N A DAGA
Rotterdam 2009

NAG/DAGA 2009

International Conference on Acoustics
Rotterdam
23 – 26 March 2009

P R O G R A M

DEGA

nederlands akoestisch genootschap
NAG

ABAV
Association Belge des Acousticiens

EAA
European Acoustics Association

NAG/DAGA 2009
International Conference on Acoustics, Rotterdam
including the
35. German Annual Conference on Acoustics (DAGA)

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- Acoustical Society of the Netherlands (NAG)
- German Acoustical Society (DEGA)

In co-operation with:
- Belgian Acoustical Society (ABAV)
- European Acoustics Association (EAA)

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- Information Technology Society (ITG in VDE)
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	Precolloquium **Array technology in audio and acoustics**
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15:20	Anton Schlesinger et al.: volumetric measurements with spherical microphone array (45)
15:40	Closing by colloquium chair
16:00	Busses leave from the Aula Congress Centre to Delft central train Station for connection to Rotterdam central train station

17:00, Willem Burger Zaal: **DEGA members general assembly**

Tuesday, March 24

9:00 Willem Burger Zaal:		**Opening Ceremony**
	followed by	**Coffee break**
11:00	Willem Burger Zaal:	**Plenary lecture T. Houtgast: “The acoustical engineer as a researcher in speech and hearing“** (35)
11:40	Willem Burger Zaal:	**Product forum** (35)
	followed by	**Lunch break, exhibition opening with sandwiches**

Room	Will. Burger	Jurriaanse	Fortis Bank	V. Cappellen	V. Beuningen	Schadee
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15:20	Coffee/poster	Taskan: heavy stairs (53)	Coffee/poster	Müller-Deile: bilateral CI (64)	Coffee/poster	Wang: Gaussian Select. (75)
15:40	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster
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					Vehicle Ac.	
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19:00: Bus starts for **Congress dinner** including music with jam-session

Room	Ruys	Mees	V.Rijckevorsel	Plate	Van der Vorm	Hudig
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15:20	Coffee/poster	Kohlrausch: Early binaural research (86)	Coffee/poster	V. Antwerpen: Window noise simul. (97)	Coffee/poster	Kuijpers: Validation Model (108)
15:40	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster
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		Teaching and educ.		**FE models**		
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Poster session Tuesday

(Authors will be present: 15:20 - 16:20)

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Mohamed Elmahdy et al.	'Survey on common Arabic Language forms from a speech recognition point of view' (113)
Christoph Fritzsche et al.	Silent Owl Flight: Setup for Fly-Over Noise Measurements (113)
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Marko Horvat et al.	Influence of Short Term Noise on Concentration and Human Performance (115)
Kristian Jambrosic et al.	The design of an in-situ absorption measuring system using the Adrienne method (115)
Igor Kirichenko et al.	To the Question of Adaptive Acoustic Systems' Synthesis (116)
F. B. Konkel et al.	Sound Radiation of Double Reed Woodwinds (116)
Helmut Lang et al.	Comparing Acoustic Model Adaption Methods for Non-native Speech Recognition (117)
Christian Lüke et al.	Feature Extraction for Speech Recognition (117)
Robert Marin et al.	Direct sound insulation of double-leaf separating walls (118)
Rein Muchall	The measurement uncertainty and equivalence of automatic unmanned noise monitoring installations (118)
Sven Öhler et al.	Effect of airborne sound on installation noise - Part 2: Practical application (119)
Marie-Agnès Pallas et al.	Nearfield noise source localisation with constant directivity arrays: a comparison (119)
Hendrik Söhnholz et al.	Controllable acoustic bubble traps (120)
Moritz Späh et al.	Noise control by hedges and woods (120)
Irina Starchenko et al.	Design and Technology Features of Mosaic Pump Transducer of Parametric Array (121)
Michael Wilsdorf et al.	Climatological and regional analysis of sound level attenuation (121)
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Wednesday, March 25

Room	Will. Burger	Jurriaanse	Fortis Bank	V. Cappellen	V. Beuningen	Schadee
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09:00	Ahrens: Coloration (123)	Geebelen: Rainfall Noise (132)	Schanda: open-plan offices (140)	Wahler: AUVA - electronic report (148)	Rychtarikova: Acoustical Categori-zation (156)	Steinert: Wideband Quality Assessment (165)
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12:40		**Lunch break**
14:00	Willem Burger:	**Plenary lecture T. ten Wolde: "Reciprocity measurements in acoustical and mechano-acoustical systems. Review of theory and applications."** (36)

Room	Ruys	Mees	V.Rijckevorsel	Plate	Van der Vorm	Hudig
	Railway noise 1	**Tram noise**	**Transferpath analysis**	**Boundary elements 1**	**Audio technology**	**Source identification 1**
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	Phy. of mus. instrum. 1					
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09:40	Maloney: The Aubrapan (175)	Groß-Thebing: Curving Tram Noise (183)	Quickert: Methods for TPA (191)	Piscoya: Acoustic/ hydrodyn. splitting (200)	Schaer: Measurement Nonlinearities (208)	Aarts: Estimating acoust. quantities (218)
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10:20	Grothe: Basson Bocal (176)	Fehndrich: Dirt on Wheel Damper (184)	Coffee break	Abboud: Time Domain BEM (201)	Coffee break	Müller: FMM SEA excitation (219)
					Bioacoustics	
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Wednesday, March 25 (continued)

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15:40	Melchior: Spherical array systems (128)	Gerretsen: Acoustic Quality Classes (137)	Buchschmid: room acoustical simulations (145)	Conter: Noise Reducing Devices (153)	Meloni: Determ. structure-borne noise (161)	Rohrer: VoIP testing (170)
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16:20	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster
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17:20	Groth: 'Sweet Spot' Adjustment, part 2 (130)	Alphei: Speech intelligibility, part 2 (138)	Goetze: System Delay Equalization (147)	Granneman: Whistling Building objects (155)	Krahé: Auditory Model (163)	Möller: Synth. Speech Quality (172)
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18:00	Klein-Hennig: Envelope based lateralization (131)	Wittstock: uncertainity of predictions (139)			Sancak: Fiber and Composites (164)	Hyder: 3D conference calls (173)

18:00: **Conference reception** in De Doelen
20:15: **Concert** in De Doelen

Room	Ruys	Mees	V.Rijckevorsel	Plate	Van der Vorm	Hudig
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15:20	Carral: Bagpipe chanter simulation (179)	Von Estorff: Nonlinear EIF-Approach (186)	Boonen: ILC-control exhaust noise (195)	Schirmer: Delivery Trucks (204)	Ocker: Sunroof Buffeting (212)	Van d. Wal: Localization with Arrays (222)
15:40	Rolf: Guitar Radiation Pattern (179)	Von Heesen: Aircraft Engine Test Stand (187)	Sobon: Noise, Impedance Control (196)	Martarelli: Characterization hoses (204)	Krampol: Turbulent noise synthesis (213)	Hundeck: Array Acoustics SONAH (222)
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16:20	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster	Coffee/poster
16:40	Fleischer: Church bell (180)	Coffee/poster	Norambuena: Active absorption systems (196)	Coffee/poster	Schram: Industry Aeroacoust. Pred. (213)	Coffee/poster
17:00	Logie: Lip-reed transient behaviour (180)	Groß: Flow Field of Cutter (188)	Misol: ASAC for Windshield (197)	Terrasse: FMM in Acoustic (205)	Clasen: Side Window Noise (214)	Wind: stiffener contributions (223)
17:20	Stevenson: Lip-reeds: loud playing (181)	Giesler: Broadband noise investig. (188)	Schirmacher: Error Microphone Placement (197)	Schram: Ducted Fan Noise (206)	Piellard: Internal Noise Modeling (214)	Van Wijngaarden: Inverse Source Strength (224)
					Cavitation 1	
17:40	Richter: Resonance frequencies (181)	Geyer: Silent owl flight (189)	Kleinhenrich: FDTD ANC Simulation (198)	Rieckh: Orthotropic media vibr. (206)	Nowak: Bubble Jetting (216)	Jaeckel: Covered Microphones (224)
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Poster session Wednesday

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Fabian Brackhane	Perception tests with a replica of von Kempelen’s speaking machine (227)
Holger Brokmann	Schulergonomie – Von der subjektiven Empfindung bis zur objektiven Beeinflussbarkeit (228)
Hans-Elias de Bree et al.	PU nosecone intensity measurements in a windtunnel (229)
Sandra Buss et al.	Transmission of noise through a pipe as structure-borne sound and fluid sound (229)
Martin Eichler et al.	Superdirective Beamforming Using an Extended Modal Subspace Decomposition (230)
Alexander Geerlings et al.	Acoustic services at Philips Applied Technologies (230)
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Guus Klamerek et al.	Good Room Acoustic Comforttm (RACTM) can be achieved by using a selection of appropriate acoustic descriptors. (231)
Milos Kodejska et al.	Ferroelectret-film accelerometers with high sensitivities (232)
Rainer Machner et al.	Eine Ära geht zu Ende – Schließung des Instituts für interdisziplinäre Schulforschung (ISF) der Universität Bremen (232)
Volker Mellert et al.	Improving transmission loss of light-weighted panels by increase of stiffness with an evacuated foil. (233)
Mitsunori Mizumachi	Smart Noise Reduction Based on Reliability of Direction-of-arrival Estimate (234)
Sergei Olfert et al.	Generation of short ultrasonic pulses using active damping (234)
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Emiel Tijs	A PU sound probe for high noise levels (235)
Isao Tokuda et al.	Estimating physical properties of vocal fold paralysis from high-speed filimg data (235)
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Huan Wang et al.	Coding of speech into nerve-action potentials (236)

Thursday, March 26

Room	Will. Burger	Jurriaanse	Fortis Bank	V. Cappellen	V. Beuningen	Schadee
	Psycho-acoustics 2	**Building acoustics 2**	**Auditory processing 1**	**Noise control 2**	**Environm. acoustics 1**	**Musical acoustics**
08:40	Blauert: Pure Psychoacoustics (238)	Breuer: Sound insulation for rock bands (246)	Weissgerber: Inverse HRTF-Filters (251)	Vos: Rating Shooting Sounds (264)	Giering: Dose-Response-Relationship (273)	Von Türckheim: Responsiveness of Violins (281)
09:00	Getzmann: Auditory motion (238)	Ruff: Lightweight gypsum walls (246)	Buchholz: room auralisation (251)	Knauss: Noise Management (264)	Probst: Environmental Noise (273)	Maloney: Modal Analysis Steelpan (281)
09:20	Schlittmeier: Modelling disturbing sounds (239)	Lentzen: lightweight junctions (247)	Le Goff: detection and spectral integr. (252)	Hammelmann: Auswirkg. Lärmmanagement (265)	Wolfert: Noise Challenges of Cities (273)	Trommer: reflection of pipes (282)
				Active noise control 2		
09:40	Haverkamp: Synesthetic Sound Design (239)	Koopman: lightweight junctions (247)	Van de Par: Tone complex lateralization (252)	Graf: ANR-Headset (265)	Desanghere: Quiet City Actions (274)	Ausserlechner: Pipe foot model (282)
10:00	Koelewijn: Auditory capture (240)	Völtl: drywall sound transmission (248)	Könsgen: Binaural Jitter ITD (253)	Kochan: ANC Controller Parametr. (266)	Luzzi: Harmonized Action Plans (274)	Angster: wooden organ pipes (283)
10:20	Coffee break	Saß: triple glass (248)	Coffee break	Hoever: active control impacts (266)	Coffee break	Mattheij: String Vibrato (283)
10:40	Menzel: Crying colours (240)	Coffee break	Dietz: Azimuthal Localiz. Model (253)	Coffee break	Van Beek: Eval. Dutch motorway-noise (275)	Coffee break
						Speech perception 3
11:00	Janse: Ignoring competing speech (241)	Wenmaekers: Sound Insulation Water (249)	May: Probabilistic Localization Model (254)	Konle: FOM Design and Applic. (267)	De Vos: global noise exposure (275)	Jürgens: Speech Intelligibility Model (284)
11:20	Treiber: Rotary Switches (241)	Peperkamp: thermally activated slabs (249)	Seeber: Density and precedence (254)	Bös: sound transmission window (267)	Condorachi: Iasi noise mapping (276)	Meyer: Speech Intelligibility Prediction (284)
11:40	Zekveld: cognition & speech compreh. (242)	Reichelt: TMD for Wooden Floors (250)	Kerber: Localisation dominance in background noise (255)	Bös: Loewe-Zentrum AdRIA (268)	De Ruiter: Urban Noise Control (276)	Friedrich: Optimising Microph. MRI (285)

12:00	Willem Burger:	**Plenary lecture: Winner of the Lothar Cremer Award**
12:40		**Lunch break**
14:00	Willem Burger:	**Plenary lecture T. Dau: “Recent concepts and challenges in hearing research“** (37)

Room	Ruys	Mees	V.Rijckevorsel	Plate	Van der Vorm	Hudig
	Railway noise 2	**Voice production 1**	**Aero-acoustics 1**	**Signal processing**	**Cavitation 2**	**Source identification 3**
08:40	Windelberg: Basic Sound Level (290)	Yang: Vocal Fold Modelling (298)	Schram: jet-airfoil noise (308)	Bach: Acoustic Object Detection (315)	Jung: Threshold Trans. Cavitation (324)	Müller-Trapet: Simul. Beamforming Meas. (332)
09:00	Salz: meas. and calculation (290)	Loch: Vocal Fold Oscillation (298)	Smith: Flow induced scattering (308)	Friedrich: Classif. therapeutic sounds (316)	Otto: Cavitation Bubble Arrays (324)	Simanowski: Acoustic Ground Tests (332)
09:20	Möhler: Railway noise prediction (291)	Zörner: Human Phonation Process (299)	König: Slat Source Identification (309)	Jaillet: STFT Phase Structure (316)	Kling: Parameters sono-chemistry (324)	Van de Vooren: rotation machinery (333)
						Sound propagation
09:40	Arndt: Calcul. wheel dampers (291)	Gömmel: glottal flow calculation (299)	Kaltenbacher: Fluid-Structure Interaction (309)	Köhler: Rotating Processes (317)	Kurz: Laser-generated cavitation (325)	Randriano-elina: Meteorolog.-acoust. model (333)
10:00	Dittrich: combined wheel/rail roughness (292)	Becker: Vocal Folds Model (300)	Boersma: jet noise (310)	Violanda: Signal Component Estim. (317)	Garbin: History Force UCAs (325)	Hirsch: classif. sound propagation (334)
10:20	Coffee break	Zörner: Human phonation analysis (300)	Coffee break	Wefers: Real-time filtering struct. (318)	Coffee break	Fischer: travel-time tomography (334)
10:40	Frese: Curve squeal (292)	Coffee break	Van der Spek: Quiet air cooling fans (310)	Coffee break	Davitt: Water cavitation density (326)	Coffee break
11:00	Verheijen: Railway Noise Monitoring (292)	Barney: voiced fricative modulation (301)	Bauers: Investig. of Mode Scattering (311)	Dörfler: Removing Components (318)	Versluis: Why buckling matters (326)	Ziemann: atmospheric sound prop. (335)
11:20	Behr: The Project LZarG (293)	Horacek: PIV Flow Measurement (301)	Weckmueller: Extended Mode Matching (311)	Franz: multi-channel noise red. (319)	Davaadorj: Trapping Bubbles (327)	Van Maercke: Harmonoise model (336)
11:40	Tinter: Standstill Measure-ments (293)	Triep: 3D glottal flow (302)	Moreau: liner ray structure (312)	Dietrich: Input Data for Auralization (319)	Vokurka: Spark bubbles (327)	Coffee break

Thursday, March 26 (continued)

Room	Will. Burger	Jurriaanse	Fortis Bank	V. Cappellen	V. Beuningen	Schadee
	Psycho-acoustics 3	**Auditory processing 2**	**Modelling in room ac.**	**Structural-ac. optim.**	**Environm. acoustics 2**	**Speech**
15:00	Hansen: Identif. Detection Tones (242)	Valkenier: Robust Vowel Detection (255)	Lautenbach: Ray-tracing Reflections (260)	Dannemann: Composite Plate Optimiz. (269)	Vogelsang: aircraft noise calculation (277)	Baumgartner: Interactive Fitting Wizard (286)
15:20	Bisitz: Mediastream noise redu. (243)	Brand: Binaural speech recognition (256)	Jeon: concert hall model (260)	Dunne: Vehicle Acoustic Design (269)	Myck: aircraft noise calculation (277)	Luneau: Wavelets Modul. Speech (286)
15:40	Scheibner: Sound Quality Evaluation (243)	Dreschler: Hearing instruments (256)	Stephenson: beam diffraction (261)	Andrä: Optimization of Absorbers (270)	Schäffer: Aircraft Noise Simul. using Radar (278)	Becker: Formant Speaker Verif. (287)
16:00	Guski: Algorithmic Psychoacou-stics (244)	Baumann: Beamform. Hearing Glasses (257)	Schröder: Optimization Simulation Param. (261)	Bless: 3D-FEM prediction (270)	Maschke: nocturnal aircraft noise (278)	Schwarz: Fujisaki-Model (287)
16:20	Coffee break	V. d. Bogaert: Hearing Aid Localization (258)	Coffee break	Hering: Structure-b. sound intensity (271)	Coffee break	Schnell: Time-varying Speech Analysis (288)
16:40	Weber: Pure tone salience (244)	Coffee break	Bos: auralization system (262)	Coffee break	Land: Aircraft noise (279)	Coffee break
17:00	Sontacchi: Expert Listening Panel (245)	Majdak: Median-Plane Localization (258)	Aretz: FE Boundary Characteris. (262)	Tcherniak: Transmiss. Method for TPA (271)	Janssens: Helicopter detection, part 1 (279)	Harwardt: speaker-spec. F0 changes (288)
17:20	Spors: Focussed Sources WFS (245)	Altinsoy: BRTF (259)	Nijs: auralization sports facilities (263)	Alzugaray: Two-Phase Flow Excitation (272)	Quesson: Helicopter detection, part 2 (280)	Loellmann: Reverb. Time Estimation (289)
17:40		Merchel: Audiotactile Touch Screens (259)	Pollow: Directiv. Mus. Instruments (263)	Blanchet: Vehicle Trim Modeling (272)	Trimpop: Elevated sound sources (280)	Winkler: Throat Microphone ASR (289)

18:00: **Farewell event** in Jurriaanse Zaal

Room	Ruys	Mees	V.Rijckevorsel	Plate	Van der Vorm	Hudig
	Loc. sources on vehicles	**Voice production 2**	**Aero-acoustics 2**	**Audiological acoustics**	**Ultrasound**	**Electro-acoucstics**
15:00	Deblauwe: source in interior enclosure (294)	Elemans: non-songbird (303)	Spehr: Broadband Noise Simul. (312)	Blau: individual acoustics model (320)	Reuter: pressure fields in cleaning (328)	Fedtke: Pressure reciprocity calibration (336)
15:20	Helfer: Acoustic Mirrors (294)	Tokuda: Modeling voice registers (303)	Tonon: Corrugated pipe model (313)	Sankowsky: Ear Drum Pressure (320)	Funke: Ultrasound field (328)	Budde: Fast distortion meas. (337)
15:40	Guidati: Beamform. Vehicle Acoust. (295)	Hütz: death metal singers (304)	Steger: Fan Noise Control (313)	Roeske: Modeling sound field (321)	Steger: Ultrasonic Laser Vibrometer (329)	Ress: Loudspeaker eval. simplif. (337)
16:00	Jakob: Area Contribution Analysis (295)	Verkerke: voice-producing prosthesis (304)	Busse: Cha-racterisation of Liner (314)	Otto: perceptual thresholds (321)	Müller: Extrusion in-line monitoring (329)	Steinbrecher: Calibrated Noise (338)
16:20	Coffee break	Kob: male/ female voice (305)	Coffee break	Hudde: Curved ear canals (322)	Coffee break	Schmitz: PA-systems sport stadiums (338)
	Noise				**Physical acoustics**	
16:40	Verheijen: Noise Political Issue (296)	Coffee break	Grabinger: Separ. Aero-/Vibro. Noise (314)	Coffee break	Hirsekorn: Near-Field Microscopy (330)	Coffee break
17:00	Beck: Flow-ind. Sound Radiation (296)	Schmidt: Vocal-Fold Model Optimization (306)	Hoevelmann: Acoustic Ground Tests (315)	Schmidt: ear canal field (322)	Deschee-maeker: porous materials (330)	Merkel: Interaction between Waves (339)
17:20	Tijs: asphalt acoustic absorption (297)	Mwangi: Effects of Vocal Aging (306)		Rader: Speech Intelligibility MSNF (323)	De Bree: source path contribution (331)	Ehrig: Flat Panel Transducers (339)
17:40	Ersoy: Noise Stage and Efficacy (297)				Rabold: Optimiz. lightweight floors (331)	

Welkomstwoord door voorzitters

Geachte collega's,

'Welkom' in Rotterdam! Ook willen wij graag zeggen 'Wilkommen' en 'Welcome'. De organisatoren van dit congres zijn zeer verheugd u allen, zowel Nederlandse, Belgische en Duitse acoustici als ook acoustici van andere landen, uit te nodigen voor het congres NAG/DAGA 2009. Het programma, welke bestaat uit 528 mondelinge presentaties verdeeld in 69 gestructureerde sessies, vele posters en een interessante expositie, wacht om vervuld te worden met leven, discussies, wetenschappelijke resultaten, technische oplossingen en producten.

Dit evenement combineert het 35ste DAGA congres van het Duitse Genootschap voor Akoestiek (DEGA) en de bijeenkomst van het Nederlands Akoestisch Genootschap (NAG). Het evenement wordt ondersteund door de Belgische Akoestische Vereniging (ABAV), door de Europese Akoestische Associatie (EAA) en door de Duitse associaties van ingenieurs en natuurkundigen (ITG, VDI en DPG). Dit congres is onderdeel van een serie van gezamenlijke congressen welke DEGA de gelegenheid geeft om haar buurlanden te bezoeken. In het verleden zijn er congressen georganiseerd in bijvoorbeeld Oostenrijk in 1990, in Zwitserland in 1998 en in Frankrijk in 2004. Het succes van de vorige gezamenlijke congressen willen wij graag voortzetten in Rotterdam. Verder vertrouwen wij erop dat u een aangenaam verblijf zult hebben in Rotterdam. Dit congres biedt een internationaal platform en zal niet alleen bijdragen aan een wetenschappelijke en technische ontwikkeling op gebied van akoestiek, maar ook aan samenwerking over de landgrenzen en talen heen, in onderzoek en technologie, productontwikkeling en in commerciële activiteiten.

Diegenen die regelmatige DAGA bezoekers zijn, zullen zeker vele welbekende details herkennen. Maar zij zullen ook een speciale touch van de NAG/DAGA opmerken vanwege het feit dat lawaaibeheersing strategieën in andere landen en steden op de agenda staan, om een voorbeeld te noemen. Het delen van ervaringen in verkeerslawaai, bouwakoestische normen en de implementaties van geluidkaarten zijn andere voorbeelden waar crossborder discussies erg vruchtbaar zullen zijn. We willen in het bijzonder uw aandacht graag richten op de twee precolloquia (pagina 5). Het ene precolloquium heeft de focus op Array Technology en wordt gehouden op de Technische Universiteit Delft. Het andere precolloquium zal de nadruk leggen op vragen betreffende lawaaibeheersing in Europese havensteden zoals Rotterdam. Verder zal de specifieke sfeer van Rotterdam en omgeving bijdragen aan een speciale ervaring op de NAG/DAGA.

De locatie waar het congres wordt gehouden is het welbekende concert en congres gebouw ’De Doelen’ in het centrum van Rotterdam, tegenover het centraal station en in de nabijheid van vele winkels en restaurants (zie pagina 359). In De Doelen hebben wij professionele faciliteiten tot onze beschikking, waaronder moderne sessiezalen, standplaatsen voor posters en een expositieruimte. Het is de moeite waard te vermelden dat De Doelen ook beschikt over een voortreffelijke concertzaal. Let u op dat u tijdig een ticket reserveert voor het concert op woensdag 25 maart.

Er is alles aan gedaan om het programma op een wijze te organiseren zodat aan algemene en individuele interesses wordt voldaan en rekening is gehouden met speciale verzoeken. Wat de avonden betreft, is er voldoende gelegenheid om de Rotterdamse cafés en restaurants te kunnen bezoeken, maar bieden we ook een sfeervol diner (zie pagina 32) aan. De sociale activiteiten en bijvoorbeeld de receptie op woensdagavond kunnen bijdragen aan het vormen van nieuwe en het versterken van oude vriendschappen. Technische excursies naar de Rotterdamse haven en de Large European Acoustic Facility in Noordwijk, bezoeken aan interessante culturele bezienswaardigheden zoals de Porseleyne Fles in Delft en een architectuur tour in Rotterdam maken het programma compleet. In het geval u nog geen hotel geboekt heeft, kunt u dit doen via onze website www.nag-daga.nl.

Tot slot spreken we onze bijzondere waardering uit naar alle auteurs voor het indienen van in totaal 560 mondelinge en poster presentaties. Rotterdam is een prettige stad en De Doelen een professioneel congrescentrum, maar zonder deze geweldige bijdragen aan wetenschappelijke en toegepaste akoestiek zou dit evenement niet mogelijk zijn geweest.

Tot ziens in maart!

Namens het NAG/DAGA organiserend comité,
Martijn Vercammen en Michael Vorländer

Grußwort der Tagungsleiter

Sehr geehrte Kolleginnen und Kollegen,

”Welkom”, wie die Niederländer sagen, in Rotterdam! Die Tagungsorganisation ergänzt dies mit einem ”Willkommen” und einem ”Welcome”, um die große Freude auszudrücken, niederländische, belgische und deutsche Akustiker sowie Akustiker aus der ganzen Welt in einer der modernsten Städte der Niederlande zur ”NAG/DAGA 2009” begrüßen zu können. Das Programm mit 528 Vorträgen in 69 Sitzungen, vielen Postern und einer attraktiven Ausstellung wartet, mit Leben gefüllt zu werden, mit Diskussionen zu wissenschaftlichen Ergebnissen, technischen Lösungen und Produkten.

Dieses Ereignis vereinigt die 35. DAGA-Tagung der Deutschen Gesellschaft für Akustik, DEGA, mit der Tagung der Nederlands Akoestiek Genootschap, NAG. Es wird unterstützt von der ABAV, der Akustischen Gesellschaft Belgiens, von der European Acoustics Association, EAA, und von den deutschen Vereinigungen für Ingenieure und Physiker, ITG, VDI and DPG. In der Reihe der Gemeinschaftstagungen mit Österreich im Jahre 1990, mit der Schweiz in 1998, und mit Frankreich in 2004, fährt die DEGA fort, ihre Nachbarn zu besuchen. Angesichts des großen Erfolgs der vergangenen Gemeinschaftstagungen fühlen wir uns verpflichtet, für alle Teilnehmer beste Bedingungen für eine fruchtbare und erfolgreiche Veranstaltung zu schaffen, für den dienstlichen und geschäftlichen Anteil und auch für einen angenehmen Aufenthalt in Rotterdam.

Auf dieser internationalen Bühne wird die Tagung nicht nur zum wissenschaftlichen und technischen Fortschritt der Akustik beitragen, sondern auch die Zusammenarbeit über Landes- und Sprachgrenzen hinweg fördern. Natürlich spiegelt dieser Aspekt die ohnehin vorliegende Vernetzung der europäischen und internationalen Aktivitäten in der Akustik in Forschung und Technik, Produktentwicklung und Wirtschaft wider.

Sicherlich werden die erfahrenen DAGA-Teilnehmer zahlreiche Details wiedererkennen. Aber sie werden zu dem einige Besonderheiten der NAG/DAGA wahrnehmen, die in der Erweiterung beispielsweise in der Lärmbekämpfung in anderen Ländern und Städten liegen. Im Austausch der Erfahrung in der Bekämpfung von Verkehrslärm, in bauakustischen Anforderungen und in Implementierungen von Lärmkarten finden sich weitere Beispiele, in denen Diskussionen über die Grenzen hinweg fruchtbar sein werden. Besonders hingewiesen sei auf die zwei Vorkolloquien (Seite 5). Eines wird an der TU Delft ausgerichtet und befasst sich mit Array-Technologie. Im anderen liegt der Schwerpunkt auf Fragen der Lärmbekämpfung in Hafenstädten wie Rotterdam. Darüber hinaus wird die reiche kulturelle Vielfalt von

Rotterdam zu einem besonderen Erlebnis der NAG/DAGA beitragen.

Der Tagungsort ist das wohlbekannte Konzert- und Konferenzzentrum ”De Doelen” im Herzen von Rotterdam, nur wenige Schritte gegenüber des Hauptbahnhofs und in unmittelbarer Nähe zu Geschäften und Restaurants (siehe Seite 359). In De Doelen haben wir professionelle Ausstattungen zur Verfügung, mit modernen Sitzungsräumen, Foyerbereichen für Poster und für die Ausstellung. Bemerkenswert ist ferner, dass De Doelen einen exzellenten Konzertsaal beherbergt. Bitte beachten Sie, dass die Reservierung für Eintrittskarten rechtzeitig gemacht werden muss, um beim Konzert am 25. März (Mittwoch) dabei sein zu können.

Die Tagungsorganisation tat ihr Bestes, das Programm so zu gestalten, dass allgemeine und individuelle Interessen gleichermaßen berücksichtigt werden. Für die Abende wurde versucht, eine Mischung aus Freizeit in Rotterdams Kneipen und Restaurants mit dem Angebot einen Banketts (siehe Seite 32) zu finden. Das Rahmenprogramm und besonders der gesellige Abend am Mittwoch werden dazu beitragen, neue Freundschaften zu knüpfen und alte zu pflegen. Technische Besichtigungen im Hafen Rotterdam, zum ”Large European Acoustic Facility” in Noordwijk, sowie Besuche interessanter kultureller Orte wie die Blaue Porzellanmanufaktur in Delft, ”Porseleyne Fles”, und eine Architektur-Exkursion in Rotterdam vervollständigen das Tagungsprogramm. Sofern Sie ihr Hotelzimmer noch nicht gebucht haben, sollten Sie das bald tun, siehe www.nag-daga.nl.

Schließlich möchten wir den Autoren für ihre insgesamt 560 Vorträge und Poster danken. Rotterdam ist eine sehr schöne Stadt, und De Doelen ist ein schönes Konferenzzentrum, aber ohne die exzellenten Beiträge zur wissenschaftlichen und angewandten Akustik wäre dieses Ereignis nicht möglich.

Tot ziens und bis bald im März!

Für das NAG/DAGA Organisationskommittee,
Martijn Vercammen und Michael Vorländer

Welcome address

Dear colleagues,

”Welkom”, as the Dutchmen say, to Rotterdam! We would also like to say ”Willkommen” and ”Welcome”. The organisers of this conference are delighted to invite you, that means Dutch, Belgian and German acousticians as well as acousticians from other countries, to visit one of the most modern cities in the Netherlands where NAG/DAGA 2009 will take place. The programme which includes 528 oral presentations in 69 sessions, many posters and an attractive exhibition is waiting to be instilled with life, discussions, scientific results, technical solutions and products.

This event combines the 35th DAGA conference of the Deutsche Gesellschaft für Akustik, DEGA, and the meeting of the Nederlands Akoestiek Genootschap, NAG. It is supported by ABAV, the Acoustical Society of Belgium, by the European Acoustics Association, EAA, and by the German associations of engineers and physicists, ITG, VDI and DPG. This conference is part of a series of joint conferences which provide DEGA with an opportunity to visit its neighbouring countries. In the past, conferences were staged, for instance in Austria in 1990, in Switzerland in 1998 and in France in 2004. Given the success of the previous joint conferences, we feel obliged and committed to create the best-possible conditions for the participants in order to make the event enriching and successful in terms of our work. Furthermore we want to ensure that you will have pleasant stay in Rotterdam. This conference provides an international platform and will contribute not only to the scientific and technical progress of acoustics, but also to cooperation across borders and languages. Of course, this aspect is also reflected by joint European and international activities in acoustics in research and technology, in product development and in acoustics business.

Those who are frequent DAGA participants will certainly recognise many well-known details. But they will also notice the special ”flavour” of the NAG/DAGA due to the fact that noise control strategies in other countries and cities are on the agenda, just to mention one example. Sharing experiences in traffic noise control, building acoustic codes and the implementation of Noise Maps are other examples where cross-border discussions will be very fruitful. We would like to draw your attention in particular to the two pre-colloquia (page 5). One is focused on Array Technology and will be held at Delft Technical University. The other will put emphasis on questions of noise control in European harbour cities such as Rotterdam. Furthermore, the rich culture of Rotterdam and surroundings will contribute to a special experience at NAG/DAGA.

The venue of the event is the well-known concert and conference centre ”De Doelen” in the middle of Rotterdam, just across the central station and in the vicinity of shops and restaurants (see page 359). In De Doelen, professional facilities are at our disposal, including modern session rooms, floor areas for posters, discussions in coffee breaks and an exhibition area. It is worth mentioning that De Doelen also includes an excellent concert hall. Please be aware that ticket reservation must be made in time for the concert on Wednesday, March 25.

The organisers of this conference did their utmost to arrange the programme in a way that common and individual interests are met and special requests were taken into account. As far as the evenings are concerned, we tried to arrange enough free time in Rotterdam’s pubs and restaurants while offering a banquet (see page 32) as well. The social events and especially the social reception on Wednesday evening might contribute to forming new and to strengthening old friendship. Technical tours to Rotterdam harbour and to the Large European Acoustic Facility in Noordwijk, visits of interesting cultural sights such as the Delft Blue factory ”Porseleyne Fles” and an ”architectural tour” in Rotterdam complement the programme. In case you have not booked your hotel yet, this is possible through the website www.nag-daga.nl.

Finally, we are particularly grateful to all the authors for submitting their in total 560 oral and poster presentations. Rotterdam is a nice city and De Doelen is a nice conference centre, but without this excellent input of contributions to scientific and applied acoustics the event would not be possible.

Tot ziens and see you in March!

For the NAG/DAGA Organising Committee,
Martijn Vercammen and Michael Vorländer

General information

NAG/DAGA 2009
International Conference on Acoustics in Rotterdam
including the
35th German Annual Conference on Acoustics (DAGA)
March 23-26, 2009
Homepage: http://www.nag-daga.nl
E-Mail: secr@nag-daga.nl

Organisers

- Acoustical Society of the Netherlands (NAG)
- German Acoustical Society (DEGA)

In co-operation with

- Belgian Acoustical Society (ABAV)
- European Acoustics Association (EAA)

Co-organisers

- German Physical Society (DPG)
- Information Technology Society (ITG in VDE)
- German Standards Committee on Acoustics, Noise Reduction and Vibration (NALS in DIN and VDI)

Conference venue

The venue for the NAG/DAGA 2009 conference will be ”De Doelen” in Rotterdam, the city’s major conference centre and in this combination the country’s largest concert hall. This monumental building is situated opposite the Central Railway Station and within easy walking distance of restaurants and the main shopping centre. A large underground parking is situated directly in front of the building, while other parkings are within walking distance.

Address

De Doelen
Willem Burger Zaal complex
Kruisplein 40
3012 CC Rotterdam, The Netherlands
www.dedoelen.nl/congresgebouw

Telephone and fax during the conference

- Telephone number registration desk: + 31 10 2171 821
- Fax number registration desk: + 31 10 2171 822

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- Ir. Martijn Vercammen, Peutz bv, Mook, The Netherlands
- Prof. Dr. Michael Vorländer, RWTH Aachen University, Aachen, Germany

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- E. Gerretsen, TNO Delft
- H.W. Gierlich, HEAD acoustics GmbH
- A. Kohlrausch, Philips Research Eindhoven
- W. Lauriks, K.U. Leuven
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- J.M. Poel, Peutz bv
- F. de Roo, TNO Science and Industry
- M.L.S. Vercammen, Peutz bv
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Languages

It is highly recommended that papers are presented in English in order to reach as much of the audience as possible. However presentations in German and Dutch are allowed. In order to create understanding with all participants the presentation slides should be in English.

Topics of the congress

Topics of the conference are all areas of acoustics, and all areas that are inter-disciplinarily related to acoustics, in particular the following:
Active noise control, Audiological acoustics, Audio technology, Building acoustics, Bio-acoustics, Education in acoustics, Effects of noise, Electro-acoustics, Environmental noise, Evaluation of noise, Flow acoustics, Hearing protection, Hydro-acoustics, Machine acoustics, Measurement engineering, Medical acoustics, Musical acoustics, Noise propagation, Noise control, Physical acoustics, Psychoacoustics, Room acoustics, Signal processing, Structure-borne sound, Speech, Soundscape, Ultrasounds, Vehicle acoustics, Vibration acoustics, Vibration technology, Virtual acoustics.

Precolloquia on March 23, 2009

- **Noise and ports: impacts on cities, humans and techniques**, De Doelen, p. 38
- **Array technology in audio and acoustics**, Delft University of Technology, lecture room C (includes technical tour), p. 41

Plenary lectures
Invited lectures will focus on hot topics of general interest:

- Tammo Houtgast: ”The acoustical engineer as a researcher in speech and hearing” (p. 35)
- Mike Barron: ”Then and now - how concert hall design of the 1960s/ ’70s compares with the present” (p. 36)
- Tjeert ten Wolde: ”Reciprocity measurements in acoustical and mechano-acoustical systems - review of theory and applications” (p. 36)
- Lothar-Cremer Award Lecture
- Torsten Dau: ”Parallel vs. serial processing of amplitude modulation and lateralization in the auditory system” (p. 37)

Structured sessions
The following structured sessions are composed of invited contributions and have been organised by experts:

- **Active noise control** (A. Jakob, R. Schirmacher, P. Sas), p. 195/ 265
- **Aeroacoustics** (S. Becker, S. Rienstra), p. 308/ 312
- **Aeroacoustics on vehicles** (M. Helfer, M. Hirschberg, B.-J. Boersma), p. 212
- **Auditory processing** (T. Dau, B. Seeber, S. van de Par), p. 255/ 251
- **Binaural systems** (J. Fels, M.M. Boone), p. 62
- **Boundary Elements** (H. Brick, M. Ochmann, R. Piscoya, H. van der Auweraer), p. 199/ 203
- **CAE methodologies for vibration and acoustics** (S. Marburg, B. Pluymers, W. Desmet), p. 95
- **Cavitation** (C. Koch, D. Lohse), p. 216/ 324
- **Finite element models** (S. Langer, J.-P. Coyette), p. 98
- **History of acoustics** (R. Hoffmann, R. M. Aarts), p. 84
- **Hydroacoustics** (J. Abshagen, C. de Jong), p. 89
- **Localisation of sound sources on vehicles** (R. Sottek, B. Roozen), p. 294
- **Low frequency noise** (D. Krahé, W. Soede), p. 160
- **Medical ultrasound** (G. Schmitz, K.W.A. van Dongen), p. 100
- **Modelling in room acoustics** (M. Vorländer, D. de Vries), p. 260
- **Music processing** (B. Schuller, A. Kohlrausch), p. 79
- **New aspects of transferpath analysis** (K. Becker, H. van der Auweraer), p. 190

- **Noise at workplace** (R. Paulsen, J. Granneman), p. 149
- **Quality classes in buildings** (C. Burkhart, E. Gerretsen), p. 136
- **Physics of musical instruments** (R. Bader, C. Nederveen), p. 175/ 178
- **Psychoacoustics** (J. Hellbrück, H. Fastl, T. Houtgast), p. 46/ 238/ 242
- **Railway noise** (D. Windelberg, M. Dittrich), p. 174/ 290
- **Sound quality and soundscapes** (A. Fiebig, D. Botteldooren), p. 156
- **Sound scaping and sound masking** (B. Schulte-Fortkamp, K. Genuit, E. Gerretsen), p. 68
- **Source identification** (A. Gerlach, B. Roozen), p. 217/ 221/ 332
- **Spatial audio** (S. Spors, D. de Vries), p. 123/ 127
- **Speech in car** (G. Klasmeyer, H.-W. Gierlich, J. Verhave), p. 73/ 165
- **Speech perception** (S. Möller, J. Beerends), p. 168/ 169/ 284
- **Structural-acoustic optimization** (J. Bös, D. Fritze, M. Dittrich), p. 269
- **Structure-borne sound in buildings** (W. Scholl, B. Ingelaere), p. 51
- **Tire-road noise** (W. Kropp, Y. Wijnant), p. 106
- **Tram noise** (F. Krüger, A. van Leuven), p. 182
- **Variable acoustics** (E. Mommertz, R. van Luxemburg), p. 57
- **Voice production** (M. Döllinger, M. Kob, M. Hirschberg), p. 298/ 303

Oral presentations

All oral presentations are scheduled as follows:

- **15 minutes lecture**
- **3 minutes discussion**
- **2 minutes break** (possibility to change rooms)

This schedule should be strictly observed. It is not possible to exceed 15 minutes for a presentation.

In each room, a laptop, a beamer with a standard resolution of 1024 x 768, and an audio loudspeaker are available. The laptops are equipped with Windows XP, MS-Office and Acrobat Reader. In case of additional technical equipment is required, please mail your request to papers@nag-daga.nl.

Presenting authors should provide their slides on CD-ROM or USB stick, and copy them on the laptop in the corresponding lecture room before the session starts. For help, technical staff is present in every room. It is not possible to use own laptops.

For PowerPoint presentations, we recommend to create an additional file which is system-independent. Please choose ”pack and go” in the file menu. Since we don’t assure the compatibility of your presentation with the installed PowerPoint version, a preview corner will be available to check your slides during the conference.

Posters

All posters will be presented on Tuesday or Wednesday (see p. 8 or p. 14). The authors of the posters will be present during their afternoon poster session (on Tuesday between 15:20 and 16:20 and on Wednesday between 16:00 and 17:00) and will be prepared for discussions and clarifications.

