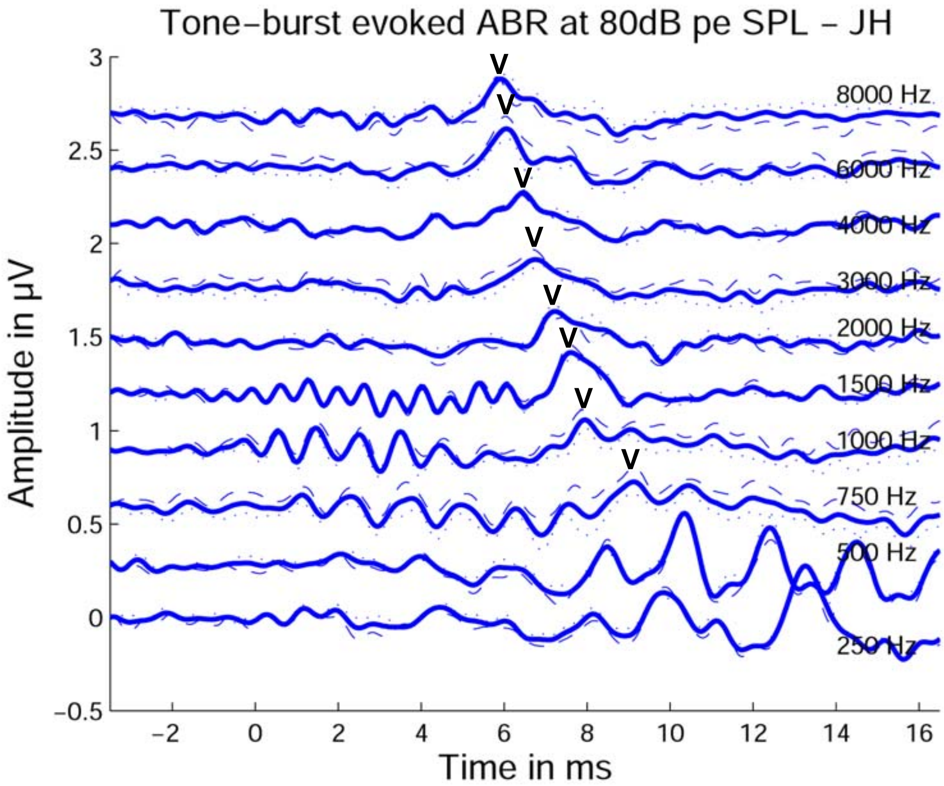
Evoked responses represent the summation of responses from many neurons, recorded from electrodes placed at the surface of the head, remote from individual neurons. Auditory evoked potentials can be recorded from all levels of the auditory system and are grouped by the time of occurrence after the onset of the stimulus.

Adaptation is the variation in response that occurs during a constant stimulus. In peripheral auditory signal processing, the change in neural activity is typically a maximum at onset and then decays or adapts to a smaller sustained change in the response. A classical method to generate and visualise an objective measure of adaptation is through averaging the responses to multiple presentations of a stimulus train. Evoked potentials require the use of time-domain conventional averaging to improve poor signal-to-noise ratio from typical recordings. If the response to each stimulus is not complete prior to the presentation of the second stimulus, then an overlapped response of limited use is acquired. This constraint obviously limits the stimulus presentation rate, and thus our ability to obtain a measure of adaptation for longer latency evoked potentials.

To circumvent these limitations it has been shown that a specially designed stimulus sequence, known as a maximum length sequence, or indeed continuous loop averaging deconvolution, is able to separate the individual responses from overlapping responses. Both of these techniques require a certain degree of stimulus jitter, ie, a variation in the rate of stimulus presentation.

This project will investigate the adaptation for auditory brainstem and middle latency responses, using a measure of adaptation based on stimulus presentation rate. By using a deconvolution algorithm to separate the overlapped responses, it is hoped to investigate adaptation through a wider range of latency values than has been historically carried out. Nonlinear residual analysis and functional series methods will be used to test the validity of using linear deconvolution based methods for this type of analysis.

Signal processing in the human auditory system: Auditory-evoked brain responses as a correlation of hearing function
Gilles Pigasse
Supervisors: *James Harte* and *Torsten Dau*



Examples of auditory evoked responses to tone bursts as investigated in Gilles Pigasse's project. Each curve represents the response to a specific frequency, and the "V" indicates the peak of the so-called wave V, which is important in clinical diagnostics. The decreasing latency with increasing frequency reflects the dispersive properties of processing in the cochlea. The data are used to estimate cochlear group delay in humans and provide constraints on models of auditory processing.

The purpose of this PhD project is to obtain a fundamental understanding of the mechanisms underlying the generation of auditory evoked potentials. The consequences of cochlear hearing impairment, on evoked-potential generation, will be investigated though the effects of loss of outer hair cell functionality on temporal nonlinearities. Various physiological measurements will be made, namely; electrocochleography, otoacoustic emissions (OAEs), and auditory evoked potentials. An 'ideal' stimulus for the generation of early evoked potentials will be developed. This stimulus will be designed to compensate for the cochlear travel-time differences across frequency and intensity, in order to maximise neural synchrony. This will highlight to what extent the evoked potential can be optimised by only considering cochlear processing.

The initial cochlear travel time estimates will be made using otoacoustic emissions. Electrocochleography will be used alongside these recordings to investigate the stimulus related responses of the inner ear and auditory nerve, ie, at a stage in auditory processing between the cochlear mechanical response and the brainstem. Offering a direct forward path measure of the basilar membrane latency and dispersion functions with frequency. This will be compared with the estimates from OAE measures, to test some underlying assumptions inherent in physiological models of cochlear processing in the literature. Further, the recorded auditory evoked potentials can be used to test the assumption that neural delay, obtained by subtracting the mechanical delay, is unaffected by stimulus frequency and/or intensity.

The project will also investigate new experimental paradigms that could provide a very effective objective indicator of hearing impairment. The last part of the project will focus on implications of the experimental findings for models of auditory signal processing.

Models of Auditory Processing and Perception

Modelling the effective signal processing in the normal and impaired auditory system
Thomas U. Christiansen and *Torsten Dau*

Coding of speech and audio signals (eg MP3) and signal processing algorithms used in digital hearing aids are examples of technical applications based on results from psychoacoustics. The goal of the present project is to develop a state-of-the-art quantitative model of the signal processing in the normal and impaired human auditory system. The idea is to obtain a functional description of the transformation from the acoustic signal into its 'internal representation' in the brain by incorporating recent knowledge of the signal processing in the inner ear, the cochlea. The challenge is to develop a model that accounts for a large variety of perceptual phenomena. Perception does not only depend on cochlear processes but also on central processes in the brain that should be described effectively within the model. Assuming that the numerical transformation from the acoustic signal into its internal representation is correct, such a model would be of great potential for advanced technical applications in audio coding, transmission quality evaluation, automatic speech recognition, and digital hearing aid.