Posters will be mounted on full boards with sizes of 2 m height and 1 m wide. The recommended poster size is A0 (portrait). The posters must be fixed on the boards with special stickers that will be provided at the conference.

Preparation of manuscripts

The manuscripts shall be submitted online via

http://www.nag-daga.nl

before or during the conference, latest March 26, 2009. They can be submitted as PDF or Word file.

The page format and the appearance (fonts, tables etc.) shall be uniform: Manuscripts shall be formatted as a two-column style document, and they can have a length of two, three, or four pages. Authors shall use the Word or LaTeX templates on

http://www.nag-daga.nl

Manuscripts submitted after March 26, 2009 cannot be included in the conference proceedings. If the online submission is not possible, a CD-ROM can also be delivered at the conference office.

Exhibition

Parallel to the congress a large European Acoustics exhibition will be held. The exhibition joins all important providers of acoustical products, services and information. At the exhibition companies, manufacturers, engineering services providers, but also research and administrative establishments, technical-scientific associations and publishers inform about the newest results in research and development concerning equipment, technologies and software solutions as well as materials, standards, regulations and publications.

Well-equipped lecture rooms and a large exhibition space provide perfect conditions for a successful meeting for everybody interested in acoustics. Interested companies can register via

igbleerdam@wxs.nl

The exhibition takes place from March 24 to 26. The exhibition booths are at the ground floor in close range to the entrance and registration area and at the first floor in the three adjacent lobbies of one of the main music halls.

Until December 31, 2008, the following companies have already registered at the exhibition:

- 01dB / AcouTronics
- Alara-Lukagro bv
- BASWA Acoustic AG
- Braunstein + Berndt GmbH
- Brüel & Kjaer bv
- BSW GmbH
- CA Software & Systems
- DataKustik GmbH
- deBAKOM GmbH
- FFT
- G.R.A.S.
- HEAD acoustics GmbH
- IAC GmbH
- Knowles Acoustics
- LMS International
- MAX Frank GmbH
- Merford Noisecontrol bv
- Microflown Technologies
- Microtech Gefell GmbH
- Ministries VROM and VWS, Den Haag
- Müller-BBM GmbH
- Norsonic-Tippkemper GmbH
- Odeon A/S
- RION Co. Ltd.
- Saint-Gobain Ecophon
- SINUS Messtechnik GmbH
- Softnoise GmbH
- Sonogamma bvba
- Soundplan
- SPEKTRA Schwingungstechnik GmbH
- TNO Science and Industry
- Wölfel Meßsysteme Software GmbH

Product forum
Like in the last years, a product forum takes place directly after the opening lecture (plenary).

Job offers
We can place job offers for you on the poster board next to the registration desk. Please ask the staff members at the registration desk. The costs are:

- € 50,- for universities and NAG or DEGA sustain. members (ex. VAT)
- € 150,- for others firms and institutions (ex. VAT)

The invoice will be sent after the NAG/DAGA 2009.

NAG and DEGA members general assemblies

On Monday afternoon, March 23, the general assembly of DEGA will take place (17:00 in the Willem Burger Zaal). The official invitation letter and the agenda will be published in the upcoming “Sprachrohr“ newsletter of DEGA (February 2009).
On Wednesday during the lunch break, March 25, the NAG assembly will take place (12:40 in the Van Beuningen Zaal).

EAA/DEGA student lunch meeting

The student lunch has the aim to bring together all participating students. After a short introduction of the ongoing student activities in DEGA, EAA and ASA (Acoustical Society of America), there will be time for discussions during lunch with senior scientists and invited representatives from companies. The participation is for free for students!
The meeting takes place on Tuesday, March 24, 12:40 in the Van Beuningen Zaal.

Technical tours

- ESTEC, Noordwijk, Large European Acoustic Facility (LEAF)
  (March, 24 and 25, 13:30 - 17:00)
- Technical (boat) tour through port of Rotterdam
  (March, 25 and 26, 9:00 - 12:00, departure: Leuvehaven)

Social events

General social events:

- Opening ceremony
  (March, 24, starting-time 9:00)
- Opening of exhibition area, with sandwiches
  (March, 24, 12:00 - 14:00)
- Congress dinner including music with jam-session (musicians should bring their instruments!)
  (March, 24, transport: bus leaves at 19:00,
  dinner: 20:00 - 23:00, ticket necessary)
- Conference reception
  (March, 25, 18:00 - 20:00)
- Concert in De Doelen
  (March, 25, starting-time 20:15, ticket necessary)
- Farewell event
  (March, 26, 18:00 - 18:30)

Accompanying persons program:

- Modern architecture city tour, Rotterdam (NAI)
  (March, 24, 14:00 - 17:00 and
  March, 25, 9:00 - 12:00)
- Delft Blue factory ”Porseleyne Fles”, Delft
  (March, 25, 14:00 - 17:00 and
  March, 26, 9:00 - 12:00)

Lunch

There are more than enough possibilities near De Doelen to go out for lunch during the lunch break. This information is provided on page 359 of this program book. On Tuesday also some sandwiches will be provided during the opening of the exhibition. Furthermore, for those who will have meetings during lunch break or prefer to stay in De Doelen to visit the exhibition, make the last preparations for a presentation, etc. we will provide a limited number of lunchboxes on Wednesday and Thursday. Tickets for lunchboxes can be bought at the registration desk in the Expohal. In the Expohal you will also find the distribution point of the lunch boxes and a lunch area. The cost for a lunchbox is € 12,00.

Conference registration

Preferably, registration should be carried out via the conference website

http://www.nag-daga.nl

Alternatively, you can register via a printed registration form which is added to this program book (see page 375) or can be found on the website. Payment is possible by debit entry (only from Germany or from Netherlands), bank transfer (IBAN), or credit card. You can choose the method of payment on the registration sheet.

Registration fees

category	member$^{(a)}$	student	retired$^{(b)}$	€ early$^{(c)}$	€ late$^{(d)}$
1$^{(e)}$	no	no	no	300,-	350,-
2	yes	no	no	230,-	270,-
3	yes	no	yes	150,-	180,-
4	yes/no	yes	no	60,-	80,-

$^{(a)}$:Member of DEGA, VDI, DPG, ITG, VdT, NAG, ABAV and all other EAA societies

$^{(b)}$:Also valid for unemployed persons. Not valid for non-members (these are cat. 1)

$^{(c)}$:Fee incl. 19% VAT if registered before Feb. 1, 2009

$^{(d)}$:Fee incl. 19% VAT if registered after Feb. 1, 2009

$^{(e)}$:Registration in cat. 1 entitles for a free membership of DEGA or NAG for the year 2009 (on request to DEGA or NAG)

The registration fee will include:

- For authors, one oral or poster presentation
- Abstract book and conference proceedings (CD-ROM)
- Refreshments
- Conference reception

Congress proceedings

The manuscripts of the contributions (oral and poster presentations) will be published as CD-ROM. Every registered participant will obtain this CD-ROM during summer 2009 automatically. An additional printed version of the proceedings will be made available at a rate of € 100,- for orders received from participants until the end of the conference. Later orders are possible at a rate of € 125,- (ex VAT).

Hotel accommodation

As the organisation realises that choosing a hotel in a foreign city is very hard, we looked for an easy but secure way for our guests to be sure to find the right place in a comfortable hotel of their choice. We found an excellent assistance for supporting you with your hotel booking in the co-operation with Hotelservice Rotterdam and their possibilities. You may choose from several suggestions and order the room you like, just by going to the online hotel booking:

http://www.hotelservicerotterdam.nl/nagdaga2009

Important dates for NAG/DAGA 2009

- February 1st, 2009: Latest date for early bird registration
- March 23, 2009: Precolloquia
- March 23-26, 2009: NAG/DAGA 2009 Congress, Deadline for submission of papers
- Summer 2009: Dispatching of CD-ROM and additionally printed proceedings

Allgemeine Informationen in deutscher Sprache

Eine deutsche Übersetzung der ”General Information” (ab Seite 26) wird im DEGA-Sprachrohr Nr. 48 veröffentlicht, welches ab Mitte Februar 2009 an alle DEGA-Mitglieder verschickt wird bzw. auf der Webseite www.dega-akustik.de heruntergeladen werden kann.

Opening ceremony and product forum

Tue 9:00 Willem Burger Zaal Opening ceremony

Tue 11:00 Willem Burger Zaal Plenary lecture

T. Houtgast: ”The acoustical engineer as a researcher in speech and hearing”; see below

Tue 11:40 Willem Burger Zaal Product forum

Plenary lectures

Tue 11:00 Willem Burger Zaal Plenary lecture Tuesday

The acoustical engineer as a researcher in speech and hearing

Tammo Houtgast
VU University medical center, Amsterdam

There is a long tradition of acoustical engineers working in the field of speech and hearing. Although it is well recognized that speech reception, and hearing in general, involves cognitive processes beyond the reach of the engineer, this engineering approach has led to interesting results which will be reviewed briefly.

For the engineer, speech is often described in acoustical terms as a succession of sounds with specific properties in terms of spectra and decibels. It has been possible to identify the acoustical cues in a speech signal which are essential for speech intelligibility. These cues are related to the dynamics, or the pattern of fluctuations, in the speech signal. By applying a specific type of speech analysis, the strength of these cues in an ongoing speech signal can be quantified. It will be shown that the effect of speech transmission and room acoustics on speech intelligibility can be estimated successfully by measuring the degree of preservation of these cues in the speech signal received by the listener.

In the field of hearing, the engineer is concerned mainly with the quality of the peripheral auditory system (i.e., the cochlea) in terms of accurate coding of the incoming sound. Sub-optimal performance (hearing impairment) may be related not only to a raised hearing threshold (hearing loss as determined by the classical tone audiogram), but also to a reduced quality of the coding process in terms of spectral and temporal resolution. The specific part of psycho-acoustical research concerned with this topic will be briefly reviewed: the design of appropriate test signals and measuring procedures to estimate the acuity in the auditory coding of the spectral, temporal and amplitude sound characteristics.

Wed 12:00 Willem Burger Zaal Plenary lectures Wednesday

Then and now - how concert hall design of the 1960s/'70s compares with the present

Michael Barron
University of Bath

The 1960s and '70s were an exciting time for concert hall design. Several important new halls were built to replace war damage, but arts provision also needed improving in many cities round the world. Concert hall design was in an experimental phase with some major successes and disappointments; the Philharmonic halls in Berlin (1963) and New York (1962) illustrate the 'extremes'. The De Doelen large hall of 1966 is rated as an acoustic success but is interesting as an example of a particular design philosophy: a large hall with highly scattering walls and ceiling. This approach has not been followed much elsewhere.
Current approaches to concert hall design are different in many ways to those of around 40 years ago. The tools available now include much better understanding of the important dimensions for concert hall listening, objective quantities corresponding to these subjective dimensions, modelling by scale or computer simulation modelling. Some designers also use auralisation. But as well as technical developments, the role of the acoustician is now generally more prominent in the design team. With greater responsibility, current designers often take fewer risks and adhere to one of two precedents: the parallel-sided hall and the vineyard terrace hall.
The paper will concentrate on the comparison between 40 years ago and the present, mainly in terms of acoustics for the listener but also with a short mention of conditions for the performers.

Wed 14:00 Willem Burger Zaal Plenary lectures Wednesday

Reciprocity measurements in acoustical and mechano-acoustical systems. Review of theory and applications.

Tjeert ten Wolde
Leidschendam

Dynamical systems are called reciprocal when the transmission of vibration from point A to point B has a simple relation with the transmission from point B to point A. The theory on this issue will be reviewed and the boundaries of the validity of reciprocity for acoustical, mechanical and electrical systems, and for combinations of these, will be discussed. When a system is reciprocal, there is an alternative possibility for the measurement of transfer functions and for the determination of source strength (acoustical, mechanical or electrical). The latter application concerns a reciprocal version of the substitution method. Reciprocal measurements can be attractive when the direct measurement offers problems, for example when there is too little space for a sufficiently strong auxiliary source in the direct set-up. Examples of such cases will be presented for acoustical and mechano-acoustical systems. The general reciprocity theorem is known since 1873 (Lord Rayleigh). It lasted nearly a century before the first article on the above applications was published (1970, Steenhoek and Ten Wolde). Since than, the methods have been developed continuously and at present they are widely used as tools in the development of quiet cars, trucks, ships, trains, aircraft and buildings.

Thu 12:00 Willem Burger Zaal Plenary lectures Thursday

Lothar-Cremer Award Lecture

Winner and lecture title will be announced in opening ceremony (Tue 9:00)

Thu 14:00 Willem Burger Zaal Plenary lectures Thursday

Recent concepts and challenges in hearing research

Torsten Dau

Centre for Applied Hearing Research, TU of Denmark

In everyday life, the speech we listen to is often mixed with many other sound sources as well as reverberation. In such a situation, normal-hearing listeners are able to effortlessly segregate a single voice out of the background, which is commonly known as the ”cocktail party effect”. Conversely, hearing-impaired people have great difficulty understanding speech when more than one person is talking, even when reduced audibility has been fully compensated for by a hearing aid. As with the hearing impaired, the performance of automatic speech recognition systems deteriorates dramatically with additional sound sources. The reasons for these difficulties are not well understood. Only by obtaining a clearer understanding of the auditory system’s coding strategies will it be possible to design intelligent compensation algorithms for the hearing impaired. This presentation highlights recent concepts of the signal processing strategies employed by the normal as well as impaired auditory system. The aim is to develop a computational auditory signal-processing model, capable of describing the transformation from the acoustical input signal into its ”internal (neural) representations”. Several stages of processing are considered to be important for a robust signal representation and a deficiency in any of these processing stages is likely to result in a deterioration of the entire system’s performance. A state-of-the-art model of auditory signal processing would be of major practical significance for technical applications, in digital hearing aids, cochlear implants, speech and audio coding, and automatic speech recognition.

Pre-colloquium
Noise and ports: impacts on cities, humans and techniques

Overview: see p. 5

Mon 13:00 Willem Burger Zaal Precolloquium: Noise and ports

Opening by colloquium chairs

Miriam Weber[a] and Christian Popp[b]
[a]*DCMR Environmental Protection Agency;* [b]*Lärmkontor, Hamburg*

Mon 13:10 Willem Burger Zaal Precolloquium: Noise and ports

The city and port dilemma

Christian Popp
Lärmkontor GmbH, Hamburg

On the one hand the ports are still expanding and even if all the available low-noise technologies are used they are a relevant noise source. On the other hand the residential areas of the cities are moving to the waterfronts. This causes conflicts between the staying port industries and the new (often very expensive) residential areas. The presentation of Christian Popp will describe Hamburg's approach to find a practicable solution for this city and port dilemma.

Mon 13:35 Willem Burger Zaal Precolloquium: Noise and ports

Noise at work in ports

Sergio Luzzi
Vie En.Ro.Se. Ingegeneria - Florence

In this paper various aspects of evaluation and reduction of noise exposure of workers in ports are investigated, starting from the minimum requirements for the protection of workers against the risks resulting from exposure to noise, fixed by 2003/10/EC Directive and other European disposals and from the state of art of its transposition in National Legislation of Member States. With reference to case studies, developed in ports of Italian cities, having different characteristics, the significant critical points present in the standard procedures for measurement of noise and risk assessment are shown, and possible solutions are proposed. Typical activities, like construction noise at night in ports, are considered and action planning experiences for the limitation of noise exposure as well as methods for the correct definition of standard annoying activities and daytime exposure calculation that have been collected, catalogued and compared according to different criteria. The general solution planning and designing methods are defined in a way that permit them to be linkable with or integrated in Good Practice Guide for ports area noise management.

Mon 14:00 Willem Burger Zaal Precolloquium: Noise and ports

Noise emission on RoRo terminals

Rob Witte
DGMR Consulting Engineers bv

The RoRo (Roll On Roll Of) terminals are a booming business, certainly on short voyages from Rotterdam to England. The terminal is characterized by activity peaks around loading and unloading times. The main equivalent noise sources are the traffic with specialized terminal tractors and the ship itself. Peak levels may arise by driving on to the ramp. The presentation gives an overview of the general operations, current sound power levels and possible mitigating measures, based on our experiences in the market.

Mon 14:25 Willem Burger Zaal Precolloquium: Noise and ports

Development and state-of-the-art of noise control in the petrochemical industry

Carl Hantschk
Müller-BBM GmbH

In Europe, a significant awareness for noise from industrial activities as a disturbing and harmful factor in everyday life has not evolved until economic growth and advancing urbanization caused industry and residential/recreational areas to move ever closer together. When legislation defined limits and requirements for the noise impact of industrial installations, practical tools and means to ensure compliance with the law became necessary. As a result, measurement methods have been developed, typical noise sources in industry have been determined and characterized, calculation methods to predict noise emissions and noise received in the neighborhood have been worked out and new noise control equipment and concepts have been elaborated. The results of this work still form the basis for modern noise control engineering. Continuous efforts over the last decades resulted in essential contributions to reducing the noise emissions associated with industrial activities and to protecting the environment from excessive noise levels. This presentation gives an overview of the above developments and some examples. Focus is set to petrochemical industry in ports, where safety requirements calling for outdoor operation of noisy equipment and the vicinity of urban areas pose a particular challenge to the noise control engineer.

Mon 15:10 Willem Burger Zaal Precolloquium: Noise and ports

Port area noise management - Lessons from 3 major European Sea-ports

Antonis Michail
DGMR Consulting Engineers bv

Noise management is an issue of increasing significance within the efforts of integrated environmental management of seaport areas. The presentation focuses on the current practices and challenges in the field of port area noise management in 3 major European Ports, namely the Ports of Rotterdam, Antwerp and Hamburg. The research aims to reveal the importance of noise as an environmental aspect of port related activities and operations, the noise impact around the selected seaport areas, and the practices and control mechanisms that are in place for integrated noise management. The research methodology includes an analysis of available noise surveys and prediction models in line with recorded complaints and applied legislative and management frameworks. Structured interviews with the port environmental managers feed to the discussion and findings.

Mon 15:35 Willem Burger Zaal Precolloquium: Noise and ports

Ports and their impacts on nature

Nico van Doorn
Royal Haskoning consultants

The NoMEPorts Project (Noise Management in European Ports) has developed a Good Practice Guide on Port Area Noise Mapping and Management. This Guide has been compiled as a synthesis and user-friendly interpretation of the European Noise Directive (END) and as a summary of the management response options available for the effective implementation of its provisions with a focus on port areas. The context of the issue of noise is set against changes in the role of ports with discussion on policy and reasons to act in response to environmental challenges. A six steps methodology for noise mapping and management has been developed and will be shown and discussed. The importance of future developments in and around the port will be addressed.
The Guide concludes with a perspective on evaluation of action plans and their implementation.

Mon 16:00 Willem Burger Zaal Precolloquium: Noise and ports

Noise Management at the port of Rotterdam

Ton van Breemen[a] and Frank Wolkenfelt[b]
[a]*Port of Amsterdam;* [b]*Port of Rotterdam*

The Port of Rotterdam is facing a huge challenge to plan and intensify the port industrial activities within strict limit values for industrial noise. PoR, DCMR and others have developed a methodology in noise management by translating the future port planning into a noise allocation map and by monitoring the actual situation. This presentation will give an overview of the methodology, stakeholders and the software tools.

Pre-colloquium
Array Technology in Audio and Acoustics,
Delft University of Technology

Overview: see p. 5
For information about the travel between Delft and De Doelen please read p. 351

Mon 11:00 TU Delft, room C Precolloquium: Array technology

Opening by colloquium chair

Diemer de Vries
TU Delft

Mon 11:15 TU Delft, room C Precolloquium: Array technology

The Laboratory of Acoustical Imaging and Sound Control of Delft University of Technology

Dries Gisolf
TU Delft

Acoustical research started in Delft around 1928 with the famous Dutch physicist A.D. Fokker. Apart from being a very good theoretical physicist, he also loved music and started investigating the room acoustics of concert halls. Room acoustics and sound control have remained the mainstay of acoustical research in Delft until the late seventies, when Berkhout introduced the concepts of array technology, developed in the seismic imaging industry, in sound analysis and reproduction. At the same time the group started to work on acoustical imaging techniques, in particular seismic imaging for the oil and gas industry, but also medical imaging and non-destructive testing.
In current work in the group, array technology is still the integrating element in all applications of acoustical imaging and sound control, because arrays sample the wave-field in space and time, allowing forward and backward extrapolation of wave-fields. The frequency range of applications based on wave-field extrapolation spans eight decades, from ultrasonic imaging for medical diagnostics or non-destructive testing, through wave-field synthesis in the audio range, to seismic imaging in the earth and listening to infra sound in the atmosphere.
In the talk an overview is given of current work in the group.

Mon 11:35 TU Delft, room C Precolloquium: Array technology

Medical acoustical array expertise at Delft University of Technology

Koen W. A. van Dongen, Erwin J. Alles and Libertario Demi
TU Delft

The history of medical ultrasound dates back as far as the mid 30s, first as a therapeutic tool (e.g. deep tissue heating) and a decade later as a diagnostic tool based on pulse-echo reflections. Nowadays, a majority of all medical imaging modalities are based on ultrasound as the technique is often non-invasive, harmless and relatively cheap and still results in high image quality.
High image quality is obtained using array technology, i.e. a large number of active elements (preferably smaller than a wavelength) are positioned either in a single line (1-D array) or in a surface (2-D array). By applying a predefined time delay to the signal of each element, electronically or via software on the measured data, it becomes feasible to focus or steer the beam to a volumetric region of interest.
At the TU Delft in the section Acoustical Imaging and Sound Control we apply array technology in various medical applications. Among the applications are breast cancer detection, hyperthermia breast cancer treatment, transesophageal echo cardiography (TEE) and intravascular ultrasound. During our presentation, we will show for each of the above applications the challenges (and solutions) of implementing array technology in clinical applications.

Mon 11:55 TU Delft, room C Precolloquium: Array technology

Plane wave decomposition: an inverse problem in array processing

Lars Hörchens and Diemer de Vries
TU Delft

The decomposition of wave fields into plane waves is a standard tool for the analysis of impulse responses measured with an array of microphones. If the geometry is appropriately chosen, the decomposition can be obtained by spatio-temporal deconvolution. Several measures can be taken to tackle the ill-posedness of the associated mathematical problem. Typical examples are given by beamforming techniques or plane wave decomposition using circular or spherical harmonics. These approaches are compared based on the presented framework. The stability of the procedure for plane wave decomposition is strongly influenced by the choice of the array setup, which is extensively used for current approaches. On the processing side, regularisation techniques and prior knowledge can be employed to overcome instabilities and improve the achievable resolution. Exemplary results based on this approach are shown, pointing out the potential of further research in this field.

Mon 12:15 TU Delft, room C Precolloquium: Array technology

High resolution reconstruction of irregularly sampled, aliased measurements

Eric Verschuur and Hannes Kutscha
TU Delft

In many multi-channel measurement situations it is not possible to acquire data with sufficiently dense sampled arrays. Furthermore, the measurement locations are not always regularly sampled or may have holes, e.g. due to obstructions. In the field of oil and gas exploration using acoustic signals this is often the case. The objective of exploration seismics is to image the subsurface of the earth based on reflection measurements made at the surface. A source at the surface emits an acoustic wavefield which propagates through the subsurface. Any inhomogeneity in the earth will cause part of the energy to scatter back to the surface, where a line or a grid of receivers is positioned. Based on the wave equation, these reflection measurements can be transformed into an image of the subsurface. Before this imaging process can take place, many preprocessing algorithms have to be applied to the data, e.g. to remove noise and other unwanted energy. These algorithms are usually designed for regularly sampled, unaliased data. Therefore, reconstruction algorithms are used to transform the measurements into a regularly sampled, aliasing free dataset. Because of the aliased nature of the data extra constraints are required to reconstruct the data at the missing locations. Usually, the assumption is that the measurements can be described with a limited number of parameters in some transform domain. Typical examples of such domain transforms are the Fourier transform and the generalized Radon transform. In this paper we will give an overview of typical transforms and the involved constraints that can be used to reconstruct measurements onto a user-defined spatial grid.

Mon 14:00 TU Delft, room C Precolloquium: Array technology

Estimation of primaries in seismic measurements by sparse inversion

Gert-Jan van Groenestijn and Eric Verschuur
TU Delft

In exploration seismics the objective is to image the subsurface structures from acoustic reflection measurements, in order to localize and monitor oil and gas reserves. The reflection measurements are usually carried out with sources and receivers positioned at the surface of the earth. Especially in the marine case the water-air interface acts as an almost perfect acoustic mirror, reflecting all upgoing energy back into the medium. As a result, the measurements suffer from multiple reflections that mask the desired primary reflections from the inhomogeneities in the earth. However, these surface multiples have a physical relationship with the primaries: each primary event will be followed by a sequence of multiple reflections. This relationship can be exploited to estimate the primary reflection response, i.e. the multiple-free transfer function of the subsurface. This is done by a full waveform inversion process, in which the primary transfer functions are parameterized by spikes and a sparseness constraint is used during the optimization process. Examples will be shown for synthetic and field data.

Mon 14:20 TU Delft, room C Precolloquium: Array technology

Optimized microphone arrays for the hearing glasses

Marinus M. Boone
TU Delft

In many cases normal hearing aids do not give sufficient speech intelligibility. This is mainly caused by bad speech to noise ratio's such as during parties and under reverberant conditions. To overcome these problems the best way is to use highly directive microphones. We developed a solution by designing a special hearing aid in which the arms of a pair of spectacles are used to house microphones arrays. Optimized beamforming is used to obtain a high directivity index over the speech range of frequencies. Normally this method is applied separately for both ears. In this paper we will give an overview of the theoretical aspects of the optimized beamforming and results are presented where theoretical directivity patterns of the hearing glasses are compared with physical measurements under anechoic and reverberant conditions. The influence of the human head has also been investigated. Our results show that a high average directivity index of 8 dB is reached in combination with a low noise sensitivity. This result is obtained under undisturbed free field conditions. The influence of the human head reduces the directivity index to 7 dB, which is still very high as compared to normal directive hearing aids.

Mon 14:40 TU Delft, room C Precolloquium: Array technology

From theory to implementation of wave field synthesis

Sascha Spors
Deutsche Telekom Labs, TU Berlin

Wave field synthesis is a spatial sound field reproduction technique aiming at authentic reproduction of auditory scenes. Its theoretical foundation has been developed almost 20 years ago by Delft University of Technology and has been improved considerably since then. This talk revisits the theory of wave field synthesis and presents a unified framework for wave field synthesis covering arbitrarily shaped loudspeaker arrays for two- and three-dimensional reproduction. The physical foundation as well as the technical implementations include a number of approximations of the basic theory, for instance approximate correction of the secondary source type mismatch and spatial sampling of the ideally continuous loudspeaker distribution. The artifacts emerging from these approximations and aspects of the practical implementation are discussed. Furthermore, a number of extensions to the basic theory that have been introduced in the past are briefly reviewed. Higher-order Ambisonics is an alternative to wave field synthesis also aiming at authentic reproduction. The commons and differences between wave field synthesis and higher-order Ambisonics are highlighted. The talk concludes with open research questions.

Mon 15:00 TU Delft, room C Precolloquium: Array technology

Deriving Room Acoustical Parameters Using Arrays and Hearing Models

Jasper van Dorp Schuitman and Diemer de Vries
TU Delft

Using microphone arrays, room acoustical parameters can be determined by measuring room impulse responses along a line, circle or sphere. With this technique the distribution of the parameters within (parts of) a room can be evaluated. The method also revealed that some of the parameters may fluctuate as a function of microphone position, in such an amount that it does not correspond to human perception. In recent research, attempts were made to apply hearing models on the measurements to yield more perceptually motivated results. Listening tests were set up to find correlations between subjective attributes and the model results.

Mon 15:20 TU Delft, room C Precolloquium: Array technology

On the applications of array technology in room acoustics

Anton Schlesinger and Diemer de Vries
TU Delft

Room acoustic parameters vary much with the position of the receiver and the source as a consequence of the interferences in the anisotropic part of the sound field. It is thus difficult to yield exhaustive information of the room properties from independent single-point measurements. Using an array measurement permits to evaluate the sound field with a high spatial resolution and leads to a more precise assessment of the room acoustic properties. We propose a technique to investigate room acoustics by reconstructing the volumetric sound field from measurements taken on a sphere. The latter are obtained by a virtual spherical array using a microphone mounted at a robotic arm. The spatially sampled room impulse response upon the sphere is used to reconstruct the sound field within a particular volume of reconstruction by applying the methods of Nearfield Acoustical Holography (NAH). Subsequently, acoustical properties are derived from the reconstructed volumetric room impulse response. A virtual spherical microphone-array was constructed and tested in a volumetric modal analysis and a volumetric sound intensity probe. The results are discussed and an outlook on the potential of the presented method is given.

Paper sessions Tuesday, 24 March 2009

Tue 14:00 Willem Burger Zaal Psychoacoustics 1

Modulation perception and threshold fine structure: Experimental evaluation of the underlying mechanisms

Stephan J. Heise, Manfred Mauermann and Jesko L. Verhey
Universität Oldenburg, Institut für Physik

Amplitude modulation detection at low levels is influenced by the fine structure of the threshold in quiet. Modulation detection thresholds are higher when a tonal carrier falls in a fine-structure minimum than when it falls in an adjacent maximum. The aim of this study is to shed some light on possible underlying mechanisms, such as (1) a change in effective modulation depth due to a spectral colouring of the stimulus by threshold fine structure, (2) a reduction of the effective modulation depth in fine-structure minima by cochlear resonances such as spontaneous otoacoustic emissions (SOAEs), (3) modulation masking caused by the beating of a SOAE with the stimulus carrier. Modulation detection thresholds of sinusoidally amplitude-modulated carriers were measured for various stimulus parameters: five carrier levels (7.5-37.5 dB SL), two carrier bandwidths (0 and 30 Hz) and two carrier and five modulation frequencies which were adapted to the individual fine structure. Fifteen subjects with different degrees of threshold fine structure participated. The experimental results favour hypothesis (2). However, the data for different modulation frequencies indicate that spectral cues could be involved in modulation detection as well (hypothesis (1)), even for low modulation frequencies. Hypothesis (3) is not supported by the results.

Tue 14:20 Willem Burger Zaal Psychoacoustics 1

Modulation perception and threshold fine structure: Simulations with a nonlinear cochlea model

Bastian Epp, Jesko L. Verhey, Stephan J. Heise and Manfred Mauermann
Universität Oldenburg, Institut für Physik

Many natural sounds show temporal level fluctuations, i.e. amplitude modulations. Thus, a detailed knowledge about mechanisms underlying modulation perception is essential for the understanding of auditory signal processing. Most psychoacoustical studies investigated amplitude modulation processing for carrier levels well above threshold in quiet. Recently, Heise et al. (presented at DAGA2008) showed that the quasi-periodic fine structure in the threshold in quiet commonly observed in normal-hearing listeners affects the modulation detection of sinusoidally modulated tones with low carrier levels. The influence of threshold fine structure on modulation detection thresholds is not predicted by current

models of modulation perception. This is essentially due to the simplification of the peripheral stages in these models. In contrast to the previous studies, the present study uses a one-dimensional active and nonlinear model of the cochlea which simulates a fine structure in the threshold in quiet. The representation of amplitude modulation in regions of fine structure at the level of the basilar membrane is analyzed and compared to the performance of subjects in the modulation detection task. The simulations can further enhance our understanding of the relative contribution of temporal and spectral mechanisms underlying modulation perception.

Tue 14:40 Willem Burger Zaal Psychoacoustics 1

Modeling the masking of tones by Schroeder-phase harmonic tone complexes

Eugen Rasumow[a], Martin Hansen[a] and Stephan D. Ewert[b]

[a]*FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie;* [b]*Universität Oldenburg*

The amount of masking from harmonic tone complexes (HTC) with Schroeder phase on a sinusoidal signal varies up to over 30 dB when the phase curvature of the masker is altered. This masking differences can be explained by the dispersion on the basilar membrane (BM) which, in interaction with the phase of the HTC components, influences the temporal "peakiness" of the BM excitation. When modeling the phase curvature dependent amount of masking the use of a linear BM-filter with optimized phase-curvature and magnitude response is insufficient in terms of the resulting dynamic range of the simulated masking thresholds. One important factor might be the inclusion of peripheral compression into the model. However, current studies using the dual-resonance nonlinear filter (DRNL) [Meddis et al., J. Acous. Soc. Am, 109(6):2852-2861,2001] have shown no improvement of the simulations. This study deals with the refinement of the response characteristic of the DRNL filter. The effect of modifications of the DRNL filter was evaluated in the framework of the CASP-model [Jepsen et al., J. Acoust. Soc. Am., 124(1):422-438, 2008]. Furthermore, the dependence of the model-parameters concerning the dynamic range of the internal representation was investigated.

Tue 15:00 Willem Burger Zaal Psychoacoustics 1

Subband instantaneous-frequency analysis to determine masking with high temporal resolution for use in audio codecs

Nils Koppaetzky[a], Stephan D. Ewert[b], Birger Kollmeier[a] and Volker Hohmann[b]

[a]*Universität Oldenburg, Medizinische Physik;* [b]*Universität Oldenburg*

Algorithms for perceptual audio coding use psychoacoustical masking models to compress audio data with minimal impact on the perceived audio quality. Latest psychoacoustical research emphasizes the excellent spectro-temporal resolution of the auditory system. Most common audio codecs, however, apply classical spectral masking models which generally reach the required high spectral resolution at the expense of an insufficient temporal resolution. An application of a current psychoacoustical model which simulates the high spectro-temporal resolution of the auditory system, might offer an increase of the coding efficiency or of the perceptual coding quality, respectively. To analyze the general effect of an enhanced time resolution, a current nonlinear auditory filterbank model with instantaneaous-frequency conrol was tested in this constribution. The applicability of the model for the classification of the tonality of the signal and for forming an estimate of the spectro-temporal masking pattern was examined. The masking model as well as the resulting perceptual coding quality which can be achieved by a combination of the model with a simple perceptual coder are presented and discussed.

Tue 16:00 Willem Burger Zaal Psychoacoustics 1

Evalution of a Nonlinear Auditory Filterbank with Instantaneous Frequency Control

Reef Eilers[a], Stephan D. Ewert[a], Jesko L. Verhey[b] and Volker Hohmann[a]
[a]*Universität Oldenburg;* [b]*Universität Oldenburg, Institut für Physik*

Psychoacoustical studies suggest that the increase in suppression as a function of suppressor level is greater for a suppressor spectrally below than above the signal frequency. This pattern was investigated by measuring suppression for a tonal signal with two different psychoacoustical methods in the same subjects. The first method was the pulsation threshold technique that was used in a two-tone suppression paradigm with suppressor frequencies below and above the signal frequency. The second method was a forward-masked paradigm where the masker was a bandpass noise and the parameter the masker bandwidths. The noise was either centred at the signal frequency or was asymmetric with a larger portion of the spectrum on either the low or the high frequency side. The results of the two experiments were consistent with the hypothesis of different suppression growth functions for the different frequency regions. The measured data was modelled with a nonlinear filter approach. The model uses a modified dual-resonance filter by including a control of the nonlinear filter gain by the sub-band instantaneous frequency. It is investigated, if the model is able to quantitatively predict suppression and the effect of suppressor frequency.

Tue 16:20 Willem Burger Zaal Psychoacoustics 1

Temporal weighting in loudness perception: Effect of bandwidth

Jesko L. Verhey[a] and Jan Rennies[b]
[a]*Universität Oldenburg, Institut für Physik;* [b]*Fraunhofer IDMT*

Recent studies indicate that for broadband stimuli the beginning is more important for the judgement of overall loudness than later temporal segments [Pedersen und Ellermeier (2008), JASA 123, 963-972]. The aim

of the present study is to investigate if the special weight to the onset is related to the higher spectral loudness summation for short signals than for long signals. An analysis of weights is used to estimate the relative contributions of individual temporal segments to the overall loudness of 1-s long noise, whose level was randomly perturbed every 100 ms. The weights are determined for the same set of subjects for a broadband noise and a 400-Hz wide bandpass noise centred at 2 kHz. In agreement with previous studies, for the majority of test subjects the first 100 ms contributed significantly higher to the overall loudness perception in the broadband condition. This effect was significantly reduced in the narrowband condition. This result is in line with the hypothesis that the higher weight at stimulus onset is at least partly due to the higher spectral loudness summation at the beginning of the signal. The experimental results are discussed in relation to current loudness models.

Tue 16:40 Willem Burger Zaal Psychoacoustics 1

Loudness perception with headphone presentation compared to loudspeaker presentation in the diffuse field

Sebastian Goossens[a], Grit Bonin[b] and Roman Stumpner[a]
[a]*Institut für Rundfunktechnik GmbH;* [b]*TU Berlin*

The same sound pressure level in the ear-canal can lead to different loudness perception for headphone and loudspeaker presentation. This effect, which is known for a long time, was examined once more for different headphone types and signal presentations. During these tests the subjects were sitting in the reverberation chamber and had to adjust a third-octave band noise to the same loudness caused by loudspeaker and headphone reproduction. The headphone reproduction was carried out in a monotic, diotic and dichotic way. Significant differences of sound pressure levels had been measured in the ear-canal for the monotic and diotic case although the same loudness had been perceived. In contrary the level differences disappear largely for a dichotic stimulation. This is valid both for a binaural noise corresponding to the loudspeaker signal with spatial information, which was taken by artificial head recordings in the reverberation room and for a simple incoherent noise without spatial information at all. In the dichotic case equal levels in the ear-canal had been measured for the same loudness.

Tue 17:00 Willem Burger Zaal Psychoacoustics 1

Just-Noticeable Roughness Differences of Technical Sounds

Alexandra Leutheuser[a], Roland Sottek[b] and Jörg Becker-Schweitzer[a]
[a]*Fachhochschule Düsseldorf;* [b]*HEAD acoustics GmbH*

Roughness has become a central focus for product sound design. The roughness-induced perception, varying from a sporty character to a very unpleasant impression, consistently proposes new questions and challenges in sound engineering. This paper is concerned with one of the important aspects of subjective roughness perception: the investigation of just-noticeable roughness differences. Listening tests using synthetic sounds (modulated sinusoids with additional noise) and technical sounds (based on engine sounds) have been performed, both dependent on parameters such as degree of modulation, sound pressure level, and signal-to-noise ratio. The influence of a single parameter and of a combination of parameters was studied using adaptive test procedures providing more reliable and precise results within a limited time for the subjective tests. The results of the studies will be discussed. Further investigations relate to using the results for optimizing the roughness calculation based on the Hearing Model according to Sottek with respect to technical sounds.

Tue 17:20 Willem Burger Zaal Psychoacoustics 1

Comparison of loudness models for artificial and technical sounds

Jan Rennies[a] and Jesko L. Verhey[b]
[a]*Fraunhofer IDMT;* [b]*Universität Oldenburg, Institut für Physik*

Different loudness models were proposed to account for temporal aspects in loudness perception. To investigate different dynamic concepts for modeling loudness, predictions were made with two current loudness models for a set of time-varying, artificial and technical sounds. A comparison to data from literature showed that both models predicted loudness of amplitude and frequency-modulated sounds about equally well. For sequences of tone pulses with different frequencies, one model agrees better with data from literature than the other, since it includes a dynamic stage which allows spectral loudness summation for non-synchronous frequency components. To further investigate this effect, the dependence of spectral loudness summation on temporal separation between the frequency components was measured. The results indicate that spectral loudness summation takes place even when different frequencies are separated by 50 ms. Although a small effect is predicted by one of the models, quantitative differences between data and predictions suggest that the underlying mechanism is still not completely understood. A comparison of the predicted loudness of technical noise showed considerable differences in the absolute loudness values between the two models. However, the high rank correlation showed that, in principle, both models are suitable for quality assessments.

Tue 17:40 Willem Burger Zaal Psychoacoustics 1

Consonance/dissonance evaluation of electrical machine noise.

Sebastian Fingerhuth
Institute of Technical Acoustics, RWTH Aachen University

In this work the result of two listening tests of consonance/dissonance will be presented. The listeners had to evaluate several sounds with high tonal components. The sounds were synthesized based on recordings of electrical machines. The stimuli represented two machines driven at different speeds generating a more or less consonant/dissonant sound,

depending on the ratio of the speeds. Each listeners had to evaluate the sounds in two different experiments, i) using an absolute magnitude estimation (AME) method, without a reference, and ii) using a category partitioning (CP) scale with 5 main divisions (from ”not at all dissonant” to ”very dissonant”) each one with 10 subdivisions.
The differences and implications of using both methods will also be presented and some conclusions of consonance/dissonance as a psychoacoustic magnitude and consequences for sound design targets will be shown.

Tue 18:00 Willem Burger Zaal Psychoacoustics 1

Precise stimulation in auditory neuroimplants

Sören Wulf Kristiansen and Hans-Heinrich Bothe
TU of Denmark

The work concerns auditory neural implants with specific application to cochlear implants. A usual cochlear implant consists of an external and internal part, with the first being the acoustic receiver, processor and transcutaneous transmitter to the internal part consisting of receiver electronics and an electrode array inserted into the inner ear or cochlea. Electrical stimulation of the auditory nerve fiber array is then conducted by pulsed stimulation with a relatively high frequency through the tissue of the inner hair cells. Thereby, some problems arise with the most obvious ones being a signal distortion by the reduction in available frequency channels and relatively strong channel aliasing or crosstalk effects due to electrode proximity in the electrically conducting tissue.
The aim of this work is to reduce the channel crosstalk by employing a new electrode stimulation method known from high frequency engineering as Phased Array Channeling.

Tue 14:00 Jurriaanse Zaal Structure-borne sound in buildings

Prediction of the sound radiation from a plate excited by a structure-borne sound source

Matthias Lievens, Pascal Dietrich and Christoph Höller
Institute of Technical Acoustics, RWTH Aachen University

The radiation of structure-borne sound into a medium is generally predicted by linear system theory. In characterising a structure-borne sound source and its radiation, the goal is to predict the sound pressure at an arbitrary point in the medium using only mobilities and frequency responses of the individual system components. This seems straight forward but in practice the prediction often fails due to one or more of three causes: an incompletely measured coupling situation, a high measurement uncertainty of the mobilities, or the non-linear source behaviour.
To investigate the validity of the multipole theory a small one-dimensional measurement setup was built. It consists of a small electrical motor connected to a plate that radiates into a receiving room of one cubic metre. In this way many positions, plates and motor excitations can be analysed with limited effort.

The analyses will compare power measured in situ to power predicted using in situ measured mobilities and source excitations. The sound pressure in the receiving room is used to verify the assumption of a one dimensional excitation process and to monitor the error in the prediction.

Tue 14:20 Jurriaanse Zaal Structure-borne sound in buildings

Experiences with characterising Simple Sources of Structure-borne Sound by the Two Plate Method

Heinrich Bietz and Volker Wittstock
Physikalisch-Technische Bundesanstalt, Braunschweig

The assessment of the ability of a vibratory source to inject sound power into a receiver is a major task in different fields of acoustics. Recently, a proposal has been made to characterise structure-borne sound sources by two properties, an activity and a mobility quantity. They are determined by connecting the source under test to two different receiving plates with very different point mobilities. From the different sound powers injected into the different receiving plates, source quantities can be calculated. At PTB, it was checked whether this method is capable to determine the source characteristics of the very easy case of an electrodynamic shaker. This source has the advantage that it can be modelled by electromechanic analogies and thus, source properties can be determined by a second method. Additionally, the power input into the receiving plates can be measured by an impedance head and the electric input power is also known. Furthermore, impulse signals can be used with shakers as well as stationary ones which gave the opportunity to investigate whether the two plate method works in all cases. Presented results will include comparisons between sound powers and source quantities determined by different methods for stationary and impulse signals.

Tue 14:40 Jurriaanse Zaal Structure-borne sound in buildings

On force- and moment mobilities of a timber joist floor

Andreas R. Mayr[a], Barry Gibbs[b] and Heinz-Martin Fischer[a]
[a]*Hochschule für Technik Stuttgart;* [b]*University of Liverpool, School of Architecture*

In wood framed buildings, floors are formed by fastening wood sheathing to joists spaced at a regular interval. The sheathing is typically fastened to the joist using screws so the resulting system is a complicated periodic point-connected plate-rib structure. It is shown that the point force mobility varies significantly with position. A machine installation generally is close to discontinuities, such as at: floor edges, joist-screw locations, joints between the sheathing plates, and due to workmanship. In such cases, moment excitation might become important at some frequencies and the neglect of moments a priori can lead to inaccurate prediction of the total emission. This paper presents measured point moment mobilities with respect to the distance to discontinuities for a timber joist floor where a single layer of chipboard forms the sheathing. It is shown that the measured point moment mobility indicates an infinite plate behaviour. This includes for positions above a joist but between screw positions. To determine the relative contribution of moments and perpendicular forces to the total structure-borne sound power, case studies of two sources, a fan unit and a whirlpool bath, are described, for various locations on a timber joist floor.

Tue 15:00 Jurriaanse Zaal Structure-borne sound in buildings

Conception of test setups for investigations on the sound transmission from landings in the staircase test facility

Christoph Fichtel[a], Tobias Schneiderhan[b] and Jochen Scheck[b]
[a]*STEP GmbH;* [b]*Hochschule für Technik Stuttgart*

The German standard DIN 4109 doesn't allow an accurate prediction of the normalised impact sound pressure level from resiliently supported heavy stairs and landings. Investigations on representative landing constructions are carried out in the staircase test facility to identify and quantify the parameters that are relevant for the sound transmission. The conception of the test setups requires an analysis of the whole transmission system including the landing, resilient layer and the test environment itself to enable a proper transformation of the laboratory data into building situations. A particular difficulty is the simulation of varying pressure conditions on the resilient elements as found in buildings. The presentation contains the latest results of the ongoing investigations.

Tue 15:20 Jurriaanse Zaal Structure-borne sound in buildings

Vibration behaviour and structure-borne sound transmission of a resiliently supported landing

Emre Taskan, Jochen Scheck and Heinz-Martin Fischer
Hochschule für Technik Stuttgart

The expected normalized impact sound level for heavy stairs and landings is presently predicted for less and obsolete constructions and insulation in the latest version of the DIN 4109. Thus the calculation of the standard impact sound level for resiliently supported heavy stairs and landings is according to DIN 4109 - Beiblatt 1 impracticable. By analyzing the structure borne transmission paths, appropriate acoustical parameters can be related to the involved elements (i.e. flight of stairs, stair landing or decoupling elements) and included into a calculation model, for example DIN EN 12354. The vibration behavior of the landing and the transmission through the resilient layer are of particular importance. The vibration behavior of a representative landing construction was investigated by means of experimental and computational modal analysis and a finite element model was set up. It is intended to advance the FEM model in order to simulate the transmission from the landing through the resilient layer into the wall to enable parameter studies.

Tue 16:20 Jurriaanse Zaal Structure-borne sound in buildings

Side Effects of the ISO Tapping Machine as a Walking Noise Source

Werner Scholl, Heinrich Bietz and Volker Wittstock *Physikalisch-Technische Bundesanstalt, Braunschweig*

Characterising noise from different floors, when people are walking on them, has become important in some countries, as there exist legal requirements. Finding an adequate measurement method was made a work item of an adhoc-group of CEN TC 126 / WG 1. Based on former research, it is proposed to use the standard tapping machine according to ISO 140 as a walking noise source. As this kind of noise has to be measured in the same room where the the walker acts, the tapping machine might unwantedly influence the sound from the floor in two ways: On the one hand its body might interfere with the sound radiation from the floor covering by shielding or resonance effects, or on the other hand it might produce noise by itself ('self-noise'). A particular problem is, that the self-noise of the tapping machine depends on the surface it hits on. So it has to be determined on a very silent floor mockup. Investigations were carried out at Physikalisch-Technische Bundesanstalt Braunschweig (PTB) and some results as well as a proposal for such a mockup are presented.

Tue 16:40 Jurriaanse Zaal Structure-borne sound in buildings

Study of the Influence of Adjacent Elements on the Sound Level Decay of Heavy Building Structures by Means of Transient SEA

Martin Schneider, Florian Mack and Heinz-Martin Fischer
Hochschule für Technik Stuttgart

The sound reduction index of heavy constructions depends on the energy losses from this element into the respective adjacent building elements. For each in-situ situation different energy offtake into the coupled elements may occur. When measuring the sound reduction index of a building element in the laboratory, it is therefore necessary to determine the loss factor of this element in its respective laboratory condition. But this loss factor measurement may not only contain energy flux from the considered element into the adjacent elements but also energy flux from the adjacent elements into the considered one. The energy flux back into the considered element depends on the energy levels and energy level difference between the elements and therefore on the internal losses of the individual elements. This time dependent energy flow is modeled with a transient SEA approach. This approach gives us an idea about the circumstances under the energy flux from the adjacent elements to the considered element will influence or determine the level decay. Furthermore measured reverberation times of heavy walls are compared with the decay calculated by the transient SEA.

Tue 17:00 Jurriaanse Zaal Structure-borne sound in buildings

Vibration reduction indices at junctions with cavity walls

Heiko Martin[a], Jan Smits[a], Jan Niggebrugge[b], Eddy Gerretsen[c] and Renz van Luxemburg[d]
[a]*Eindhoven University of Technology, Acoustics Laboratory;* [b]*Kupers& Niggebrugge, Utrecht;* [c]*TNO;* [d]*DHV B.V., Eindhoven*

By applying concrete or brick cavity walls in dwellings instead of single homogeneous walls, the sound insulation between dwellings can be increased. However, then the role of flanking sound transmission is becoming more important. Flanking transmission depend partly on the properties of the building junctions formed by the cavity walls and other building elements. The properties of junctions made of single homogeneous building elements are known for some time. Few data are available of junctions of which cavity walls are part. In calculation models empirical modules are used for the vibration reduction indices of these specific junctions. Their accuracy is not known yet because of missing data from practice. Therefore, vibration reduction indices of several junctions with cavity walls have been determined by field measurements, also in dwellings under construction. For this purpose, a quick and reliable measurement method had to be developed. Also the invariance of the vibration reduction indices concerning the same junction in different surroundings has been investigated.

Tue 17:20 Jurriaanse Zaal Structure-borne sound in buildings

A Wave Based Model to predict the Airborne and Structure-Borne Sound Insulation of Finite-Sized Multilayered Structures

Arne Dijckmans[a], Gerrit Vermeir[a] and Walter Lauriks[b]
[a]*KU Leuven, Afdeling Akoestiek en Thermische Fysica;* [b]*KU Leuven*

In this paper, a Wave Based Method (WBM) is used to predict the airborne and impact sound insulation of finite-sized multilayered structures. The WBM is a Trefftz-based deterministic prediction technique for the steady-state dynamic analysis of coupled vibro-acoustic systems. The field variables (sound pressures and plate displacements) are expanded in terms of acoustic and structural wave functions, which are exact solutions of the homogeneous dynamic equations. The contributions of the wave functions depend only on the boundary and continuity conditions. Because the modal behaviour of plates and rooms is taken into account, the WBM is not limited to the high-frequency range like Statistical Energy Analysis. The enhanced computational efficiency, as compared to other deterministic methods like finite element models, allows predictions in a broad frequency range (typically 50-5000 Hz in building acoustics). As many problems in building acoustics can be simplified to a rectangular geometry, the WBM is developed for this specific case. Predictions for multilayered structures containing air layers and porous layers are

compared with analytical prediction models and measurements. The influence on the sound reduction index of different parameters, like concentrated absorption in sending or receiving room, is discussed.

Tue 17:40 Jurriaanse Zaal Structure-borne sound in buildings

Combining Experiments for the Identification of the Parameters of Viscoelastic Materials

Martin Schmelzer
Physikalisch-Technische Bundesanstalt, Braunschweig

Certain memory effects, that show up during the deformation of many materials of technical importance, can be described by a law of linear viscoelasticity. Using inner variables, arbitrary signals in time can be handled easily. In the special case of harmonic signals, this naturally leads to frequency dependend complex moduli.
The identification of the associated parameters can be performed by a process of optimisation using iterative simulation of the experiment and the comparison of the calculation with the experimental data. Several experiments with one material can be combined to yield a united set of parameters. Results of this process will be presented.

Tue 18:00 Jurriaanse Zaal Structure-borne sound in buildings

Building SEA Predictive Models to Support Vibro-Acoustic Ship Design.

Denis Blanchet
ESI GmbH, Munich

In the past, the shipping industry has used empirical models extensively to predict the vibro-acoutic response aboard ships. This method has proven effective for reasonable ship size, shapes and typical materials. However, empirical approaches offer little flexibility and are limited to known construction types and materials. Nowadays, ships are getter larger and use more sophisticated material such as composite. Regulation and ship owners impose more stringent vibro-acoustic performance of their vessels and the luxury yacht segment is no exception. In addition, the time available to design the vibro-acouctics package aboard a ship is shrinking forcing designers to seek industrially efficient methods to build ship models and perform design analysis.
This paper presents an efficient method for quickly building a detailed predictive vibro-acoustic model from 2D or 3D data. This model includes all SEA structural subsystems of the hull, superstructure, interior bulkhead, etc. It also includes SEA acoustic subsystems of all cabins and other living areas. Structureborne and airborne sources are defined for all types of excitation encountered in a ship such as engine, generator, HVAC, bow thrusters, propeller.... Vibration, SPL and power inputs are available at any locations and serve as basis for vibro-acoustic design analysis.

Tue 14:00 Fortis Bank Zaal Variable acoustics

Why Variable Acoustics - and What Variable Acoustics?

Eckhard Kahle
Kahle Acoustics, Brussels

Why do we need variable acoustics, and what needs to be varied, both in terms of physical elements and in terms of acoustic criteria? In terms of why, we feel that there are three complementary reasons: (1) musical needs, as contemporary concert halls (and other venues) have to cater for music composed for different spaces and musical ensembles; (2) user needs, for communities that cannot afford to build several single purpose halls, and therefore need to build a single multi-purpose hall; (3) sociological-economical needs that have been developing over the last couple of decades, as most hall operators want to offer their public an eclectic programme including different types of music within the same, ”their” hall. Concerning the question of what variable acoustics needs to achieve, we feel that reverberation time is only one, and probably not even the most important criterion for variable acoustics. Which other parameters do we need to change in order to achieve successful variable acoustics in a hall?

Tue 14:20 Fortis Bank Zaal Variable acoustics

Variable Acoustics means Variation of Reverberation Time - does it?

Ben Kok
Nelissen Ingenieursbureau B.V., Eindhoven

It is generally recognized that the acoustics of a room are not defined by its reverberation time only. For variable acoustics, however, the variation in reverberation time still is the commonly used quantifier of the effectiveness of the variation. This leads to the impression that variation of reverberation time is essential for effective variable acoustics. Acting as the acoustic consultant for a theatre renovation project, we were confronted with a client who required optimum acoustic performance for both opera and non-reinforced drama in the same house, but did not want to implement variable acoustics. Discussions with the client learned that their opposition to variable acoustics was in the expected complexity of the provisions, not only during construction, but also in operation and maintenance. A simple system was proposed to influence the early reflections, but without change of reverberation time. The resulting variation of about 1 dB in C50 and 1.5 dB in C80, at a reverberation time of 1.4 s, effectively changes the auditorium from a speech theatre to an opera house. After the renovation, the acoustics have been received very favourably, which has been expressed by the critics of ’Opernwelt’ by electing the theatre ’Opernhaus des Jahres’ in 2007.

Tue 14:40 Fortis Bank Zaal Variable acoustics

Developments in the Acoustic Design of Theatres with Natural Variable Acoustics

Maarten Luykx, Martijn Vercammen and Rob Metkemeijer
Peutz bv, Mook

In designing variable acoustics in theatres in the 1970's in the Netherlands Peutz' philosophy was to start with a hall size necessary for symphonic music. These halls are therefore usually quite large (9,000-11,000 m3), including an orchestra shell. To reduce the reverberation time and volume down to 5,000-6,000 m3 for the theatre mode curtains can be unrolled from ceiling boxes. These 1000 seat halls work well, the only drawback being the added sound absorption that compromises the strength for speech. A next step was therefore to use movable ceiling panels to obtain a more effective volume variation. A final step in this development was made with theatre "De Spiegel" (Zwolle (NL) 2006), which contrary to all others has a compact and intimate theatre as a starting point. It has 850 seats, a room volume of 3,500 m3, and 2 horse-shoe shaped balconies. Using a movable ceiling up to 20 metres height and a gallery, an additional reverberant volume of 4,000 m3 can be added to the hall, creating a total volume of 11,000 m3 for symphonic music. Reverberation times range from 0,9 s up to 2,0 s, without compromise to the strength of the sound in the theatre mode.

Tue 15:00 Fortis Bank Zaal Variable acoustics

Acoustics of Muziekkwartier Enschede (NL).

Niek Janssen and Renz van Luxemburg
DHV B.V., Eindhoven

November 22, 2008 the new building "Muziekkwartier" (Music Quarter) in the town of Enschede (NL) is inaugurated. The building will be the new home of the Opera (Nationale Reisopera), the Theatre (Podium Twente), the School of Music (Muziekschool Twente) and the Stage for Popmusic (Atak).

The main auditorium is to be used for drama, opera and as a special: for opera production. The auditorium will be the second hall specific for opera in the Netherlands, after Amsterdam. The capacity of the hall is 1000 seats, the volume of the hall is 9500 m3. Varying the acoustics in the hall is solved mechanically. Movable sound reflectors moved in height, ground plan and vertical angle are the ingredients for variable acoustics meant for speech or opera. The variation in reverberation time is T = 1.3 to 1.6 sec. in combination with strong support of early reflections.

The combination of functions, performing simultaneously without mutual disturbance, leads to a high aim on the sound insulation in the building.

The acoustics of both the halls for popmusic were not only well-damped, but also pinpointed to performance in low frequencies and neutralizing the absence of visitors during sound checking.

Tue 16:00 Fortis Bank Zaal Variable acoustics

Relationship between room shape and Early Lateral Energy Fraction in rectangular concert halls

Andrzej K. Klosak[a] and Anders C. Gade[b]
[a]*Cracow University of Technology;* [b]*Technical University of Denmark*

The acoustic properties of 24 rectangular rooms representing ”shoebox” type concert halls with volumes between 8000m^3 and 16000m^3 have been analysed using Odeon (v.8) computer simulation software. Among several objective measures calculated in numerous receiver positions (between 300 and 850) in each hall, this paper concentrates on the behaviour of Early Lateral Energy Fraction (LF_{80}). The dependency of LF_{80} on source-receiver distance, receiver location within the room and room dimensions will be discussed. It is found that the distribution of LF_{80} in rectangular rooms can be described as a three-dimensional ”tongue-shape”. The simulation results have also been compared with regression models based on measurements in real halls. In rectangular halls, the variation in LF_{80} values can be described by the influence of the overall room dimensions and their mutual ratios. Formulas for prediction of LF_{80} in rectangular halls have been proposed, which takes into account both the width and the length of the hall.

Tue 16:20 Fortis Bank Zaal Variable acoustics

Variability and Adaptability of the Acoustics in the new Conservatory of Amsterdam

Peter Heringa, Debby Isbrücker and Marten Valk
Peutz bv, Zoetermeer

Last summer the newly built conservatory for music of Amsterdam was delivered. The conservatory has 5 larger halls. The primary use of these halls is rehearsal and exams. They are also used for musical performances. The halls are quite different in order to learn the students to cope with different acoustics. They range from a hall suitable for rehearsal of an symphonic orchestra to a recital hall, a hall for jazz and pop and a hall for opera and musical. The acoustics of three halls can, to a certain extend, be adapted to the use of the halls. The acoustics aimed and achieved and the variation possibilities will be subject of the invited paper. Furthermore the conservatory has more then 100 education rooms and about 70 study rooms. The acoustics of these rooms can, by means of ”cushions”, be adapted to the lessons scheduled. The needed adaptability was predicted by means of experiments together with students and teachers in a mock-up of ”new rooms” in the former conservatory building. They had to express their preference and give their evaluation of different subjective parameters. The adaptability and the evaluation of it will be shown.

Tue 16:40 Fortis Bank Zaal Variable acoustics

Variable Acoustics in rehearsal rooms and concert halls

Gernot Kubanek and Georg Jansen
ISRW Dr.-Ing Klapdor GmbH

To optimize acoustic conditions in rooms with a certain change of use or requirements, flexible approaches are possible, according to practical realization.

For rehearsal rooms with more or less small dimensions and the need for simple constructions, often just constructional approaches are needed. For the user (vocalist - percussionist) is essential, that these mobile methods are unique and easy to handle with in daily use.

For rooms with greater dimensions are not only the constructional methods, but also electroacoustic systems of great importance. Not just to create simple enhancement of reverberation; the wave field synthesis can be an appropriated strategy to create different acoustic situations for example at the National Conservatory of Music Detmold (Germany). The system is dedicated to sound reproduction for artistic purposes

The system comprises 346 independent loudspeaker channels, including a horizontal loudspeaker array all around the auditory (500 seats) and ceiling loudspeakers. Since the hall is used for a broad repertoire comprising chamber music, romantic orchestra instrumentations, organ concerts, contemporary music, etc., the hall will be equipped with a variable room acoustic system.

The paper presents aspects of system design concerning the direct sound and diffuse field as well as practical implementations for WFS rendering.

Tue 17:00 Fortis Bank Zaal Variable acoustics

Creating Temporary Venues for High Demanding Classical Concerts using a Room Enhancement System

Gunter Engel and Marcus Blome
Müller-BBM GmbH

In the field of classical concerts an increasing tendency of arranging concerts in unusual and sometimes unsuitable venues can be observed. There are various possibilities to acoustically improve these venues, but most of them turn out to be impracticable due to the limited preparation time and an unacceptable appearance of the acoustical measures. In many cases the only solution can be provided by the use of room enhancement systems, which counteract the acoustic deficiencies with electro acoustical means. The lecture presents case studies of two festivals in St. Moritz and in Abu Dhabi, where ball rooms, dining halls or auditoria of first class hotels are used as venues for top level orchestras and soloists. Beside the benefit for the musicians and concert audience the use of enhancement systems helps acousticians to improve the knowledge about the correlation between measurement results and the audible impression.

Tue 17:20 Fortis Bank Zaal Variable acoustics

Proposal for a new parameter for acoustic absorption taking amplification into account

Cees Mulder
ing. Cees Mulder, Rotterdam

In Sabine's equation for reverberation time and the equation for the sound pressure level of the diffuse sound field, the acoustic absorption A [m2 open window] is a major parameter. This absorption is obtained from the product of the surfaces S [m2] that enclose the room and the associated absorption coefficient α [-] of those individual surfaces. However, the effective absorption A in a room changes when amplification systems are used in that room. This phenomenon of a change in acoustic properties of a room has been described when using amplification (Franssen, Acustica 1967), but never or seldom it has been described as a change of absorption. When rooms are designed where the use of a high gain amplification system is intended, like artificial reverberation systems, surround sound systems or speech amplification systems that incorporate decorrelation techniques, the change of absorption has to be taken into account. A new parameter for the acoustic absorption will be presented where the electro-acoustic gain is included. This will make it possible to determine the architectural absorption A when a certain electro-acoustic gain is expected.

Tue 17:40 Fortis Bank Zaal Variable acoustics

Large scale in situ acoustic absorption properties measurements in a theatre.

Emiel Tijs
Microflown Technologies

The acoustic absorption properties of materials used in a theatre are a major concern, both in the initial design stage and during renovation. The PU surface impedance method allows the simultaneous measurement of both sound pressure and acoustic particle velocity, providing the opportunity to measure materials as they are installed on seats, floors, walls and ceilings, assessing their sensitivity to both normal and oblique angles of sound wave incidence. Using a portable piece of equipment, actual testing time is short, allowing large scale testing for comparisons. The method was applied at the Musis Sacrum theatre in Arnhem, the Netherlands. Results will be presented and findings will be discussed.

Tue 18:00 Fortis Bank Zaal Variable acoustics

A Comparison of the Variable Acoustics of Two Foley Studios for Sound Effects Recording

Remy Wenmaekers[a] and Constant Hak[b]

[a]*Level Acoustics, Eindhoven;* [b]*Eindhoven University of Technology*

The sound effects and voices that are heard in motion pictures are often replacements or additions of sounds recorded in a Foley Studio. In this type of studio, named after Jack Foley the pioneer in sound effects recording, many objects like wooden floors and doors are present to record sounds from. These recordings need to sound as they are heard in the real world or even as an exaggeration of that world. Therefore it should be possible to create a lot of different acoustical conditions in a Foley Studio. The variability of acoustical properties of two existing Foley Studios have been compared: one typical studio and one with a good reputation. The impulse responses of typical Foley recording setups have been measured and analyzed with DIRAC software in both studios. The results show that in order to achieve sufficient acoustical variability in a Foley Studio it is important to make the critical distance and individual reflections controllable. It should also be possible to make the room semi-anechoic.

Tue 14:00 Van Cappellen Zaal Binaural systems

Parametric Binaural Synthesis: Background, Applications and Standardization

Jeroen Breebaart

Philips Research, Eindhoven

The synthesis process of binaural sounds has been subject to research for several decades. The employment of head-related transfer functions (HRTFs) has greatly progressed both binaural research as well as binaural applications. At the same time, the amount of information present in HRTFs and the required processing capabilities for real-time binaural rendering have long been a challenge for many applications in the consumer domain. More recently, parametric methods to capture the perceptually-relevant information from HRTFs have been developed. By means of extracting perceptually-relevant attributes from HRTF pairs, binaural rendering can be performed at lower complexity compared to the employment of HRTF convolution. Moreover, parameter-based binaural rendering can be efficiently integrated with parametric audio coders, it can provide a more convincing spatial image when converting conventional stereo to binaural signals, and can be used to provide means to independently adjust the spatial position of individual sound sources present in a down mix. This paper provides an overview of the perceptual consequences and limitations of HRTF parameterization, its applications, and relevant standardization efforts.

Tue 14:20 Van Cappellen Zaal Binaural systems

Preserving binaural hearing of hearing impaired subjects with binaural noise reduction systems for hearing aids

Tim van den Bogaert[a], Jan Wouters[a] and Marc Moonen[b]
[a]*KU Leuven, ExpORL;* [b]*KU Leuven, ESAT-SCD*

Hearing aid users experience great difficulty understanding speech in noisy environments. Therefore noise reduction algorithms are introduced in hearing aids. The development of these algorithms is typically done monaurally. However, the human auditory system is a binaural system. Providing two monaural, independently operating, noise reduction systems to the hearing aid user may disrupt binaural information needed to localize sound sources correctly and to improve speech perception in noise.

We first examined the influence of commercially available, bilateral, noise reduction algorithms on binaural hearing. Objective and perceptual evaluations with normal hearing and hearing impaired subjects show that fixed and adaptive directional microphone (ADM) configurations, two of the most commonly used noise reduction configurations in hearing aids, can significantly distort the binaural properties of the sound signals. As a second step, binaural noise reduction schemes based on a multichannel Wiener filter (MWF) approach were developed and evaluated. It is observed that a binaural hearing aid design significantly increases noise reduction performance owing to the larger number of microphones used. Moreover, the binaural MWF and the binaural MWF with partial noise estimation provide a better combination of noise reduction performance and preservation of binaural cues compared to the bilateral ADM algorithm.

Tue 14:40 Van Cappellen Zaal Binaural systems

On the application of auditory scene analysis in hearing aids

Anton Schlesinger and Marinus M. Boone
TU Delft

A big challenge in hearing aid (HA) design is the improvement of speech intelligibility (SI) in adverse acoustical situations. Improving SI in noise is a main demand for the majority of hearing impaired people who are afflicted by a sensorineural hearing loss. Approaches to compensate for this dysfunction do not pose a general solution to the problem. There are encouraging approaches that rather replace auditory functions, in particular the function of spatial filtering. Today's HA's are thus often equipped with directional microphones or microphone arrays, which successfully enhance the signal-to-noise ratio (SNR). As the hearing apparatus is a superior speech processor, a further developmental leap in HA design is expected from the emulation of models of auditory scene analysis (ASA). Common to ASA-models is the categorization of sound sources in a feature-space. Using this amended representation, the target-speaker and interfering sound sources are separably displayed. In analogy with the psychological attention to one perceptual stream, it is possible to facilitate these ASA-models to enhance a given target. Several of these functional ASA-models showed a considerable improvement of SNR in a broad range of acoustical situations. In this contribution we discuss representative functional ASA-models and sketch their implementation in HA’s.

Tue 15:00 Van Cappellen Zaal Binaural systems

Source Positioning in a Two listener Crosstalk Cancellation System

Bruno Masiero

Institute of Technical Acoustics, RWTH Aachen University

It is long known that to correctly reproduce a binaural signal through a pair of loudspeakers, it is necessary to pre-filter the binaural signal to compensate for the loudspeakers’ crosstalk. To study the performance of crosstalk cancellation (CTC) systems, or to answer the question about the ideal positioning of loudspeakers, several analytical models with varying degrees of complexity have been used. However, so far, all implemented CTC systems were designed for a single listener use. When it comes to two (or more) listener CTC systems, the conditioning of the transfer matrix becomes more irregular than the conditioning of the transfer matrix for a one listener CTC system, making the task to find the optimal source positioning considerably more complicated. By modelling the listeners’ heads as two rigid spheres in free-field, the optimal positioning of the four sources in a two listeners CTC system was analysed, using for that two potential source distribution arrangements, namely a linear and a circular scheme.

Tue 15:20 Van Cappellen Zaal Binaural systems

Evaluation of Binaural Hearing with Cochlear Implants

Joachim Müller-Deile

Christian-Albrechts-University of Kiel

Bilateral cochlear implantation is a well established clinical procedure in rehabilitation of deaf or residual hearing patients. Aims of the bilateral cochlear implantation are the improvement of localisation of sound sources, the improvement of speech intelligibility in noise, an improvement of quality of live and the implantation of the better ear. The direct labour necessary to fit the speech processors of a bilateral implanted patient are more than double the expenditure of work for a unilateral implanted patient. Methods to adjust loudness and pitch for both sides are shown. It will be discussed to what extend the clinically used speech processors can transmit the cues used with normal hearing in binaural directional hearing. Results are presented for just noticeable differences of interaural time delay and interaural level differences by direct electrical stimulation of single electrodes and by using the speech processors. There is hope that better speech processing algorithms will at least in some patients improve their auditive localisation abilities. Early bilateral implantation of congenitally deaf children ends up in better results in directional hearing tests than with children with sequential implantation with long times between the operations. Findings with adults indicate that bilateral with actual speech processors supplied patients improve more from interaural level difference than form time differences.

Tue 16:20 Van Cappellen Zaal Binaural systems

Application of binaural technology in an adaptive auditory speech test for children

Janina Fels[a], Frans Coninx[b] and Wolfgang H. Döring[c]
[a]*Institute of Technical Acoustics, RWTH Aachen University;* [b]*Institut für Audiopädagogik at the University of Cologne;* [c]*RWTH Aachen, Otorhinolaryngology, Plastic Head and Neck Surgery*

This contribution reports on the development of a screening procedure for the identification of hearing loss in children by using a speech test in various spatial conditions.

A study looking at the performance of the Adaptive Auditory Speech Test (AAST) under spatial locatable noise as a possible diagnostic procedure for Auditory Processing Disorders (APD) has been carried out. The AAST was developed to determine the Speech Recognition Threshold (SRT) in a fast and reliable way under quiet and noisy conditions for children of 3 to 4 years of age.

In the study the AAST is used to develop a diagnosis method with a noise made by a talk between children (Two-Talker-Noise) or talking noise (person reading a newspaper) to test the intellectual abilities, separation and localization.

The test was carried out under various spatial conditions with the signal and noise coming from different positions in the horizontal plane. The signals were generated using head-related transfer functions which are appropriate for this age group. Therefore, the test words and the talker-noise were convolved with the corresponding children HRTFs. The scenarios were presented with the help of headphones.

Significant differences were detected between the different spatial situations. Results and experiences from current screening programs will be presented.

Tue 16:40 Van Cappellen Zaal Binaural systems

Spatial coherence in binaural applications

Roman Scharrer
Institute of Technical Acoustics, RWTH Aachen University

Spatial coherence and correlation have a well known behaviour for different types of sound fields. There are analytic expressions for free field conditions as well as diffuse sound fields. This makes spatial coherence an interesting indicator in signal processing for reverberated and noisy situations as well as acoustic situation recognition. The behaviour in real sound fields is not that well known and there are a lot of parameters influencing analysis results. Some examples are signal to noise ratio, analysis block size and averaging. Former research results show a good

agreement between theoretic calculations and real measurements, but usually they are using long time or spatial averaging which makes that approach unusable for real time signal processing purposes.
Interaural coherence in binaural recordings is analysed in different room acoustic situations and compared to theoretic values. Measurements were carried out in a reverberation chamber, under free field conditions and in different rooms. The measurement results are compared to those of diffuse and free field models as well as room acoustic simulation and the influence of analysis parameters is examined. Finally typical binaural coherence behavior is analysed for different room sizes, reverberation times and sources based on a binaural room acoustic model.

Tue 17:00 Van Cappellen Zaal Binaural systems

A Boundary Element model to calculate HRTFs. Comparison between calculated and measured data.

Wolfgang Kreuzer and Piotr Majdak
Austrian Academy of Sciences, Acoustics Research Institute

Head-related transfer functions (HRTFs) play an important role in spatial sound perception. With the boundary element method (BEM) it is possible to numerically calculate these functions. As a rule of thumb six BEM elements per wavelength should be used, thus it is necessary to have very fine meshes when calculating HRTFs at high frequencies. In that case, the solution of the linear system of equations generated by the BEM needs a lot of computing time and memory. Advanced algorithms like the fast multipole method (FMM) speed up calculations and reduce memory requirements which allows a computation of HRTFs even for frequencies as high as 20 kHz in feasible time.
In this work, we give a short overview of a BEM model coupled with the FMM. We present calculated results for several persons and compare the results with measured data of these persons.
This work was supported by the Austrian Science Fond (Pr. P18401-B15)

Tue 17:20 Van Cappellen Zaal Binaural systems

The influence of different microphone mounts on measured HRTFs

Christopher Haut[a], Jacqueline Rausch[a] and Volker Mellert[b]
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When measuring HRTFs, that contain the directional information for localization in the acoustic domain, the insertion of microphones into the ear canal changes the acoustic impedance at its entrance. Hence different microphone mounts cause different alterations of the HRTFs. The influence of different microphone mounts on HRTFs, measured in the horizontal and medium plane, have been investigated systematically. The mounts used are common foam- and silicone-plugs together with a newly developed one, which was to minimize the influence on the original ear-canal entrance impedance. The HRTFs, gained with different

mounts, will be presented together with a comparison to some HRTFs in open-access databases.

Tue 17:40 Van Cappellen Zaal Binaural systems

Just Noticeable BRIR Grid Resolution for Lateral Head Movements

Frank Schultz, Stefan Weinzierl and Alexander Lindau
TU Berlin, Fachgebiet Audiokommunikation

The angular grid resolution of binaural room impulse response datasets can be considered as an important factor regarding the quality of binaural auralisations. Just noticeable grid resolutions for horizontal and vertical head movements have been investigated by different authors, mainly for frontal sound sources. We will present the results of a new listening test, using free field (HRTF) BRIR datasets for a sound source placed *above* the head, assuming that this is the most critical configuration for lateral head movements. The listening test was conducted using a 3AFC procedure with adaptive *The Best PEST* procedure. The results reveal a remarkably high sensitivity of listeners towards a missing interaction of binaural simulations and lateral head movements.

Tue 18:00 Van Cappellen Zaal Binaural systems

Blind and Non-Blind Spatial Signal Processing Using Head-Related Impulse Responses

Hendrik Kayser[a], Birger Kollmeier[b] and Jörn Anemüller[b]
[a]*Universität Oldenburg;* [b]*Universität Oldenburg, Medizinische Physik*

Spatial signal processing for hearing aids is of particular importance in acoustically challenging environments. The development and testing of algorithms performing digital signal processing, e.g., blind source separation (BSS), beamforming and direction-of-arrival estimation requires realistic test conditions. For the evaluation of such blind and non-blind algorithms, virtual sound fields are generated by convolution of sound signals with impulse responses measured for different spatial source positions. By the use of head-related impulse responses measured in real scenarios, a simulation is achieved that approximates real-world recordings. For this purpose, multi-channel hearing aids were mounted on a head and torso simulator. Hence, the number and type of superimposed sound sources can be chosen arbitrarily. Background noise recorded in the underlying scene can be added with a selectable SNR. Due to the knowledge of the entire transmission characteristics from each source to each microphone, optimal linear filtering techniques are applicable. The results achieved by multi-channel Wiener filtering serve as benchmark for practical filtering algorithms having less knowledge about the underlying situation. A BSS approach is applied to multiple mixtures of speech signals in anechoic conditions and an office room. The quality of separation is evaluated subject to the spatial arrangement of the sources.

Tue 14:00 Van Beuningen Zaal Sound scaping and sound masking

Informational masking and attention focussing on environmental sound

Dick Botteldooren and Bert de Coensel
Ghent University

Appreciation of a soundscape depends on the cognitive and emotional evaluation of the sonic environment with particular emphasis on the meaning associated to the perceived sound. Before meaning can be attributed, a sound has to be detected and recognised. Energetic masking due to the working of the inner ear has been mentioned as a reason for not hearing an environmental sound. The central nervous system however also plays role. Informational masking may prohibit recognising or understanding the sound. In this context attention focussing is always around the corner. As attention gets focussed on an auditory stream, informational masking could be reduced even if the signal to noise ratio drops considerably after initial detection. It is for this reason that we include attention shifting in our computational model for environmental noise perception. The model starts from a separation of auditory streams based on spectro-temporal characteristics. A robust mechanism using activation and inhibition models the ability of auditory streams to attract attention. Top down attention is related to activities. The outcome of the model includes the time that particular environmental sounds are heard, their level, saliency, ... This could allow to validate designs based on masking unwanted components of the sonic environment.

Tue 14:20 Van Beuningen Zaal Sound scaping and sound masking

Soundscape Design of Urban Spaces

Giovanni Brambilla[a], Maria di Gabriele[b], Luigi Maffei[b] and Patrizio Verardi[a]
[a]*CNR-Institute of Acoustics, Roma;* [b]*Built Environment Control Laboratory, Second University Naples*

In designing urban spaces perceptually more acceptable and comfortable, the available techniques of auralization and visual rendering are a valuable tool as they enable to create a virtual environment which can be experienced and evaluated by people. This procedure can provide useful hints to select the most effective solution, and hopefully less expensive as well, to either building new urban spaces or improving existing ones, taking into account the human perception and preference. The application of the above techniques in the restoration of some urban squares in Rome and Naples are described in this paper. Different lay-outs and corresponding sonic environments have been considered and submitted to a group of subjects in laboratory for their visual and audio evaluation. Foliage and natural sounds, such as fountains and bird twittering, have been included in the proposed plans as objects and sounds masking the road traffic noise. Pedestrian zones and substitution of petrol buses with electric ones, have been also considered in the proposed lay-outs.

A questionnaire has been design to collect the subjective ratings on the audio and visual environments separately in order to evaluate their relationship and their influence on the overall rating. The subjective tests are in progress.

Tue 14:40 Van Beuningen Zaal Sound scaping and sound masking

Realistic Prospects for Sound Source Description and Recognition for Complex Soundscapes

Tjeerd Andringa, Dirkjan Krijnders, Maria Niessen and Maarten van Grootel
University of Groningen

Automatic analyzes and descriptions of soundscapes, which are qualitatively similar to human descriptions, are very important to objectify soundscape research and to develop convincing tools for soundscape analysis. One possible way to develop such an automatic soundscape analysis system is to start with an annotation system for unrestricted real-world sounds and enrich this system with more and more automated functions. We will demonstrate an annotation system that automatically selects coherent time-frequency regions of similar spectro-temporal properties and similar acoustic ”textures”. The artificial system will learn through the interaction with a user whose task it is to assign names to regions. According to prototype theory of categorization, naming would structure the acoustic signals into meaningful relevant categories integrating textures and properties along family resemblance. After a low number of examples the system is able to associate an annotation with a time-frequency region. In cognitive terminology, the system will be able to use the emerging categories for evaluating new signals through top-down processing and membership decisions. We propose that systems like this can be used to bootstrap a more general, and eventually automatic, soundscape analysis system that complies with the aim stated above.

Tue 15:00 Van Beuningen Zaal Sound scaping and sound masking

Physical and Perceptual Masking: one or two Phenomena?

Danièle Dubois
CNRS, LCPE/LAM IJLRDA, Paris

While masking may be accurately defined within physical theories and described in acoustic parameters, the ”same” (or corresponding?) phenomenon seems more difficult to grasp from the perspective of cognitive (human) processing. In this case, masking actually depends on theories of perception that are not unified within the field of cognitive psychology. One main point is that masking points to the limits of a strict bottom-up conception of acoustic information processing and therefore to the necessity to identify and to take into account the differences between the way the signal is recorded and measured and the way human cognition processes it. One of the main differences is that cognition involves top down processing, which requires processes such as assigning relevance and meaning to the ”signal”. We sketch how different psychological theories account for differences in terms like figure/ground separation, attentional/incidental processing, analytic/holistic processing, or as relevant/irrelevant, and local / global properties of soundscapes. Some applied examples will be given to introduce not only theoretical discussions but also as methodological guidelines for soundscaping with psychological masking in designing environmental sounds.

Tue 16:00 Van Beuningen Zaal Sound scaping and sound masking

Sound masking in Soundscapes - decisions by the new experts

Brigitte Schulte-Fortkamp
TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics

In a current project the evaluation process shows the success through implementing the Soundscape approach with its new methods and innovative approaches for the understanding of human response to acoustical environments. Moreover, factors, among others the meaning of the noise, the composition of the diverse noise sources, the attitude and the listener’s expectations and experiences, are validated as significant parameters which were considered to comprehend the different perceptions and evaluations with regard to the chosen space. Harmony, compatibility, assimilation, and acoustical home are the new indicators found with respect to expertise of the people involved- the new experts. Moreover, the chosen Soundscape approach here assures meaningful sound masking with respect to the rebuilding of the chosen area. The related project is a module of the Project ”Nauener Platz - Remodelling for Young and Old” in the framework of the research program ”Experimental Housing and Urban Development (ExWoSt)” of the ”Federal Ministry of Transport, Building, and Urban Affairs (BMVBS)” by the ”Federal Office for Building and Regional Planning (BBR)”. The projectexecuting organization is the Regional Office Berlin-Mitte.

Tue 16:20 Van Beuningen Zaal Sound scaping and sound masking

Potential and Benefit of Audible Noise Maps

Klaus Genuit, André Fiebig and Philipp Marla
HEAD acoustics GmbH

The importance of several (psycho-)acoustic parameters to noise evaluation and noise rating was demonstrated in numerous studies. However, in conventional noise maps neither information about further acoustic parameters nor comprehensive auralization options are available. In the context of the European research project ”Quiet City Transport” the development of a traffic noise synthesizer was initiated, which deals with the idea of generating time signals representing specific traffic scenarios. On the basis of such generated time signals various acoustical parameters can be calculated and in consideration of sound propagation effects the noise for certain receiver positions can be determined. The advantage of noise maps providing audible examples of noise exposure in important immission points would be that experts and decision makers can listen to the environmental noise and can base their decisions with respect to required noise mitigation measures on a more significant basis. Of course, several criteria like accuracy, uncertainty, practicability, feasibility, interpretability of the maps must be critically discussed before considering the implementation of more extensive auralization tools in noise maps. The presentation will show some case study results. Moreover, the benefit of advanced noise maps will be highlighted and realization impediments will be discussed.

Tue 16:40 Van Beuningen Zaal Sound scaping and sound masking

Sound masking - ein Design Tool für Architekten?