A functional point-neuron model simulating cochlear nucleus ideal onset responses
Torsten Dau

Cochlear nucleus neurons revealing ideal onset-type peri-stimulus time histograms encode temporal features of acoustic stimuli with very high precision. These neurons are therefore assumed to be involved in the recognition of natural sounds with temporally varying envelopes such as speech. A functional point-neuron model has been developed for the simulation of onset-unit responses. The model assumes a biphasic response of the membrane potential to a current impulse (the membrane impulse response) and a dynamic spike-blocking mechanism. The model is tested and compared to recordings from the literature using a variety of stimuli, including pure tones at low and high frequencies, amplitude modulated stimuli covering a wide range of modulation frequencies and levels, and depolarising and hyperpolarising current steps. The model accounts for the main response properties in the data using the same small set of parameters for all experimental conditions. The assumed biphasic shape of the membrane impulse response allows for a description of onset-responses that cannot be accounted for by an integrate-to-threshold unit in connection with a coincidence-detection mechanism. The presented functional model may be useful as a processing module in more complex models of auditory signal processing and perception.

The study was carried out in collaboration with Ulrike Dicke, University of Oldenburg, Germany. Most of the study took place in Oldenburg.

A neural circuit transforming temporal periodicity information into a rate-based representation in the mammalian auditory system

Torsten Dau and *Stephan D. Ewert*

Periodic amplitude modulations (AM) of an acoustic stimulus are presumed to be encoded in temporal activity patterns of neurons in the cochlear nucleus (CN). Physiological recordings indicate that this temporal AM code is transformed into a rate-based periodicity code along the ascending auditory pathway. Especially, the bandpass shaped rate modulation transfer functions (rMTF) found in neurons at the level of the inferior colliculus (IC) are assumed to represent a rate-based code of AM information, arising between CN and IC. The model developed in this project provides a neural circuit that transforms temporal periodicity information, provided by ideal onset units in the CN, into a rate-based periodicity representation at the level of the IC. Due to the neural connectivity of the present circuit, bandpass shaped rMTFs with different best modulation frequencies (BMF) are obtained that correspond to recorded rMTFs of IC neurons. In contrast to previous modelling studies describing the formation of bandpass rMTFs in the IC, the present circuit employs no continuously changing temporal parameter in order to obtain different BMFs in different IC bandpass units. Instead, different BMFs are yielded from varying the number of input units projecting onto different IC bandpass units within the present circuit. The model also suggests an explanation of how rMTFs with a region of suppression (a further rMTF-type observed in the IC) might result from neural connectivity within the auditory system.

The study was carried out in collaboration with Ulrike Dicke and Birger Kollmeier, University of Oldenburg, Germany. Most of the study took place in Oldenburg.

Nonlinear time domain cochlear modelling: Limit-cycle oscillators as sources of cochlear nonlinearity

James Harte

In a previous investigation by Johansson and Elliott at the Institute of Sound and Vibration Research (University of Southampton, UK), a comparison was made on the performance and characteristics of frequency- and time-domain models of the motion of the basilar membrane within the cochlea. Later collaboration by Harte and Elliott investigated models of the macro-mechanical interaction between cochlear fluids and the basilar membrane. Limit-cycle oscillators were used in the governing equations in an attempt to account for the cochlear active process, compressive nonlinearity and the generation of spontaneous otoacoustic emissions.

A simple time-domain model was developed, making historically used simplifying assumptions. Namely, that the cochlea could be modelled as a straight fluid-filled box structure that was split in the middle by the basilar membrane (BM). A discretised cochlea model was developed with the assumption that no longitudinal coupling of the BM existed except through fluid interactions. Each location or node on the BM was represented as a mass-spring-damper system. In order to produce the sharp tuning inherent in a healthy cochlea a damping term that is negative over a certain frequency range must be used. The dynamic system can then generate as well as dissipate energy and thus becomes active. A model of this type based on Van der Pol’s equation displays a wide variety of dynamic behaviours including limit cycle oscillations whose properties have striking similarities to those of spontaneous otoacoustic emissions, and is described by a ‘Hopf-bifurcation’. Dynamic systems of this type operate from a stable equilibrium to a limit cycle oscillation of increasing amplitude as the damping parameter is varied. They can be shown to account for well documented essential nonlinearities of the ear, for example the compression of dynamic range, sharper cochlear tuning for softer sounds and generation of combination tones.

The majority of work for this project was carried out at the Institute of Sound and Vibration Research, Southampton University, in collaboration with Stephen J. Elliott and Torbjorn Johansson;

however a technical report was written at Acoustic Technology on the use of Van der Pol Oscillators in cochlear models. Some initial numerical simulations were carried out comparing well known spontaneous otoacoustic emission experimental findings to the modelling efforts.

PUBLICATIONS

Journal Papers

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D. Junius and *T. Dau*: Effects of cochlear traveling wave and neural adaptation on auditory brainstem responses. *Hearing Research* **205**, 53-67, 2005.

M. Müller-Wehlau, M. Mauermann, *T. Dau* and B. Kollmeier: The effects of neural synchronization and peripheral compression on the acoustic-reflex threshold. *Journal of the Acoustical Society of America* **117**, 3016-3027, 2005.

A.J. Oxenham and *S. Ewert*: Estimates of auditory filter phase response at and below characteristic frequency. *Journal of the Acoustical Society of America* **117**, 1713-1716, 2005.

C. Füllgrabe, B.C. Moore, L. Demany, *S. Ewert*, S Sheft and C. Lorenzi: Modulation masking produced by second-order modulators. *Journal of the Acoustical Society of America* **117**, 2158-2168, 2005.

A. Stein, *S.D. Ewert* and L. Wiegrebe: Perceptual interaction between carrier periodicity and amplitude modulation in broadband stimuli: a comparison of the autocorrelation and modulation-filterbank model. *Journal of the Acoustical Society of America* **118**, 2470-2481, 2005.

J.M. Harte and S.J. Elliott: A comparison of two methods for obtaining derived, noise-evoked otoacoustic emissions. *Acta Acustica united with Acustica* **91**, 880-891, 2005.

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J.M. Harte and S.J. Elliott: Using the short-time correlation coefficient to compare transient- and derived, noise-evoked otoacoustic emission temporal waveforms. *Journal of the Acoustical Society of America* **117**, 2005, 2989–2998.

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Conference Papers

J.M. Buchholz: Simultaneous room reflection masking: effect of forward fringe and direct sound duration. *Forum Acusticum 2005*, Budapest, Hungary, 2005.

N.M. Kwok, *J.M. Buchholz*, G. Fang and J. Gal: Sound source localization: microphone array design and evolutionary estimation. *International Conference on Industrial Technology*, Hong Kong, China, 2005.

T. Zarouchas, *J.M. Buchholz* and J. Mourjopoulos: An audio quantizer based on a time domain auditory masking model. *AES 118th Convention* (preprint 6378), Barcelona, Spain, 2005.

T.U. Christiansen and *S. Greenberg*: Frequency selective filtering of the modulation spectrum and its impact on consonant identification. *Proceedings of 21st Danavox Symposium 'Hearing Aid Fitting'* (eds. A.N. Rasmussen and *T. Poulsen*), Kolding, Denmark, 2005.

T. Dau, *S. Ewert* and A.J. Oxenham: 'Effects of concurrent and sequential streaming in comodulation masking release' in *Auditory Signal Processing: Physiology, Psychoacoustics, and Models*, eds. D. Pressnitzer, A. de Cheveigné, S. McAdams, and L. Collet (Springer, New York, 2005).