Juergen Bauer

Architekt, Tramore Co Waterford (IRL)

Als Architekt und Entwurfslehrer denke ich seit einiger Zeit unter anderem darüber nach, wie das Hören in den Entwurf kommt. Architektur kommuniziert man am besten durch Modelle, Zeichnungen und Photos; ist es dann nicht folgerichtig, Akustik über Klangbeispiele, Klang-”Bilder” zu vermitteln? Im zweiten Studienjahr der Architektenausbildung am Waterford Institute of Technology in Irland haben wir mit einfachen Klangbildern gute Erfahrungen gemacht: Mit Hilfe von Hörproben aus einem Klassenzimmer und unterschiedlichen Nachhallzeiten erfahren Studenten, dass Raumklang nicht nur etwas mit Lautstärke, sondern auch etwas mit Sprachverständlichkeit und Nachhall zu tun hat. Dem Akustiker mag das banal erscheinen, für einen jungen Designer zu Beginn des Studiums kann dies jedoch zum Schlüsselerlebnis werden. Wie kann man nun Architekten an das Phänomen verschiedener (Umwelt-)klänge in ihrer Wechselwirkung heranführen? Findet eine ”sprachverständliche”,technische Definition von Soundscaping und Soundmasking automatisch ”Nachhall” im Kopf des Entwurfsarchitekten? Wie stimulierend ist eine ”White-Noise-Hörprobe” für einen Designer, bei der man 10 Sekunden nichts als stupides Rauschen hört? Wie kann Soundmasking als Gestaltungsmoment jenseits von White- und Pink Noise-Rezepten im (Architektur-)Entwurf genutzt werden? Wie wird die technisch präzise Hörprobe zum inspirierenden Klangerlebnis? Standards und Dezibels - nein danke, Strategien und Klangbilder - ja bitte! (Paper will be presented in English)

Tue 17:00 Van Beuningen Zaal Vehicle acoustics

Beamforming with high Dynamic Range

Dirk Püschel
Akustik Technologie Goettingen

A new method of beamforming is used which allows for high dynamic range of acoustic pictures. The method works in real time without any constraints on the type of noise source. Analysis results are shown for different applications in comparison to classical beamforming.

Tue 17:20 Van Beuningen Zaal Vehicle acoustics

SEA vehicle model for rolling- and engine noise

Fred Nentwich, Thorsten Bartosch and Gregor Müller
Magna Steyr Fahrzeugtechnik AG & Co KG

The task is to optimize the acoustic trim in balance with cost and weight during the medium project phase. The SEA model should predict how much a trim measure will reduce the interior noise at mid and high frequencies. The body structure has a predictive SEA description based on the latest FEM model. The trim and its acoustic behaviour are either measured or estimated via sub-model. The real excitations are modelled by defining the energy levels of their adjoining subsystems as boundary conditions. These levels are measured on a forerunner model with the dynamometer test rig and subsequently imposed on the model's matching subsystems. Excitation and structure together yield a simulation of the coast down rolling noise and run up engine noise. This enables a speedy simulation of trim variations and their acoustic impact.

Tue 17:40 Van Beuningen Zaal Vehicle acoustics

Vehicle noise reduction of multiple load cases using vibro-acoustic potential analysis

Fred Nentwich, Thorsten Bartosch and Gregor Müller
Magna Steyr Fahrzeugtechnik AG & Co KG

The engine noise and rolling noise are to be reduced with trim measures derived from a SEA model. After a SEA model with 370 subsystems has been set up, there are some 25400 non-zero coupling and internal loss factors to choose from - too much for manual labour. Therefore MAGNA STEYR developed the VAPA algorithm. It investigates how a loss factor variation affects the cabin's sound pressure level. The algorithm calculates the theoretically possible reduction presuming an unlimited loss factor und thus sets the upper limits for the noise reduction expectable from practical measures with a limited loss factor. With this paper we show an easy extension of the VAPA in order to consider a multi-load case scenario. The VAPA identifies the most over all promising vibro-acoustic potentials. The engineer translates these into construction measures and finally reruns the model. Using this procedure a trim package is designed for a small car. It weighs 4kg and successfully reduces the mid and high frequency engine noise and rolling noise up to 3.5 dB(A).

Tue 18:00 Van Beuningen Zaal Vehicle acoustics

Acoustic Challenges of Rolling Bearings in Combustion Engines

Peter Genender, Klaus Wolff and Christoph Steffens
FEV Motorentechnik GmbH

In addition to the efficiency of gas exchange and combustion, fuel consumption is influenced to a large extent by engine friction. The distribution of mechanical losses shows that with one third, the main and conrod bearings have the largest share in engine friction apart from the piston subassembly. From a given 1,6 L 4-cylinder plain bearing engine changed to roller bearings a proved 5,4% (NEDC) improvement of the fuel consumption resulted from reduced friction. The challenges resulting from the use of roller bearings in internal combustion engines are related to the disciplines acoustics, durability, and fabrication. Within this article the main influencing factors on NVH will be identified by means of simulation and experimental investigations. It will be shown in which way the critical engine NVH as a matter of principle can be improved with the help of design modifications. Based on this knowledge, an advanced test engine was set up. Vehicle measurements, which have examined the development stages of the rolling bearing engine, verify the significant NVH improvements. Finally, interior noise level as well as psycho-acoustic parameters are on the same level as in the series production engine with plain bearings.

Tue 14:00 Schadee Zaal Speech in car 1

Performance Analysis of Wavelet-based Voice Activity Detection

Marco Jeub[a], Dorothea Kolossa[b], Ramon Fernandez Astudillo[b] and Reinhold Orglmeister[b]
[a]*RWTH Aachen University;* [b]*TU Berlin*

The objective of this paper is to analyse the performance of wavelet-based voice activity detection algorithms (VAD) and to contrast it with that of the VAD standardized for the AMR-WB (Adaptive Multi-Rate Wideband) codec. Experimental results show that wavelet approaches lead to good results with respect to speech clipping and offer a much lower computational complexity. Integration of these algorithms into a Hidden Markov model (HMM) speech recognizer shows that the recognition performance using the AMR VAD can also be obtained or improved upon by wavelet based approaches, again at a notably reduced computational effort.

Tue 14:20 Schadee Zaal Speech in car 1

Optimal projections between Gaussian Mixture Feature Spaces for Multilingual Speech Recognition

Martin Raab[a], Olaf Schreiner[a], Tobias Herbig[a], Rainer Gruhn[a] and Elmar Nöth[b]
[a]*Harman Becker Automotive Systems;* [b]*University Erlangen-Nuremberg*

Multilingual speech recognition is increasingly gaining attention for in-car speech controlled applications. An example is a media player that allows selection of music by voice command, requiring speech recognition for multiple languages in order to cover the languages of artist names and music titles in a given music database.
There are two traditional approaches how a system can support the recognition of multiple languages. The first is to run a set of monolingual recognizers in parallel; the second one is to train an integrated multilingual recognizer dataset for the required set of languages.
The first approach has the disadvantage that there is no parameter sharing between the different recognizers, thus needing large amounts of processing power and memory. The second approach has the disadvantage that one multilingual recognizer has to be trained for every combination of languages.
In our paper we present a scheme to create a multilingual recognizer out of monolingual trained recognizers. A formula is given for an optimal projection of emission probabilities between Gaussians Mixture feature spaces. With this formula, we can project each HMM state of all languages to one set of Gaussians without retraining the system.

Tue 14:40 Schadee Zaal Speech in car 1

On the Estimation and Use of Clean Speech Feature Posteriors for Noise Robust Speech Recognition

Reinhold Häb-Umbach and Volker Leutnant
University of Paderborn, Fachgebiet Nachrichtentechnik

Traditionally, front-end methods for noise-robust speech recognition aim at denoising the observed feature vectors to obtain a point estimate of the clean, uncorrupted feature vector sequence. These estimates are then ”plugged” into the Bayesian decision rule as if they were perfect estimates. In an optimal decoding rule, however, the reliability of the clean feature estimate would have to be taken into account, a method termed ”Uncertainty Decoding” in the literature. We have shown that the optimal uncertainty decoding rule involves an integration over the posterior density of the clean feature given the observed corrupted feature sequence. This can lead to significant improvements in error rate if perfect feature reliability information is available.
In this paper we show how the clean speech feature posterior can be estimated for noise robust speech recognition in the car. We employ a state space model of speech and noise and track both of them jointly.

This model can be extended by allowing for multiple observations. In doing so, the Kalman filter serves as a sensor fusion algorithm which fuses clean speech or noise estimates obtained from different enhancement methods. Experiments conducted on the AURORA databases illustrate the potential of this approach.

Tue 15:00 Schadee Zaal Speech in car 1

Improving Context Modeling for Phoneme Recognition

Daniel Vasquez[a], Guillermo Aradilla[b], Rainer Gruhn[b] and Wolfgang Minker[a]
[a]*University of Ulm, Institute of Information Technology;* [b]*Harman Becker Automotive Systems*

In this paper we investigate a novel technique for modeling long temporal context in phoneme recognition. Conventional techniques take a global context consisting of consecutive concatenated acoustic features as input for a phoneme classifier. Thus, the classifier must learn different variations corresponding to different parts of the global context. In contrast to conventional techniques, our scheme decomposes the temporal context of each phoneme into a set of slides. Each temporal slide is input for a non-linear classifier given by a Multilayered Perceptrons (MLP). Every MLP is trained using the central label of the global context. Thus, the assumption that the current phoneme occupies the global context is preserved. Furthermore, each MLP can robustly model different temporal context inside each phoneme. The outputs of the classifiers are combined to estimate posterior probabilities of phoneme-level classes. These posteriors are employed in a hybrid HMM/MLP framework. Experiments have shown an absolute phoneme error reduction of 3.6% compared to a baseline classifier with the same context length. A further improvement is achieved when a hierarchical approach is implemented, which introduces information across different phonemes.

Tue 15:20 Schadee Zaal Speech in car 1

Speed improvements in a Missing Data-based speech recognizer by Gaussian selection

Yujun Wang and Hugo van Hamme
KU Leuven, ESAT/PSI

Speech recognition performance in noisy environments such as cars is degraded due to the mismatch between the feature vector and the speech model. To improve the noise robustness, we apply Missing Data Techniques (MDT) to Hidden Markov Models (HMM) using a mixture of Gaussians for the emission densities. Traditionally, MDT uses diagonal Gaussian covariance matrices to model spectral features, but due to correlation in the feature vector, this leads to a loss of accuracy at high SNR. To overcome this problem, we previously proposed an MDT-based system that takes the feature correlation into account to regain the accuracy. However, Gaussian evaluation in the acoustic model now

requires solving a Non Negative Least Square problem, which increases the computational requirements by an order of magnitude. Hence several Gaussian selection paradigms are extended to be used in an MDT-based speech recognizer and subsequently compared. Hereto, a modified Symmetric Kullback-LeiblermDivergence (KLD) metric is proposed for Gaussian selection methods based on clustering. Experimental results over Dutch and Flemish databases show that on average, only about 35% of the mixtures need to be evaluated and about 60% CPU time are saved, while maintaining the accuracy of a system that evaluates all Gaussians.

Tue 16:20 Schadee Zaal Speech in car 1

Evaluation of missing data techniques for in-car automatic speech recognition

Yujun Wang[a], Rudi Vuerinckx[b], Jort Gemmeke[c], Bert Cranen[c] and Hugo van Hamme[a]
[a]*KU Leuven, ESAT/PSI;* [b]*Nuance, Merelbeke (B);* [c]*Dep. of Language and Speech, Radboud University Nijmegen*

Background noise has a detrimental effect on the accuracy of automatic speech recognition (ASR) systems because it changes the statistical properties of the spectrally related features extracted from the speech. If the recognizer's acoustic model that describes these statistics is trained under noise conditions different from those of the test data, the likelihoods of speech hypotheses will deviate and recognition errors result. Many practical systems therefore compensate the spectral features for noise with techniques like spectral subtraction, modify the statistics by acoustic model adaptation or train the acoustic model under multiple noise conditions. In our ASR system which is based on missing data techniques (MDT), spectral regions that are dominated by noise are replaced by an estimate that depends on the speech hypothesis of the decoder. In this paper, we discuss the pro's and con's of this approach in terms of computational requirements, training data requirements and versatility. In a benchmark study using in-car recorded data, as well as other non-stationary noise types, we compare the accuracy of a MDT system with that of the commercially available Nuance VOCON 3200.

Tue 16:40 Schadee Zaal Speech in car 1

Reworking spoken dialogue systems with context-awareness and information prioritisation to reduce driver workload

Jose Luis Blanco[a], Álvaro Sigüenza[a], David Díaz[a], Mauricio Sendra[b] and Luis Hernández[a]
[a]*Universidad Politécnica de Madrid;* [b]*Telefónica R&D, Barcelona*

An entire host of new information, communication and entertainment applications and security systems are assaulting our cars. Interaction with onboard systems is moving beyond the classic switches, knobs, sticks, pedals and dashboard lights and sounds, to incorporate speech, touchscreens and a variety of other displays. While some of these systems are

intended to help with the task of driving (e.g., navigation aids, lane departure and vehicle-in-blind-spot warnings), a constellation of displays, messages and requests threaten to overload the driver with information that is too rich and may distract him. Speech being eyes- and hands-free, it is increasingly being considered as a safe communication channel. However, interactive dialogue can increase the driver's mental workload, especially when robustness problems occur (e.g., recognition errors). Here we present our research results on dialogue as a secondary task, to gain understanding on how it may affect driving, on what dialogue strategies foster safer driving and on which applications, messages and interaction modalities should be prioritised over which others in each driving context. The proposed contribution is related to our current research activities within MARTA (Mobility for Advanced Transport Networks; www.cenitmarta.org), a Spanish, publicly-funded project. Experimental results from tests with a simulated driving platform are presented.

Tue 17:00 Schadee Zaal Speech in car 1

Position Estimation of Car Occupants by Means of Speech Analysis

Timo Machmer, Alexej Swerdlow, Benjamin Kühn and Kristian Kroschel
Universität Karlsruhe, Lehrst. Interaktive Echtzeitsysteme

The interaction between man and machine via acoustic analysis systems gains more and more in importance for next-generation cars. Thereby, the positions of car occupants are of particular interest. This kind of information is useful not only for generation of position specific properties, e.g. seat and air conditioning settings, but can also be utilized for manipulations of entertainment as well as other internal car systems. An in-car speaker localization system presented on the DAGA 2008 enabled the detection of passengers by means of a one-step approach for sound source localization using SRP-PHAT for the position estimation. In so doing, promising localization results were achieved in real experiments, which were carried out in an exemplary up-to-date car. However, the biggest constraint was the circumstance that the car was not driving during the evaluation, but was parking in a typical public road.
In the current paper, the influence of the environmental noise, which inevitably occurs while driving, on the localization accuracy is investigated. Therefore, different driving situations are taken into account and measurements are done under various conditions. Furthermore, an altered microphone array configuration is evaluated as well.

Tue 17:20 Schadee Zaal Speech in car 1

Robust Superdirectional Beamforming for Hands-Free Speech Capture in Cars

Edwin Mabande, Adrian Schad and Walter Kellermann
University Erlangen-Nuremberg

The application of multichannel signal processing algorithms in the front-ends of hands-free systems facilitates a better extraction of a desired

source signal and allows for a suppression of unwanted noise and interference. In this paper, a novel superdirective beamforming design method called the Robust Least-Squares Frequency-Invariant Beamformer (RLSFIB) is investigated for microphone arrays in cars. The RLSFIB design offers significant spatial selectivity at low frequencies even for small array apertures and some variability in microphone positioning. To cope with the high sensitivity inherent in superdirectional beamformers, this method directly incorporates a white noise gain (WNG) constraint for controlling the beamformer sensitivity. The resulting problem is formulated as a convex optimization problem which can be solved by conventional iterative solvers and the convexity guarantees a globally optimal solution. As the WNG constraint can be chosen freely, the method can match desired robustness levels, and therefore can be adapted to given prior knowledge on microphone mismatch, positioning errors, and microphone self-noise. Design examples for typical car requirements will be presented.

Tue 17:40 Schadee Zaal Speech in car 1

Challenges in the introduction of wideband hands-free in cars

Hans Wilhelm Gierlich

HEAD acoustics GmbH

The introduction of wideband speech services in the new generation networks will be one of the major steps in the near future. The support of such services by the different networks and services also forms the basis for the integration of wideband speech hands-free services in cars. Wideband speech quality would be mostly welcomed in cars since especially on the receiving side the poor (narrowband) speech quality is more obvious in cars than in other type of terminals. The narrowband speech is compared directly to the typically high quality audio presentation of all other audio equipment in the car and even less sensitive users ob-serve the difference in quality immediately. To achieve a superior speech quality inside the car all elements involved in the transmission have to provide adequate speech quality. The challenges faced when connecting the hands-free car system via Bluetooth to the mobile terminal, via the mobile terminal to the network and connecting the mobile network to the wideband NGN type network is introduced, the users expectation with respect to the different quality dimensions of wideband speech and the wideband presentation of narrowband speech signals is discussed. An overview of the standardization work in the ITU-T Focus

Tue 18:00 Schadee Zaal Speech in car 1

Improving the User Experience with Spatial Auditory Displays

Scott Pennock
QNX Software Systems (Wavemakers)

For most non-music applications today, users perceive sound coming from the same position within the vehicle. However, spatial auditory displays can cause users to perceive sounds coming from different locations in 3-dimensional space. Spatial auditory displays can improve the user experience by creating a better ”sense of presence” for speech communications, making identification of the current application/object effortless, and naturally conveying information within the current application. Spatial auditory displays can also help the driver perform tasks more effectively by reducing errors, enabling faster reaction times, and making it easier to hear in noisy vehicle environments (the ”cocktail party effect”). Potential applications of spatial auditory displays include navigation, multi-party conferencing, simultaneous applications/agents (for instance, navigation, radio, traffic), speech-interface menus, alerts, and car-to-object communications (for instance, other car, toll booth, wireless device). Spatial cues used by listeners include onset, Inter-aural Time Differences (ITDs), Inter-aural Level Differences (ILDs), spectrum, overall level, direct-to-reflected sound ratio, and deltas from reference objects. This paper will discuss how spatial auditory displays can be used to improve the user experience and increase efficiency of tasks performed by the driver. It will also discuss different ways to implement spatial auditory displays and the constraints of each implementation.

Tue 14:00 Ruys Zaal Music processing

Bridging Music Information Retrieval and Folk Song Research - The Computational Setup of the WITCHCRAFT Project

Jörg Garbers
Utrecht University

The WITCHCRAFT project sets as its objective to develop a fully functional content-based retrieval system for folk song melodies. Besides of making the Netherlands folk song collection of the Meertens Institute available for the general public, we aim to support their research in oral variation.

Folk song researchers traditionally know many songs by heart and intuitively find correlations between them. For large or international collections, however, researchers cannot know all the song instances, so computer aided support is indispensable to reveal related items. This in principle means to take existing models about oral variation and design and implement similarity measures that can be used in a retrieval system. This, however, proves to be difficult in practice.

In this paper we present our information retrieval setup consisting of performed folk song recordings, their symbolic encoding, an expert classification of variant tunes (ground truth), annotation data about the particular expert classification reasons and our approaches to similarity. As an example we present how we translated the expert concept of ”pitch stability” into an alignment-based similarity measure and how we evaluated our method and the original ”pitch stability” hypothesis with respect to the given annotations.

Tue 14:20 Ruys Zaal Music processing

Preprocessing methods for rhythmic mid-level features

Christian Dittmar
Fraunhofer IDMT

Rhythmic mid-level features are an important pre-requisite in Music Information Retrieval. They can be deployed to describe the rhythmical gist of songs for various tasks, such as genre classification and music similarity search. Although different computation strategies have been proposed in the literature, rhythmic mid-level features commonly represent the most salient rhythmic periodicities in a music signal. Major shortcomings of rhythmic mid-level features are their dependency on the actual tempo of the songs and their susceptibility to degradation caused by interference with non-rhythmic signal components, such as melodic-sustained instruments. This publication describes preprocessing strategies applicable to rhythmic mid-level features. The problem of interference with melodic instruments is addressed by emphasizing the rhythmic content via a signal decomposition related to Drum Transcription. Tempo changes are tackled by logarithmic re-sampling of the features\’ lag a xis. An evaluation of the proposed methods is conducted via genre classification using both artificial and real-world music signals. The evaluation results are presented and discussed accordingly.

Tue 14:40 Ruys Zaal Music processing

Everything in its right place? - Learning a User’s View of a Music Collection

Sebastian Stober, Korinna Bade and Andreas Nürnberger
Otto-v.-Guericke-Universität Magdeburg

Keeping one’s personal music collections well organized can be a very tedious task. Fortunately, today, many popular music players (such as AmaroK or iTunes) have an integrated library function that can automatically rename and tag music files and sort them into subdirectories. However, their common approach to stick with some hierarchy of genre, artist name, and album title barely represents the way a user would structure his collection manually. When it comes to organizing a music collection according to a user-specific hierarchy, three things are required: First, the music files have to be described by appropriate features beyond simple meta-tags. This includes content-based analysis but also incorporation of external information sources such as the web. Second, knowledge about the user's structuring preferences must be available. And third, and most importantly, methods for learning personalized hierarchies that can integrate this knowledge are needed. We propose for this task a hierarchical constraint based clustering approach that can weight the importance of different features according to the user perceived similarity. A hierarchy based on this similarity measure reflects a user's view on the collection.

Tue 15:00 Ruys Zaal Music processing

Assessment of music similarity perception

Alberto Novello[a], Armin Kohlrausch[b] and Martin McKinney[c]
[a]*Eindhoven University of Technology;* [b]*Philips Research, Eindhoven;* [c]*Starkey Laboratories (USA)*

We present the results of a Web-based listening experiment on music similarity that used triadic comparisons of a relatively large set of stimuli: 78 song excerpts selected from 13 genres of Western popular music. The high correlation across the similarity rankings of the 78 subjects suggests the presence of a common and stable model underlying the participants' perception of music similarity. The three control variables used in the excerpt selection, genre, tempo and timbre, show statistically significant saliency and a hierarchical degree of impact on participants' pair rankings (genre $>$ tempo $>$ timbre). A combination of scaling techniques and discriminant functions distilled the data to gain insight into the important factors underlying the organization of the participants' perceptual space. Through multidimensional scaling we deduced that six dimensions were sufficient to represent the participants' original data. Quadratic discriminant analysis was used to quantitatively evaluate the influence of each control variable on the organization of the participants' perceptual space. We searched for axes that maximized the separation of the excerpt classes. We identified three axes that were a posteriori labeled ”slow-fast”, ”vocal-non vocal”, and ”synthetic-acoustic”, showing significant discriminability of the stimulus classes.

Tue 16:00 Ruys Zaal Music processing

An Evaluation of using Chroma- and MFCC-based Features for Classifying Radio Transmissions

Frank Kurth and Dirk von Zeddelmann
FGAN-FKIE, Abteilung KOM, Wachtberg

In this contribution, we evaluate the usability of chroma- and MFCC-based features for the task of classifying radio transmissions containing music and speech content. In our system, an incomming audio stream is analyzed by a cascade of binary classifiers, each based on a different type of audio feature. Here, the type of feature used by an individual classifier is choosen according to the particular classification task, based on the hypothesis that MFCC-based features better characterize speech contents whereas chroma-based features better characterize music audio. The latter hypothesis is validated in a series of experiments which we describe subsequently. As an overall result of the classification process, the audio stream is segmented into temporal regions and a class label is assigned to each region. We evaluate our system on a larger-scale corpus of manually annotated audio material recorded from live radio transmissions.

Tue 16:20 Ruys Zaal Music processing

Handling Scanned Sheet Music and Audio Recordings in Digital Music Libraries

Christian Fremerey[a], David Damm[a], Meinard Müller[b], Frank Kurth[c] and Michael Clausen[a]
[a]*Bonn University;* [b]*Saarland University and MPI Informatik;* [c]*FGAN-FKIE, Abteilung KOM, Wachtberg*

Significant digitization efforts have resulted in large music collections, which typically contain various types and modes of data. In particular, such collections may comprise audio data in form of CD recordings as well as image data in form of scanned sheet music, thus concerning both the auditory and the visual modalities. In this paper, we review various techniques for automatically processing, analyzing, and organizing such multimodal music collections. We then present a music player system along with various interfaces that allow a user to conveniently search, navigate, and browse across the two types of music data. For example, during audio playback our system synchronously highlights the corresponding musical measures within the sheet music. Furthermore, a user may formulate a query by marking a region of measures within the scanned score to trigger an audio search. The system retrieves all available audio recordings along with the temporal positions of the audio segments that correspond to the queried measures. The user may then freely navigate across the various retrieved interpretations. Our techniques and interfaces are prototypically integrated into the library service system of the Bavarian State Library as part of the PROBADO project.

Tue 16:40 Ruys Zaal Music processing

Towards timbre-invariant audio features for harmony-based music

Sebastian Ewert[a], Meinard Müller[b] and Michael Clausen[c]
[a]*Bonn University, Informatik III;* [b]*Saarland University and MPI Informatik;* [c]*Bonn University*

In the context of automatic music processing, chroma features have turned out to be a powerful midlevel representation with a wide range of applications such as content-based retrieval, music synchronization, and chord transcription. An important step of the feature calculation is the grouping of spectral energy components that belong to the same pitch class or chroma of the equal tempered scale. Here, the octave identification introduces a high degree of invariance to changes in timbre and instrumentation. In this paper, we introduce a strategy to further increase this invariance by combining the concept of chroma features with the well-known concept of mel-frequency cepstral coefficients (MFCCs).

The lower MFCCs are known to capture information on timbre. Therefore, to enhance robustness to timbral changes, we discard this information by only keeping the upper coefficients. Furthermore, using a pitch scale instead of a mel scale allows us to group the remaining coefficients into twelve chroma bins. The resulting audio features, as indicated by our systematic experiments, have indeed gained a significant boost towards timbre invariance without a degradation of discriminative power.

Tue 17:00 Ruys Zaal Music processing

Extracting Expressive Tempo Curves from Music Recordings

Verena Konz, Meinard Müller and Andi Scharfstein
Saarland University and MPI Informatik

Musicians give a piece of music their personal touch by continuously varying tempo, dynamics, and articulation. Instead of playing mechanically they speed up at some places and slow down at others. Similarly, they continuously change the sound intensity and stress certain notes. The automated analysis of different interpretations, also referred to as performance analysis, has become an active research field. Here, one goal is to find commonalities between different interpretations which allow for the derivation of general performance rules. A kind of orthogonal goal is to capture what is characteristic for the style of a particular musician. Algorithms for automated performance analysis rely on accurate annotations of the audio material by means of suitable musical parameters. Here, the annotation process is often done manually, which is prohibitive in view of large audio collections. In this paper, we present a fully automatic approach for computing tempo curves that reveal the relative tempo difference between two performances. First experiments indicate that our automated methods yield good estimations of the overall tempo and, for certain classes of music such as piano music, even of finer tempo nuances.

Tue 17:20 Ruys Zaal Music processing

Combining Onset Features for high-resolution Music Synchronization

Peter Grosche[a], Meinard Müller[a] and Sebastian Ewert[b]
[a]*Saarland University / MPI Informatik;* [b]*Bonn University, Informatik III*

Many different methods for the detection of note onsets in music recordings have been proposed and applied to tasks such as music transcription, beat tracking, tempo estimation, and music synchronization. Most of the proposed onset detectors rely on the fact that note onsets often go along with a sudden increase of the signal's energy, which particularly holds for instruments such as piano, guitar, or percussive instruments. Much more difficult is the detection of onsets in the case of more fluent note transitions, which is often the case for classical music dominated by string instruments. Here, one has to develop methods that allow for detecting smooth temporal and spectral changes in the audio signal. In this paper, we present various methods for onset detection including

energy-based and phase-based techniques, which account for different properties of the audio signal. In particular, we present a recent audio feature that combines the high temporal accuracy of onset features with the robustness of chroma features. Finally, we show how a combination of various onset features can be used to significantly improve the temporal resolution in music synchronization tasks.

Tue 17:40 Ruys Zaal Music processing

Blind Enhancement of the Rhythmic and Harmonic Sections by NMF: Does it help?

Björn Schuller, Florian Eyben and Gerhard Rigoll
TU München, Institute for Human-Machine Communication

Non-Negative Matrix Factorization is well known to lead to considerable successes in the blind separation of drums and melodic parts of music recordings. Such splitting may well serve as enhancement when it comes to typical Music Information Retrieval tasks as automatic key labelling or tempo detection. In this respect we introduce the combination of an NMF based blind music separation into several isolated audio tracks in combination with Support Vector classification to assign each obtained track to be either rhythmic or melodic. Thereby optimal parameterization and performances are discussed. Next, stereophonic information is further used to eliminate the key melody for chord labelling. We then analyze the potential for the named tasks by an extensive number of experiments carried out on the MTV Europe Most Wanted of the 80ies and 90ies in MP3 format.

Tue 14:00 Mees Zaal History of acoustics

What do the acoustics of the 'occidental' Europe owe the Arabs?

Peter Költzsch
TU Dresden, Inst. f. Akustik und Sprachkommunikation

Whilst the Arabs translated the results of the ancient Greek science almost without exception into Arabic until the middle of the 9th Century, and therefore made them known for the Arab world, the Greek heritage was largely unknown in Central Europe at the beginning of the 2nd Millennium. About from this time on, a transfer process began towards Central Europe, primarily for the scientific and philosophical discoveries of the Greeks. This happened because of the Arab occupation of Sicily (9th to 11th century) and of Spain (8th to 15th century), through the translations from Arabic into Latin and through multi-transfer paths (universities, libraries, etc.). However the Arab scientists were not only mediators, in an impressive way they also commented on the works of the Greeks and introduced their own new scientific ideas. Consequently the ancient Hellenistic and the Arab knowledge were spread in the 12. /13. Century in Central Europe. The lecture tries to answer the following questions for the area of acoustics: Which works of ancient Greek scientists have been passed on to the Arabs? Which Arab scientists were involved and how have they inspired the ”occidental” culture in Europe through their own scientific achievements? [Lecture in German]

Tue 14:20 Mees Zaal History of acoustics

History of the Compact Disc

Kees Schouhamer Immink

Turing Machines Inc., Rotterdam

An audio compact disc (CD) holds up to 74 minutes, 33 seconds of sound, just enough for a complete mono recording of Ludwig von Beethoven’s Ninth Symphony (’Alle Menschen werden Brüder’) at probably the slowest pace it has ever been played, during the Bayreuther Festspiele in 1951 and conducted by Wilhelm Furtwängler. Each second of music requires about 1.5 million bits, which are represented as tiny pits and lands ranging from 0.9 to 3.3 micrometers in length. More than 19 billion channel bits are recorded as a spiral track of alternating pits and lands over a distance of 5.38 kilometers (3.34 miles), which are scanned at walking speed, 4.27 km per hour.
We will discuss the various crucial technical decisions made that would determine the technical success or failure of the new medium.

Tue 14:40 Mees Zaal History of acoustics

Acoustical and electro-acoustical sound fields - the beginnings in Gravesano 1954

Ernst-Joachim Völker

Institute for Acoustics and Building Physics, Oberursel

It was a courageous start in 1954. Professor Dr. Herman Scherchen had invited experts from all over Europe to discuss the new and special relationship between music and technology. The UNESCO supported the work. The place was the small Swiss City of Gravesano, a quiet and peaceful location. It was just the beginning and starting point for many investigations. New concert halls were discussed such as the Berlin Philharmonic Hall, Radio studios, concert halls, reverberation and first reflections, the ear as a time-measuring device. Today, it is well-known: The discussions and publications were the roots and the foundation of many developments. In the years to come, universities became strongholds of acoustics, e.g. E. Meyer in Göttingen, L. Cremer and F. Winckel in Berlin, J. Blauert in Bochum, K.H. Weisse in Darmstadt, A. Moles in Paris, the school of Le Corbusier and W. Kuhl of the Institute for Radio-Technique in Hamburg. The acoustical installations and measuring equipment in Gravesano were outstanding. Different aspects of stereophony were discussed with experiments in radio-stations. Orchestras were recorded and transmitted via two radio-channels at SFB Broadcasting Corporation in Berlin 1956. The paper describes these roots and follows the paths taken from the beginning up to today.

Tue 15:00 Mees Zaal History of acoustics

Air acoustics: Development of listening equipment in the Netherlands

Aad van der Voort[a] and Ronald Aarts[b]
[a]*curator Museum Waalsdorp;* [b]*Philips Research, Eindhoven*

From the first world war until the 30's air acoustics played an important role in the air defence. Air vehicles carrying a weapon could not be located from the ground e.g. at night time or under cloudy conditions. As radar was still to be discovered, vision had to be supplemented by hearing using the sound of the engines. At least a dozen different acoustic location equipment's for aeroplanes from different countries were in the 1920's available on the military market. They served as said as supplement to visual means preceding the use of artillery. These equipments contained most times four sound pick-up elements such as curved shells or horns used in pairs. Two elements at a fixed mutual distance were adjusted in the horizontal plane for the determination of the chart angle and the remaining two in the vertical plane for the measurement of elevation. The transport of sound from each element of a pair to a corresponding pair of human ears was carried out by means of metal or rubber tubes. Each pair is adjusted in the correct direction when the two ears receive the sound signals at the same time.

Tue 15:20 Mees Zaal History of acoustics

Early research on spatial hearing in Helsinki and Eindhoven

Armin Kohlrausch
Philips Research, Eindhoven

In this contribution, I want to present the work from two Ph.D. theses on spatial hearing. The first one, by Alvar Wilska, was performed at the Institute of Physiology of the University of Helsinki and was in 1938 published under the title: Untersuchungen über das Richtungshören (Experiments on directional hearing). The other, by Kornelis de Boer, was performed at the Philips Research Laboratories in Eindhoven and appeared in 1940 as Ph.D. thesis at the Technische Hogeschool Delft under the title: Stereofonische geluidsweergave (stereophonic sound reproduction). The work of de Boer, who also published extensively in Philips Technical Review, is widely cited and is considered an influentual contribution to stereophonic sound reproduction. In contrast, Wilska's work on just noticeable differences in sound source direction and interaural time differences is hardly known, and I am only aware of a single reference to it, in the Ph.D. thesis of Franssen (1960). One interesting commonality is that both authors used early realizations of artifical heads in their experiments.

Tue 16:20 Mees Zaal History of acoustics

Historical developments of Organs in the Netherlands

Cor Doesburg
retired NOB, Hilversum

An introduction will be given about the history and development of Pipe Organs in The Netherlands and their place in the Organ History of Europe.
Starting with the oldest Church Organ installed in the 15th Century, we talk about Organs in The Golden Age, influence of the German Baroque in the 18th Century, influence from the Romantic from Germany and France in the 19th Century, influence of the Modern Times from the United States concerning electric action and sound especially concerning Concert- and Theatre Organs, influence from The Organ Reform from the Elzas and the Organ Movement from North Germany. State of the Art in the Netherlands about Organ Building in this 21th Century.
Some essential information will be given about the changes in sound and technical solutions in Organ Building. Some examples with pictures and sound of the most remarkable Organs in the Netherlands will be presented.

Tue 16:40 Mees Zaal History of acoustics

Measuring Frequencies with Historic Resonators from SCHAEFER

Rüdiger Hoffmann, Dieter Mehnert, Rolf Dietzel and Günther Fuder
TU Dresden, Inst. f. Akustik und Sprachkommunikation

Measuring frequencies of acoustic processes within the audible range required large technical expense in the early times of experimental phonetics and acoustics. Indirect methods like KUNDT's tube, the monochord, the interpretation of kymographic recordings, or the interference tube from QUINCKE allowed such kind of measurement, but had been too complicated for practical purposes. Therefore simpler measurement methods have been looked for in early times. E. g., it is possible to apply one-sided closed pipes of variable length as $\lambda/4$ resonators. The acoustic perception of the experimentator was used as indicator.
This paper investigates this measuring method at first within the historic arrangement. Then, the transfer behaviour of single tube resonators from SCHAEFER will be measured in the free sound field.
These results as well as the ability of judgement of the listener are used to estimate the measuring uncertainty of such historic methods.

Tue 17:00 Mees Zaal Teaching and education

Introduction of a Laboratory Technical Acoustics at Heilbronn University (Germany)

Gerald Landstorfer, Alexander S. Treiber and Gerhard Gruhler
Heilbronn University

Currently, the education in acoustics at Heilbronn University is only representing by a lecture technical acoustics (2 ECTS) that gives each winter semester an overview of acoustic issues. Under the aspect of expanding the acoustic course offer, a laboratory technical acoustics (3 ECTS) will be introduced for prospective summer semesters.

The most important requirements for the laboratory concept are to achieve the following aims:

a) The students' comprehension of basic acoustical phenomena shall be a focus, b) The experiments shall deal with actual problems close to our areas of research, as well as general tasks given from surrounding industrial companies, c) The expansion of knowledge in our core competence fields with student support in projects, bachelor or master thesis';

According to the number of expecting participants in conjunction with available locations (anechoic measuring room, jury test room, echo corridor, auditorium, laboratory electronic circuit technology), groups of two or three students may work for their own of a total of six different experiments.

It will make sense to correlate the topics of the laboratory with the topics of the lecture. Therefore, the practical tasks will be located in psychoacoustics, room acoustics, electro-acoustics, acoustic quality control, multi-channel playback/recording and passive/active noise control.

Tue 17:20 Mees Zaal Teaching and education

ADP: Practical experience in acoustics education

Sebastian Buckert and Adam Skowronek
TU Darmstadt, System Reliability and Machine Acoustics SzM

Conveying interdisciplinary competences is an important aspect of the master degree program at the Technische Universität Darmstadt. A so called "Advanced Design Project" (ADP) largely contributes to the communication of these skills. The ADP of the Department of System Reliability and Machine Acoustics (SzM) preferably takes place at industrial sites (e.g. manufacturing plants, test and research facilities), tasking real problems. Thus the students get prepared for the challenges of their prospective profession. The problems have to be solved by the students within a given time frame working in teams of three to four people. Successfully accomplishing a project like this requires not only professional knowledge but also a high degree of teamwork, communication and presentation skills as well as a good time management. Besides these soft skills, professional knowledge in the field of measurement techniques, use of data acquisition and analysis software, noise control engineering, silencer design, etc. is attained. This project complements the courses in the field of machine acoustics offered by the department of System Reliability and Machine Acoustics.

Tue 17:40 Mees Zaal Teaching and education

ADP: Baseline analysis of the acoustic emission of induction furnaces

Adam Skowronek and Sebastian Buckert
TU Darmstadt, System Reliability and Machine Acoustics SzM

An ADP is an industry-oriented course, in which students are to solve a described task as independently as possible, enhancing their ability to work as a team. In collaboration with several industrial partners the Department of System Reliabiltiy and Machine Acoustics SzM of the Technische Universität Darmstadt has accomplished various ADPs in recent years.

As an example for such an ADP, this paper describes the baseline analysis of the acoustic emission of induction furnaces at SMS Elotherm in Remscheid, Germany. Alternating magnetic fields and Lorentz forces cause oscillatory impulses in the induction furnaces as well as in the metal components which are to be heated up, partly creating sound pressure levels far above 100 dB.

After a week-long introduction in the handling of the available measuring instruments, the students were to perform acoustic measurements and analyses on these furnaces. For the most part independently, the students hence measured several air-borne and structure-borne sounds.

The necessary preparation, measurement procedures, and some selected results are presented.

Tue 14:00 Van Rijckevorsel Zaal Hydroacoustics

Measurements of Construction Noise During Pile Driving of Offshore Research Platforms and Wind Farms

Rainer Matuschek and Klaus Betke
itap GmbH

To support technological developments and improve our knowledge of the impact of offshore wind energy technology on the marine flora and fauna, the research project FINO (research platforms in the North and Baltic Seas) was started in 2002. The platforms as well as the research activities are funded by the german ministry for the Environment, Nature Conservation and Nuclear Safety (BMU).

One aspect of active research is the influence of the very high sound pressure levels on fish and marine mammals during pile driving with large hydraulic hammers.

We present measurements of sound pressure during the construction phase of all three platforms. Data were recorded at different distances to the sound source with autonomous measurement buoys and from aboard a ship.

During the pile driving of FINO 3 in 2008, a closed bubble curtain around the monopile was installed, in order to reduce the sound pressure levels in the vicinity. The influence of this bubble curtain on the sound exposure level, as well as on the spectral content of the emitted sound will be investigated.

Tue 14:20 Van Rijckevorsel Zaal Hydroacoustics

Temporal and Spectral Characteristics of a Marine Piling Operation in Shallow Water

Paul Lepper[a], Stephen Robinson[b], Justin Ablitt[b] and Simon Dible[a]
[a]*Loughborough University;* [b]*National Physical Laboratory (UK)*

Analysis of the underwater radiate acoustic characteristics for marine piling operations for two pile diameters, 2m and 4.74m, in a relatively shallow water site are presented. Measurements of the entire piling sequence for several piles were conducted at ranges from 10m to 22km for piles in 10-20 m water depth. Variations in the temporal and spectral characteristics of radiated energy are analysed in context of pile size, range from source, hammer energy used and pile penetration depth.
The hammer energy used shows a strong interdependence between mechanical strike 'hammer' energy and underwater radiated acoustic energy. This process appears 'coarsely' linear for individual piling operations although considerable variation in overall gradient were observed between operations. Temporal and spectral variations in radiated energy due to pile penetration are also examined for fixed hammer energy and range.
Simultaneous recordings of radiated energy made at increasing distances from the pile showed evidence of temporal and spectral dispersion effects consistent with relatively shallow water propagation. Correlation of received levels at various ranges in differing seabed topographies were made suggesting complex shallow water modal propagation dependant on both the source and environment characteristics including seabed topography, sediment type and water column acoustic properties.