T. Dau, *T. Piechowiak*, *S.D. Ewert* and *A.J. Oxenham*: Auditory object formation affects modulation perception. *Proceedings of 21st Danavox Symposium 'Hearing Aid Fitting'* (eds. A.N. Rasmussen and *T. Poulsen*), Kolding, Denmark, 2005.

J.B. Nielsen: Construction of a Danish HINT. *Proceedings of 21st Danavox Symposium 'Hearing Aid Fitting'* (eds. A.N. Rasmussen and *T. Poulsen*), Kolding, Denmark, 2005.

T. Piechowiak, *S.D. Ewert* and *T. Dau*: Principles of modulation processing in monaural versus binaural processing. *Fortschritte der Akustik*, DAGA '05, Munich, Germany, 2005.

E.R. Thompson and *T. Dau*: Monaural and binaural processing of amplitude modulated sounds in the auditory system. *Proceedings of 21st Danavox Symposium 'Hearing Aid Fitting'* (eds. A.N. Rasmussen and *T. Poulsen*), Kolding, Denmark, 2005.

Published Abstracts

J.M. Buchholz: Binaural effects in simultaneous room reflection masking. *149th Convention of the Acoustical Society of America*, Vancouver, *Journal of the Acoustical Society of America* **117**, p. 2484, 2005.

O. Hau, *S.D. Ewert* and *T. Dau*: A unified view of the temporal-window and the adaptation-loop model in conditions of forward- and simultaneous masking. *149th Convention of the Acoustical Society of America*, Vancouver, *Journal of the Acoustical Society of America* **117**, p. 2536, 2005.

H.-H. Bothe: Self-organizing maps for measuring similarity of audiovisual speech percepts. *149th Convention of the Acoustical Society of America*, Vancouver, *Journal of the Acoustical Society of America* **117**, p. 2571, 2005.

P.C. Nelson, *S.D. Ewert*, L.H. Carney and *T. Dau*: Comparison of intensity discrimination, increment detection, and comodulation masking release in the envelope versus audio-frequency domain. *149th Convention of the Acoustical Society of America*, Vancouver, *Journal of the Acoustical Society of America* **117**, p. 2535, 2005.

T. Piechowiak, *S.D. Ewert* and *T. Dau*: Modeling comodulation masking release using an equilization cancellation mechanism. *149th Convention of the Acoustical Society of America*, Vancouver, *Journal of the Acoustical Society of America* **117**, p. 2536, 2005.

2. TEACHING ACTIVITIES

The Danish system for tuition at a university level prescribes close connections between research and teaching. Consequently, university scientists must devote time for teaching students at undergraduate and graduate level. As a rule, their teaching will be related to their personal research activities; thus, nearly all the research projects described in chapter 1 are reflected in the basic and advanced tuition described below.

At the Technical University of Denmark, MSc and PhD degrees will be awarded to successful students after nominal periods of study of five and additional three years. Undergraduate courses are scheduled to fit into the University's existing tuition modules, thus permitting the students to compose a sensible curriculum encompassing their professional requests. DTU also offers a BEng degree, and an increasing number of BEng students follow AT's regular courses.

In the year 2000 DTU launched a series of two-year international MSc programmes. Acoustic Technology, Ørsted•DTU, offers an MSc programme in Engineering Acoustics, coordinated by Mogens Ohlrich. See www.msc.dtu.dk for information on the application procedure.

Tuition related to preparation for academic degrees (MSc and PhD) is individual, based on the expertise of an academic member of the staff as adviser to each student, often supplemented with assistance from industry or other research institutions. Nineteen students carried out their MSc thesis work at AT in 2005, and two did their BEng thesis work. Supervision was received by a total of ten PhD students enrolled at DTU and ten visiting PhD students from other institutions.

2.1 SCHEDULED COURSES

Acoustic Technology offers a series of courses every year. These courses are taught to groups of students, and most courses are supported by written material prepared for the purpose by staff members. Each course comprises a series of lectures and laboratory exercises or excursions. The courses offered are listed and briefly described below. Detailed descriptions of all the courses on acoustics are available on the internet (http://www.oersted.dtu.dk/English/education/courses/at.aspx).[3]

Additional 'special courses' may be given to small groups of students or to individuals.

INTRODUCTORY LEVEL

Fundamentals of Acoustics and Noise Control (*Finn Jacobsen* and *Jens Holger Rindel*)

The purpose of this course is to introduce the students to fundamental acoustic concepts, simple sound fields, psychoacoustics, acoustic measurements, architectural acoustics and structureborne sound, and thus to give the necessary background for more specialised courses in acoustics.

ECTS credit points: 5.

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[3] Enquiries from students interested in the courses are encouraged. It can be recommended to contact the relevant teacher directly (e-mail addresses are available at http://www.oersted.dtu.dk/English/research/at/at_staff.aspx). Rules for admission are available at the address http://www.dtu.dk/English/education/admission.aspx. A request for a Guest Student Form should be sent to International Affairs:

International Affairs,
Technical University of Denmark, Building 101A,
DK-2800 Kgs. Lyngby, Denmark
Tel.: +45 4525 1023
Fax: +45 4587 0216
E-mail: pwi@adm.dtu.dk

Environmental Acoustics (*Jens Holger Rindel*)

This course is intended for students with an interest in sound as an environmental factor, wishing to become able to solve problems as they appear in traffic, in industry and under domestic conditions, including noise predictions and measurements.

ECTS credit points: 5.

ADVANCED LEVEL

Acoustic Communication and Audiology (*Torben Poulsen*)

The aim of the course is to give the student an understanding of normal and impaired human sound perception. This understanding may be used to evaluate communication systems, for example auditoria, loudspeaker installations, hearing aids, and telephones. The student shall thus be able to expound on concepts, theories and methods of sound perception and speech intelligibility for normal and hearing-impaired persons. Major topics in the course include the function of the ear, psychoacoustics (hearing threshold, loudness, masking etc.), psychoacoustic measuring methods, speech and speech intelligibility, hearing impairment, principles of hearing aids, and hearing aid fitting.

Four laboratory exercises with reports are a substantial part of the course. Forty students participated in 2005. An excursion was made to the acoustics department at Aalborg University and to Bang & Olufsen, Struer. The evaluation was made in groups starting with a presentation of one of the exercises/reports.

ECTS credit points: 10.

Technical Audiology (*Torben Poulsen* and *Torsten Dau*)

The aim of this course is to introduce the participants to measurements in technical audiology. In 2005 the measurements comprised: Clinical tests (high frequency pure-tone audiometry, bone conduction calibration and measurements), hearing aid measurements using pure tones, hearing aid measurements using broad band signals, evoked response audiometry. Twelve students participated in 2005. Excursions were made to the audiological departments at Widex and at Oticon (both in Copenhagen), and to Technical Audiological Laboratory, DELTA, Odense. The evaluation was made by means of a poster presentation and an oral presentation from each group.

ECTS credit points: 5.

Auditory Signal Processing and Perception (*Torsten Dau*, *Stephan D. Ewert* and *Oliver Fobel*)

The purpose of this course is to give the participants an understanding of the processing mechanisms in the auditory system and the perceptual consequences. The topics include the psychophysics and physiology of the auditory system, models of auditory signal processing and perception, neurophysiological measuring methods, and neural imaging techniques.

ECTS credit points: 10.

Architectural Acoustics (*Anders Christian Gade* and *Jens Holger Rindel*)

The aim of this course is to give student a profound knowledge of the theories and methods of room acoustics and sound insulation. The topics include reflection and absorption of sound, design of absorbers and of rooms for speech and music, structureborne sound and sound insulation of building constructions, building codes and test methods.

ECTS credit points: 10.

Electroacoustic Transducers and Systems (*Finn Agerkvist*)

This course provides the student with knowledge of the fundamentals of acoustic transducers. Analogies between mechanical, acoustic and electrical systems are considered and applied to loudspeakers, microphones and communication systems. The student will also gain insight into the fundamental principles of sound recording and reproduction, as well as audio coding techniques. Laboratory exercises are a substantial part of the course. The course has been significantly revised in 2005; the changes include less time spent on lectures and more time for student activities and learning. The changes have been carried out as a part of a course on teaching methods run by LearningLab, DTU.

ECTS credit points: 10

Advanced Loudspeaker Models (*Finn Agerkvist*)

This new course (offered for the first time in January 2006) aims to introduce advanced elements in loudspeaker models in order to improve the validity of the models at high frequencies and/or high levels. The main part of the course is related to the measurement and modelling of the dominant sources of distortion in the loudspeaker. Methods for compensation of the distortion are also introduced. In addition the course introduces micro-loudspeakers as used in mobile phones and headsets. The final element in the course is the modelling of 'break-up' vibrations in the loudspeaker diaphragm, which are modelled by Finite Element Model calculations.

ECTS credit points: 5

Advanced Acoustics (*Finn Jacobsen*)

This course is intended to give the student an insight into the fundamental methods of theoretical acoustics. Among the topics taught are: sound fields in ducts, design of silencers, the Green's function, radiation of sound, classical room acoustic modal analysis, statistical room acoustics, numerical methods of sound field calculation, fundamentals of active noise control, sound intensity and other advanced acoustic measurement techniques. Laboratory exercises and simulation tasks have been a substantial part of the course from 2002.

ECTS credit points: 10.

Sound and Vibration (*Mogens Ohlrich*)

By introducing the principles and laws governing the generation, transmission and radiation of structureborne sound, the course will enable the student to analyse technical noise and vibration problems, design practical measures for reduction of structureborne noise in machines, vehicles and buildings, and apply advanced measurement techniques in the field. Mini-projects, laboratory exercises and reports are a substantial part of the course.

ECTS credit points: 10.

PhD COURSE

Attenuation and Damping of Structure Borne Sound (*Mogens Ohlrich*)

The aim of this course is to provide the student with an advanced level of knowledge and understanding of vibro-acoustic problems that involve sound waves in solid structures. Among the topics that are covered are: structureborne sound in beams, plates and shells, transmission and attenuation of longitudinal waves and bending waves, damping mechanisms, structural intensity, sound radiation, analytical methods, numerical methods, and statistical energy analysis.

The course was not given in 2005.

LECTURE NOTES ISSUED IN 2005

A.C. Gade: Room acoustic engineering. *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 4213, 2005.

F. Jacobsen: An elementary introduction to applied signal analysis. *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 7001, 2005. (45 pp.)

F. Jacobsen and P.M. Juhl: Radiation of sound. *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 31263, 2005. (58 pp.)

F. Jacobsen, *T. Poulsen*, *J.H. Rindel*, *A.C. Gade* and *M. Ohlrich*: Fundamentals of Acoustics and Noise Control. *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 31200, 2005. (172 pp.)

T. Poulsen: Acoustic communication. Hearing and speech (version 2.0). *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 31230-05. (94 pp.)

M. Ohlrich: Structure-borne sound and vibration, Part 1a. *Acoustic Technology, Ørsted-DTU, Technical University of Denmark*, Note no 7016/0116, 2005. (84 pp.)

M. Ohlrich: Structure-borne sound and vibration, Part 1b. *Acoustic Technology, Ørsted-DTU, Technical University of Denmark*, Note no 7017, 2nd ed., 2005. (93 pp.)

T. Poulsen: Psychoacoustic measuring methods (version 2.2). *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 31230-08, 2005. (59 pp.)

J.H. Rindel: Sound insulation in buildings. *Acoustic Technology, Ørsted•DTU, Technical University of Denmark, Note no* 4214, 2005.

2.2 BEng PROJECTS IN 2005

DC/AC converters in mobile phones
Sune Berentsen
Supervisor: *Finn Agerkvist*

In mobile phones the voltage supplied from the battery is quite limited. This limits the output level and to some extent also the sound quality from the phone as the loudspeaker must be very efficient. In this project a voltage doubler in the form of a DC-DC charge pump was developed for use with an existing class D mobile phone amplifier. The charge pump design has the advantage of not needing any inductor that takes up a significant amount of physical space. The total efficiency of DC-DC charge pump and amplifier was as high as 80%.

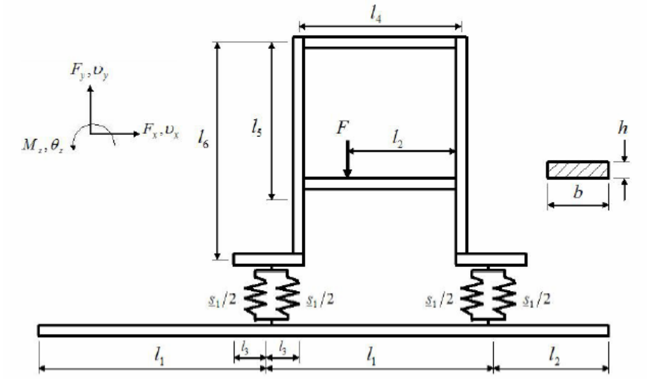
Investigation of microtransducer boxes
Rasmus Olesen
Supervisor: *Finn Agerkvist*

Since space is very limited in mobile phones, the options available for the acoustic design are also quite limited. In this project the influence of back volume and front cover has been investigated. The acoustic impedance of the slits and holes was measured, and the frequency extension that can be achieved by using a vented box was also investigated.