Tue 14:40 Van Rijckevorsel Zaal Hydroacoustics

Risk Mitigation for Sea Mammals - The Use of Air Bubbles Against Shock Waves

Edgar Schmidtke and Bernd Nützel
FWG, Kiel

Many thousand tons of ammunition were disposed in the Baltic after World War II simply by sinking. Today these explosives (e.g. torpedoes and mines) pose a risk mainly to shipping traffic. Some ammunition cannot be salvaged without risking human life. In this case blowing up is the only solution. Any underwater explosion causes a shock wave which is dangerous for humans and sea mammals even over long distances.

To investigate the possible attenuation of the acoustic level caused by a shock wave two kinds of experiments were realised. The first series of experiments was to blow up small amounts of different explosives in a shallow basin to learn how shock waves propagate. The second experiment was to measure these shock waves in the Baltic close to the shore, directly and after propagation through an air bubble curtain.
Results of both experiments will be presented and discussed.

Tue 15:00 Van Rijckevorsel Zaal Hydroacoustics

Characterization of Ships as Sources of Underwater Noise

Christ de Jong
TNO Science and industry

There is a growing interest in the possible impact of anthropogenic underwater noise on marine life. One of the concerns is the increasing contribution of shipping noise, with the growing number and size of commercial ships. Traditionally, underwater radiated noise control was only of interest for naval and research vessels. Due to the potential environmental impact, it becomes relevant for commercial shipping also. The challenge is to bring the expertise from the naval and research area to the maritime industry. One of the issues is the measurement of ship radiated underwater noise. The definition of the ship as a noise source, the measurement set-up and the environment in which the measurements are taken all have a large impact on the results of ship underwater noise measurements. The various influences are discussed. The elimination of environmental effects from the radiated noise characteristics is illustrated by means of numerical simulations and the analysis of measurement trials with a point source in different environments.

Tue 16:00 Van Rijckevorsel Zaal Hydroacoustics

Acoustic Monitoring of a Failing Dike

Tom Basten
TNO Science and Industry

The ’ijkdijk’ in Groningen, The Netherlands, is an international fieldlab for new inspection and monitoring techniques for water barriers. The goal is to learn more of their behavior and to increase confidence in dikes. The dike of the future is filled with a lot of sensors which are used for monitoring and to predict failure of the dike. During the initial experiments a lot of parameters are measured, with all kind of sensors. The current paper deals with the acoustic experiments in the dike during the macrostability experiment in September 2008 while a complete collapse of the dike was forced. The water barrier, 100 meters long and 6 meters high was equipped with two tubes in which microphones and hydrophones were placed. Besides, an array of hydrophones was used, listening to an array of loudspeakers on top of the dike. The goal of this array was image processing based on tomographic inversion. Furthermore a fibre optic sensor was applied as a prototype hydrophone for future applications. A lot of data was generated during the experiments up to the moment of complete collapse of the dike. The paper describes the experiments and some initial results of the analysis.

Tue 16:20 Van Rijckevorsel Zaal Hydroacoustics

Using Multivariate Methods (PCA) for the Online Prediction of Underwater Radiated Sound

Carsten Zerbs[a] and Ingmar Pascher[b]

[a]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics;* [b]*Müller-BBM GmbH*

The acoustical signature is one of the crucial criteria for the detection and classification of vessels. For the estimation of the underwater radiated sound of marine vessels, a procedure was developed and implemented into an operational monitoring system. It is used to calculate an online-prediction of the radiated sound in combination with propagation models of submarines in the far field based on structure borne sound measurements.

With Principal Component Analysis (PCA) the components can be separated and a transfer function to the water borne sound can be calculated. Similarities to the transfer path analysis (TPA) and especially operational TPA as well as cross talk cancellation (CTC) are illustrated.

The paper describes the necessary parallel measurements of structure and water borne sounds to derive and calibrate the model. The possibilities of the integration into an acoustical monitoring system and into a general signature management are illustrated.

Tue 16:40 Van Rijckevorsel Zaal Hydroacoustics

Model Scale Measurements of Surface Ship Radiated Flow Noise

Christ de Jong[a], Johan Bosschers[b] and Hans Hasenpflug[c]

[a]*TNO Science and industry;* [b]*MARIN, Wageningen;* [c]*GE/NL Centre for Ship Signature Management*

Advances in weapon and sensor capabilities are driving an increased interest in the control of underwater signatures of naval platforms. The control of machinery and propeller noise is well understood but there is a shortfall of knowledge of the mechanisms that govern noise due to the flow around the hull of a surface ship. The subject has been investigated by the collaborative US-NL project Mechanisms and Prediction of Surface Ship Radiated Flow Noise. The project aimed to determine and quantify the sources of flow noise generated by surface ships in terms of ship speed and hull shape. The focus was on the underwater noise generated by turbulence excited hull plating vibration and noise resulting from breaking bow waves. A combination of full scale, large scale, model scale tests and computational fluid dynamics was used. The paper will discuss the test set-up and results of the model scale experiments which were made in a (highly reverberant) towing tank. A special silent towing carriage was developed to reduce the background noise, while an acoustic antenna was used to measure the noise due to the breaking bow waves. The pressure fluctuations due to the turbulent boundary layer were measured at several hull locations.

Tue 17:00 Van Rijckevorsel Zaal Hydroacoustics

Estimation of Underwater Flow Noise by Wave Number Decomposition

Jan Abshagen
FWG, Kiel

Flow noise limits the capability of a SONAR system operated on any moving platform at sea. Its physical origin differs qualitatively from other underwater noise sources, such as from ambient noise, yet it is difficult to measure. Flow noise results from pressure fluctuations in the turbulent boundary layer which are generated by the movement of the SONAR platform relative to the surrounding sea. In particular at higher frequencies this noise source differs in wave number distribution from sources which originate in wave propagation processes, e.g. from ambient noise. Measurements of a wave number distribution require an array of spatially separated hydrophones. A method based on wave number-frequency analysis is presented that allows to discriminate between flow noise and noisy sound sources. It therefore enables, for example, estimations of frequency dependencies of underwater flow noise. The method is validated by flow noise measurements performed with an equally spaced line array mounted on a towed system.

Tue 17:20 Van Rijckevorsel Zaal Hydroacoustics

Radiation of ship windows induced by structure-borne sound

Christof Weißenborn and Stefan Semrau
Germanischer Lloyd, Hamburg

Nowadays there is a trend in exterior super-yacht design for the use of larger and larger glass surfaces. When we talk about acoustics here, we are not referring to the occasional noise coming from outside the vessel. Rather, the issue is structure-borne noise (SBN), which is present at all times onboard ship and can stimulate large ship windows to radiate annoying sound into representative passenger areas.

To gain basic knowledge on this issue and to give tangible acoustical consulting for both the shipyard/owner as well as the window system suppliers, several window designs were investigated full-scale in terms of their radiation efficiency. For this purpose, a test-rig was set up to represent typical ship structural elements and the respective SBN-propagation.

In parallel, FEM-calculation were carried out with the aim to predict the window's acoustic behavior from simple modal analysis to the radiated sound power or radiation efficiency.

The paper presents the measurement strategy, incorporating a scanning-laser vibrometer and a sound intensity probe. The results for different window designs are compared and discussed. The validated vibro-acoustic calculation practice is presented as well.

Tue 17:40 Van Rijckevorsel Zaal Hydroacoustics

Investigation of the resonance frequency of a cavitating vortex

Johan Bosschers
MARIN, Wageningen

The cavitation pattern of a modern ship propeller is often characterized by a cavitating vortex. This cavitating vortex may generate (low frequency) hull pressure fluctuations which cause noise and vibration hindrance on board the ship. Existing measurement data of the pressure fluctuations suggest the presence of a resonance frequency, but detailed investigation is hindered by the highly intermittent character of the cavity. A distinct resonance frequency has been observed in the sound generated by a cavitating vortex trailing from a wing of elliptical planform in a cavitation tunnel by Maines & Arndt [JFE199-1997]. The paper will present a theoretical formulation for such a resonance frequency. The agreement between theory and experiment is good for low cavitation numbers but becomes less with increasing cavitation number. The theorically derived dispersion relation is based on the convected Helmholtz equation for a disturbance velocity potential and assumes an inviscid cavitating vortex which shows small deformations for prescribed oscillation modes. The dispersion relation is investigated with respect to phase speed and group speed. The first mode, a breathing mode, may have a zero group speed which is used to derive a unique relation between the non-dimensional (resonance) frequency and the cavitation number.

Tue 18:00 Van Rijckevorsel Zaal Hydroacoustics

Frequency Dependent Attenuation of Oscillations in Fluid-filled Pipes and Orifices

Jürgen Koreck[a] and Otto von Estorff[b]
[a]*Robert Bosch GmbH;* [b]*Inst. of Modelling & Comp., TU Hamburg-Harb.*

High frequency pressure pulsations in injection systems require an extended modeling approach to capture all physical effects. In general, such hydraulic systems, including their pipes, diameter changes, orifices and volume chambers, are modeled by a 1D approach, and steady state pipe friction, being well established and validated, is assumed. Oscillating pressure pulsations, however, lead to unsteady frequency dependent attenuation, which has approved solutions in time and frequency domain. Taking into consideration the geometry, the fluid properties and the frequency, even complex systems can be modeled rather accurately. However, common approaches so far do not account for flow-related effects. Even in the case of no mean flow, oscillating pressure pulsations create nonlinear velocity-dependent resistance at orifices and diameter changes. In the case of high frequency oscillations, the steady state orifice law according to Bernoulli underestimates real effects. Therefore, in this contribution an unsteady orifice law is investigated. Further on the limits of validity of the no flow assumption are inspected by means of an experimental setup and compared to published results.

Tue 14:00 Plate Zaal CAE methodologies

Limits of Finite Element Methods in the Mid-Frequency Range

Andreas Hüppe and Manfred Kaltenbacher
Alps-Adriatic University of Klagenfurt, Applied Mechatronics

Field simulation methods like the Finite Element Method (FEM) in computational acoustics offer some advantages in comparison to energy based methods. For example, they can be applied in the frequency as well as in the time domain and the solution for acoustic pressure and/or velocity is available at every point inside the simulation domain. On the other hand, especially when considering higher frequencies, a FEM simulation requires a huge amount of computational time and memory, thus limiting its range of application to smaller geometries or lower frequencies.
In our contribution we focus on the simulation of the acoustic field in the time domain as the solution provides the whole frequency range of interest in one simulation. Therefore, we consider h-FEM and p-FEM applied to the wave equation and the Spectral Finite Element Method (s-FEM) applied to the governing equations of acoustics. We perform simulations in the mid-frequency range to show the limits of the methods in terms of accuracy of the solution and computational costs. As a practical application, the methods are used to simulate the sound field in the environment of a noise barrier.

Tue 14:20 Plate Zaal CAE methodologies

Wave Based Prediction Technique for Sound Radiation Analysis

Jan Rejlek[a], Tamas Mocsai[a], Petra Silar[a], Achim Hepberger[b] and Hans-Herwig Priebsch[a]
[a]*Virtual Vehicle Competence Center, Graz;* [b]*AVL LIST GmbH*

Recently, the wave based prediction technique (WBT) has been developed as an alternative method for solving steady-state acoustic problems. The WBT has proven to be a robust tool for interior acoustics. However, a considerable class of real-life acoustic applications involves the analysis of problems in unbounded spatial domains, such as sound scattering and sound radiation problems. Besides the methods based on boundary integral formulation of the governing differential equation various strategies employing the standard finite element schemes were developed in order to tackle unbounded problems. Although based on different approaches, all these concepts have the same basic idea in common, namely introducing an artificial truncation boundary that divides the infinite domain into bounded and unbounded region. This paper outlines the recent developments of the novel wave based approach and its application for three-dimensional sound radiation analysis by adopting similar strategy of dividing the model into two parts, as described above. The conventional interior formulation is used in the bounded part, while some new additional functions, which inherently satisfy the Sommerfeld radiation condition are adopted in the unbounded part of the wave model.

Application to industry-sized problem demonstrates the enhanced computational efficiency, which allows the practical computational limitation to be shifted towards higher frequencies.

Tue 14:40 Plate Zaal CAE methodologies

Hybrid FEM/SEA model of an aircraft fuselage section

Alexander Peiffer and Stephan Tewes
EADS Innovation Works

This paper deals with the SEA- and hybrid model generation derived from a detailed finite element model of a fuselage structure. Starting from a modal analysis as first check of the dynamic behaviour several subsystem configurations are investigated using the energy flow method as SEA post-processing. These results provide all criteria that are required for the evaluation of each subsystem configuration, i.e. the modal density and the coupling behaviour between the subsystems. Based on that, the frequency ranges for different hybrid FEM/SEA models are selected leading to a full SEA model for high frequencies. The models are checked concerning plausibility and SEA rules validity. Finally a primary insulation is realized by trim coupling of the fuselage shell and the interior cavities are realized as SEA cavities.

Tue 15:00 Plate Zaal CAE methodologies

Energy resolution of complex vibro-acoustic problems

Nicolas Totaro and Jean-Louis Guyader
INSA of Lyon

The present article deals with an energy resolution of complex vibro-acoustic problems. This method is based on SmEdA approach previously published for extending Statistical Energy Analysis (SEA) to systems where modal equipartition of energy is not achieved. Modal energies are calculated from modal injected powers and modal coupling loss factors whatever the excitation even for systems with modal behaviour. From the computing time point of view two basic advantages appears, the calculation is made directly for frequency averaged energies and modal coupling loss factors are simply extracted from modal characteristics of the uncoupled systems obtained with standard Finite Element method. The approach provides also an analysis tools to detect the modal transmission path, very useful in early design stage. It is possible to easily identify couple of structural and acoustical modes most likely to transmit energy and estimate contributions of each structural mode to energy into the acoustical domain One interesting point is the possibility of calculating energy maps into systems with very small increase of computation, additional assumption is necessary that will be discussed. An application on a test case will be presented and compared with standard Finite Element method.

Tue 15:20 Plate Zaal CAE methodologies

Vibro-Acoustic Simulation of Structure-borne Induced Radiation of Ship Windows

Bernard van Antwerpen[a], Diego d'Udekem[a] and Christof Weißenborn[b]
[a]*Free Field Technologies;* [b]*Germanischer Lloyd, Hamburg*

Large ship windows seem to represent a weak point in terms of acoustic radiation within a ship cabin. There are is a emerging need to predict the behavior of those components prior to their final design, construction and assembly on board.
In this framework, numerical methods offer a good platform for predicting the acoustic radiation and structural vibration of such windows or to compare the performance of different windows system designs. For instance, it allows predicting the radiation efficiency, which represents a key point in reducing the global sound pressure level within the cabin. This paper presents a vibro-acoustic study performed with the finite element code ACTRAN. The radiation efficiency of different window configurations is estimated. The focus is placed on the determination of the most sensitive system parameters.
In this paper, the simulation method is first briefly described. The obtained results are also compared to a set of full scale experimental measurements for validation. Different window designs approaches are compared in order to guide the choice of an optimal design to select.

Tue 16:20 Plate Zaal CAE methodologies

Automated modelling procedure for acoustic Wave Based Technique models

Gregor Müller and Thorsten Bartosch
Magna Steyr Fahrzeugtechnik AG & Co KG

In modern vehicle acoustic development, fast and easygoing modelling and simulation procedures are required to meet the today's demands in automotive industry. To close the gap in the mid frequency range, several years ago, the Wave Based Technique, a deterministic vibro-acoustic simulation technique for the mid-frequency range, was established. Instead of meshing the fluid domain of interest into a vast number of small volumes (as required for FEM), the model has to be partitioned into convex polyhedron shaped sub-domains. This paper intends to present an algorithm for the automated creation of acoustic WBT models of passenger compartments out of hull models. In addition, relations between geometric model parameters and modelling/simulation time and quality are investigated. From these investigations, conclusions are drawn and automated modelling strategies are established. Furthermore, alternative modelling approaches are discussed and proposed.

Tue 16:40 Plate Zaal CAE methodologies

Design and Tests of Thermoplastic Textile-reinforced Composite Trays for Vibro-acoustic Relevant Applications

Werner Hufenbach, Martin Dannemann, Stefan Friebe, Frank Kolbe and Michael Krahl
TU Dresden, Institute of Lightweight Eng. and Polymer Techn.

Driven by increasing customer demands, environmental and economical requirements, lightweight engineering gains in importance for various industrial and transport applications. Here, composite materials based on thermoplastic hybrid yarns offer new possibilities for multifunctional design due to its adjustable properties. Requirements concerning high stiffness and damping as well as low weight and short cycle times during manufacture can be met by an adapted composite lay-up and manufacturing process. In the presented work, the vibro-acoustic behaviour of selected textile-reinforced thermoplastics was characterized dependent on the kind of reinforcement, number of layers and its fibre orientation. Therefor, models based on Finite Element Method and Boundary Element Method for the calculation of modal and acoustic parameters of textile-reinforced structures were developed. For the verification of this simulation models as an example composite trays were manufactured by a novel hot pressing technology. The modal parameters and transmission loss of the produced trays were measured. Subsequently, the experimental results were used for the verification of the numerical simulations. The performed investigations illustrate that the vibro-acoustic behaviour of composite trays is influenced by the complex interaction of geometry and material parameters. Thus, an optimal sound radiation behaviour can only be achieved by a comprehensive material-adapted vibro-acoustic design.

Tue 17:00 Plate Zaal Finite element models

Simulation of acoustic transmission characteristics of complex multilayered structures using a finite element method

Jean-Pierre Coyette, Gregory Lielens and Benoit van den Nieuwenhof
Free Field Technologies

The paper focuses on a finite element simulation technique for the evaluation of the acoustic transmission characteristics of complex multilayered structures. The key ingredients of this technique are finite element models for the bounded/unbounded acoustic and structural domains, a refined sampling procedure for distributed random excitations (diffuse field or turbulent boundary layer) and a particular homogenization procedure which enables the reduction of the complex multi-layered structure (plate or shell) to an equivalent homogeneous structure. The technique is illustrated by various examples. Computational performances are highlighted.

Tue 17:20 Plate Zaal Finite element models

Acoustic Modal Analysis of a Recorder

Stefanie Fuss and Steffen Marburg
TU Dresden, Institut für Festkörpermechanik

This talk presents the three-dimensional numerical simulation of the sound spectrum and the propagation of the acoustic noise inside and around a recorder. The fluid inside and close to the recorder is meshed by Lagrangian tetrahedral finite elements. To optain results in the far field of the recorder, complex conjugated Astley-Leis infinite elements are used. To apply these infinite elements the finite element domain has to be meshed either in a spherical or an ellipsoidal shape. Advantages and disadvantages of both shapes regarding the recorder will be shown in this talk. Examples of the modal analysis of the fluid for different tones will also be presented.

Tue 17:40 Plate Zaal Finite element models

Computational Acoustics based on Conservation Equations using Taylor Hood Finite Elements

Manfred Kaltenbacher and Andreas Hüppe
Alps-Adriatic University of Klagenfurt, Applied Mechatronics

Within this contribution we concentrate on the efficient solution of enhanced wave problems, which are mathematically modeled by their conservation equations, namely the mass and momentum conservation. Such formulations arise in computational aeroacoustics or in nonlinear acoustics. Even for acoustic wave propagations, where an acoustic wave equation exists, formulations based on the conservation equations have the advantage, that the numerical solution provides directly the two main engineering quantities, the acoustic particle velocity and the acoustic pressure.

The application of the standard Finite Element (FE) method to the acoustic conservation equations will result in an instable formulation. In order to obtain stable finite elements, we need a mixed FE formulation. The way to obtain such a formulation is to use H(div) functions for the approximation of the particle velocity and L2 functions for the acoustic pressure. However, it can be proofed, that approximating the acoustic particle velocity by Lagrangian finite elements one order higher than those for the acoustic pressure will also provide us a stable formulation.

We have applied this so-called Taylor-Hood elements to the acoustic conservation equations and will present their applicability to general wave problems. In detail, we will present a study of the accuracy and computational efficiency.

Tue 18:00 Plate Zaal Finite element models

Fast FE-Analysis and Measurement of the Hydraulic Transfer Function of Pipes with Non-Uniform Cross Section

Jan Herrmann, Michael Spitznagel and Lothar Gaul
University of Stuttgart, Applied and Experimental Mechanics

Automotive piping systems are often characterized by a non-uniform cross-section. Orifices in fuel charge assemblies are a typical example. The goal of this research is to quantify the influence of diameter changes on the hydraulic transfer function between the dynamic pressure at the beginning and at the end of the pipe. A finite element based substructuring and component mode synthesis technique is used to assemble the spatial piping system including the orifice. The adaptation of the Craig-Bampton method on fluid-structure coupled piping systems with an additional reduction of the remaining interface degrees of freedom leads to a considerable model order reduction. A local, frequency dependend fluid damping model is integrated in the finite element code and the result of the harmonic analysis is compared to measurements. A hydraulic test bench with a dynamic pressure source is used to estimate the hydraulic transfer function for different orifice diameters.

Tue 14:00 Van der Vorm Zaal Medical ultrasound

Bubble vibrations at medical diagnostic frequencies

Nico de Jong[a,b], Rik Vos[a], Marlies Overvelde[b], Valeria Garbin[b], Marcia Emmer[a], Michel Versluis[b] and Detlef Lohse[b]
[a]*Biomedical Engineering, Erasmus MC, Rotterdam;* [b]*Physics of Fluids, University of Twente, Enschede*

Ultrasound is the most widely used medical imaging modality. Ultrasound contrast agents (UCA, consisting of small (encapsulated) gas bubbles), will increase the reflection of ultrasound by the blood pool, after intravenously administration, and by that make it possible to provide perfusion images. The diagnostic frequency used (1-10 MHz) together with the bubble size ($\sim 3\ \mu$m) result in interesting bubble vibrations which can be imaged with the ultrahigh speed Brandaris-128 camera and predicted with newly developed models. The vibrations can be classified as follows: 1. Spherical vibration of bubbles 2. Mode vibration of bubbles 3. Compression-only vibration of bubbles 4. Onset of the bubble vibration 5. Vibration of bubbles close to a wall Studying the dynamics of the bubble upon insonification can be done optically, e.g. under a microscope, or acoustically with the aid of piezo transducers. Optical studies are performed with a high-speed camera, Brandaris (25 million frames/s). To predict the behaviour of encapsulated bubbles in an ultrasound field we have adapted the Rayleigh-Plesset equation to include the behaviour of the phospholipid shell. Detailed analysis of the high speed recordings have shown to be mandotory for a complete model.

Tue 14:20 Van der Vorm Zaal Medical ultrasound

Modeling Pulsed Nonlinear Ultrasound for Medical Applications: the INCS Method

Martin Verweij
TU Delft, Laboratory of Electromagnetic Research

The higher harmonics in nonlinear ultrasound fields gain importance for improving the image quality of medical echoscopy. Consequently, the optimization of medical imaging techniques has become highly dependent on the accurate simulation of pulsed nonlinear ultrasound fields in tissue. In this context, a successful simulation method should be able to compute the pulsed ultrasound field of a phased array transducer in a nonlinear and attenuating medium. Moreover, the method should cope with a large-scale domain (approximately 100 wavelengths in three spatial directions and 100 periods in time) and wide-angle propagation. Current methods are either based on approximations that favor a specific propagation direction or require prohibitive large grids. This paper presents the Iterative Nonlinear Contrast Source (INCS) method, which meets all above requirements and provides a full-wave solution of the Westervelt equation. This method treats the nonlinear term from the Westervelt equation as a distributed contrast source. The field is iteratively updated by convolution of an estimate of this nonlinear contrast source with the Green's function of the linear wave equation. Moreover, the performance of the INCS method is discussed by showing the iterative generation of the higher harmonics, the directional independence, and the simulation of a highly steered beam.

Tue 14:40 Van der Vorm Zaal Medical ultrasound

Application of Volterra Series to Ultrasound Imaging

Martin Schiffner, Michal Mleczko and Georg Schmitz
Ruhr-University Bochum, Institute of Medical Engineering

Diagnostic ultrasound imaging provides a real-time imaging modality which features anatomic and, with the help of ultrasound contrast agents, molecular imaging capabilities. Ultrasound contrast agents consist of microbubbles with diameters in the micron range. Under ultrasound insonification, these bubbles oscillate nonlinearly. Furthermore, the nonlinearity of sound propagation in tissue cannot be ignored for diagnostic ultrasound imaging. To improve current imaging modes and to develop novel techniques, a manageable model which enables the investigation of these combined effects is necessary.

Models which enable an accurate description of the nonlinear effects encountered in ultrasound imaging are known. These usually require numerical solutions of nonlinear differential equations, and consequently, they are of limited use for the development of novel imaging techniques. Although numerical solutions enable an accurate description of a system's response, they do not give insight into the general input-output behavior. Thus, lacking a simple yet sufficiently accurate description form, nonlinearities are either ignored or modeled as characteristic curves. Volterra series provide a manageable means to model nonlinear propagation of sound and the oscillation of microbubbles with sufficient accuracy. Applications of Volterra series to the modeling of the signal chain of ultrasound imaging systems are presented.

Tue 15:00 Van der Vorm Zaal Medical ultrasound

Results of an Experimental Study for 3D Ultrasound CT

Nicole Ruiter, Gregor Schwarzenberg, Michael Zapf, Alexander Menshikov and Hartmut Gemmeke
Forschungszentrum Karlsruhe

Our 3D Ultrasound Computer Tomograph (USCT) records the interaction of ultrasonic waves with the imaged object from many different angles with approx. 2000 unfocussed ultrasound transducers. The main advantage of such a system is simultaneous recording of reflection, absorption and speed of sound images, and, in future, higher image quality with faster data acquisition.
For many years, it was not feasible to build a device for 3D USCT due to the large number of required transducers, high data rate and time consuming post-processing. The aim of our first experimental setup for 3D USCT was to analyze the feasibility of today's technology and draw conclusions for application in breast cancer diagnosis.
The results of our system are presented and discussed in regard to data acquisition, image quality and duration of image reconstruction. The main conclusion is, that 3D USCT is feasible with today's technology, if a sparse aperture or long DAQ times can be accepted. The next question is, if the resulting images have clinical relevance and can improve early breast cancer diagnosis. Based on a second generation of 3D USCT, imaging volunteers and carrying out a small pre-clinical study is under preparation.

Tue 16:00 Van der Vorm Zaal Medical ultrasound

Contrast Enhanced Optoacoustics for Molecular Imaging

Marc Fournelle, Kirsten Maass, Heinrich Fonfara, Wolfgang Bost and Robert Lemor
Fraunhofer Institut für Biomedizinische Technik

Optoacoustic imaging can provide valuable information about tissue structures by combining the high intrinsic contrast of optical modalities with the good resolution and the high penetration depth of ultrasound. When biological tissue is irradiated with ultrashort laser pulses of durations of a few nanoseconds, the light is absorbed according to the local optical properties of the tissue, and is converted into pressure by means of the thermoelastic effect. The resulting pressure waves can be acquired and analyzed with techniques similar to conventional ultrasound and provide an image of the distribution of absorbed light in tissue. Although endogenous chromophores such as melanin or haemoglobin result in remarkably high optoacoustic contrast, the administration of additional targeted nanoparticular contrast agents becomes necessary if a more selective molecular imaging is needed. Different types of gold nanoparticles are particularly suitable in this context since their strong plasmon resonance results in a high absorption coefficient. An overview on results regarding the usability of the different particle types as optoacoustic contrast agents will be given and the technology and its applications will be discussed. Further, first results on contrast-enhanced in-vivo imaging with a system for real-time optoacoustics will be presented.

Tue 16:20 Van der Vorm Zaal Medical ultrasound

Comparison of Coding Techniques for Photoacoustic Coded Excitation

Martin Mienkina[a], Annika Eder[a], Claus-Stefan Friedrich[b], Nils Gerhardt[b], Martin R. Hofmann[b] and Georg Schmitz[a]

[a]*Ruhr-University Bochum, Institute of Medical Engineering;* [b]*Ruhr-University Bochum, Inst. of Photonics and Terahertz-Techn.*

In photoacoustics (PA) ultrasound is generated by laser irradiation of tissue. Usually, PA signals are generated by Nd:YAG lasers but it would be attractive to use less expensive laser diodes instead. Although laser diodes exhibit low pulse energy, the pulse repetition frequency (PRF) of pulsed laser diodes is much higher than the PRF of Nd:YAG lasers thus averaging can be used to increase the SNR. The applied PRF for averaging is limited by the acoustical time-of-flight. To further increase the SNR we propose to use coded excitation for PA imaging. In optical-time-domain-reflectometry, which is similar to PA imaging, Simplex Codes (SC) are widely used for coding. In ultrasound imaging Golay Codes (GC) are successfully employed. Therefore, we compare the performance of SC and GC for PA coded excitation (PACE) by a simulation study. For both codes, the coding gain normalized to equivalent averaging procedures is computed as a function of code length and PRF of the laser diode. Assuming a time-of-flight of 60 μs and e.g. a PRF of 250 kHz the coding gain of GC is 1.6 dB higher than the coding gain of SC. Consequently, GC is more suitable for PACE than SC.

Tue 16:40 Van der Vorm Zaal Medical ultrasound

Output Beam Characterization of Medical Diagnostic Ultrasound Systems Using a Thermal Technique for Intensity Measurements

Volker Wilkens

Physikalisch-Technische Bundesanstalt, Braunschweig

Ultrasound exposure measurements for medical ultrasound systems are essential as regards safety and quality assurance. International standards require the characterization of output beams regarding, for instance, power, intensity, peak pressure, and lateral dimension. Spatial-peak temporal-average intensities are particularly important since they are related to thermal hazards for the patient. Local intensity parameters are commonly derived from hydrophone measurements. Alternatively, a simple and low-cost thermal intensity sensor technique recently developed at PTB can be applied to precise ultrasound time-averaged intensity determination. The applicability of these novel sensors is demonstrated by extensive exposure measurements on a typical diagnostic ultrasound system. Output beams of different operation modes and working frequencies are characterized by line scans along the beam axis and by beam profile measurements. For non-scanning modes of the diagnostic device, these results are compared with additional hydrophone-based intensity measurements. With respect to typical uncertainties of intensity parameters within the range of 20% to 30%, the agreement is very good. However, the advantages of the thermal sensor technique are especially evident for intensity measurements in the case of scanning and combined modes of the diagnostic device, where the synchronization between hydrophone measurements and the complex pulse emission pattern can be difficult.

Tue 17:00 Van der Vorm Zaal Medical ultrasound

Design of an Ultrasound Cylindrical Phased-Array for Hyperthermia Breast Cancer Treatment

Koen W. A. van Dongen[a], Jurriaan F. Bakker[b], Maarten M. Paulides[b], Inge-Marie Obdeijn[b] and Gerard C. Rhoon[b]
[a]*TU Delft - Faculty of Applied Sciences;* [b]*Erasmus MC, D. den Hoed Cancer Center, Rotterdam*

The objective of this theoretical study is to design an ultrasound (US) cylindrical phased-array that can be used for fever-range hyperthermia (40^{o}C to 44^{o}C) treatment of tumors in intact breasts. Hereto, design parameters such as frequency, number of transducers per ring, ring distance and number of rings are optimized to obtain an interference pattern with a small primary focus, while suppressing secondary foci.
In addition, we have investigated the influence of acoustic and thermal heterogeneities on the specific absorption rate (SAR) and temperature patterns in order to determine the necessity of using heterogeneous models for US applicator design and treatment planning.
From the simulations it is shown that an US cylindrical phased-array can be used to obtain 44^{o}C in the center of tumors located anywhere in the intact breast. In addition, it is demonstrated that due to the low frequency content of the signal (100kHz) acoustic heterogeneities do not disturb the SAR and temperature patterns.

Tue 17:20 Van der Vorm Zaal Medical ultrasound

Metrology of HITU fields

Klaus-Vitold Jenderka[a], Klaus Beissner[b] and Julian Haller[b]
[a]*Physikalisch-Technische Bundesanstalt, Braunschweig;* [b]*Physikalisch-Techn. Bundesanstalt, Braunschweig / Berlin*

High Intensity Therapeutic Ultrasound (HITU) is a non-invasive technique for the treatment of various types of cancer, as well as non-malignant pathologies by inducing localized necrosis of the tissue. Even though HITU is already in clinical use, this technique is still an emerging therapeutic procedure, with a lack of knowledge on reliable and stable treatment planning and monitoring procedures. A crucial step for the prediction of the desired temperature rise in a HITU-treated region is the measurement of the ultrasound output power. The evaluation of tissue thermometry methods, using MR or ultrasound itself, is important for the treatment monitoring, as well as for choosing the appropriate ultrasound power for a desired local intensity and temperature rise respectively. While ultrasound fields are well-investigated for output powers smaller than 50 W, this paper describes first results of measurements in much stronger, highly focused fields ($>$ 300 W ultrasound power at 1.5 MHz) using an absorbing target arranged close to the transducer. The measurement uncertainties are discussed.

Tue 17:40 Van der Vorm Zaal Medical ultrasound

Simulation of Large Spatial Domain Ultrasound Scattering Problems

Erwin J. Alles and Koen W. A. van Dongen
TU Delft - Faculty of Applied Sciences

Currently, many ultrasound simulations of medical imaging modalities are performed using the Field II-software. This software models a scattering body as a collection of point-scatterers and evaluates the resulting total pressure field within the Born approximation. Consequently, it assumes a constant speed of sound throughout the volume. Therefore, modelled reflections arrive at incorrect times, and errors are introduced if thickness measurements are performed. In addition, multiple scattering is neglected in this approximation, which means that for strong contrasts results obtained with Field II are incomplete.
To circumvent these issues, we have developed software which efficiently computes the total pressure field within a volume due to scattering off contrasts of arbitrary shape and magnitude. The software uses a conjugate gradient scheme to solve the scatter integral equation for the total field, given known contrasts and incident field. Spatial domain decomposition is used to allow for large domains of which the memory load would otherwise exceed the amount of memory present in current machines.
During our presentation, we will compare the simulated received signals, using both programs, for an acoustic probe measuring plaque thickness in an array.

Tue 18:00 Van der Vorm Zaal Medical ultrasound

Noniterative second harmonic ultrasound filed simulations: an axisymmetric approach

Nicola Testoni[a], Karin Hensel[b], Monica Siepmann[b], Nicolò Speciale[a] and Georg Schmitz[b]
[a]*Università di Bologna;* [b]*Ruhr-University Bochum, Institute of Medical Engineering*

Nonlinear numerical modeling of finite amplitude acoustic beams has a key role in the design of state-of-art ultrasonic systems. The Khokhlov-Zabolotskaya-Kuznetsov (KZK) equation is one of the most accurate model describing diffraction, absorbtion and nonlinearity combined effects in ultrasound wave propagation.
KZK solutions usually follow two different approaches: the spectral and the time domain method; the former is well suited for periodic ultrasound excitation, while the latter is more efficient in propagation of short pulses. As finite difference methods and stepping techiques are usually employed in most realizations, prediction accuracy and computational burden of ultrasound field simulations are strongly related to the depth.
The work described in this paper points to develop a model for simulating second harmonic ultrasound fields by featuring equations that can be solved without the use of iterative techniques. Using a domain change and addressing the integration along the axial direction with a proper approximation, robust estimations of nonlinear sound fields at arbitrary depth are produced for both focused and unfocused axisymmetrical sources. Predicted profiles for nonlinear propagation in water from real transducers will be presented and compared both visually and numerically with measurements from water tank experiments, showing the good agreement and artifact rejection obtained.

Tue 14:00 Hudig Zaal Tire-road noise

A Ring-based Tyre Model for the Prediction of Structure-borne Interior Tyre/Road Noise.

Peter Kindt, Paul Sas and Wim Desmet
KU Leuven

The tyre/road interaction causes a structural excitation of the tyre. This vibrational energy is transmitted through the suspension towards the vehicle body. The resulting body panel and window vibrations cause noise radiation in the vehicle passenger compartment. In order to predict this interior noise during the vehicle development phase, an accurate tyre model is required. This paper presents a structural tyre model, based on a three-dimensional flexible ring on an elastic foundation. The ring represents the belt and the elastic foundation represents the tyre sidewall. The model is valid till 300 Hz and includes a submodel of the wheel and the air cavity. Unlike most ring models, which only consider in-plane modes, the presented model also predicts modes that involve torsion of the belt in circumferential direction. The parameterization of the model, which does not require detailed knowledge of the tyre construction, is based on the main geometrical properties of the tyre and a limited modal test. The unloaded tyre response is calculated and validated with test data. As the presented model is physical, it can be applied to describe other operational conditions such as loading and rotation. In this paper, the response of the loaded tyre is also considered.

Tue 14:20 Hudig Zaal Tire-road noise

Contributions to a Better Understanding of Tire Cavity Noise

Christoph Bederna and Ernst-Ulrich Saemann
Continental AG

The air cavity as source of tire/road noise was discovered in the beginning of the 1990th when the vehicle industry gave the suspension a more sportive condition and when it became popular to use tires with low aspect ratio. As a solution to lower the transmission of that sound into the vehicle interior, mainly secondary measures like absorbing material in the tire were proposed by the tire manufacturers. Unfortunately the vehicle industry rejected the use of those solutions for cost reasons and asked the tire manufactures to include primary measures directly into the tire construction. A literature review reveals limited information and offers simple models to predict only the first eigenfrequency of the tire cavity and a qualitatively prediction of the forces at the vehicle spindle based mainly on FEM techniques which do not allow conclusions how the tire constructions has to be changed to reduce the air cavity. The paper presents current FEM simulation results based on real tire constructions to understand the problem and additional sound measurements in the tire with a hydrophone to validate the simulations and contribute to a better understanding of the sound generating mechanism.

Tue 14:40 Hudig Zaal Tire-road noise

Road Traffic Noise of Expansion Joints - Get a Grip on it

Willem-Jan van Vliet[a], Jan Hooghwerff[b], Nico Booij[c] and Ronald van Loon[b]
[a]*Rijkswaterstaat - Dienst Verkeer en Scheepvaart;* [b]*M+P - raadgevende ingenieurs, Vught;* [c]*Rijkswaterstaat Bouwdienst, Utrecht*

Expansion joints used in road traffic bridges can be a serious source of noise annoyance. This is caused by the impulsiveness of the sound of vehicles passing over the joints and sometimes the presence of low frequencies in the sound. In the Netherlands large-scale application of low noise road surfaces creates a need for low noise expansion joints. Therefore Rijkswaterstaat has created a guideline to deal with noise of expansion joints. The goal is to avoid excessive noise problems by setting noise limits in contracts. The approach in the guideline is based on existing calculation and measurement methods with which there is already considerable experience in the field of low-noise road surfaces. The application of these methods to the noise caused by expansion joints is new.

Tue 15:00 Hudig Zaal Tire-road noise

Uncertainty of Close-Proximity (CPX) Tire-Road Noise Measurements

Foort de Roo[a], Gijsjan van Blokland[b], Hans van Leeuwen[c], Jos Reubsaet[d], Jan Telman[a] and Willem-Jan van Vliet[e]
[a]*TNO Science and Industry;* [b]*M+P - raadgevende ingenieurs, Vught;* [c]*DGMR Industrie, Verkeer en Milieu BV;* [d]*Adviesburo Jos Reubsaet, Heerlen;* [e]*Rijkswaterstaat - Dienst Verkeer en Scheepvaart*

Practical experience in the Netherlands with the Close-Proximity (CPX) measurement method for the influence of road surfaces on traffic noise (ISO/CD 11819-2) showed that the measurement uncertainty of this so-called trailer method is relatively large. In order to gain more knowledge about the determining factors a well-coordinated Round-Robin test was organised and combined with a statistical investigation of the primary parameters that contribute to the uncertainty. Five organisations that perform CPX-tests on a commercial basis participated in the Round-Robin test and made their measurement trailer and its technical specifications available for this research. This paper will describe the test set-up of the Round-Robin test and the results achieved. Based on these results recommendations for a stricter specification of the test conditions and the design parameters of measurement trailers will be presented.

Tue 15:20 Hudig Zaal Tire-road noise

Validation of the Next Generation SPERoN Model

Ard Kuijpers
M+P - raadgevende ingenieurs, Vught

The SPERoN model is a hybrid tyre/road noise prediction model. It was developed in the lastdecade with the aid of several national and European R&D programs. Lately the model has undergone major changes and we can now speak of a next generation model. The model was extended with a completely new truck tyre contact and radiation model. In addition, a horneffect/propagation model was introduced which describes the influence of road absorption on the radiation characteristics of both passenger car and truck tyres. The original passenger car model was reworked to allow prediction of near-field (CPX) levels besides pass-by levels.

Validation of the model showed that the predictions are accurate with a standard error of approximately 1 dB. For the validation, an extensive measurement program was carried out on five different road surfaces on urban, secondary and highway roads.

This paper presents the characteristics of the new SPERoN model, the range of application for the model and the results of the validation.

Tue 16:20 Hudig Zaal Tire-road noise

Silent Road Trafic 2

Wolfram Bartolomaeus
Bundesanstalt für Straßenwesen (BASt)

After finishing the research project ”Silent Road Traffic 1” in 2003 the follow-up project ”Silent Road Traffic 2” was started in 2005. Under the leadership of the Bundesanstalt für Straßenwesen - BASt a budget of 4.5 Mio. e was shared with ten other partners from Industry (Continental AG, RW Sollinger Hütte GmbH and Maurer Söhne GmbH & Co KG) , Research Institutes (Müller BBM GmbH, Forschungsinstitut für Pigmente und Lacke, Bundesanstalt für die Materialforschung und -prüfung - BAM) and Universities (Universität Stuttgart - Institut für Straßen- und Verkehrswesen, Technische Universität München - Fachgebiet Hydromechanik, Universität Hannover - Institut für Baumechanik und Numerische Mechanik and Universität Hamburg-Harburg - Institut für Modellierung und Berechnung). The research project consists of three parts: ”Quiet Tires”, ”Quiet Roads” and ”Result Checking”. The end of the project will be at the end of 2009. But nevertheless there are already a lot of results available, specially from the part ”Quiet Tires”.