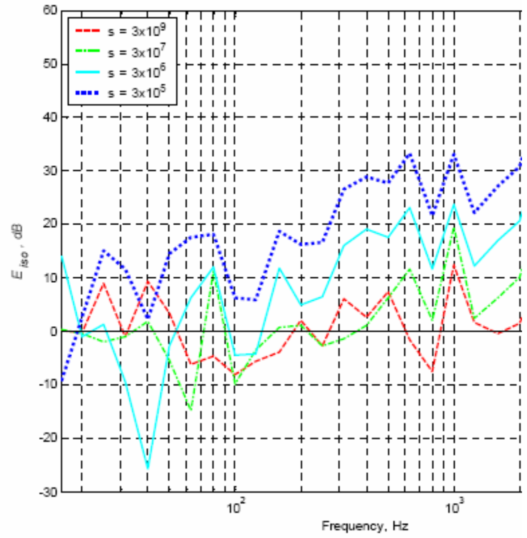
2.3 MSc PROJECTS IN 2005

Minimisation of vibration transmission via machine feet and mountings
Waseim Alfred
Supervisor: *Mogens Ohlrich*

This project has investigated the transmission of structureborne sound power from a non-rigid source structure to a receiving beam structure. Initially the source was modelled as a flexural vibrating beam, and this beam and the receiver beam were connected at two points by vibration isolators with translational and rotational spring properties. It was found that the total transmitted power is almost equal to the translational power component in the cases of medium and soft coupling springs. Thus, disregarding the power associated with the rotational transmission coordinate has no noticeable effect on power transmission in the connected beam structures. Moreover, the power calculations for this arrangement with and without cross-coupling terms gave virtually the same result at medium and high frequencies. This indicates that the local and global transfer mobilities can be neglected without sacrificing the accuracy in the associated much simpler calculations. The power transmission was also examined for a more complex two-dimensional source structure, which was modelled by using ANSYS-software. With this model the vibration isolation effectiveness was calculated for various values source feet thicknesses and tri-axial isolator properties, although modelled as discrete, massless springs.



Two-dimensional machine-source structure mounted resiliently onto a receiving beam.



Vibration isolation effectiveness for a source structure with feet thickness of 10 mm and for different values of axial spring stiffness; the receiver beam has a thickness of 25 mm.

Evaluation of criteria for echo detection

Silvia Alvaredo-González

Supervisors: *Anders Christian Gade* and *Finn Jacobsen*

Reflections in large rooms are sometimes perceived as disturbing echoes, and it might be useful if such echoes could be detected objectively. Two criteria for detecting echoes based on analysis of impulse responses have been implemented: the Dietsch and Kraak criterion, which is based on analysis of the steps due to reflections in the build-up function, and a ‘new criterion’ developed at Norwegian University of Science and Technology in Trondheim, based on the convolution between energy room impulse response and a Hanning window which simulates the integration properties of the ear. Simulations with the Odeon room acoustic model of an auditorium in Vigo, Spain, and a sports hall at DTU were carried out, and measurements were also made in the sports hall. Furthermore, listening tests were carried out in order to be able to compare the objective measurements and simulations with subjective judgements. The comparisons showed that adjustment of the algorithms calculating the risk of echo was needed both for use with simulated and measured impulse responses. Therefore, modifications of the ‘new criterion’ were suggested that seem to solve the mismatch problem. Nonetheless, problems due to the curve that simulates the sound pressure decay in the ear also need to be addressed.

Compensation for nonlinearities in transducers
Martin Andersen
Supervisor: *Finn Agerkvist*

This work was concerned with the topic of compensation algorithms for the nonlinearities in transducers for use in high-fidelity loudspeaker. The general ideal loudspeaker model has been examined. Furthermore, nonlinearities that influence the performance of the loudspeaker have been described, and the effects they have on loudspeaker performance have been measured. The loudspeaker model was then transformed into the digital domain, in preparation for constructing a control system. In order to make the model complete, appropriate functions were fitted to the nonlinearities and added to the model. Furthermore, a text based toolbox for Matlab was made for simulations and evaluations of different properties in both the loudspeaker and the compensation algorithm. Finally, two feedforward controller systems were developed. The first was the state space compensator and the second was based on the mirror filter derived by Wolfgang Klippel.

Most of the project was carried out at Informatics and Mathematical Modelling, DTU, with Jan Larsen as the main supervisor.

Compression in hearing aids and speech intelligibility
Karsten Raahauge Bonke
Supervisors: *Torben Poulsen* and *Erik Schmidt*

Two models of dynamic range compression systems have been constructed and tested. The first model uses 'conventional' multi-channel dynamic range compression implemented in most commercially available hearing aids today. The second model draws inspiration from cochlear modelling and comprises an adaptive filterbank based on the gammachirp approximation to the human auditory filters.

The functionality of the two models was verified with regard to level dependence and timing issues. A test matrix was decided upon such that the first model would be used to generate mildly compressed signals (compression rate (CR) = 1.5:1) and relatively strongly compressed signals (CR = 3:1). Both CRs were combined with two sets of time constants. The second model was used for signal synthesis with configurations roughly equivalent to CRs of 1.5:1 and 2:1.

Two listening tests were performed with normal hearing listeners. The first was a speech intelligibility test with sentences in noise, and the second was a subjective rating of sound quality and speech intelligibility using scales from 0 to 10. Relative positions of the estimated speech recognition thresholds and the 70%-intelligibility points on a fitted psychometric function were used for the evaluation. Furthermore the average ratings of sound quality and speech intelligibility were also used for the evaluation. It was not possible to determine which of the models that performed better in a general sense from the speech intelligibility investigation. However, it was clear that fast compression with large compression ratios, processed by the conventional model, is most detrimental to speech intelligibility, compared with all the other processing forms.

The sound quality investigation indicated that the signals processed by the second model were rated similar to the strongly compressed signals from the first model. The tests subjects' subjective assessment of speech intelligibility in comparison with the data from the speech intelligibility test revealed that the signals processed by the second model were rated worse than they actually performed.

Acoustics on orchestra stages
Anna-Clara Culmsee
Supervisor: *Anders Christian Gade*

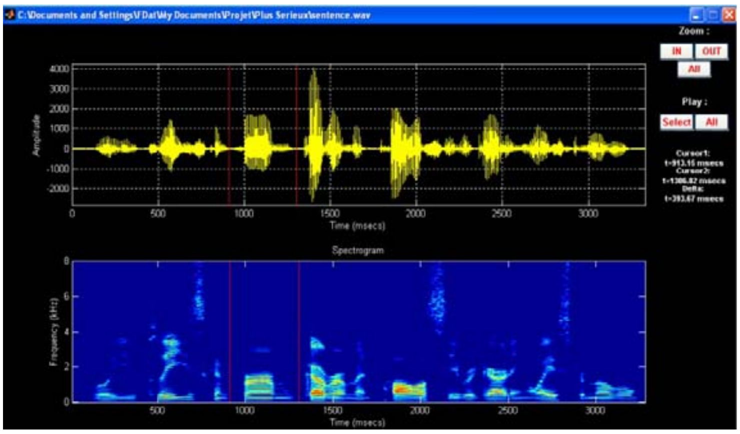
The background for this project was the design of the new arena shaped concert hall in Copenhagen that will have an orchestra stage with few early reflections being transmitted between the orchestra members. According to current knowledge, this should result in challenging conditions for ensemble playing in this hall. However, experience from other halls seems to indicate that this might not necessarily be the case. (Fortunately!) Thus, it might be possible that an efficient transmission of the direct sound—facilitated by placing most of the musicians on a semi cone shaped riser system—can provide sufficient cues for ensemble playing. Studies of the literature and modelling of various hall shapes in the ODEON program formed the basis for experiments in which violin players listened to each other while playing in simulated acoustic conditions created with electroacoustic equipment in the anechoic room. Because of problems with the equipment and shortage of suitable test subjects, the results were not very clear; but a tendency for preference of sound fields in which the direct sound was efficiently transmitted was observed.

Testing micro-electro-mechanical microphones
Olivier Desplechin
Supervisor: *Finn Jacobsen*

There are no established methods of testing and calibrating micro-electro-mechanical (MEMS) microphones. In this project a number of different methods of calibrating MEMS microphones have been examined theoretically and, in most cases, experimentally. The calibration arrangements include a small cavity, a plane wave tube, and a free field method that requires an anechoic room. Methods of determining the distortion of the microphones, the directivity, and the influence of vibration have also been examined.

The project was carried out in cooperation with Sonion, with Patrich Scheeper as the supervisor from the company.

Software analysis tool of audiovisual speech signals
Sébastien Duchaine and Florian Paez
Supervisor: *Hans-Heinrich Bothe*



Audio-visual speech analysis.

The goal of this project was the design and implementation of a software system as a frame-work program for automatic or interactive analysis of video sequences with audiovisual speech signals. Video sequences with ‘speaking faces’ are already at hand from previous projects for English and for German. The new software tool should be used to analyse acoustic speech signals and corresponding movements of the speaking face, and to correlate the corresponding acoustic and visual data.

The project is embedded in a larger concept of designing a training aid for speech-reading for hearing-impaired or deaf people, together with designing algorithms for automatic audiovisual speech recognition. There is also the possibility of using an audiovisual speech synthesis system to support the tuning process of hearing-aids or cochlea implants.

Estimating room reverberation time with a minimal knowledge of the excitation signal
Pierre Guiu
Supervisor: *Finn Jacobsen*

The purpose of this work was to examine a method of estimating reverberation time of rooms without knowledge of the excitation signal. An estimate of the reverberation time would be advanta-geous for many applications, eg intelligent hearing aids and mobile phones. However standard measurement techniques make it difficult to acquire room acoustic parameters under real-life situations. With this in mind an automatic algorithm for estimating the reverberation time has been developed. The algorithm requires no prior knowledge of the surrounding space and operates with real-life signals such as speech or music. The basic idea of the estimation procedure is first to locate suitable signal segments and then analyse the reverberation process. This second step is inspired by the works of Ratnam *et al*. and consists of fitting, in a maximum-likelihood sense, a model to a signal recorded under reverberant conditions. The performance of the algorithm has been evaluated both with simulated signals and with real recordings. The results show a very good agreement between the estimated reverberation time and measurements, even though there is some degree of variability between results obtained with different test signals.

A nonlinear signal processing model of the auditory system
Ole Hau
Supervisors: *Stephan D. Ewert* and *Torsten Dau*

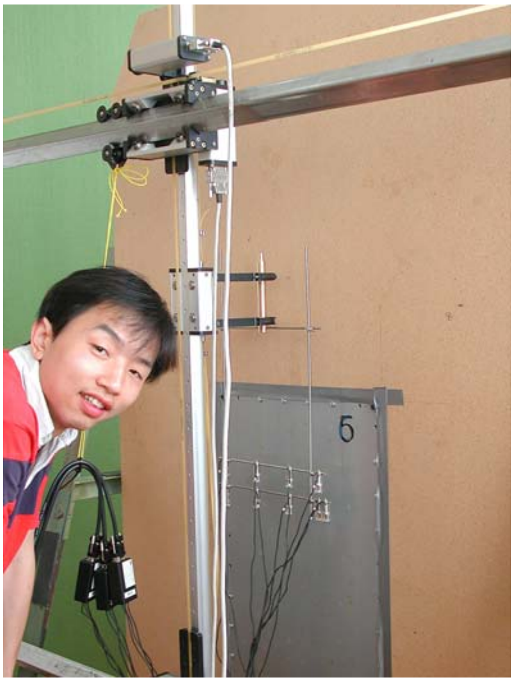
Two well established models of auditory processing, the perception model (PEMO, Dau *et al*. 1997a, b) and the temporal window model (TW, Oxenham and Moore 1994) were integrated in a common framework and compared. The common framework shares a nonlinear auditory filter stage using the dual-resonance, nonlinear filter proposed by Meddis *et al*. (2001) and the optimal-detector decision stage of the PEMO. The choice of a realistic nonlinear auditory filterbank is a prerequisite for the usability of the resulting model when sensorineural hearing loss is studied. The models were tested in conditions of simultaneous and forward masking to characterise the model's performance. The results provided valuable insights in similarities of the basic structure of the two models that have been so far regarded as very different approaches in the literature. It was found that the adaptation loops of the PEMO and the temporal window mechanism in connection with the detection criterion used in the TW model show essentially equivalent behaviour. The results of this project might serve as a basis for a new, more refined model of auditory processing.

Non-uniform meshing for improving boundary element calculations on head and torso simulators
Isaac Iglesias-Faustino
Supervisor: *Finn Jacobsen*

Different approaches to improving the quality of boundary element method (BEM) calculations of head and torso simulators have been examined. The quality improvement consists of reducing either the error or the calculation time. The meshes used for these purposes were uniform with respect to the size, degree and order of the elements employed. The work has also involved several implementations and simulations with nonuniform meshes. The advantages and disadvantages of the non-uniform meshes have been investigated for some simple test cases. An existing software package running over MatLab, OpenBEM, was used, letting not only general three-dimensional but also axisymmetrical bodies be analysed. Modifications and extensions to the existing OpenBEM code were also implemented and tested during the development of the project. Throughout the work, the global nature of the BEM method has been confirmed. It also is concluded that improving the mesh in the areas exposed directly by the sound field, either by increasing the amount of elements or the degree of them, yields satisfactory results.

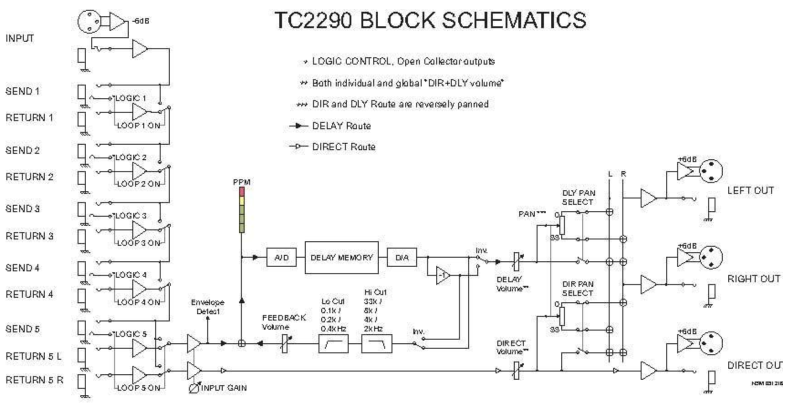
Near field acoustic holography using a velocity array
Yang Liu
Supervisor: *Finn Jacobsen*

Near field acoustic holography makes it possible to reconstruct three-dimensional sound fields from data measured in a plane close to a source. The usual method involves measuring the sound pressure in a plane with a microphone array. However, a particle velocity transducer called the Microflown has recently become available, and thus the purpose of this investigation has been to compare the performance of velocity-based holography with conventional pressure-based holography. Both simulations and experiments showed that the normal component of the particle velocity decays faster towards the edges of the measurement plane than the sound pressure. This is an advantage because less spatial filtering is needed. Moreover, when the particle velocity is predicted close to the source on the basis of the sound pressure in a plane further away, higher spatial frequency components corresponding to evanescent modes are not only amplified by the distance, but also by the wavenumber in z-direction, and thus there is a danger of aliasing. By contrast, when the sound pressure is predicted close to the source on the basis of the particle velocity in a plane further away, higher spatial frequency components are reduced. For the same reason particle velocity-based near field holography is less affected by transducer mismatch errors than conventional pressure-based holography. Thus all in all a velocity array will generally perform better.