Tue 16:40 Hudig Zaal Tire-road noise

DEUFRAKO Prediction and Propagation of Rolling Noise - Results

Markus Auerbach
Bundesanstalt für Straßenwesen (BASt)

This project aims at the development of new road pavement concepts with respect to tyre-road noise in a wider context of the abatement of road traffic noise. French and German institutes headed by the ”Laboratoire Central des Ponts et Chaussées” (LCPC) and the ”Bundesanstalt für Straßenwesen” (BASt) cooperate throughout this project. The theoretical approach is based on the SPERoN (”Statistical Physical Explanation of Rolling Noise”) model developed by one of the partners. In a first stage, the model was validated for different kinds of French road surfaces. In a second step, the model was applied as an adapted tool to design new textures for low noise road surfaces. Finally, in a third step, the results issued from SPERoN were used as an input for outdoor sound propagation models developed in France and Germany in order to estimate the effect of this optimised model-aided low noise pavement in the far field, close to the façades. A ranking with other typical German and French pavements has lead to a common database (DEUFRABASE). The results of the project will be presented.

Tue 17:00 Hudig Zaal Tire-road noise

Tread pattern optimization by means of simplified rolling model

Wolfgang Kropp
Chalmers University of Technology

While on very rough roads the tread pattern is of minor importance on typical European road surfaces the tread pattern plays a dominant role for the generation of tyre/road noise. Therefore the optimization of the tread pattern design is essential in order to minimize rolling noise. Different approaches are utilized to evenly distribute spectral energy due to tread pattern variation along the circumference of the tyre. Here an approach is suggested which is based on a simplified rolling model as implemented in the SPERoN model. The approach has the advantage that it includes the structural response of the tyre, while many other approaches only consider the geometrical variation of the tread pattern. In addition the approach also allows for taking into account the influence of different road surfaces in order to investigate the performance of the tread pattern "in field". In the paper the model is presented and it functioning is demonstrated for typical design changes in tread pattern.

Tue 17:20 Hudig Zaal Tire-road noise

Effect of road surface characteristics on rolling resistance in relation to rolling noise

Gijsjan van Blokland[a], Ines Lopez Arteaga[b], Stijn Boere[b] and Jurek Ejsmont[c]
[a]*M+P - raadgevende ingenieurs, Vught;* [b]*Eindhoven University of Technology;* [c]*Technical University Gdansk*

It is generally acknowledged that road surfaces have a significant effect on the noise of a tyre rolling on the surface. This has led to extended study ands application of low noise surfaces in the Netherlands, Germany and many other countries. The increased awareness of climate change generated the question if low noise surfaces might also exhibit low rolling resistance. This is studied on the Kloosterzande testing area where several test section with varying properties were laid. The rolling resistance (RR) is determined by the TUG through measuring the force generated by a free rolling passenger car tyre on these surfaces. These data is then related to the rolling noise levels found on these surfaces where we found that low noise correlates well with low rolling resistance. A special case were surfaces with a flexible top where large variations in rolling resistance were found. Surprisingly the lowest value were observed on a flexible surface. In a next step we related the RR properties to surface characteristics, mainly the surface texture and the mechanical impedance. The paper will present data of RR of the surfaces, the relation with rolling noise levels and a first explanation of RR by the surface properties.

Tue 17:40 Hudig Zaal Tire-road noise

The development of a CPX measurement set-up capable of measuring tire-road noise effectively

Dirk Bekke
Vredestein Banden B.V., Enschede

In the Directive 2001/43 limits are put on the maximum noise level of a pass-by noise measurement on an ISO 10844 surface. This prescribed measurement set-up only gives an L_{Amax} level of the total passage of a whole vehicle equipped with four tires and does not provide enough information for a problem analysis or model validation. Measuring on one tire with (closed or open) CPX-trailers or -cars gives a better insight into tire-road noise. The difficulties with these measurement set-ups can be the noise generation of the set-up itself, the acoustics of the trailer-interior or vibrations directly disturbing the microphone signals. A CPX measurement set-up has been developed keeping these considerations in mind in order to be able to do proper problem analysis and model validation.

Tue 18:00 Hudig Zaal Tire-road noise

On Solving the Tyre-Road Contact Problem at High Frequencies.

Ysbrand Wijnant
University of Twente

The contact between tyre and road is non-linear and most accurately described in the time domain. Currently however, accurate finite element models are too large to be solved at the small time scales required to capture tyre/road noise. At the Structural Dynamics and Acoustics group of the University of Twente an alternative contact algorithm has been developed. The algorithm solves the dynamic equations and, while solving, satisfies exactly the contact condition (the condition that states that there is no penetration of the tyre into the road). As a result, there is no need for contact elements and no additional parameters are required to stabilize the solution process. The possibility to optimize and speed up the algorithm by means of multigrid is the major advantage of the new approach. In this paper the contact algorithm is explained for a finite element discretization.

Poster session Tuesday, 25 March 2009

Authors will be present: 15:20 - 16:20

Tuesday Poster session 1

Following of Inverse Music Sequencer Operation – Detection of Music Components from Wave-table in a Complex Music Signal

Stepan Albrecht
University of West Bohemia in Pilsen

We made an attempt to a novel approach in automatic music transcription (i.e., in detection of pitch, loudness and timing of all sound events in a complex music signal automatically) working without any constraints on the observed music signal in general. It follows the reverse working of music sequencers. That is, we have a wave-table comprising arbitrary sounds as drums, melodic sounds, etc. Given the wave-table with sounds and a piece of a complex music signal for analysis, the sounds in the wave-table (or their modifications) are identified in the music signal. The sound events (as component IDs, their position in time and their truncation) are the output of the identification process. When we try to put what was identified into the track, we should obtain the same or rather similar song up to some point. The core part of the algorithm is the sequential Monte Carlo method. Tests of functionality will be demonstrated.

Tuesday Poster session 1

Effect of airborne sound on installation noise - Part 1: Basic investigations

Mirco Ebersold[a], Lutz Weber[a], Sven Öhler[a] and Matthias Blau[b]
[a]*Fraunhofer Institut für Bauphysik;* [b]*FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie*

The noise excitation of buildings by service equipment takes place by both structure-borne and airborne sound. Mostly sound transmission is determined by the structure-borne portion. For some sources, however, the contribution of airborne sound can't be neglected. A typical example is the EMPA-hammer, defined in the swiss standard SIA 181 as a standardized source for the simulation of user noise. If airborne sound contributes substantially, conventional measures for noise reduction based on elastic isolation between source and building won't work properly. In order to develop efficient measures for that kind of excitation, structure-borne and airborne sound must be separated and investigated in context with each other.

The separation of structure-borne and airborne sound was performed by means of a simplified experimental set-up consisting of two parallel plates acoustically connected by an adjustable structure- borne sound bridge and by the air gap between the plates. Since the receiving plate was bedded on an elastic support the transmitted sound power could be determined directly from the mean velocity level on the surface of the plate. For investigation of sound transport the experimental conditions such as distance of the plates or damping and division of the air gap were varied systematically.

Tuesday Poster session 1

’Survey on common Arabic Language forms from a speech recognition point of view’

Mohamed Elmahdy[a], Rainer Gruhn[b], Wolfgang Minker[c] and Slim Abdennadher[a]

[a]*German University in Cairo;* [b]*Harman Becker Automotive Systems;* [c]*University of Ulm, Institute of Information Technology*

Arabic language is the largest still living Semitic language based on the number of speakers which exceeds 250 million first language speakers. Arabic language has two main forms: Standard Arabic and Dialectal Arabic. Standard Arabic includes ”Classical Arabic” and ”Modern Standard Arabic (MSA)” while dialectal Arabic includes all forms of currently spoken Arabic in day life and it vary among countries and deviate from standard Arabic to some extent. While there are many forms of Arabic, there still many common characteristics on the acoustic level and the language level. In this paper we investigate the main differences between Arabic forms from a speech recognition point of view to focus on similar and different acoustic properties among Arabic forms as well as the language properties. For standard Arabic we have chosen both classical and MSA forms and for dialectal Arabic we have chosen Egyptian colloquial Arabic as an example since it is the most popular dialect among Arabic speakers. The main purpose of this paper is to summarize all characteristics of Arabic and previous research results in Arabic speech recognition from a speech recognition point of view in one paper so researchers in this field can use it as one reference.

Tuesday Poster session 1

Silent Owl Flight: Setup for Fly-Over Noise Measurements

Christoph Fritzsche, Ennes Sarradj and Thomas Geyer

BTU Cottbus, Aeroacoustics Group

It is common knowledge that most genera of owls fly silently in order to be able to catch their prey. To investigate the mechanisms leading to that quiet flight, several basic fly-over measurements were carried out by biologists in the past. But the very low noise that is produced when the owls are in gliding flight makes it very hard to do exact acoustic measurements.

The presentation describes an attempt to do fly-over noise measurements of trained birds on natural conditions. It is focused on the measurement setup which consist of a small linear microphone array on a ground plate. The microphone array data is processed using a simple weighted-sum beamforming algorithm. Birds pass over the array in gliding flight at very low speed. A camera setup is used for flight speed measurement. A number of tests were conducted in a wildlife park. Preliminary results are presented from several raptor genera, including owls. Necessary improvements of the measurement setup will be discussed.

Tuesday Poster session 1

An appropriate BGM as a room acoustics: the interaction between BGM and interior design

Yasuhiro Goto
Faculty of Psychology and Appl. Comm. School, Sapporo

Two experiments were performed in order to examine an interaction of room acoustics and the interior design of the room. The room used in these experiments was ”interior-designed room.” In Experiment 1 two types of BGM were prepared and a change of room evaluation was investigated in terms of types of BGM: Used BGM as a room acoustics were ”relaxation music” and ”no-relaxation music.” Participants were asked to rate the degree of harmony between music and room interior design. Result was that a relaxation music was judged more appropriate for room interior design than no-relaxation music. In Experiment 2 the change in the impression and the likes and dislikes concerning this room were examined in terms of BGM by using two types of lightening: relaxation lighting and ordinary lighting. 60 participants were asked to rate the impression and the likes and dislikes concerning this room. The result was that the change in the impression from the BGM and lighting didn’t change the atmosphere formed by the interior design. This study showed that BGM as a room acoustics and lighting can play an additional role in a highly-designed indoor room.

Tuesday Poster session 1

Joint Speaker Identification and Speech Recognition for Speech Controlled Applications in an Automotive Environment

Tobias Herbig and Franz Gerl
Harman Becker Automotive Systems

In recent years speech recognition and speaker identification have increasingly obtained attention for a variety of speech controlled applications. Even though speech and speaker recognition apply the same statistical methods to similar features both have been clearly separated in literature for a long period. Thus an integrated approach for speech and speaker recognition is investigated in this paper. The interaction of speech and speaker recognition enhances recognition rates for speech and for speaker identity. Unlike most existing speaker identification systems this approach requires no explicit training phase for new speakers. Speaker-dependent codebooks for speech and speaker recognition are obtained by unsupervised online speaker adaptation starting from a speaker-independent codebook. During the speech decoding process the identity of the current speaker is estimated and the speaker dependent codebook is further adapted after each identified utterance. Thus the speech controlled system has to handle speaker change detection, speaker identification and speaker adaptation in an unsupervised manner. Furthermore the system accounts for different codebook training levels by providing a smooth transition between short-term and long-term speaker adaptation dependent on the number of speaker-dependent utterances. Results are presented for an in-car application and a limited number of speakers.

Tuesday Poster session 1

Influence of Short Term Noise on Concentration and Human Performance

Marko Horvat, Kristian Jambrosic and Hrvoje Domitrovic
University of Zagreb, Faculty of EE and Computing

It is well known that short-term exposure to noise causes fatigue, insomnia, headache, loss of concentration and the occurrence of negative emotions. In order to examine certain aspects of this problem, a subjective testing has been made with the emphasis on the ability of human beings to maintain concentration and to perform well in a noisy environment. The test itself included the evaluation of the subjects' psychophysical state before and after completing the required task in order to examine the changes in psychophysical state caused by exposure to noise. In the central part of the test the subjects were asked to solve a simple task that did not require any specific knowledge, only a certain amount of concentration. Simultaneously, the subjects were exposed to four types of artificially generated noise. The type of noise varied from group to group and the loudness of the signal varied from one test séance to the next. The results of the tests reveal that the subjects were able to maintain their concentration and a certain level of performance when exposed to short-term noise, although they experienced significant changes in their psychophysical state.

Tuesday Poster session 1

The design of an in-situ absorption measuring system using the Adrienne method

Kristian Jambrosic, Marko Horvat and Mia Suhanek
University of Zagreb, Faculty of EE and Computing

The Adrienne method, standardized in the CEN/TS 1793-5, is a frequently used measurement technique developed for outdoor in-situ measurements of sound reflection and airborne sound insulation. The data given in the standard was used as the basis for the design of a measuring system to be used for determining the absorption coefficient and sound insulation of noise barriers and built-in roofs and wall systems. In this paper the design process is described, specifically, solving of encountered construction problems and system optimization in order to reduce the measurement errors and to eliminate elements that introduce uncertainty. As stipulated in the standard, a fixed source-microphone combination has been achieved, in our case by mounting the microphone holder at the end of a bar kept tightly in position by cables attached to the loudspeaker cabinet. The unit is mounted on a swivel that allows variations of both azimuth and elevation in the range of ±40 degrees from the reference axis in 10-degree steps. The excitation signals used in the measurements included the MLS and the sine sweep. The post processing of the measured impulse responses for various rotation angles and source positions was done in Matlab, developed especially for the Adrienne method.

Tuesday Poster session 1

To the Question of Adaptive Acoustic Systems' Synthesis

Igor Kirichenko and Irina Starchenko
Taganrog Institute of Technology

Design and perfection of adaptive acoustic systems (AAS) appear to be important problems determining solution the level of development of modern acoustic investigation. In natural framework on given restrictions design of optimal AAS with minimal errors on time-variable influences has become necessary. Input signals and influences are regarded as stochastic in time functions. Adaptability conditions of measuring systems with parametric arrays are of special interest. The main concerned quality characteristics of AAS are the next: accuracy, reliability, stability, comfort exploitation, overall dimensions, electrical consumption, etc. Terms of AAS, its general structure and extra structure of measuring system are considered in the article.

Tuesday Poster session 1

Sound Radiation of Double Reed Woodwinds

F. B. Konkel[a], André Jakob[b], Frank Heintze[c] and Michael Möser[a]
[a]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics;* [b]*TU Berlin;* [c]*Staatsoper unter den Linden, Berlin*

In the recent past the verification of radiating properties by woodwinds was measured by using microphones next to the woodwind holes. Hence, it was of particular importance to use suitable microphone positions between the woodwind holes and free space. In the present contribution the radiation properties of the woodwind holes and the bell of a bassoon are investigated by using the "acoustic camera". For the acoustic testing the instrument is observed by three directions. The examination of the results verify the known "cut off" frequency of woodwinds at medium frequencies. From the experimental investigation, two distinct new frequency ranges are found for the radiation properties of the bassoon. The first located range between the medium and the high frequencies of the bassoon modes shows that the central radiation is emitted by the bell, but the last open woodwind holes are also a part of the sound radiation. The second noticed range is located at high frequencies and it becomes clear that the sound radiation is only emitted by the last open woodwind holes. However, there is no sound radiation by the bell at these frequencies.

Tuesday Poster session 1

Comparing Acoustic Model Adaption Methods for Non-native Speech Recognition

Helmut Lang[a], Martin Raab[a], Rainer Gruhn[a] and Wolfgang Minker[b]
[a]*Harman Becker Automotive Systems;* [b]*University of Ulm, Institute of Information Technology*

In order to enable voice control for in-car navigation- and entertainment-systems the necessity of recognizing non-native speech arises. Examples are Germans uttering destinations in France or selecting English audio tracks.
When confronted with non-native speech, recognizers trained on native speakers do not achieve high accuracy on such a task.
The explosion of possible native and foreign language combinations and – as a consequence thereof – the lack of databases covering appropriate non-native speech, make the training of separate models unfeasible.
A common approach for non-native adaption in speech recognition is to utilize a baseline model trained on native speech of target language T and to perform some kind of model reestimation – utilizing a small set of utterances in T spoken by native speakers of the source language S.
Due to the necessary data collection effort, most previous work was limited to evaluate the results of this reestimation on one or two different accents.
With the help of new non-native databases that have recently become available, we can fully focus on evaluating and comparing different reestimation techniques on different language pairs. Our results show how stable different techniques are when they are applied to different language combinations.

Tuesday Poster session 1

Feature Extraction for Speech Recognition

Christian Lüke and Karl Schnell
J.W.Goethe-Universität Frankfurt, Inst. für Angewandte Physik

For speech recognition, the representations of the speech signals are usually sequences of feature vectors. The features are often spectral-based such as MFCCs. If the type of the features is inappropriate, the pattern recognition task is more difficult, especially in the case of varying environments. Therefore, this contribution discusses some different variations of the standard feature extraction methods. This concerns the features themselves as well as a postprocessing of the features such as normalization methods. One focus of the investigations is to use different weightings of stationary and instationary regions of the speech signals.

Tuesday Poster session 1

Direct sound insulation of double-leaf separating walls

Robert Marin, Martin Schneider, Jochen Scheck and Heinz-Martin Fischer
Hochschule für Technik Stuttgart

The simple calculation of the sound insulation of stiff double-leaf separating walls according to DIN 4109: 1989 Beiblatt 1 allows only a rough estimation of the in-situ sound reduction index. Construction parameters like wall material, cavity thickness, absorption material in the cavity and the flanking transmission are only considered by general requirements. An alternative approach is to obtain the in-situ sound insulation from the (undisturbed) direct sound insulation and the situation-dependant flanking sound reduction. The direct sound insulation of the double-leaf wall is obtained from the direct sound insulation of the first leaf and by adding the improvement of a second leaf with respect to the design of the cavity. Measurement results for the direct transmission are compared with results obtained by various prediction models.

Tuesday Poster session 1

The measurement uncertainty and equivalence of automatic unmanned noise monitoring installations

Rein Muchall
geluidconsult bv

Since the last ten years the use of unmanned automatic noise monitoring installations is increasing. Most of them are used to measure the noise of aircrafts, industrial areas, race circuits and pop concerts. Because of its attractive properties: better monitoring at lower costs, it can be expected that in the future more of this kind of installations will be used. An important aspect of this activity is the question: Are the results of the automatic measurements equivalent to the results of manned measurements according the national measurements standards so that they can be used in legal procedures and what is the procedure to validate this equivalence”? There are at this moment no Dutch or international standards how to determine the measurement uncertainty and the equivalence of automatic noise measurements installations. In this article we try to answer this question.

Tuesday Poster session 1

Effect of airborne sound on installation noise - Part 2: Practical application

Sven Öhler, Lutz Weber and Joachim Mohr
Fraunhofer Institut für Bauphysik

The noise excitation of buildings by service equipment takes place by both structure-borne and airborne sound. Mostly sound transmission is determined by the structure-borne portion. For some sources, however, the contribution of airborne sound can't be neglected. A typical example is the EMPA-hammer, defined in the swiss standard SIA 181 as a standardized source for the simulation of user noise. If airborne sound contributes substantially, conventional measures for noise reduction based on elastic isolation between source and building won't work properly. In order to develop efficient measures for that kind of excitation, structure-borne and airborne sound must be separated and investigated in context with each other.
The practical consequences of airborne sound transmission were investigated using water installations (bath tubs, etc.) excited by an EMPA-hammer as example. Since the resulting sound level in the installation room can reach up to 100 dB(A), special measures for noise reduction are required. Apart from the description of appropriate measures as e.g. damping of the vibrating surface by coating, the sound generated by the EMPA-hammer is compared with real user noise. Furthermore the measuring results are discussed with respect to basic investigations using a simplified experimental set-up consisting of two parallel plates.

Tuesday Poster session 1

Nearfield noise source localisation with constant directivity arrays: a comparison

Marie-Agnès Pallas and Régis Perrier
INRETS

The localisation and analysis of noise sources on fixed or moving objects is a preoccupation in many different areas, for instance for passing-by vehicles. Most approaches use microphone arrays, and classical array processing relies on beamforming, possibly adapted to match nearfield conditions and moving sources. However this procedure suffers from inconstant spatial properties with frequency, mainly variable beamwidth of the array spatial response. Several methods have been proposed in the literature to correct this behaviour, using frequency-variable shadings, which may be implemented through FIR filtering of the microphone signals. Specific microphone distribution may sometimes be used jointly. The present poster will present a comparison of the performance of these methods, on the basis of 3dB- beamwidth, frequency uniformity and overall sidelobe levels, with simulations of nearfield moving point sources.

Tuesday Poster session 1

Controllable acoustic bubble traps

Hendrik Söhnholz, Thomas Kurz and Werner Lauterborn
Drittes Physikalisches Institut, Universität Göttingen

Acoustic bubble traps are used to levitate an oscillating bubble at a fixed position near the pressure antinode of the sound field. In this way the bubble dynamics and light emission can be investigated with high resolution. Typically, to obtain high pressure amplitudes, a high-Q resonance of the trap's container is exploited. This makes the trap susceptible to changes of external parameters like the temperature. Furthermore, the back-reaction of the bubble on the container may become large, making it difficult to maintain constant driving conditions when the bubble parameters change. In this paper the construction of bubble traps with medium- to low-Q resonators utilizing flexural disk and sandwich transducers is investigated. One design goal is to be able to control the bubble position to some extent by suitable driving. The transducers are operated at frequencies of about 20 kHz and are designed with the aid of FEM simulations. The sound pressure inside the water-filled prototypes is measured with a hydrophone. The transducer displacement is obtained from measurements with a vibrometer.

Tuesday Poster session 1

Noise control by hedges and woods

Moritz Späh, Lutz Weber and Timo Oesterreicher
Fraunhofer Institut für Bauphysik

Hedges and woods perform important duties in our environment: filtering of air, production of oxygen, acting as fencings, blinds and windbreaks and providing habitat for birds and insects. However, for sound barriers in urban planning they are rarely utilised. This is due to the fact, that the damping of plants is regarded very low in the relevant planning directives, so that a noticeable sound reduction can only be achieved by large-area green space. On the other hand, the damping coefficients given in the directives are only minimal values for not specified situations. In practice under optimized conditions the damping coefficients of hedges can be much larger, so that they can provide an interesting alternative for environmental noise control.

Until now, very little knowledge on the acoustic properties of hedge plants is available. Therefore the basic acoustic properties of hedges and woods are investigated in a research project that comprises measurements and computer simulation. The focus of the project is to gain information on the damping and absorption coefficients of hedges and to provide reliable data on their acoustic properties for urban and landscape planning. First results of the investigation are reported.

Tuesday Poster session 1

Design and Technology Features of Mosaic Pump Transducer of Parametric Array

Irina Starchenko and Igor Kirichenko
Taganrog Institute of Technology

The questions of aperture's filling influence on beam pattern of parametric array are considered. On application of two-channel scheme every signal must be radiated by the separate system of pump transducers. Elements of mosaic or sectioned transducer must be situated in such a manner providing effective interaction of acoustical waves. While antenna is discrete and different parts require excitation on different frequencies, the several cases of aperture filling by discrete transducers were considered and their beam patterns were analyzed. Dependence of beam pattern width from transducer's lines shift was obtained. Recommendations for general antenna construction were formulated.

Tuesday Poster session 1

Climatological and regional analysis of sound level attenuation

Michael Wilsdorf, Astrid Ziemann and Armin Raabe
University of Leipzig, Institute for Meteorology

The properties of sound propagation in the atmosphere primarily depend on the meteorological parameters temperature and wind. Vertical gradients of these quantities result in refraction of sound and have thus an important influence on the sound level attenuation.
By means of the two-dimensional sound ray model SMART (Sound propagation Model of the Atmosphere using Ray-Tracing), the sound level attenuation is calculated, whereas in this calculation the effects of the meteorological conditions on the sound propagation in the atmosphere are respected.
So, the calculation of attenuation enables a fast sound immission forecast for currently measured or modelled situations of sound propagation on the one hand, otherwise it gives the possibility for a classification of an area (e.g. the state area of Germany) in sound climatologically ”likewise” regions (regionalization). The defined regions are considered as statistically ensured sound climatologic units. For the generation of such a sound climatology, it is necessary to consider the special atmospheric conditions at the respective observation points.

Tuesday Poster session 1

Working Group Noise Eurocities

Henk Wolfert
DCMR EPA Rijnmond

EUROCITIES is the network of major European cities. Founded in 1986, the network brings together the local governments of more than 130 large cities in over 30 European countries. EUROCITIES provides a platform for its member cities to share knowledge and ideas, to exchange experiences, to analyse common problems and develop innovative solutions, through a wide range of Forums, Working Groups, Projects, activities and events. EUROCITIES gives cities a voice in Europe, by engaging in dialogue with the European institutions on all aspects of EU legislation, policies and programmes that have an impact on cities and their citizens. Within the EUROCITIES network Working Group Noise (WGN) is established in 2006. The objectives of the WGN are (a) exchange of experiences and knowledge on Noise, Noise effects and Noise Abatement, (b) influencing the European legislation and requirements, (c) execute projects to gain more knowledge about noise and noise effects and last but not least (d) gaining awareness among public, policymakers and politicians on all levels. Currently the WGN is presided by the city of Rotterdam. The poster presentation will raise more understanding of the mission and vision of the WGN.

Paper sessions Wednesday, 25 March 2009

Wed 8:40 Willem Burger Zaal Spatial audio 1

The Perception of System Latency in Dynamic Binaural Synthesis

Alexander Lindau and Stefan Weinzierl
TU Berlin, Fachgebiet Audiokommunikation

In an interactive virtual acoustic environment latency plays an important role for the authenticity of simulations. In dynamic binaural synthesis this is the time until the simulation responds to head movements of the listener. Since several elements contribute to the total system latency such as head tracker update rate, port latency, messaging and scheduling of cross fading, response times are typically distributed around a mean and have to be determined empirically. Whereas just detectable latency values as low as 38 ms have been found when using head related transfer functions, no reliable values exist for the binaural simulation of reverberant acoustic environments. Therefore a new study has been conducted on the just noticeable latency in a system for dynamic binaural synthesis. Data sets were used from measurements of different acoustic environments made with an automated head and torso simulator with free head orientation in azimuth and elevation over its torso. For different room acoustic environments the detection of latency was examined in an adaptive 3AFC listening test procedure.

Wed 9:00 Willem Burger Zaal Spatial audio 1

Timbral Coloration in High Resolution Sound Field Reproduction Due to Spatial Bandwidth Limitation

Jens Ahrens and Sascha Spors
Deutsche Telekom Labs, TU Berlin

Sound field reproduction methods based on orthogonal expansions like e.g. higher order Ambisonics always introduce a limitation of the spatial bandwidth of the loudspeaker driving signals. As a consequence, the desired component of the reproduced wave field is similarly bandlimited. This leads to a pronounced sweet area around the centre of the loudspeaker distribution. This area is sweet both in terms of spatial aliasing artefacts as well as in terms of accuracy of the desired component of the reproduced wave field. The sweet area gets significantly smaller with increasing temporal frequency. This means that outside the centre of the loudspeaker distribution, the sweetest point, frequency dependent artefacts arise. In this paper, we objectively investigate the resulting artefacts for circular loudspeaker arrangements with special attention to timbral coloration.

Wed 9:20 Willem Burger Zaal Spatial audio 1

Quality Evaluation of Ambisonics Recordings

Johann-Markus Batke
Deutsche Thomson OHG

Higher Order Ambisonics (HOA) is a sound field description techniques that is capable to represent a sound field with all three spatial dimensions. The coefficients for such a HOA representation are usually acquired by a spherical microphone array when natural sounds should be captured.
The mechanical size of such an array and the used number of microphone capsules limit the accuracy of the acquired HOA coefficients. Furthermore, the spatial extension of the microphone diaphragm can be taken into account. In this contribution, these influences are discussed with regard to the spatial quality of the sound field representation.
The theoretically calculated representation of a plane wave is compared to the output of a simulated microphone array under ideal conditions. As the major result it should be pointed out that the quality of HOA coefficients acquired by a spherical microphone array shows a strong dependency from the angle of incidence. Variations of the array size, the number of capsules and the microphone capsules' size exhibit possibilities for improvements. The simulations made are confirmed by measurements using a commercially available sound field microphone.

Wed 9:40 Willem Burger Zaal Spatial audio 1

Externalization in data based binaural synthesis: effects of impulse response length

Florian Völk
AG Technische Akustik, MMK, TU München

Databases of head related impulse responses (HRIRs) for binaural synthesis can be measured either in anechoic or reflective environments. The impulse response length ranges from some milliseconds under anechoic conditions to some seconds in reflective environments. The necessary computational load for dynamic binaural synthesis increases significantly with the impulse response length. For that reason short impulse responses would be preferable. If an authentic binaural simulation of an everyday listening situation is desired, an impulse response length in the dimension of the reverberation time would be necessary. A shortening of the impulse responses might lead to perceptual aberrations.
This paper considers the dependence of the perceived distance of auditory images (externalization) from the impulse response length. Impulse response databases of different lengths are used for dynamic binaural synthesis of a frontal virtual source. The distances of the resulting auditory images are measured in a psychoacoustic localization experiment. Statistical analysis of the resulting relative externalization differences suggests that the impulse response length influences the externalization of auditory images.

Wed 10:00 Willem Burger Zaal Spatial audio 1

Sound Synthesis and Spatial Reproduction by Physical Modeling

Rudolf Rabenstein and Alexander Müller
Universität Erlangen-Nürnberg, Multimedia Comm. and Sign. Proc.

Physical models of wave propagation or mechanical vibrations are becoming increasingly important for the design of algorithms for sound reproduction and sound synthesis. For sound reproduction, wave field synthesis has been established as a physically well-founded method for determining the loudspeaker driving signals of massive multi-channel systems. On the other hand, algorithmic models for sound synthesis are derived from detailed physical descriptions of strings, membranes, etc. by the so-called functional transformation method. This contribution combines these methods to form a joint synthesis method of the sound production and sound propagation properties of real or virtual musical instruments. Three different building blocks are addressed: The simulation of string vibrations, the radiation pattern of the generated acoustical waves, and the determination of the driving signals for a multi-channel loudspeaker array. From physical descriptions of string vibrations and sound waves by partial differential equations, follows an algorithmic procedure for synthesis-driven wave field synthesis. The design of filters for sound generation, propagation, and reproduction is presented. The proposed method differs from previous wave field synthesis approaches since no pre-recorded source tracks are required.

Wed 10:40 Willem Burger Zaal Spatial audio 1

Qualitative evaluation of Wave Field Synthesis with expert listeners

Amandine Pras[a], Etienne Corteel[b] and Catherine Guastavino[a]
[a]*McGill University, CIRMMT, Montreal;* [b]*sonic emotion, Oberglatt (CH)*

A qualitative evaluation of Wave Field Synthesis sound quality was conducted with a panel of 5 expert listeners to investigate the perceptual effect of partial decorrelation at high frequencies and of reducing the number of active loudspeakers. The experiment was carried out on a 24 loudspeaker array. In each trial, participants were presented with different versions of the same auditory scene (using 24 or only 8 speakers, with and without partial decorrelation in high frequencies) and asked to freely describe the perceived differences. The verbal data was analyzed to derive semantic scale for sound quality evaluation.

Wed 11:00 Willem Burger Zaal Spatial audio 1

Analysis of Enclosed Sound Fields using Multichannel Impulse Response Measurement

Michael Strauß[a], Johannes Nowak[a] and Diemer de Vries[b]
[a]*Fraunhofer IDMT;* [b]*TU Delft*

When evaluating the performance of spatial audio systems, the knowledge about the sound propagation process inside the listening area gives additional information to assess the influence of room acoustics on the reproduction quality. Wave field analysis based on multichannel impulse response measurement delivers insight to the temporal and spatial properties of a sound field. Closely spaced impulse responses are usually recorded along a line, a circle or a grid resulting in a coherent acoustic data set. This enables the possibility for analyzing the spatial characteristics of a sound field. By applying a Fourier transform or Radon transform it is possible to identify room reflections.
As an example a set of impulse responses was collected at a two dimensional grid inside the listening plane of a small enclosure. Because of the fact that a 2D array recording contains 3D information, it was expected to identify the strongest reflections and to obtain directional information from the data.The analysis methods were validated against a multitrace impulse response set calculated from the simulation of a simple room model. In a second step the functions were applied to real measurement data of a car compartment.

Wed 11:20 Willem Burger Zaal Spatial audio 1

Binaural Sky - Examination of Different Array Topologies

Klaus Laumann[a,b], Günther Theile[a] and Hugo Fastl[b]
[a]*Institut für Rundfunktechnik GmbH;* [b]*AG Technische Akustik, MMK, TU München*

The ”Binaural Sky” system is known as a novel reproduction technology for binaural signals. The essential part can be characterized by the term ”virtual headphone”. Movements of a listener are compensated using dynamic WFS-based rendering of focussed sources. A first prototype was constructed with a circular speaker array mounted above the listener. Details are given in: Menzel, Wittek, Theile, Fastl - ”Binaurale Raumsynthese mittels Wellenfeldsynthese - Realisierung und Evaluierung", DAGA 2006. Several parameters influence the quality of reproduction, one of them being the array topology. For a versatile practical application of the concept it is necessary to quantify this influence. A number of experiments are presented in this paper, examining different array topologies with up to 24 speakers.

Wed 11:40 Willem Burger Zaal Spatial audio 1

Adaptive Adjustment of the ”Sweet Spot” to the Listener’s Position in a Stereophonic Play Back System - Part 1

Sebastian Merchel and Stephan Groth
TU Dresden, Inst. f. Akustik und Sprachkommunikation

The spatial reproduction of sound in a conventional stereo system works in a small area which is located on the symmetry axis between the loudspeakers - the so called ”sweet spot”. Beyond this area, the spatial perception collapses and the stereo image moves to the nearer loudspeaker since the signal arrives both louder and sooner. Finally, the stereo image is completely located in the nearer loudspeaker due to the precedence effect. However, in desktop applications or virtual environments, image and sound have to correlate even though the listener is out of the ”sweet spot”. Although stereophony has a long history, the listener has not yet been released from its static hearing position. Most methods try to broaden the area of stereophonic perception by the adjustment of loudspeaker characteristics. This leads to contradicting localization cues. This paper presents a playback system which adjusts both loudspeaker signals depending on the listener’s position in real-time. In this study, the x-y position of the listener is tracked by a stereo camera and a face recognition system. This information is used to calculate the amount of delay and amplification in the corresponding audio channels. Thereby, a correct localization over the whole off-center listening area is achieved.

Wed 15:00 Willem Burger Zaal Spatial audio 2

Binaural Monitoring of Massive Multichannel Sound Reproduction Systems Using Model-Based Rendering

Matthias Geier, Jens Ahrens and Sascha Spors
Deutsche Telekom Labs, TU Berlin

In model-based spatial sound reproduction a number of input signals is positioned in a virtual space according to a scene description. Using soundfield reproduction methods like Wave Field Synthesis or Ambisonics a scene can be rendered with a loudspeaker array. Suitable loudspeaker arrays typically consist of a huge number of loudspeakers and are often installed in heavily booked venues.

During the creation of an audio scene for playback in such a system it is desirable to listen to a representation of the scene via headphones without needing access to the actual loudspeaker system. This can save production resources because the target system is only needed for final adjustments. Alternatively, the same scene can also be monitored in a studio with much fewer loudspeakers using Vector Base Amplitude Panning.

We present different methods for reproducing a given audio scene via headphones, either by binaural simulation of a loudspeaker-based (possibly virtual) system or by binaural rendering of the original scene. Binaural simulation can be implemented efficiently by combining the loudspeakers' driving functions with Head Related Transfer Functions.

Wed 15:20 Willem Burger Zaal Spatial audio 2

A Sound Reproduction System for Spatial Audio in a Driving Simulator

Dieter Leckschat and Patrick Pogscheba
Fachhochschule Düsseldorf

In the framework of the Driving Simulator at Duesseldorf University of Applied Sciences, the realization of the audio system will be performed in steps. The objective is to reproduce driving noise as well as the sounds of functional components, including their spatial position. In the first step a binaural, loudspeaker-based sound reproduction system was developed using a CTC (cross talk cancellation) system which is robust even without head tracking. The system is completed by subwoofers and an additional shaker. In the presentation, we report on how an optimized positioning of tweeters and woofers facilitates a relatively simple digital filter setup for CTC. Measurement results will be shown, and subjective qualities of the system discussed. Moreover, we will reveal some localization problems and discuss methods to solve them.

In einer ersten Realisierungsstufe des Düsseldorfer Fahrsimulator-Projekts wurde ein binaurales, lautsprecherbasiertes Audio-Wiedergabesystem aufgebaut, das bereits ohne Kopftracking eine robuste Übersprechkompensation beinhaltet. Das System wird durch Subwoofer und Schwingungsgeber ergänzt. Im Beitrag wird berichtet, wie durch eine optimierte Anordnung von Hochton- und Tieftonlautsprechern eine vergleichsweise einfache digitale Filterung zur Übersprechkompensation ausreicht. Neben der messtechnischen Verifikation wird auch über die subjektiven Eigenschaften des Audiosystems referiert. Weiterhin werden räumliche Abbildungsprobleme aufgezeigt und deren Lösungsansätze diskutiert.

Wed 15:40 Willem Burger Zaal Spatial audio 2

Spherical Array Systems - On the Effect of Measurement Errors in Terms of Perceived Auralization Quality

Frank Melchior[a], Kuang Zheng[b] and Diemer de Vries[c]
[a]*TU Delft / Fraunhofer IDMT;* [b]*Fraunhofer IDMT;* [c]*TU Delft*

The analysis of acoustic fields using an open cardioid spherical microphone array enables the extrapolation of the measured field to arbitrary positions inside the reconstruction area of the array. An application of such a system is the generation of virtual stereo main microphone setups, which can be used for auralization purposes. Practical limitations of a array system will lead to several types of errors. In this work the effects of measurements errors on the perceptual characteristics of diffuse fields is investigated. In a listening experiment two channel ideal microphone setups are compared with corresponding virtual sensor setups based on spherical microphone array measurements.

Wed 16:40 Willem Burger Zaal Spatial audio 2

Characterization of Sampling Strategies on the Sphere

Franz Zotter
Institute of Electronic Music and Acoustics, Graz

In discrete spherical loudspeaker and microphone arrays, sampling strategies for the sphere need to be considered. Even more, since spherical arrays have been applied in several sound field analysis and synthesis tasks recently. However, sampling the sphere is not as easy, and hardly ever as simple and efficient, as equi-spaced sampling the circle. In fact, there are only a few equi-spaced sampling layouts, which have narrow limitations in terms of Fourier (spherical harmonics) analysis. As sampling causes aliasing in the Fourier domain, and most applications in acoustics are based on this domain, it has to be payed considerable attention. This paper discusses the properties of several sampling strategies on the sphere found in literature. Essentially, the impact of aliasing on incident and radiating field representations, as well as practical issues (numerical conditioning, sampling efficiency) are briefly discussed.

Wed 17:00 Willem Burger Zaal Spatial audio 2

Approximation of directional characteristics of the human head with a microphone configuration.

Natalia Podlaszewski[a], Volker Mellert[b] and Christopher Haut[a]
[a]*Universität Oldenburg;* [b]*Universität Oldenburg, Institut für Physik*

Classical method of head-related recording uses the dummy head with acoustical properties of the geometry of some average human head. An adaptation to an individual listener in order to improve an ear-related recording is difficult if not impossible and the measurements are sometimes laborious by the weight and the size of the device. In this case a light-weight microphone configuration which is able to approximate different averages and also individual HRTF's by the adjustment of filters and appropriate geometric parameters is a desirable substitution. A method to process plane waves received by a microphone configuration such that the transfer functions approximate the HRTFs of a dummy head was presented earlier [N. Podlaszewski, V.Mellert. Forschritte der Akustik, DAGA 2001, pp. 278/ 279; J.Becker, M.Sapp. Forschritte der Akustik, DAGA 2001, pp.262/263]. The procedure, which allows for modifying the binaural cues according to different head geometries, is now revised and optimised. The algorithm to adjust the microphone configuration for an individual HRTF is presented. Methods to evaluate the quality of an individualised binaural recording are reviewed and discussed.

Wed 17:20 Willem Burger Zaal Spatial audio 2

Adaptive Adjustment of the ”Sweet Spot” to the Listener’s Position in a Stereophonic Play Back System - Part 2

Stephan Groth and Sebastian Merchel
TU Dresden, Inst. f. Akustik und Sprachkommunikation

One major disadvantage of stereophonic play back systems is the narrow ”sweet spot” in which correct audio localization is possible. Due to this limitation, the listener’s freedom of movement is drastically restricted. The present article focuses on a system designed to adaptively adjust the ”sweet spot” to the listener’s position. The primary concern is the analysis of sound localization during the adjustment of the ”sweet spot” via time delay and level differences. Therefore, different models of binaural hearing are used to validate the utility of the system. As a first approximation, an analytical approach by Lipshitz (1986) is used. In a more advanced approach, a binaural model after Braasch (2005) is applied. Both models lead to the same conclusion: They show that an adjustment of stereo signals allows a good localization over the whole off-center listening area. The results of several simulated hearing and play back situations are plotted in quality and vector maps to clearly show the effectiveness of a system with combined time delay and level adjustment. Furthermore, different stereo recording techniques are evaluated regarding their suitability for play back systems with signal adjustment. Thereby, intensity stereo turns out to be most favorable.

Wed 17:40 Willem Burger Zaal Spatial audio 2

Efficient Parametric Audio Coding for Interactive Rendering: The Upcoming ISO/MPEG Standard on Spatial Audio Object Coding (SAOC)

Leonid Terentiev, Cornelia Falch, Oliver Hellmuth and Jürgen Herre
Fraunhofer Institute for Integrated Circuits IIS

Spatial Audio Object Coding (SAOC) is one of the recent standardization activities in the ISO/MPEG audio group. SAOC is a technique for bitrate efficient parametric coding and flexible, user-controllable rendering of multiple audio objects. An extremely high audio compression efficiency is reached due to transmission of only a backward compatible mono (or stereo) downmix signal and a small amount of additional side information describing the perceptual properties of the sound objects. At the SAOC decoder, all sound objects can be rendered interactively to different spatial reproduction set-ups of choice. The new technology extends the MPEG Surround standard in a natural way and inherits its advantages and spatial rendering capabilities. Both bitrate efficiency and rendering interactivity makes SAOC attractive for broad adoption in a variety of interactive applications, for example music 2.0 (personal remixing), next generation teleconferencing and broadcasting. The publication provides an overview of the SAOC reference model architecture and its basic underlying concepts. It describes modes of operation, shows typical

application scenarios and reports the current status of the ISO/MPEG standardization process.

Wed 18:00 Willem Burger Zaal Spatial audio 2

The Effect of Envelope Waveform on Lateralization

Martin Klein-Hennig[a], Mathias Dietz[b], Volker Hohmann[b] and Stephan D. Ewert[b]
[a]*Universität Oldenburg, Medizinische Physik;* [b]*Universität Oldenburg*

Interaural timing disparities are important cues for lateralization in the human auditory system. The auditory system is sensitive to timing disparities in the fine-structure and the envelope of sounds. At high frequencies, only envelope disparities can be exploited due to the lack of phase-locking to the fine-structure. In previously published data [e.g. Bernstein and Trahiotis in "Hearing - from basic research to applications", Springer 2007] parameters like the modulation frequency, or the power of a modulating sinusoid have been altered in order to investigate the role of the envelope waveform on lateralization. However, these "classical" parameters simultaneously change several "secondary" envelope parameters like attack and decay times, as well as pause and hold durations, which might conceal the fundamental cues exploited by the auditory system. In this study, psychophysical measurements were conducted with customized envelope waveforms in order to investigate the effect of isolated secondary parameters on lateralization. For high-frequency tones centred at 4 kHz with systematic envelope modifications, the just noticeable differences of interaural envelope time differences were measured. The results indicate that attack times and pause durations prior to the attack are the most important envelope features. The results are compared to predictions by a binaural auditory model.

Wed 8:40 Jurriaanse Zaal Building acoustics 1

Comparison of artificial and natural rainfall

Klaus Naßhan
Fraunhofer Institut für Bauphysik

There are two important differences between natural and artificial rainfall. While drops in natural rain have reached their terminal velocity, they are still in the phase of acceleration in the rain noise laboratory. The maximum number of drops per unit area and unit time in natural rain is at drop diameters around 2 mm, whereas artificial rain according to IS0 140 [1] consists mainly of 5 mm thick drops. It is shown that an accurate consideration of aerodynamic resistance is essential for properly modeling the motion of free falling drops. In particular, the dependence of the drag coefficient on the Reynolds number, e. g. on diameter and actual velocity, has to be included. The impact energy of a drop on a plate or membrane is calculated according to B. A. T. Petersson [2] from the velocity and the shape of the drop. First estimates show a difference of 15 dB to the results obtained with the assumption of a constant drag

coefficient. This might explain the difference between measured and calculated rain noise levels encountered in previous investigations.

Wed 9:00 Jurriaanse Zaal Building acoustics 1

The Use of Reciprocity in determining Rainfall Noise

Nathalie Geebelen and Hans Cauberg
Cauberg-Huygen Raadgevende Ingenieurs

The theorem of reciprocity exists in many forms and is therefore applied in a wide range of scientific fields. With regard to building acoustics, a relation exists between the sound radiation and the vibration response of a structure. From this, a formula can be derived that connects the air-borne and structure-borne sound insulation of the structure, being only dependent on frequency. This way it is possible to calculate the air-borne sound insulation when the structure-borne sound insulation is known and vice versa, creating lots of practical possibilities. Rainfall noise comes from the mechanical excitation of light structures by rain drops. The measurement procedure for determining rainfall noise is however quiet elaborate. The question rises if reciprocity can be used to simplify this specific set-up? Or else said, when the type of rain (number of drops, intensity, etc.) is known, is it possible for a light structure to predict rainfall noise from the air-borne sound insulation? The use of reciprocity creates many practical advantages, but the relations and formulas have a lot of preconditions. In this paper it is investigated if these preconditions are met in the case of rainfall noise or if they can be overcome be using certain techniques?

Wed 9:20 Jurriaanse Zaal Building acoustics 1

A simplified measurement method for the determination of impact sound reduction

Marc Sommerfeld
Physikalisch-Technische Bundesanstalt, Braunschweig

The reduction of transmitted impact sound by floor coverings on heavyweight floors is usually determined according to ISO 140-8. This procedure requires a test facility with two rooms with a volume of at least 50 m^3 each. The rooms are separated by a standardised concrete slab which is excited by the ISO tapping machine. The sound pressure levels in the receiving room are measured twice, once without and once with the floor covering. From that difference, the impact sound reduction of the covering can be calculated. Manufacturers of floor coverings in Germany - mainly small and medium enterprises - expressed their interest in a lower measurement effort and thus a joint research project of Forschungsinstitut für Leder und Kunststoffbahnen FILK and PTB was initiated to develop a simplified measurement method. It involves measurements of vibration levels on a concrete slab of 0,8 X 1,2 m, once with and once without the floor covering. The vibration level differences are treated like sound pressure level differences, and it turned out that the procedure works well for locally reacting coverings. Besides these results, possible extensions to one of the standardised timber joist floors according to ISO 140-11 and to plate-like coverings like laminate will be discussed.

Wed 9:40 Jurriaanse Zaal Building acoustics 1

Technical aspects in the qualification of free-field environments

Christian Bethke and Volker Wittstock

Physikalisch-Technische Bundesanstalt, Braunschweig

Free-field environments are usually qualified according to annex A of ISO 3745. The sound pressure decrease with an increasing distance from a point source is compared to the ideal free-field behaviour. The tolerated deviations of $\pm$1,0 dB to $\pm$3,0 dB depend on the frequency but also on the type of the room to be qualified. In the last years, PTB has developed and optimised a measurement setup to perform this test. In view of the small tolerances, measurement uncertainties must be small. This requires special technical solutions which will be overviewed in the presentation. One focus is on the different sources and the achieved directivities and stabilities. Other important points are the development of the multi-sine test signals, the automatic microphone traversing system, the background noise handling and the data acquisition and analysis. Finally, some selected results of room qualification measurements will be presented.

Wed 10:00 Jurriaanse Zaal Building acoustics 1

How to improve the accuracy of the absorption measurement in the reverberation chamber?

Martijn Vercammen

Peutz bv, Mook

Sound absorption measurements of building materials such as suspended ceilings and other products are performed in a reverberation chamber according to ISO 354. It is known that the inter laboratory reproducibility of these measurements is not very well. At this moment the differences of results between laboratories are much larger than can be accepted, e.g. from a jurisdictional viewpoint in case of building contracts and liability. Actions should be taken to reduce the spread. An ISO working group has started to investigate possibilities to improve the method. Due to the insufficient diffuse sound field in a reverberation chamber with the test sample, the shape of the reverberation room and the placing of diffusers will influence the result. A round robin research containing 13 laboratories is performed to get information on the spread and if it is possible to reduce this by correcting for the mean free path or by application of a reference material.

Wed 10:20 Jurriaanse Zaal Building acoustics 1

Measurement of sound insulation in laboratory - comparison of different methods

Lutz Weber, Henning Schreier and Klaus-Dieter Brandstetter
Fraunhofer Institut für Bauphysik

For measuring the sound insulation of building elements in laboratory three standardized methods based on different acoustic principles are available. According to ISO 140-3 the element under test is installed between two reverberant rooms. The measurements acc. to ISO 140-5 combine free field excitation with a reverberant receiving room and ISO 15186-1 uses sound intensity for determination of transmission loss. All methods assume idealized acoustic conditions that differ more or less from the situation found in practice. In addition to the inherent statistic variation of the measuring results this gives rise to systematic deviations between the different measuring methods.
Since the deviations are so far only partly understood, they were investigated by means of an extensive series of tests. The tests comprised three different types of building elements (a chipboard, a lightweight double-leaf construction and a membrane partition) and were performed under well-defined conditions in the test facilities of the Fraunhofer-Institute for Building Physics. Apart from comparing the different measuring methods additional investigations on the influence of the most important measuring parameters (reverberation time, number and position of microphones, etc.) are presented. The results of the investigations contribute to a better understanding of accuracy and reliability in measuring sound insulation.

Wed 11:00 Jurriaanse Zaal Building acoustics 1

Measuring Sound Insulation under Extreme Conditions using Deconvolution Techniques

Nicole van Hout[a], Constant Hak[b], Esther Slaat[b] and Eddy Gerretsen[c]
[a]*Level Acoustics, Eindhoven;* [b]*Eindhoven University of Technology;* [c]*TNO*

To compare the sound insulation performance of a building element with the given requirements, measurements have to be done. Generally, a broadband noise source is used, according to international standards. This method does not always work in practice due to high sound insulation values or high background noise levels. From a practical point of view it is very inconvenient or even impossible to perform an accurate sound insulation measurement for all frequency bands. A solution to this problem can be found in deconvolution techniques using MLS or sweep signals. It is possible to increase the signal to noise ratio using these techniques by averaging measurements and spreading out the spectral sound energy in time. As a result an efficient use of available sound power is possible. In a laboratory it can be investigated how to use MLS

or sweep as a source signal and deconvolution as a measurement technique to obtain the sound insulation under noisy conditions. This has been investigated under moderately reverberant conditions, presented at the conference ’Acoustics’08 Paris’, while this paper describes the same investigation under more extreme conditions.

Wed 11:20 Jurriaanse Zaal Building acoustics 1

Room Acoustics, Sound Insulation Design and Supervision for a Multimodal Measuring Laboratory

Axel Roy[a] and Juergen Landgraf[b]

[a]*Akustik Bureau Dresden GmbH;* [b]*TU Dresden, Inst. f. Akustik und Sprachkommunikation*

For analyzing sound design in acoustics measuring labs have become essential. The integration of acoustic equipment in existing laboratory rooms and the build-up of acoustic labs still remain challenging. The IAS as part of the Dresden University of Technology set up a new laboratory for measurements and subjective assessment of multimodal stimuli. The laboratory rooms had to be integrated into an existing faculty building. The demands to the acoustic quality in terms of reverberation time and absence of noise were very high since the intended experiments shall include presentations of acoustic stimuli as well as vibrations. Both have to be connected with optical informations on a projection screen. One issue was to prevent vibrational stimuli from exciting floor and wall structures to avoid unwanted radiation of disturbing airborne noise. For optimum airborne and impact sound insulation a room-in-room construction with solid inner shell was chosen. The platform for vibrational stimuli received a separate foundation isolated from the remaining floor area. The acoustic design will be explained and the various issues of the building process will be reported in the paper. Measuring results obtained in the completed laboratory will be presented proving the excellent acoustic quality.

Wed 15:00 Jurriaanse Zaal Quality classes in buildings

Sound Classification of Dwellings - Comparison of Schemes in Europe

Birgit Rasmussen
Aalborg University - SBi, Danish Build. Research Institute

National sound classification schemes for dwellings exist in nine countries in Europe, and proposals are under preparation in more countries. The schemes specify class criteria concerning several acoustic aspects, the main criteria being about airborne and impact sound insulation between dwellings, facade sound insulation and installation noise. The quality classes reflect different levels of acoustical comfort. The paper presents and compares the sound classification schemes in Europe. The schemes have been implemented and revised gradually since the 1990es. However, due to lack of coordination, there are significant discrepancies. Descriptors, number of quality classes, class intervals, class levels and status of the classification schemes in relation to legal requirements vary. In some countries the building code and the classification standard are incoherent. In other countries, they are strongly ”integrated”, implying that the building code refers to a specific class in a classification standard rather than describing requirements. Although the schemes prove useful on a national basis, the diversity in Europe is an obstacle for exchange of experience and for further development of design tools. The current variety of descriptors and classes also causes trade barriers. Thus, there is a need to harmonize concepts and other characteristics of the schemes.

Wed 15:20 Jurriaanse Zaal Quality classes in buildings

DEGA Sound insulation certificate - a concept for more transparency

Christian Burkhart
Akustikbüro Schwartzenberger und Burkhart

The German society for acoustics has published the DEGA recommendation 103 ”sound insulation in the housebuilding - sound insulation identity card”. The two main targets of the DEGA recommendation 103 are:
(1) creating a multi-level system for the differentiated designing and marking of the structural sound insulation between room-situations regardless of the kind of the building, (2) developing a classification system on this basis for the simple marking of the sound insulation performance of whole housing units or buildings.
The introduction of a multi-level requirement- and assessment-system is meaningful and necessary for a clear differentiation and evaluation of the sound-related quality of buildings. The system is co-ordinated with the today usual building methods and with the today’s build-up-obviously introduced minimum requirements according to DIN 4109. A differentiated and practical classification is possible by the organization into altogether

7 levels both for new buildings and for the old building existence. Unfortunately, the identity values of the sound insulation are very badly clear for planners and users. With the ”DEGA sound-insulation certificate” a simple system is given to mark the sound insulation. The user and the consumer can carry out high-class comparisons without deeper professional knowledge and make purchase decisions.

Wed 15:40 Jurriaanse Zaal Quality classes in buildings

Model-based Assessment Scheme for Acoustic Quality Classes in Buildings

Eddy Gerretsen
TNO

In the past the acoustic quality in buildings was mainly specified by minimum requirements deduced from the performance of existing qualitatively good buildings. Later changes and additions were often only partly based on additional research and social surveys but mainly on ’political’ negotiations. During the last decade the interest has grown in specifying classes of quality, especially for dwellings, preferably for the buildings as a whole. A class could than still be used as minimum - legal - requirement while at the same time a desirable better quality for the future can be clearly indicated and quantified. In order to avoid endless discussions about a decibel less or a decibel more, a model for the subjective assessment of the acoustic climate in buildings has been deduced from the available information. Discussion can than focus on the various aspects of the subjective assessment and their quantification and the final requirements simply follow from the model. This model has been used to specify acoustic quality classes in the Dutch standard for the assessment of sound reduction in buildings, NEN 1070. The presentation will focus on that model and its backgrounds.

Wed 16:00 Jurriaanse Zaal Quality classes in buildings

DEGA Sound Insulation Certificate - Application of DEGA Recommendation with Practical Examples

Frank Schnelle[a], Roland Kurz[b] and Christian Burkhart[c]
[a]*Kurz und Fischer GmbH, Halle;* [b]*Kurz u. Fischer GmbH;* [c]*Akustikbüro Schwartzenberger und Burkhart*

According to the current German jurisdiction newly constructed dwellings with standard comfort usually have to meet higher sound insulation requirements than those demanded in the valid building regulations of DIN 4109: 1989-11. Consumer without acoustical experience faces difficulties in comprehending the acoustic requirements for improved sound insulation, e.g. according to Supplement 2 to DIN 4109 or VDI 4100.

The method of classifying dwelling units into sound insulation classes A* - F according to the DEGA sound insulation certificate can provide the consumer with a comprehensible acoustic rating. The criteria list of the DEGA sound insulation certificate can be used either for the planning of new buildings or the acoustic valuation of existing dwelling units in

houses. The sound insulation certificate can be issued based on results of estimations and measurements.
We present the practical use of the criteria list by showing examples for dwelling units in different buildings (apartment houses, semi-detached and terrace houses). The effects of the individual criteria on the overall rating of the sound insulation classes A* - F will be explained.

Wed 17:00 Jurriaanse Zaal Quality classes in buildings

About Speech Intelligibility depending on different Sound Insulations (part 1)

Thomas Hils[a] and Henning Alphei[b]
[a]*hils consult gmbh;* [b]*Akustikbüro Göttingen*

In the course of the development of the DAGA recommendation ”Schallschutz im Wohungsbau - Schallschutzausweis” (sound protection in residential construction - certificate of noise protection), in a former study theoretical considerations were made on the basis of the calculated loudness of transferring noises that served, among other things, as a basis for the predefinition of the degree levels in the different classes of noise protection.
The recommendation also includes a list of verbal descriptions of the subjective perception of (typical) sounds in housing space and speech from neighbouring living areas. In another list the perception descriptions are linked with the levels of noise protection for giving an orienting evaluation of sound transfer in the seperate classes of noise protection. However, the application for the planning in practice is usually difficult, as these experimental data are only correct under special conditions, which have not been sufficiently investigated so far.
The research in hand starts here. The perception of different situations of sound insulation shall be experimentally investigated under controlled conditions. Thereby, the speech intelligibility is first investigated with sample tests for different sound insulations.

Wed 17:20 Jurriaanse Zaal Quality classes in buildings

About Speech Intelligibility depending on different Sound Insulations (part 2)

Henning Alphei[a] and Thomas Hils[b]
[a]*Akustikbüro Göttingen;* [b]*hils consult gmbh*

In the first part (see previous abstract Hils/Alphei, 17:00), the performed hearing tests were described, and first results were shown. The aim is the verification of the expressions of the list based on the measured intelligibility for as real as possible situations.
In this second part it is tried to create a correlation between intelligibility and sound insulation at a specific initial level in the emitting room.
It is looked at the loudness more closely that, on the other hand, verifies the relation between loudness and the levels of sound protection.

Wed 17:40 Jurriaanse Zaal Quality classes in buildings

Certified sound insulation of different types of building elements

Wim Beentjes
Lichtveld Buis & Partners

In order to develop new building products one needs to know sometimes the sound insulation in situ. Since 1980 special procedures are developed for different types of building products in the Netherlands. The two key-issues of these procedures are :

1. The statistics of predicting the sound insulation of a large series on basis of a random test, to get reliable results. The measurements were made according to ISO 140 part 4 and 7.
2. The contribution of the building element to the total sound insulation must be determined and/or the test situation must be described in order to get comparable results.

The procedures for inner walls, complete building systems and roof elements are presented and discussed. Ground floors and other floor elements and facade-elements are acoustically classified according to Dutch codes of practice and/or measurements acc. to ISO 140 part 3.

Wed 18:00 Jurriaanse Zaal Quality classes in buildings

Determination of the uncertainty of predicted values in building acoustics

Volker Wittstock and Werner Scholl
Physikalisch-Technische Bundesanstalt, Braunschweig

According to the European Construction Products Directive, sound insulation belongs to the six main requirements a building has to fulfil. Usually this is verified by a calculation in which the properties of a building are predicted from the properties of the building products used. Besides the predicted value itself, its uncertainty plays a major role in the planning of a building. A transparent consideration of uncertainties must include the entire chain of effects, from measurements in laboratories via product scatter, to the prediction and the verification by measurement. After comprehensive investigations of the single uncertainty contributions in the past, it was still open as to how these uncertainties affect the uncertainty of the predicted value. This is the topic of the project introduced here. The starting point of the investigations was the compilation of the equations used for the prediction. Continuing from that point, the uncertainty was calculated according to the ”Guide to the expression of uncertainty in measurement”. This was followed by the implementation into a calculation software program (www.ptb.de/en12354). The uncertainty of the weighted sound reduction index turned out to be between 1.5 and 2.5 dB. These uncertainties explain the differences between measured and predicted values very well.

Wed 8:40 Fortis Bank Zaal Room acoustics 1

Acoustical environment in open-space offices - How to achieve the field of confidence?

Sabine A. Fischer
modern-life-design, Königstein

The open-space office has a sensitive acoustical behavior and has therefore to be designed carefully. There is e.g. disturbing noise from adjacent working places. The falling off of noise-level with increasing distance is too small. Rooms are reverberant and noisy. Intelligibility of speech is too high and leads to a lack of concentration. Many sound reflections from walls, from the ceiling, and the furniture arrive at the working place and increase the sound levels. The open space is a sensitive system with many parameters which form an acoustical field of confidence. It must be provided at every working-place. It includes the criterions of privacy defined for different requirements and expectations such as team-workers or single working places. Background noise may be introduced to mask interfering noise from adjacent places. It should be a non-informative noise comparable with pink noise. The paper deals with the different parameters and their application in today's office design. Measurement results will be included, such as reverberation times, sound reflections, decrease of sound levels, and background noise levels. Evaluations will be presented.

Wed 9:00 Fortis Bank Zaal Room acoustics 1

Investigations on room acoustical parameters for open-plan offices

Ulrich Schanda[a], Elmar Schroeder[b] and Susanne Wulff[a]
[a]*FH-Rosenheim;* [b]*Müller-BBM GmbH*

Procedures to plan open-plan offices have to take into account many acoustical aspects like background sound level, room damping, sound insulation and speech transmission. Nowadays human voices tend to be the most annoying sound source in open-plan offices. Hearing tests confirm, that especially the informational content of human speech has a big impact on working efficiency. Planers suffer by a lack of relevant acoustical parameters to treat these many-fold dependencies. The results of hearing tests, carried out with auralized situations of specially selected, acoustically measured open-plan offices will be presented. Working efficiency as a result of the hearing tests will be correlated with measured, objective room acoustical parameters of these open-plan offices.

Wed 9:20 Fortis Bank Zaal Room acoustics 1

Balanced Acoustics in Class Rooms A Question of Sound Field Diffusivtiy and Absorber Dimensions

Andreas Niermann

Knauf AMF GmbH & Co. KG

Reverberation time for the required frequency range in cubic rooms is in generally calculated using the Sabine formula. Assembling the room with different materials with different absorbing properties, including furniture and people, can lead to a satisfactory result. Reality can be different.

Even if the calculating procedure follows standards and well known theories, comparing calculated results with measured values show variations especially at low frequencies. The rather simple calculation of reverberation time does not provide information about local distribution and the diffusing effects of elements such as furniture.

National Standards propose a mixture of absorbent and reflecting zones underneath the massive ceiling. Reflecting zones, executed with dense panels in front of a cavity can support sound transmission and intelligibility by early, powerful reflections in the frequency range of human voice. It can also behave as low frequency absorber.

Actual test results, taken on site, show that low frequency absorbers need a certain extension to affect the absorbance compared to laboratory tests. Class rooms, differently equipped, but with the same total surface dimensions of absorptive/reflective ceiling elements but with different local distribution and extension differ at low frequency reverberation. A number of samples show the effects found on site.

Wed 9:40 Fortis Bank Zaal Room acoustics 1

Room acoustics in schools - Experiences using DIN 18041

Wolfgang Teuber

Institute for Acoustics and Building Physics, Oberursel

For planning, construction or modernization of schools acoustical requirements have to be considered to achieve high standards as summarized in DIN 18041 of 2004. Absorbing materials must be found for practical use under special conditions in schools. The advantage of reduced reverberation times for a better speech intelligibility and lower noise levels have in the meantime been proven through numerous studies. Now we often find discussions about the justification of such standards on one side and much stronger requirements including further reductions of reverberation time and suppressed reflections on the other. Some examples of rooms, materials and test results will be presented.

Wed 10:00 Fortis Bank Zaal Room acoustics 1

Early decay development in inside and outside reverberating spaces

Berit Janssen
Univ. Hamburg, Institut für Systematische Musikwissenschaft

Reverberation is a complex phenomenon, which can be approached from various different physical approaches. Inside and outside spaces alike show distinct reverberant behaviour, which is analyzed and categorized cognitively to distinguish spaces. Studies investigating which aspects of reverberation are cognitively most relevant exist, but mostly restrict themselves to inside spaces. The question whether there is a cognitive continuity in the handling of inside and outside reverberating spaces led to a listening experiment using impulse responses from various spatial settings. In order to achieve a presentation of the perception space, 49 subjects were presented with 36 pairs of impulse responses, stemming from nine inside and outside spaces. They were asked to state the difference between the stimuli, and these judgements were processed by multidimensional scaling. In the resulting perception space, the first dimension was found to be most important, and related to the early decay development, which is very different for inside and outside situations. This implies that an investigation of the early decay process is a rewarding direction to follow, which may indicate new aspects to consider in the evaluation, but also simulation of reverberation.

Wed 10:40 Fortis Bank Zaal Room acoustics 1

Room Acoustic Scale Model Measurements Using a ”Spark Train”

Constant Hak[a] and Kjell Bijsterbosch[b]
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The spark gap, used as a sound source in scale model investigations, has advantages and disadvantages. One of the advantages is its limited size, which is particularly important for investigations at small scales. Also, the spark gap is omnidirectional and can be constructed such that the shape has a negligible influence on the sound field to be measured, for any scale factor. A disadvantage of an excessive discharge is the appearance of a shockwave. This leads to non-linear effects that make it impossible to acquire reliable impulse responses at small distances from the spark gap. A second disadvantage is the spread in successive separate impulse levels as a result of the micro-climate near the spark gap. Experiments were performed with a low energy spark gap (maximum electrode separation of 8 mm). In these experiments the energy was spread over a longer time interval through the use of spark trains. This allows for high INR values and reliable measurements of sound pressure levels. The period stability (jitter) of the spark train and spread in strength of the individual discharges were studied.

Wed 11:00 Fortis Bank Zaal Room acoustics 1

Methods to characterize the acoustic properties of periodic surfaces

Klaudius Hengst[a], Horst Drotleff[b], Roman Wack[b] and Matthias Blau[a]

[a]*FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie;* [b]*Fraunhofer Institut für Bauphysik*

Sound striking a periodic surface is absorbed and scattered depending on the surface properties. In comparison to non-periodic surfaces, periodic surfaces cause a higher sound absorbtion, due to their nearfield: If the nearfield is large, the scattered soundfield will be small and the amount of the absorbed sound high. If the scattered sound field is large, the nearfield is becomming smaller and the periodic surface is less absorbing because more sound energy is reflected. The mechanisms of sound absorption and acoustic scattering by periodic surfaces consisting of bands of alternating impedance are demonstrated first by means of a numerical model. Subsequently, measuring methods are presented to determine the acoustic properties of periodic surfaces at normal, diffuse and directed sound incidence. These measuring methods are assessed with regard to accuracy, informational value and practicability and compared to the theoretically calculated values. It is shown that it is possible (with varying accuracy) to determine the structural absorption at normal, diffuse and directional incidence. The geometrical reflection can be separated from the scattered field at diffuse and directional sound incidence. None of the measuring methods presented allows the separate examination of the near and farfield. The measurements essentially confirm the calculated values.

Wed 11:20 Fortis Bank Zaal Room acoustics 1

Evaluation of the sound field in small fitted enclosures in the modal range

F. B. Konkel and B. A. T. Petersson

TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics

This paper deals with the sound field in small fitted enclosures. In previous work the sound field is divided into three distinguishing ranges. The range at low frequencies is the quasi statistically range, where the system is controlled by the mass or the stiffness of the air. The 2nd established range ist the modal range encompassing the first few well separated modes. Finally, the third range is the multi-resonant range. This paper presents and considers a new calculation model for the modal range of the sound field. Therefore, wave guides are modelled, which vary in cross section and length. The given free volume is filled with these different wave guides. The transfer impedance in the small fitted enclosures between the source and the receiver is solved by the utilization of a Monte Carlo Simulation. In parallel, the acquired calculated results are compared with measured quantities. In engineering applications the exact positions of the objects are not known in the small enclosures at an

early point of design. With this novel approach it is feasible to predict the sound field at the modal range for an early design stage.

Wed 11:40 Fortis Bank Zaal Room acoustics 1

Acoustical Survey of 25 European Concert Halls

Klaus-Hendrik Lorenz-Kierakiewitz[a] and Martijn Vercammen[b]
[a]*Peutz Consult GmbH;* [b]*Peutz bv, Mook*

In order to collect the acoustical data of European concert halls, an acoustical survey of impulse response measurements of European concert hall was performed in the years 2000-2007. The data were analysed to reveal differences, similarities and averages of the common acoustical parameters, but also for the new parameter G5-80. Furthermore, the measurements were correlated with subjective judgments of concert acoustics, stage acoustics and recording acoustics. The correlation results of these different aspects were confirmed: for concert and recording acoustics, no uniform sound ideal exists, but different groups of similarly judging subjects, the size of which is influenced by the chosen stimulus, whereas for podium acoustics a trend can be found highly intelligible and well strength-supported stage acoustics to be preferred by the musicians.

Wed 15:00 Fortis Bank Zaal Room acoustics 2

How to discuss the measurement uncertainties of room acoustical measurements

Ingo Witew and Renzo Vitale
Institute of Technical Acoustics, RWTH Aachen University

The measurement specification for room acoustical measurement, as explained in ISO 3382, has found acceptance of the acoustic community. The discussion of measurement uncertainties of ISO 3382 measurements, however, has not reached that stage yet. At times the discussion may appear unorganised since a straightforward procedure has only recently been developed. Nevertheless the means to determine the measurement uncertainty of room acoustical measurements require efforts that may turn out to be quite comprehensive. In this contribution it is discussed how the work to derive the measurement uncertainty can be reduced by modelling the measurement chain using the tools provided by the ”Guide of the expression of Uncertainty in Measurements” (GUM). The strategy is described using the example of the uncertainty that is introduced to the measurement due to the directivity of the sound source.

Wed 15:20 Fortis Bank Zaal Room acoustics 2

Model for assessing the influence of an omnidirectional source's directivity in room acoustics measurements

Renzo Vitale
Institute of Technical Acoustics, RWTH Aachen University

Omnidirectional sources represent an integral part of acoustic measurement systems as their acceptance, particularly for dodecahedron loudspeakers, has become almost universal. For higher frequencies (above 1 kHz), the radiation pattern of an omnidirectional source loses homogeneity, since its directivity diverges significantly from an ideal spherical shape. As a consequence, the final measurement is affected by inaccuracies that can be expressed in terms of uncertainty.
This paper presents a model developed to quantify the influence of the source directivity on the measurement uncertainty. The propagation of sound in rooms is simulated according to the image sources and radiosity methods. The characteristics of dodecahedron sound sources are implemented in this model with their statistical properties and characteristics. These two core concepts are combined by means of Monte Carlo Simulations, which produces a set of synthesized impulse responses, both of them modelled according to the physical characteristics of a test environment and to the statistical distribution of the measured levels around the loudspeaker. The results are compared with measurement results, thus allowing a quality assessment of the model. In conclusion the input quantities that affect the measurement uncertainty that is introduced by the source directivity are identified.

Wed 15:40 Fortis Bank Zaal Room acoustics 2

A semi-analytical model for rooms with absorptive boundary conditions

Martin Buchschmid[a], Gerhard Müller[a] and Martina Pospiech[b]
[a]*Lehrstuhl für Baumechanik - TU München;* [b]*Lehrstuhl für Numerische Mathematik - TU München*

The prediction of sound-fields in closed volumes with vibrating delimiting surfaces typically is carried out by the use of energy methods. Those methods are robust for subsystems with high modal density. However their performance is limited if the spatial resolution of the response has to be described and if boundary conditions are investigated in detail. In order to gain spatial information about the sound pressure in such a volume, the coupled system of fluid (acoustic volume) and structure (boundary conditions) has to be considered in the frequency range of interest. First the components are calculated decoupled. A modal approach is used for the fluid, where analytical results are available for simple geometries. For an efficient numerical implementation, especially for analyzing more complex geometries, spectral approaches are considered in addition to FEM. The boundary conditions are modeled with the help of impedances. The angle of inclination of the incoming waves is considered

by a wavenumber dependent description. Plate resonators and arbitrary plate-like compound absorbers are treated using the Theory of Porous Media for modeling porous foam structures and the Lamé Equation for homogeneous materials. The coupling of fluid and structure is carried out with the help of the component mode synthesis.

Wed 16:40 Fortis Bank Zaal Room acoustics 2

A Passive Method for the Determination of Acoustical Parameters in Occupied Rooms

Reto Pieren
applied acoustics GmbH, Gelterkinden (CH)

A passive measurement method for the determination of acoustical parameters in occupied rooms is presented. In a defined performance setting the music signal of a natural source is used as the test signal. The method is based on cross-correlation and uses additional information about the setting which can be found using a calibration measurement of the unoccupied space. The new method was first tested within a virtual environment. During subsequent practical testing, the method was applied to the well known concert hall of the "Stadt-Casino" Basel. The resulting acoustical parameters were compared to conventional measurements. It could be shown that the new method is suitable for the determination of acoustical parameters (RT, EDT, C80) in occupied rooms with a solo singer as acoustical source. Measurements with a grand piano as acoustical source showed promising results.

Wed 17:00 Fortis Bank Zaal Room acoustics 2

Investigation of Scattering Coefficient of Everyday Furniture

Renzo Vitale and Dirk Schröder
Institute of Technical Acoustics, RWTH Aachen University

In recent years, the scattering coefficient has been more and more integrated in room acoustics computer simulation as it offers a handy description of the ratio between specular and diffuse reflected energy on surfaces and structures. However, there is still a lack of scattering coefficient data, especially for commonly used furniture, such as desks, bookshelves and chairs.

This paper presents a preliminary study for quantifying the influence of furniture on the sound field reflections in terms of scattering. The research aims to obtain a database of scattering coefficient values for typical furniture, applicable to any room acoustics simulation software which supports scattering coefficients as input data.

In a first stage, furniture of interest are properly designed and built in a scaled-down version. The scattering coefficient is then evaluated through measurements in a scaled reverberation chamber. Afterwards, acoustical measurements in a real furnished room are carried out. Finally, the selected environment is simulated with room acoustic software. Comparisons between real and simulated results are presented.

Wed 17:20 Fortis Bank Zaal Room acoustics 2

Estimation of the Optimum System Delay for Speech Dereverberation by Inverse Filtering

Stefan Goetze[a], Markus Kallinger[b], Alfred Mertins[c] and Karl-Dirk Kammeyer[d]
[a]*Fraunhofer IDMT;* [b]*Universität Oldenburg, Signal Processing Group;* [c]*University of Lübeck;* [d]*University of Bremen*

Equalization of room impulse responses is an straightforward approach for dereverberation of speech signals in a hands-free scenario. The well-known least-squares equalization filter minimizes the euclidian distance between the concatenated system of room impulse response (RIR) and equalizer and a given target system. This target system usually is chosen as a delayed discrete pulse, a delayed band-pass or a delayed high-pass filter. In this contribution we address the choice of the this delay which has to be introduced w.r.t. maximum dereverberation performance. Since designing one equalizer (EQ) for each possible delay and choosing the best one is computationally inefficient we investigate the dependence of the optimum EQ delay on various parameters describing RIRs. A high correlation was found between the so-called central time of the room impulse response and the optimum EQ delay. Since the central time can be determined based on estimates of the initial peak of the RIR and the room reverberation time, we propose to use a very short filter for system identification and an estimate of the room reverberation time to identify the optimum equalizer delay.

Wed 17:40 Fortis Bank Zaal Room acoustics 2

The renovation of the Doelen Concert Hall

Martijn Vercammen[a] and Margriet Lautenbach[b]
[a]*Peutz bv, Mook;* [b]*Peutz bv, Zoetermeer*

In the venue of the conference is the Doelen concert Hall located. The hall is well known for its clear / transparent acoustic. The hall is built in 1967, and a renovation will start shortly after this congress. The renovation includes (a.o.) the integration of theatre light, replacement of the ceiling and new seats. In this renovation process the acoustic has to be monitored and it is aimed to improve some acoustic properties, such as the ensemble conditions on stage and enhancement of the low frequency reverberance. This presentation gives a short history of the hall and an overview of the results of the investigations executed in the design process. In the design process an investigation is made of the stage acoustic by measurements and questionnaires in 3 concert halls, a scale model 1:10 is built and a computer model is used. To improve stage acoustic a canopy is (re-) introduced. In the presentation the measurement and calculation results will be shown.

Wed 8:40 Van Cappellen Zaal Noise at workplace

The Calculation of Sound Propagation in Rooms to determine Noise Exposure at Workplaces

Wolfgang Probst
DataKustik GmbH

Based on measurements of sound propagation in about 150 industrial workrooms, the calculation method based on the work of Jovicic and Kuttruff has been standardized in VDI 3760 as the general method to determine sound pressure levels at workplaces from the emission values of machines and other sources. The core of the method is to calculate the sound pressure levels at receivers distributed on a path that are caused by a single point source and then to apply this special sound decay curve to determine the contribution of all sources at a given receiver position for each relevant frequency band. The method is based on the uniformity of sound propagation in all directions, uses a mean absorption of each wall and neglects screening effects by partitions and even large objects. In the frame of a research project financed by BAuA, methods have been developed to overcome these restrictions. Some typical applications like the acoustic optimization of a large office and the determination of noise exposure at workplaces of machines based on the emission values declared by the manufacturer according to ISO 4871 are demonstrated.

Wed 9:00 Van Cappellen Zaal Noise at workplace

The new electronic report of noise-exposure-measurements and audiometric-examinations in the AUVA

Wilhelm Wahler, Karl Körpert and Ahmed Gaafar
Allgemeine Unfallversicherungsanstalt, Wien

The AUVA is the Austrian Social Insurance for occupational risks for approximately 3 million employed persons and 1.3 million school children and students. Therefore, a central motivation of the AUVA is the prevention of occupational accidents and diseases, e.g. the noise-induced hearing loss. One major element of the prevention activities are the visits of prevention workers in companies to improve the employees' conditions of hearing by an audiometric examination. To pick out the employees which are heavily exposed, it is necessary to determine the noise exposure level at work places by measurement and calculation. The data entry of the detailed noise-exposure and audiometric studies can be now electronically supported by a new computer application called EPOS. The objective of this paper is to report on the challenges and experiences gained during a five year multidisciplinary software development project and the essential components of the software will be presented.

Wed 9:20 Van Cappellen Zaal Noise at workplace

Web-based Toolbox Noise and Provisions in Metal Working Industry

Jeroen van Hees and Jan Granneman
Peutz bv, Zoetermeer

Employees in metal working industry are often exposed to high noise levels. European noise limits regarding occupational noise are related to the daily dose of workers. In this type of industry the daily dose is hard to obtain by measurements due to great variety in activities and equipment. A specific web-based toolbox has been developed to enable firms to determine the daily dose of their workers. Based on an estimate of activities and their duration this toolbox derives the daily dose and highlights the most dominant sound sources and activities. In the toolbox also the effect of room acoustics and contributions due to activities of co-workers are taken into account. If noise limits are exceeded, the toolbox provides specific measures for each activity to reduce the noise levels, giving detailed information about the amount of reduction of the daily dose to be expected. The web-based aspect of the toolbox makes its content easy accessible and up-to-date. The toolbox is part of a program issued by the government and branch organisations that also deals with other occupational aspects. By simple means the toolbox can also be used for other branches dealing with occupational noise problems.

Wed 9:40 Van Cappellen Zaal Noise at workplace

Relevance of the Peak Sound Pressure Level at Industrial Work Places

Jürgen H. Maue
BGIA - Inst. for Occup. Safety, Sankt Augustin

The risk assessment according to the European Directive 2003/10/EC on occupational noise requires the determination of the noise exposure level LEX,8h and the C-weighted peak sound pressure level LCpeak, if relevant. The noise exposure level describes the long-term noise exposure including exposure to impulsive noise and intermittent noise, whereas the peak sound pressure level is a measure for single noise events which may occur only few times each day.
The measurement of the C-weighted peak sound pressure level is explained and the relation to the sound pressure level with the time-weighting ”F” (fast) and ”I” (impulse) is examined for several examples of noise impulses in industry. It is shown, that the action values of 135 dB(C) and 137 dB(C) may occur at rather few industrial work places. But if there are extremely high noise events, just a few of them may sum up to daily noise exposure levels of 80 dB(A) and more. The measuring results are listed in a table and will assist in the decision, if the peak level may be relevant and shall be determined for the risk assessment at a work place.

Wed 10:00 Van Cappellen Zaal Noise at workplace

Problems of the real world sound attenuation of hearing protectors with respect to the peak sound pressure level

Peter Sickert
BG Metall Nord Süd

With the release of the European directive 2003/10/EC on noise at the workplace the relevance of high peak sound pressure levels at the workplace (and protection against) has been strengthened. Also for peak sound pressure levels two action values exist (lower and upper): 135 and 140 dB(C).
Additionally, the noise directive defines an exposure peak limit value. Under no circumstances shall the exposure of the workers exceed the exposure limit value. Hereby, the attenuation of hearing protectors has to be taken into account. The problem is to select an appropriate hearing protector that complies with both exposure limit values: daily noise exposure limit value (87 dB(A)) and peak exposure limit value (137 dB(C)). In general, it is difficult to determine the actual sound pressure level individually at the ear of the user in the practical use of hearing protectors. The problem is still much more complicated in the case of peak sound pressure levels.
This presentation will discuss the assessment of hearing protectors with regard to peak levels. Possibilities to check the noise at the work place concerning peak action and peak limit values will be covered.

Wed 10:20 Van Cappellen Zaal Noise at workplace

Methods for the determination of individual sound attenuation of ear plugs

Sandra Dantscher
BGIA - Inst. for Occup. Safety

The European directive 2003/10/EC on noise at the workplace defines exposure limit values that may not be exceeded at the ear of the worker. Thereby, also the attenuation of hearing protectors has to be taken into account. For checking the compliance with this requirement usually the attenuation data specified by the manufacturer is used. It is based on the measurement for the EC type examination with a group of test subjects. But the individual values of attenuation may differ from the group average. To cope with this problem some manufacturers of hearing protectors and especially custom moulded ear plugs have developed measurement systems for the determination of individual sound attenuation. Different techniques are used: objective measurements without active participation of the wearer of the hearing protector (e.g. with two microphones inside and outside of the plug) or subjective methods (e.g. hearing thresholds with and without hearing protector). This presentation gives an overview of the different measurement systems on the market. Moreover the calculation (and correction) methods will be discussed that are necessary to determine an individual value of sound attenuation that is comparable to the laboratory data.