Yang Liu and his experimental setup with a robot borrowed from Brüel & Kjær and an array of p-u sound intensity probes borrowed from Microflown.

Simulation of an analogue music effects control processor
Jan Marguc
Supervisor: *Hans-Heinrich Bothe*



Block diagram of TC2290.

A software simulation of a well-known analogue musical device from 1986 has been designed and implemented in real-time. It runs within software-based music production environments for PCs supporting the VST standard. The typical distortions of the original analogue electronics realisation turned out to be very successful in music production and were thus to be emulated by a software solution. The modelling is based on non-intrusive black-box measurements using carefully selected input signals for each conceptual part of the device. The prototype runs so far on Intel microprocessors under MS Windows, but the source code is prepared to be implemented on a DSP-based hardware platform.

The project was carried out in cooperation with TC Works, Hamburg.

Comparison of binaural and monaural loudness summation for different of signals
Antoine Maugy
Supervisor: *Torben Poulsen*

The project was concerned with monaural and binaural loudness. The method of magnitude estimation was used to evaluate the loudness of signals presented binaurally and monaurally. Three noise bands (40 Hz, 160 Hz and 1000 Hz) with centre frequency 2 kHz, a speech signal, and two music signals were used as stimuli. Nine test subjects participated in the investigation, and both headphone and free field presentation was used. Fitted loudness functions showed different exponents. Because of large individual differences and some calibration problems, no firm conclusions could be drawn from the investigation.

Adaptive control of room modes
Esteban Olmos
Supervisors: *Finn Agerkvist* and *Finn Jacobsen*

The unevenness of the low frequency response in small rooms is a problem for critical listening purposes, and this problem has been approached using many different techniques. The use of passive means is prohibitive because it requires the use of large absorbers at low frequencies. Active control methods have been widely used for the improvement of the reproduction in high-quality sound spaces. Modal equalisation, a novel technique, addresses the reduction of the decay time of target normal modes by means of second order IIR filters. Simulations of the sound field in a room generated by a point source were implemented for modelling the performance of the method. The sensitivity and robustness of the method were analysed, and measurements were conducted at the facilities of AT with a view to recording pre and post filtered signals after an offline signal processing. The technique was successful and had an adequate spatial robustness but proved to be highly sensitive to the centre frequency of the resonance to be controlled.

Vibration isolation in hand-held power tools
Mark Christopher Orton
Supervisor: *Mogens Ohlrich*

The hand arm vibration syndrome, sometimes referred to as 'vibration white fingers', is a problem facing many industries. Operators of hand-held power tools are especially at risk, and manufacturers of such tools are making efforts to reduce the vibration transmitted to the hand. Today, a frequency-weighted acceleration of less than 2.5 m/s^2 allows for a maximum exposure of 8 hours per working day (ISO 5349), but as the frequency-weighted acceleration of most angle grinders fitted with a standardised imbalance is higher than this, the maximum exposure is necessarily less than 8 hours per working day.

The objective of this study has been to reduce the hand-transmitted vibration of a specific angle grinder. A lumped parameter model with a vibration isolated handle was developed, and this model was used for simulating different isolator designs. The model included inertia properties and gyroscopic effects of rotating parts as well as the frequency dependent grip-impedances of the human hand-arm system. Since both the handle and the tool have six coordinates of motion the resulting system has twelve degree-of-freedom. Values for translational motion grip-impedances are provided by ISO 10068; values for rotational motion impedances were determined experimentally in the project.

The results showed that the hand-arm system has both a mass-like and damper-like behaviour at low frequencies, but becomes mostly damping controlled above 200 Hz. Six different vibration isolated handle setups were tested experimentally, and the vibration isolation effectiveness was found to be in reasonably good agreement with theoretical predictions based on the lumped parameter model. All the test-handles reduced hand-transmitted vibration when the tool was gripped by an operator, even when one of the test handles had a resonance frequency similar to the operational speed of the tool. The hand-arm system was found to be necessary for the prediction of the tool motion, whereas the gyroscopic effect was shown to be negligible.

This project was carried out in cooperation with Black and Decker Inc., USA, with Jeff Mahn as the industrial contact.

Relation between speech intelligibility, temporal auditory perception and auditory-evoked potentials
Alexandra Papakonstantinou
Supervisors: *Torsten Dau* and *Oliver Fobel*

Measurement of the auditory brainstem response.

The relationship between psychophysical experiments, speech intelligibility in noise and evoked brainstem responses in high-frequency hearing-impaired listeners and normal-hearing listeners was investigated. Psychophysical experiments related to temporal processing and the band width of auditory filters were performed in the low-frequency region where hearing-impaired listeners showed normal pure-tone thresholds. The auditory brainstem responses (ABR) elicited by a click and a chirp that compensates for cochlear travel-time difference across frequency were recorded. Hearing-impaired listeners with reduced temporal processing abilities showed reduced speech intelligibility in noise. Click-evoked responses were similar between listeners with significant differences in speech intelligibility, while the chirp-evoked responses were severely reduced for the listeners with reduced temporal processing performance. The chirp might therefore be useful in clinical ABR diagnostics since it demonstrates the temporal processing abilities of the auditory system better than the click.

Melody recognition with binaural-pitch stimuli in normal-hearing and hearing-impaired listeners
Sebastien Santurette
Supervisors: *Torsten Dau* and *Oliver Fobel*

When two white noises differing only in phase in a particular frequency range are presented simultaneously each to one of our ears, a pitch sensation may be perceived inside the head. This phenomenon, called ‘binaural pitch’ or ‘dichotic pitch’, can be produced by frequency-dependent interaural phase-difference patterns. Their evaluation depends on the functionality of the binaural auditory system and the temporal fine-structure information at its input. Several experiments were performed with normal-hearing and hearing-impaired listeners, including the detection and discrimination of binaural pitch, and melody recognition using different types of binaural pitches. For the normal-hearing listeners, all types of binaural pitches could be perceived immediately and were musical. The hearing-impaired listeners could be divided into three groups based on their results: a) some perceived all types of binaural pitches, but with decreased salience or musicality compared to normal-hearing listeners; b) some could only perceive the strongest pitch types; and c) some were unable to perceive any binaural pitch at all. The performance of the hearing-impaired listeners was not correlated with audibility. The experimental results were only to some extent consistent with predictions from the modified equalisation-cancellation model (Culling *et al*., 1998). Overall, binaural pitch stimuli could be very useful within clinical diagnostics for detecting auditory deficits that are linked to temporal processing of sounds.

Monaural and binaural processing of fluctuating sounds in the auditory system
Eric Thompson
Supervisor: *Torsten Dau*

The temporal resolution of the monaural system has been well studied in the literature, but there have been few similar studies on the ability of the binaural system to follow fluctuations in sound intensity and those studies have often failed to differentiate between monaural and binaural cues. In this project, psychoacoustic measurements were made of the temporal resolution and frequency selectivity in envelope processing of the monaural and binaural systems. These measurements showed similarities that suggest that mechanisms already used in models of the monaural auditory system might be employed in extending those models into the binaural domain. Preliminary tests with a widely accepted binaural model also showed that further development is required in order to account for the measured binaural temporal resolution.

Synchronous compression in binaural hearing aids
Jinjing Xu
Supervisor: *Torben Poulsen*

The purpose of this project was to test the benefits from binaurally fitted hearing aids. Recordings of speech and noise were made with a head and torso simulator. The recordings were used in headphone based speech intelligibility tests using the DANTALE II speech material. A non-significant difference in the speech recognition threshold was found between the two models. The report is confidential.

The work was carried out in cooperation with Lars Bramsløw, Oticon.

2.4 TEACHING OUTSIDE OF DTU

Transducers and Distortion (*Finn Agerkvist*)

One week course at New Jialian Electronics, in Jiashan, China, 17-21 October 2005, given as a part of the ongoing collaboration with New Jialian Electronics on transducers for mobile phones.

Room Acoustic Scale Models (*Anders Christian Gade*)

Lecture at University of Milano, Italy, 23 May 2005.

Concert Halls and Other Halls, Theory and Examples (*Anders Christian Gade*)

Half day course at Lund University, Sweden, 11 October 2005

Short Course on Sound Intensity (*Finn Jacobsen*)

Half day course for staff and students of Departamento de Mecânica Computacional, Universidade Estadual de Campinas, Brazil, on 16 August 2005.

Appendix A: Extramural Appointments

Finn Agerkvist
Member of Danish Acoustic Society's Technical Committee on Electroacoustics

Hans-Heinrich Bothe
Member of the International Academic Advisory Council, Natural and Artificial Intelligence Systems Organization (NAISO), Canada
Member of the Editorial Board of *International Journal of Computer Research*, NOVA Science Publishers, Commack, NY, USA
Senior member of Institute of Electrical and Electronics Engineers (IEEE)

Torsten Dau
Member of the Technical Committee for 'Psychological and Physiological Acoustics' in the Acoustical Society of America
Member of the Technical Committee for 'Hörakustik' in the German Acoustical Society (DEGA)
Member of the Scientific Advisory board of the 'Hanse Institute for Advanced Study' in Germany.
Co-Organiser of regular 'Hanse workshops on hearing research'.

Anders Christian Gade
The Rockwool Price, Member of the Board

Finn Jacobsen
Member of the Editorial Board of *International Journal of Acoustics and Vibration*
Member of the Scientific Committee for *Twelfth International Congress on Sound and Vibration*, Lisbon, Portugal, July 2005
Member of the Scientific Committee for *Thirteenth International Congress on Sound and Vibration*, Vienna, Austria, July 2006
Member of the Board of Directors of *International Institute of Acoustics and Vibration*
Section leader of *Handbook of Signal Processing in Acoustics*, Springer Verlag, ed. by D. Havelock, S. Kuwano and M. Vorländer
Assessor for Grant Agency of the Academy of Science of the Czech Republic (evaluation of proposal)
Assessor for RMIT University, Melbourne, Australia (academic promotion)

Mogens Ohlrich
Member of ISO/TC 43/SC 1/WG 22, 'Characterisation of Machines as Sources of Structure-borne Sound'
Member of the Organising Committee for *1st International Operational Modal Analysis Conference*, Copenhagen, April 2005

Torben Poulsen
Member of the Board of the Danish Acoustical Society
Member of ISO/TC 43/WG l, 'Threshold of Hearing'
Convener of ISO/TC 43/SC 1/WG 17, 'Methods of measurements of sound attenuation of hearing protectors'
Chairman of the board of *Danavox Jubilee Foundation* (Research in Audiology).
Member of the organising committee for the *21st Danavox symposium* 'Hearing aid fitting'
Official opponent at Per Hiselius PhD defence, 'Acoustical properties of earplugs'. Lund University, Sweden (January 2005)

Knud Rasmussen
Chairman of IEC Technical Committee 29, 'Electroacoustics'
Convener of IEC TC29/WG5, 'Measurement microphones'
Member (via DFM) of CCAUV, Consultative Committee for Acoustics, Ultrasound and Vibration (of the BIPM)
Contact person of EUROMET Acoustics
Technical Manager of DPLA, Danish Primary Laboratory of Acoustics
Chairman of S529, 'Electroacoustics', Danish Standards Association
Technical assessor for United Kingdom Accreditation Service (UKAS)
Technical assessor for National Agency for Testing and Accreditation (NATA), Australia

Jens Holger Rindel
Chairman of the Technical Committee for Building and Room Acoustics of EAA (the European Association of Acoustical Societies)
Director of Odeon A/S
Member of the Editorial Board of Applied Acoustics
Convener of ISO/TC 43/SC 2/WG 19, 'Measurement of reverberation time'
Member of ISO/TC 43/SC 2/WG 24, 'Application of new measuring methods in building acoustics'
Convener of ISO/TC 43/SC 2/WG 25, 'Measurement of the random-incidence scattering coefficient of surfaces'
Convener of CEN/TC 126/WG 8, 'Field measurement of reverberation time'
Member of ISO/TC 43/SC 2/WG 7, 'Measurement of absorption coefficient in a reverberation room' (revision of ISO 354)
Member of ISO/TC 43/SC 2/WG 18, 'Measurement of sound insulation in buildings and of building elements' (revision of ISO 140)
Member of CEN/TC 126/WG 1, 'Methods for measuring the sound insulation of building elements and the acoustic performances of buildings'
Member of the International Advisory Committee for *Inter-Noise 2006*, Honolulu, USA, December 2006
Official opponent at Ann-Charlotte Johansson's Doctoral disputation, 'Drum sound from floor coverings—Objective and subjective assessment'. Lund University, Sweden (June 2005)
Official opponent at Klas Hagberg's Licentiate disputation, 'Evaluation of sound insulation in the field'. Lund University, Sweden (September 2005)
Member of the thesis committee for Cheol-Ho Jeong's PhD defence, 'Prediction of acoustical characteristics of the enclosed space using the phased beam tracing method'. Korean Advanced Institute of Science, Korea (July 2005)
Member of Supervisory Committee for Jonathan Rathsam's PhD study at University of Nebraska, USA.
Appointed nominator for INDEX: Award 2005
Assessor for the Dutch Technology Foundation STW, the Netherlands
Assessor for the Engineering and Physical Sciences Research Council, UK
Assessor for Helsinki University of Technology, Finland (academic promotion)

Appendix B: Principal Intramural Appointments

Mogens Ohlrich
Head of Acoustic Technology, one of the six sections of Ørsted•DTU
Coordinator of the International MSc Programme in Engineering Acoustics

Torsten Dau
Member of DTU's PhD Progamme Committee *Electronics and Communication*
Member of DTU's PhD Study Committee