Wed 11:00 Van Cappellen Zaal Noise at workplace

Earprotection Project Military Music

Wilhelm Wahler[a], Marion Logar-Holzer[b], Michael Emich[b], Christof Guber[b], Andrea Graf-Langheinz[b] and Susanne Spendelhofer[b]
[a]*Allgemeine Unfallversicherungsanstalt, Wien;* [b]*Occupational health Center VIE / Army Hospital, Vienna*

The past experiences in the Austrian military music band with the adaption of a customized earprotectors have so far been unsatisfactory. Different aspects felt negatively by the musicians (subjectively) lay particularly in the range between the false interpretation of the sound image and/or the difficulty in the assessment of the dynamics, volume and Intonation of the sounds/tones.This observation study was conducted in three representative Austrian Military Music Bands. The results were gained out of a total of 146 persons by means of a standardised questionnaire in various subjects with regard to personal handling and the use, deployment and acceptance of the personal protection of the ears.The daily length of music performance was documented during the period of 8 weeks. Parallelly volume decibel measurements in different practical-relevant sound situations of the musicians (orchestra, single exercises,...) were gathered.

Wed 11:20 Van Cappellen Zaal Noise at workplace

Safe and Sound - Exposure Control for Workers in Music and Entertainment

Georg Brockt
Bundesanstalt für Arb.-schutz und Arb.-medizin, Dortmund

The EU-Directive 2003/10/EC on occupational noise refers to all workers expressly including those in music and entertainment industry. In this sector sound is intended and indispensable but may be harmful at the same time. Nevertheless, the fundamental principles of noise control e.g. the general obligation for noise reduction at source or the priority of collective protection measures over individual protection measures are implemented in the directive. Where noise exposure exceeds action levels further measures have to be applied: use of hearing protection, health surveillance, marking of noisy work places and the implementation of noise reduction programmes. Moreover a limit of 87 dB(A) for the noise exposure level, taking into account the attenuation of hearing protection, shall be complied. The general way of noise control is noise reduction at source, on the transmission path, by organizational measures and the application of hearing protection. But this procedure appears to be a challenge in music and entertainment. This contribution covers the exposure in different sectors of the entertainment industry like orchestras, clubs or discotheques. Principles and examples for exposure limitation are described. Obviously, there exists no general solution, but

only the combination of several individually chosen measures can yield a practicable exposure control.

Wed 11:40 Van Cappellen Zaal Noise at workplace

Evaluation Of Acoustical Suitability Of Lecture Rooms With Respect To The Quality Of The Sound Reinforcement System And The Level Of Background Noise

Hrvoje Domitrovic, Marko Horvat and Sanja Grubeša
University of Zagreb, Faculty of EE and Computing

In order to evaluate acoustical suitability of several lecture rooms located on the Faculty of EE and Computing in Zagreb, Croatia, a questionnaire was given to the students who take their classes in those particular rooms. The first part of the investigation is focused on the quality of sound reinforcement system located in two largest lecture halls, to be replaced in the near future with a new one of significantly better quality. The goal is to examine whether this change is reflected in the grades of acoustical suitability given by the students. The second part of the examination deals with two smaller lecture rooms of identical size and acoustic treatment, but with significantly different levels of background noise due to positions of these rooms relative to the nearby street as the source of traffic noise. The overall level and spectral content of noise are measured in these lecture rooms. In addition, speech intelligibility is examined in all four rooms. The results of these measurements are then interpreted with respect to the subjective grades given by the students.

Wed 15:00 Van Cappellen Zaal Noise control 1

A View on Geometry of Noise Barrier Edges

Judith Kokavecz and Michael Möser
TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics

Noise barriers are a widely spread noise prevention measure. Although a lot of studies spend time on reseach about edge design, noise barriers are still simple single wall systems in most cases. Because replacement is not on the list of urban and regional planing, reconstruction might be a cost saving alternative. By using attachments on the edge of existing barriers, the acoustical behaviour mainly caused by diffraction could be improved.

The studied attachments are simple flaps. In the computer simulations different open angles for a single flap were tested. To improve the insertion loss a second flap was integrated into the model. The variations are numerous with two flaps depending on the open angles, the length of the flaps and the location of the source. First results of these simulations for different parameter combinations will be shown.

Wed 15:20 Van Cappellen Zaal Noise control 1

In-situ measurements of road barriers made of natural stones

Andre Isele[a], Gerrit Höfker[a] and Christian Nocke[b]
[a]*Hochschule Bochum;* [b]*Akustikbüro Oldenburg*

With an increasing disturbance of traffic noise the length of barriers along highways and railways grows year by year. Besides the well-known materials like concrete, metal, glass or synthetic stones also natural stones are more and more in use because of its appearence.
A minimum value for sound absorption coefficient and transmission loss is necessary for the acoustic efficiency of the barrier. Measurements of the acoustic properties of gabions in the reverberation chamber and laboratory are often impossible because of the construction or the weight respectively. To avoid these problems the reflection index RI and the sound insulation index SI were determined by in-situ measurements according to DIN EN ISO 1793, part 5. Results of the reflection index for several materials, grain sizes and layer thicknesses of the charge of the wire mesh are presented.
Earlier research have shown that a minimum value of the transmission loss could be reached with additional elements inside the gabion only. Different constructions of concrete with and without slits between the elements have been tested. A mathematical description of the problem is given, which shows the essential parameters of sound insulation and allows approaches to optimisation.

Wed 15:40 Van Cappellen Zaal Noise control 1

Noise Reducing Devices: an Austrian Experience with the new European Standard for Measurements of in-Situ Sound Diffraction

Marco Conter and Manfred Haider
arsenal research, Vienna

Traffic noise is one of the major environmental concerns within the European Union. The most important solution to noise abatement along roads and railways is the construction of noise barriers. In order to measure the acoustic performance in real life conditions and the long-term performance of these devices, the working group CEN/ TC226/ WG6 has developed the so called Adrienne method. This measurement method was introduced in 2003 with the technical specifications CEN/TS 1793-5, dealing with in-situ sound absorption and sound insulation, and CEN/TS 1793-4, dealing with in-situ sound diffraction. Added devices for sound diffraction are becoming more relevant in the last years to increase the acoustic performance of barriers. This paper describes the first Austrian research dealing with measurements of in-situ sound diffraction.This study, carried out in 2005 by arsenal research, compares the acoustic behaviour of aluminium cassettes, alternately equipped with reference barriers, partially non flat products, added devices on the top of the barrier, transparent insertions and digital and digital prints on the surface. In detail, the authors have measured and analysed the in-situ

acoustic properties sound absorbtion, sound insulation and sound diffraction of each variant. The research has been carried out for the company Forster Metallbau GmbH.

Wed 16:00 Van Cappellen Zaal Noise control 1

Concepts of Global Noise Control in Cabins

Julian Greßkowski and Delf Sachau
Helmut-Schmidt-University Hamburg, Inst. Mechatronics

Nowadays people have to be protected for high sound pressure levels at low frequencies. An example is a propeller driven aircraft, where the propellers induce periodic noise inside the cabin. Therefore active noise reduction systems are applied to reduce these unwanted disturbances. To reach an acceptable noise level in the whole cabin actuators and sensors have to be allocated global in the distributed area. Also large numbers of actuators and sensors are required for attenuation of unwanted sound in a large volume with a high density of acoustic modes. This results in a high computational load for the controller. In order to reduce the number of components and consequently the effort of the controller unwanted sound transmission in cabin can be reduced in the propeller plane by controlling the radiated sound power. This innovative concept is investigated using an A380 structure as primary source which radiated noise due to excitation. The control sources are allocated in front of the panel. Also the microphones were placed in the vicinity of the structure. The possibility and first experimental results is shown in this research.

Wed 17:00 Van Cappellen Zaal Noise control 1

Investigation of the Sound Transmission of a Partial Discharge through a Solid Dielectric Multilayered Device

Uwe Buchholz[a] and B. A. T. Petersson[b]
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Acoustic emission (AE) sources caused by partial discharges in solid dielectric devices like transition joints or outdoor terminations are investigated. Together with a prepared high voltage cable end this device forms a multilayer structure. -The hypothesis is that due to the high damping of polymeric insulation materials no AE signal can be detected on the surface of the prescribed structures. -An analytical technique is used to determine the transmission for a typical multilayer insulation. For quantitative computations it is necessary to know the material properties such as complex Young's modulus, density and the Poisson's ratio. Since the properties found in the literature were not fully satisfactory, measurements had to be done. The experimental setup and the results are shortly presented. -Finally it will be shown that the transmission path plays not the significant role as was assumed in the beginning.

Wed 17:20 Van Cappellen Zaal Noise control 1

Whistling Building Objects, Origins and Solutions

Jan Granneman

Peutz bv, Zoetermeer

Widespread complaints often occur due to whistling sounds caused by building objects, such as (high rise) buildings and viaducts, under specific meteorological circumstances, especially wind speed and direction. In many cases this is related to the application of specific grating. Much is already known about the relevant parameters for this phenomenon from measurements in practice and in laboratories. To obtain more understanding into this subject and to be able to prognosticate the possible occurrence and intensity of it in the design phase of building objects, systematic research has been done under laboratory conditions. The sound power level and frequency characteristics of whistling sounds dependent of wind speed, angle of incidence and grid configurations were studied. Also the sound reducing effect of different and practically feasible modifications of gratings was part of the study. The paper presents the results of this study.

Wed 17:40 Van Cappellen Zaal Noise control 1

Traffic noise annoyance on road and rail (TNAR) in an experimental laboratory setup

Michael Cik

Graz University of Technology

At present, according to the current standards and specifications, the impact of road and rail traffic noise is represented by the A-weighted energy-equivalent sound level (LA,eq). This quantity does not account for the subjective annoyance of sound events as it is perceived by the affected persons.

Pass-by road and rail vehicle noise samples were binaurally recorded under predefined conditions with a dummy head measurement system and synthetically composed to defined vehicle ensembles. A total of 200 persons were selected and exposed to these defined vehicle ensembles in different sessions in a hearing laboratory. Annoyance was rated on an 11-graded interval scale and by a hand power-dynamometer which measures the intensity of the applied strength. Psychoacoustic parameters of the defined vehicle ensembles were calculated. Annoyance ratings were combined with psychoacoustic parameters by multiple regression analysis. The relevant parameters were statistically combined into the index TNAR.

The analyses revealed that loudness is the parameter describing annoyance best, significantly better than the LA,eq ($P<0.0001$). Further significant influencing parameters are sharpness and a new designed parameter representing the frequency characteristic of traffic noise. These three parameters together (R^2=0.807), correlate significantly better with the subjective estimation of noise-induced annoyance than the LA,eq (R^2=0.672) does ($P<0.0001$).

Wed 8:40 Van Beuningen Zaal Sound quality and soundscapes

Psychoacoustic Evaluation of Traffic Noise

André Fiebig[a], Sandro Guidati[a] and Alexander Goehrke[b]
[a]*HEAD acoustics GmbH;* [b]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics*

In the European research project ”Quiet City Transport” the quantitative characterization of annoyance caused by traffic noise using psychoacoustic descriptors for a perception-related environmental noise description was a major objective. In the first project stage single pass-by noises have been evaluated by subjects. The aim was to discover components and patterns, which are responsible for specific annoyance effects. The results showed that certain noise properties such as diesel knocking are particularly annoying regardless of their absolute sound pressure levels. In the second part of the project complete traffic noise scenarios were evaluated to detect indicators which allow for an adequate assessment of environmental noise quality. Using a traffic noise synthesizer as well as artificial head recordings a wide variety of stimuli were created and evaluated by subjects. By means of a principal component analysis links between specific noise properties and subjective evaluation patterns were found and provided the basis for the development of a metric. The results of the study dealing with the evaluation of traffic noise will be presented and discussed under the perspective of their applicability in environmental noise policy and for soundscape studies.

Wed 9:00 Van Beuningen Zaal Sound quality and soundscapes

Acoustical Categorization of Urban Public Places by Clustering Method

Monika Rychtarikova and Gerrit Vermeir
KU Leuven, Afdeling Akoestiek en Thermische Fysica

Development of urban public places in sustainable way requires multidisciplinary approach where acoustic is only one of many aspects taken into account. To be able to make the proposal for acoustical improvement in the city in accordance with a long-term perspective, deeper understanding of correlations between urban soundscape parameters and other non-acoustical disciplines is necessary. This paper shows our first attempt to cluster streets with similar acoustical properties, based on several acoustical parameters such as statistical noise values, psychoacoustical parameters and some newly developed variables, obtained from binaural recording 10-15 minutes long. 27 acoustical parameters are calculated for each of the 91 recordings and clusters are created by using the hierarchical agglomerative clustering method.

Wed 9:20 Van Beuningen Zaal Sound quality and soundscapes

Soundscape characterization in the city of Bilbao

Gianluca Memoli[a], Igone Garcia[b] and Itziar Aspuru[b]
[a]*Memolix Environmental Consultants;* [b]*LABEIN Tecnalia*

The first round of noise mapping and action plans has left a lot of unresolved questions, mainly addressing the design of perception-oriented action plans and the classification/preservation of quiet areas. While the interest in the practical use of the soundscape concept to address this problems is now growing all through Europe, psychoacoustic indicators linked to perceived noise are the key to determine the path to follow. This work presents the soundscape characterization of two different green areas in the Municipality of Bilbao (E), with in common the presence of a heavily trafficked road nearby. The acoustical experience of passers-by will be mapped using an indicator related to the time history of sound energy, related in previous studies to people's perceptions. Comparison with the same description performed by classical psychoacoustic analysis and perspectives for innovative, positive soundscape based actions and measurement techniques will be discussed.

Wed 9:40 Van Beuningen Zaal Sound quality and soundscapes

Research database for everyday listening

Dirkjan Krijnders, Maarten van Grootel and Tjeerd Andringa
University of Groningen

This paper describes the design process and content of a publicly available database for everyday listening. "While databases for musical sounds and studio recordings are readily available, databases for everyday sounds are lacking. Such a database is essential as more and more research is aimed at environmental sounds and everyday listening. The current lack of a standardized collection of properly recorded and annotated sounds in realistic acoustic environments limits the comparability of results between research groups, and hampers interdisciplinary cooperation. The collection described in this study must to help solve these issues. The collection currently consists of approximately 120 fragments of 60-second recordings of real-life situations. The use of head-mounted binaural microphones ensures a realistic capture of the environment for headphone playback. The recordings differ in location (e.g. indoor, rural, urban) and activity (e.g. taking the bus, dish washing, typing). Detailed annotations, provided for all recordings, describe the environment of the recording as well as the sources and onset-offset information of each of the sounds present. We aim to expand and improve the collection with more recordings and more specific information. We hope that many researchers in the auditory field will benefit from this database.

Wed 10:00 Van Beuningen Zaal Sound quality and soundscapes

Speech Communication in Outdoor Soundscapes

Bill Davies[a], Peter Mahnken[a] and Chris Plack[b]
[a]*University of Salford;* [b]*University of Manchester, Human Communication & Deafness*

The positive soundscape project is a large multi-disciplinary researching the perception of soundscapes. Recent findings indicate that speech communication is a principal factor in users' perceptions of urban soundscapes. The project has therefore explored how this factor might be quantified. This paper reports on an attempt to use speech intelligibility index (SII) to characterise time-varying speech intelligibility in real outdoor soundscapes. The relationships between SII, subjective intelligibility and subjective quality will be explored, as functions of signal-to-noise ratio. Possible applications in sound quality mapping of soundscapes will be discussed. Potential implications for rational planning of soundscapes will be outlined.

Wed 10:40 Van Beuningen Zaal Sound quality and soundscapes

Vertical Profile of Community Noise in High-rise Built Environment

Mahbub Alam Sheikh[a], Siew Eang Lee[a] and Johnny Liang Heng Wong[b]
[a]*Department of Building, National University of Singapore;* [b]*Housing and Development Board, National Univ. Singapore*

High-rise public housing apartments in Singapore are often subjected to different types of "Community Noise Sources" in close proximity. Community noise in housing estates comprises noise sources such as "Food centre noise", "Children playground noise", "Soccer court noise", "Basketball court noise", "Waste disposal truck noise" etc. It was found as one of the major noise source categories in residential environment. With the enhanced performance quality in many aspects of public housing in recent years, it is imperative that the acoustics performance of the apartments can be enhanced to match overall quality of public housing environment through appropriate characterization of these noise sources. A scientific and reliable approach for characterizing noise propagation from these sources is vital in the implementation of projects.
Software modeling and simulation were carried out to study noise propagation characteristics of five types of community noise sources. Five sixteen-storey residential buildings were investigated with noise measurement from these sources, conducted on each level of the buildings. The predicted results were validated with field measured data. This study observes that noise level from these sources drops between 3.7 dBA and 12.8 dBA at a height between 43m and 46.5m above the ground with the increase in building height.

Wed 11:00 Van Beuningen Zaal Sound quality and soundscapes

Is there a necessity for noise caused by motor bikes?

Uwe Ritterstaedt
Ingenieurbüro für Schallschutz, Neuss

It is obvious that in free flowing traffic two types of vehicles are specially noticeable as to their loudness: i.e. trucks and motor bikes, respectively scooters. Much effort has been spent in the last decades an how to reduce truck noise, but bikes remain as loud as before: the passing by of a bike is sometimes ten decibels louder than that of a motor car. On the other hand not all bikes are equally loud and noisy; sometimes one may even find bikes which are not Iouder than motor cars: yet it seems to be possible o construct even less noisy bikes.
In recreation areas, in residentlal areas where citizens and inhabitants require silence, motor bikes can be heard over kilometres when passing by and that as the only technical noise which definitely is strongly disturbing and can be heard in a natural soundscape.
The following article intends to describe both the problems and the possible solutions as given by technical innovations and by administrative means in a European context.

Wed 11:20 Van Beuningen Zaal Sound quality and soundscapes

Footprint of a vehicle - the Swiss Contribution

Irène Schlachter
Federal Office for the Environment, Bern

Footprint is a European collaborative research project within the pan-European EUREKA framework for development and exploitation of technologies crucial tho global competitiveness and better life quality. The aim of the project is to relate the environmental footprint of a heavy road or rail vehicle to the lifetime cost of maintaining the infrastructure and environment. As a project partner Switzerland installed in 2005 the first Footprint Measuring station in Europe for road traffic on the North-South-transit motorway A1. Objectives include developing techniques to measure and characterise impacts including noise. Methods of analysing the data have been established to develop strategies to minimise these impacts.
For noise, methods to determine the acoustical footprint of a vehicle passing have been developed in Switzerland by evaluation of the in situ measured maximum sound pressure level. The impact of various vehicle types and traffic flows can also be recorded. Successfully evaluated pass-by events reach 95% for dense traffic. The Greening Transport Package announced 8 July 2008 promotes measures to reduce the environmental impact of transport, including reallocating external costs from society to users. A further stage of Footprint is to monitor the effectiveness of noise reduction strategies and to cost an environmental index for different vehicle types.

Wed 11:40 Van Beuningen Zaal Sound quality and soundscapes

Background noise: an increasing environmental problem

Jan Jabben[a], Dick Bergmans[b], Eric Schreurs[a] and Tom Koeman[a]
[a]*National Institute of Public Health and Environment (RIVM);* [b]*National Aerospace Laboratory NLR*

Noise quality in most studies is related to the average noise level (Lden) only, although this indicator merely expresses the average intensity of the soundscape. A complementary feature of increasing importance is the background noise level, which is the noise level that is exceeded during 95 % of the observation time. Inhabitants of areas with a high background noise level may be more vulnerable to annoyance or health effects, as recuperation times are narrowed. Due to strong growth of road, railway and air traffic, background noise levels have steadily increased in the Netherlands. In order to assess the impact of environmental noise levels from transport sources, RIVM in cooperation with the Dutch National Aerospace Laboratory have started a research project in 2007 with the aim of setting up background noise maps for road-, railway- and air traffic. This paper discusses the mapping and characteristic behavior of background noise. The implications of the background noise levels on health and annoyance will be discussed. The results indicate that a further increase of background noise levels in the near future is to be expected that may affect the Dutch population. Hence an intensified reduction policy of noise emission is most urgent.

Wed 15:00 Van Beuningen Zaal Low frequency noise

An acoustic vector based approach to locate low frequency noise sources in 3D

Hans-Elias de Bree[a] and Carel Ostendorf[b]
[a]*Microflown Technologies;* [b]*Cauberg-Huygen Raadgevende Ingenieurs*

Although low frequency noise is an issue of huge societal importance, traditional acoustic testing methods have limitations in finding the low frequency source. It is hard to determine the direction of the noise using traditional microphones.

Three dimensional sound probes capturing the particle velocity vector and sound pressure offer a novel possibility to locate the low frequency noise sources in 3D. An overview on the benefits of acoustic vector sensors will be given and a low frequency case study will be presented and discussed.

The method has been tested during low frequency noise measurements in an urban environment. In this case a very specific frequency of the source was known. This frequency was used to tune the measurement system. The paper will describe the results of the test.

Wed 15:20 Van Beuningen Zaal Low frequency noise

Vibroacoustics and sound emission characteristics of thin-walled, oil-immersed transformer vessels.

Michael Ertl
Siemens Energy Sector - Transformers

The fluid-structure-interactions, vibro-acoustics and sound emission characteristics of power transformers are investigated by numerical methods.
Oil-immersed power transformers consists of thin-walled, rib-stiffened tank structures. This geometric setup and the three-phase acoustical excitation results in a complex spatial sound emission characterisics in the low frequency range of 100Hz to400Hz. Sound emission lobes are observed in measurement and simulation.
The effect of additional damping, mass load and stiffening to the vibroacoustic behaviour and overall sound level is investigated. Further, the effect of cavity modes inside the vessel to the sound emission is quantified. The influence of these parameter to the structure-borne sound level, the transmission ratio and radiation loss factor of the vessel construction is analysed.

Wed 15:40 Van Beuningen Zaal Low frequency noise

Determination of structure-borne noise based on the vibration signal

Tommaso Meloni
Federal Office for the Environment, Switzerland

Structure-borne noise is perceived in rooms of buildings as a dull rumbling. A vibrating building structure is a pre-condition on the load side. Structure-borne noise and vibration often occur combined. Rating levels for both vibration and for structure-borne noise can be based on the measured or on the calculated vibration signal. This paper presents a determination procedure for structure-borne noise based on the vibration signal. This procedure presumes knowledge of a characteristic value of the velocity of vibration which in turn depends on the vibrating room's specificities. A transfer function is implicitly integrated in this characteristic value, respectively in the weighted vibration signal. This transfer function describes the relationship between the velocity of vibration on the room surfaces and the structure-borne noise radiated into the room depending on the characteristics of the building. Finally, the proposed determination procedure helps to evaluate indoor perceived structure-borne noise and thus to protect people from harmful effects.

Wed 16:40 Van Beuningen Zaal Low frequency noise

Finding the source of Low Frequency Noise by measuring ground-born vibrations

Gerard Busscher and Carel Ostendorf
Cauberg-Huygen Raadgevende Ingenieurs

The residents of an old farmhouse in the countryside were suffering from nuisance by low frequency noise. Although the residents could hear the noise all the time, the noise was heard better during specific moments in the evening and night. The combination of vibration measurements on a wall in the farmhouse and sound measurements outside the house, did not reveal the direction of the source of low frequency noise. During a second measurement session, the vibration measurements in the farmhouse were performed in combination with ground-born vibration measurements on several positions outside the farmhouse. Based on the phase-shift of the vibrations on the different measurement positions, it was possible to determine the direction of the source of low frequency noise.

Wed 17:00 Van Beuningen Zaal Low frequency noise

Investigation on the Stimulation of the Human Auditory Cortex by Low-Frequency Sound and Infrasound Using Functional Magnetic Resonance Imaging: Stimulus Generation and Control, and Noise Assessment

Johannes Hensel[a], Esther Dommes[b] and Günther Scholz[b]
[a]*Physikalisch-Technische Bundesanstalt, Braunschweig;* [b]*Charité, Universitätsmedizin Berlin*

Following the demonstration of the impact of infrasound in the cochlea [Hensel et. al. 2007, Hear. Res. 233, 67-76], in the present study the stimulation of the auditory cortex down to 12 Hz was investigated by means of functional magnetic resonance imaging (MRI).

The neuroradiological evidence of cortical excitation will be reported elsewhere. In this report, details of stimulus application and noise assessment are described.

In order to exclude the other sensory paths, and to avoid the difficulties of free-field stimulation, the stimuli were delivered directly into the ear canal.

This would ideally have required a metal-free earphone, which was not available. 12 meters of silicon tube and an ear-plug were used instead. To control the influence of standing waves, the sound pressure at the ear was measured in-situ by a metal-free optical microphone, embedded at the interface of tube and ear-plug. At very low frequencies, where high sound pressures were needed even near the hearing threshold, it turned out to be a demanding task to keep the harmonics small.

The microphone position near the ear canal entrance allowed to measure the MRI noise entering the system up to that point. Spectral analysis showed that this noise already contained some infrasound.

Wed 17:20 Van Beuningen Zaal Low frequency noise

Reflection of LFN Perception on an Auditory Model

Detlef Krahé
University of Wuppertal

Former investigations with an auditory model of Ray Meddis showed that special effects in hearing low-frequency noise (LFN) might possibly be explained by synchronized activities in the nervous system. It is also known that visual stimuli can support and increase the auditive perception in the case of synchronisation. Can the assumption be confirmed that a strong annoyance caused by LFN is indicated by a kind of synchronism in the nervous system? Former investigations were done with a monaural model and synthetic stimuli. In continuation, the model is extended to a binaural one, because low-frequency sound meets special conditions regarding interaural time and level differences. Further, real life noise is processed by the model now. The reaction of people on this noise is compared with the reaction of the model on these stimuli. The paper reports about the results of this comparison and the extended model.

Wed 17:40 Van Beuningen Zaal Low frequency noise

Prediction of Load Noise of Power Transformers by 3D FEM and Comparison to Measurements

Andreas Hauck[a], Martin Meiler[b] and Reinhard Lerch[a]
[a]*University Erlangen-Nuremberg;* [b]*Simetris GmbH*

In absence of reliable theoretical formulas for noise emitted by electric power transformers, finite element simulation play an important role in the design of low-noise transformers. A major challenge in developing a FE model is to resolve the small scale windings structure, as well as the tank with the oil and the surrounding air.

In this work we make us of a two-scale homogenization approach for calculating the mechanical deformations of the windings by exploiting the periodicity of the winding structure. In a first step, a unit cell (circular ring) problem is solved on a very fine grid resolving the winding structure, which yields the mechanic eigenmodes. On a very coarse model of the complete setup, these eigenmodes are augmented with the standard finite element basis functions to model the homogenized winding structure.

The full structure-acoustic interaction of the windings, the oil and the tank is taken into account, as well as the acoustic radiation into air. The applicability of our method is demonstrated for a 3-phase power transformer. Comparisons of computed the sound pressure level, as well as the acoustic power with measurements are performed and show good agreement.

Wed 18:00 Van Beuningen Zaal Low frequency noise

An Investigation of Acoustic Properties of Multiaxial Glass Fiber Fabric and Hazelnut Rind Reinforced Polyester Resin Composites

Erhan Sancak[a], Sezgin Ersoy[b] and Mehmet Akalin[a]
[a]*Marmara University, Dep. Textile, Istanbul;* [b]*Marmara University, Dep. Mechatronics, Istanbul*

Sound absorption constitutes one of the major requirements for human comfort today. Sound insulation requirements in automobiles, manufacturing environments, and equipment, generating higher sound pressure drive the need to develop more efficient and economical ways of producing sound absorption materials. Industrial applications of sound insulation, generally includes the use of materials such as glass fabrics, foam, mineral fibers and their composites. Composite materials have been using in the industry for many years.

Nowadays, waste materials are coming into question place, so we used together waste material and warp-knitting fabrics to make reinforced composites. In order to improve the mechanical properties of composites fibrous reinforcement materials are used. Glass fiber is the mostly used reinforcement material for composites. In order to improve sound absorption properties of composites used hazelnut rind as a filled materials. This has been broken into pieces with two kind of size. And then, hazelnut rind and multi-axial glass fiber reinforced composites produced with different size of hazelnut rind.

In this study, we investigated that Hazelnut Rind / Biaxial Warp knitted fabric / Polyester resin composite structures produced and their sound absorption at the low frequency.

Wed 8:40 Schadee Zaal Speech in car 2

Application of the Binaural Speech Intellegibility Model for prediction of intelligibility in in-car noise

Rosa-Linde Fischer[a], Sandra Meyer[a], Jörn Otten[b] and Cristina Meinecke[a]
[a]*Universität Erlangen-Nürnberg;* [b]*Audi AG*

Conventional speech prediction models are based on monaural input, which limits the application in spatial conditions. However, in vehicles signal and noise sources are spatially distributed. Hence, an application of a binaural model is likely to increase the accuracy of speech intelligibility prediction. The Binaural Speech Intelligibility Model (BSIM, Beutelmann & Brand, 2006) was used to check, whether this binaural SII gives advantage over monaural approaches as SII and STI (speech transmission index). The advantage of the BSIM is the combination of the monaural speech intelligibility index (SII, ANSI, 1997) with the binaural equalization- cancellation processing (Durlach, 1984). The model is taking into account the binaural release of masking, if there is a directional separation of speech and interfering noise. The evaluation of speech prediction accuracy was based on measurements with normal hearing subjects. The collected speech related threshold data for several in-car noise conditions are compared with speech indices predic- tions named above. Results of the study will be presented and discussed concerning implications for future work.

Wed 9:00 Schadee Zaal Speech in car 2

A Comparison of Instrumental Measures for Wideband Speech Quality Assessment of Hands-free Systems in Echoic Condition

Kai Steinert[a], Suhadi Suhadi[b] and Tim Fingscheidt[b]
[a]*Siemens AG;* [b]*Braunschweig Technical University*

Instrumental speech component quality assessment presumes the availability of the processed speech portion of the enhanced output signal mixture along with its clean input counterpart. As the filtered speech component is unavailable in the context of black box measurements of speech enhancement systems, we have proposed a signal separation technique in earlier publications along with instrumental and subjective evaluations for the narrowband case. In this paper we apply the technique for a black box speech component quality assessment of two wideband hands-free systems with some instrumental measures. In particular, the echo reduction performance in the double-talk case will be investigated. Our findings are compared with the results of white box measures and of a subjective listening test.

Wed 9:20 Schadee Zaal Speech in car 2

About the Measurement of the Hands-Free Transmission Quality - An Experience Report

Martin Herrenkind and Holger Pastillé
IAV GmbH

About the Measurement of the Hands-Free Transmission Quality - An Experience Report
The method of evaluation of the transmission quality by car-integrated hands-free telephones, presented on DAGA 2006, was tested in practice. First results from investigations will be shown. The performance of the perception is illustrated in a multidimensional judgement. To produce a decision memo for the management and definition of the ”state-of-the-art” in car-integrated phones, a pool of references and benchmarks are helpful. The reduction on a singular value will be discussed.

Wed 9:40 Schadee Zaal Speech in car 2

Automatic Evaluation of Intercom Systems

Arthur Wolf, Tim Haulick and Gerhard Schmidt
Harman Becker Automotive Systems

Due to high background noise and sound absorbing materials the communication between front and rear passengers in a car is often difficult. The in-car communication (intercom) system distributes the seat dedicated microphone signals via the sound system in order to improve speech intelligibility. Due to interfering signals and the closed loop operation of the intercom system, various signal processing techniques are required to reduce echoes and noise and to prevent feedback instability of such a system.

In this contribution a methodical evaluation of an intercom system and its incorporated signal processing schemes is presented. It considers different requirements for both speaking and listening passengers. The evaluation is performed with three types of systems or setups. The first one is an ”ideal” intercom system. Here, a simulated system without any noise or feedback problem is computed in real-time and presented to listeners and to the measurement equipment. This ”ideal” system is used to obtain suitable parameter settings like the maximum desired gain for instance. Secondly a real intercom system is evaluated, in order to analyze the achieved speech intelligibility and system quality. Finally, all measurements are performed without any intercom system, to obtain a basis for comparison.

Wed 10:00 Schadee Zaal Speech in car 2

Psychoacoustic Evaluation of Music Reproduction in Passenger Cars

Hugo Fastl[a], Martin Ammler[a,b] and Adam Sulowski[a,b]

[a]*AG Technische Akustik, MMK, TU München ;* [b]*now: Audi AG, Ingolstadt*

A method to assess the quality of music reproduction in passenger cars is developed and evaluated. In essence, the subject sits in a standing car with the motor off, and is presented from a CD music of different genres via the car's audio system. The subject has to rate different aspects of the performance on a 10 step scale using a PDA. From the ratings, a net like a spider net, with different attributes like natural, squeeking, dull, reverberant, acceptable, good etc. on its rays is constructed. The goal is a net with a large area, i.e. as many as possible of the attributes should attain the maximum rating.

Examples from a pilot study are given for car audio systems of different sophistication, and aspects like gender differences or possible biases due to product relationship will be discussed.

Wed 10:40 Schadee Zaal Speech in car 2

Loudness Controlled Dynamic Compression for Automotive Fidelity Systems

Gottfried Behler

Institute of Technical Acoustics, RWTH Aachen University

The most common place for listening to music today is our private car. However, it is accepted that in a noisy environment like a car it is not possible to appreciate programs with rather high dynamics, like classical music, jazz and the like. Connoisseurs of that kind of music either must accept to continuously adjust the volume or to miss the pianissimo passages of the music. A very meaningful method would be to adjust the level of the program automatically with respect to the accepted maximum level of the listener. Once he settled a defined loudness for the reproduction of the fortissimo passages this loudness will be automatically maintained. Pianissimo passages therefore will be raised in level for a certain amount so not to be lost in background noise. Loudness means that the subjectively assessed loudness shall be used as the steering property rather than the peak- or rms-level. A dynamic compression method was investigated that uses loudness as a steering parameter. The aim is to make the listening to music in a noisy environment like a car easier. Listening test have been performed to verify the potential of the method.

Wed 11:00 Schadee Zaal Speech in car 2

Navigation System: An Experiment

Tomas Nestorovic
University of West Bohemia in Pilsen

This article describes results of a multi-modal navigation system experiment conducted on N=11 people. The goal was to decide whether the designed interface of the system is suitable to provide enough functionality and robustness. Moreover, undertaking this experiment, several of common human-computer interaction paradigms were proven - for example, our volunteers tended not to perceive the whole system prompt, and expressed themselves vastly briefly. However, averagely, participants took approximately twice longer time to accomplish given tasks, compared to graphical interaction only. The research in this field continues and, thus, at the end of this article, currently undertaking, in short upcoming and future work is presented.

Wed 11:20 Schadee Zaal Speech perception 1

PESQ Based Speech Intelligibility Measurement

John Beerends[a], Ronald Buuren[b] and Jeroen van Vugt[a]
[a]*TNO Information and Communication Technology;* [b]*TNO Defensie en Veiligheid*

Several measurement techniques exist to quantify the intelligibility of a speech transmission chain. In the subjective domain the CVC score is mostly used. In the objective domain the Articulation Index and the Speech Transmission Index (STI) are standardized for predicting intelligibility. The STI uses a signal that contains spectro-temporal characteristics similar to natural speech. Comparing intensity fluctuations of degraded and reference signals, the modulation transfer function is derived from which the STI can be calculated. In speech transmission, coding is used that may behave differently for speech then for the STI test signal, implying a possible incorrect intelligibility prediction. A more fundamental approach in assessing intelligibility is to take natural speech signals, and derive internal representations of the reference and degraded from which the difference can then be used to estimate the intelligibility. In this paper we investigate the idea to use a testsignal composed of concatenated CVC words, that are also used in the subjective CVC test, in combination with PESQ to measure the intelligibility. Normally testing a speech link requires long subjective listening sessions with CVC words in carrier sentences, while a compact testsignal approach in combination with PESQ allows a fast measurement.

Wed 11:40 Schadee Zaal Speech perception 1

Feature Decomposition and Modeling of Speech Quality for Various Wideband Conditions

Marcel Wältermann, Alexander Raake and Sebastian Möller
Deutsche Telekom Labs, TU Berlin

The integral quality of transmitted speech can be seen as a combination of features which are recognized by the listener in the auditory domain. It has been shown recently that traditional narrowband as well as wideband speech quality can sufficiently well be quantified by three orthogonal dimensions: “Discontinuity”, “noisiness”, and “coloration” (DNC). These dimensions can not only be used to model the integral quality, they also provide perceptually adequate diagnostic quality information.
In wideband speech transmission (50-7000 Hz), the perceptual effects of degradations due to transmission impairments are even more spread across the perceptual space than in the narrowband case. In order to investigate common wideband transmission scenarios with respect to the DNC framework, a series of real-world conditions (including a variety of codecs and codec tandems) are quantified in the study presented here. This quantification was done by directly scaling the dimensions “discontinuity”, “noisiness”, and “coloration” following a new test method. Furthermore, the integral quality was assessed. The obtained results allow to characterize the considered conditions in terms of the aforementioned dimensions which, in turn, helps to improve existing quality models.

Wed 15:00 Schadee Zaal Speech perception 2

Attribute-based Instrumental Assessment for Speech- Transmission Quality

Lu Huo and Ulrich Heute
Christian-Albrechts-University of Kiel

Recently, Scholz et. al. presented an initial realization of an attribute-based instrumental measure for end-to-end speech transmission quality. Related publications by Wältermann et al. identified three orthogonal quality dimensions, namely, ”discontinuity”, ”noisiness”, and ”coloration”, through multidimensional analysis. Instrumental measures were developed by Huo et al., Kühnel et al., and Scholz et al., for all dimensions, which were indexed by so-called ”dimension impairment factors” due to Wältermann et al., and a prediction model was also given. Here, we further develop these instrumental measures and prediction models. The following modifications are included: (1) The pre-processing is refined, and efforts have been given to separate different attributes so that their mutual influences are suppressed. (2) The algorithms, as well as the prediction models, are extended to wide-band speech, so that narrow-band and wide-band stimuli can be equally treated on the signal basis. (3) The extracted parameters and the prediction model of each dimension are modified and improved, based on the implication of so-called

”sub-dimensions” identified by follow-up auditory tests and multidimensional analysis; especially, packet loss is now included. Finally, the new prediction model is evaluated, using the same narrow-band database as in the previous papers, as well as a wide-band database.

Wed 15:20 Schadee Zaal Speech perception 2

Intelligibility Assessment Method for Semantically Unpredictable Sentences in German

Juan-Pablo Ramirez, Alexander Raake and Dina Reusch
Deutsche Telekom Labs, TU Berlin

We propose a new intelligibility test method in German, based on the French test method described in Raake and Katz (2006). It measures the Speech Reception Threshold (SRT), the target-to-masker level ratio (SNR) yielding 50% word intelligibility. The SRT is often determined in an adaptive procedure using lists of sentences. The present paper describes the development of the German sentence material, the speech recordings, the test to derive a first SRT-estimate, the level-calibration of the sentences and the derivation of the slope of the psychometric intelligibility function as a function of the SNR. The resulting speech corpus comprises 24 lists of 12 sentences each, yielding a total of 288 semantically unpredictable sentences (SUS, cf. Benoît et al., 1996). Each sentence is composed of four keywords set within a basic syntax. A stationary, speech-shaped noise masker was generated from the recordings. Three calibration tests were performed. A first test to yield preliminary SRT and slope estimates, a second test evaluating intelligibility at the SNR to calibrate - within limits - individual sentence levels, and a third test for more exact slope evalution. The paper discusses the results of these tests in the light of the particular sentence material used.

Wed 15:40 Schadee Zaal Speech perception 2

New developments in VoIP testing

Nils Rohrer, S. Poschen and Frank Kettler
HEAD acoustics GmbH

The market for IP based speech transmission increases rapidly. Integrated access devices (IAD) are offered by network providers for end users. VoIP gateways provide the interface between NGN (Next Generation Networks) and traditional PSTN (Public Switched Telephone Networks). Speech quality measurements are also further developed in order to cover more realistic test scenarios and more efficient test conduction. This contribution discusses new tests and test results on different devices. Information are provided about ongoing international testing activities e.g. the ETSI Speech Quality Test Events (SQTE).

Wed 16:00 Schadee Zaal Speech perception 2

Performance of Instrumental Speech Quality Measures for Next Generation Wireless Networks

Blazej Lewcio, Marcel Wältermann, Pablo Vidales, Alexander Raake and Sebastian Möller
Deutsche Telekom Labs, TU Berlin

Wireless transmission networks of the next generation (NGNs) provide new perspectives of VoIP service distribution and the possibility of seamless mobile service usage. The quality of the speech transmitted over such networks will contain new kinds of degradations with time-varying characteristics. In order to provide high-quality VoIP calls, speech quality monitoring is necessary. Even though quality prediction models designed for today's networks already exist (e.g., the E-Model and PESQ), the accuracy of these models in wireless networks of the next generation have not been proven yet.
On the basis of new auditory and instrumental test data, we address limitations of these models in NGNs, and we show how accurate they handle time-varying quality parameters (i.e., non-constant packet loss rate, audio bandwidth changes between wideband and narrowband, link quality changes within ongoing voice calls). Only if quality prediction models are validated in a realistic NGN environment, they can be considered and will be accepted as reliable tools for speech quality monitoring. In this way, they can contribute to the improvement of mobile VoIP user experience.

Wed 17:00 Schadee Zaal Speech perception 2

Evaluation of aurally-adequate analyses for echo assessment

Frank Kettler[a], Marc Lepage[a] and Matthias Pawig[b]
[a]*HEAD acoustics GmbH;* [b]*RWTH Aachen, Inst. für Nachrichtengeräte*

The migration towards NGN (Next Generation Networks) is supposed to introduce higher propagation delay in telecommunication. This emphasizes the need for reliable echo control. Wideband telephony will further change speech perception - and echo perception. New investigations on echo perception demonstrated the necessity to renew tolerances for wideband echo attenuation. These trends also motivate new echo assessment methods. A new approach based a hearing model analysis is introduced and discussed in this presentation suitable to extend existing echo analysis methods.

Wed 17:20 Schadee Zaal Speech perception 2

Quality Prediction for Synthesized Speech: Comparison of Approaches

Sebastian Möller[a] and Tiago H. Falk[b]
[a]*Deutsche Telekom Labs, TU Berlin;* [b]*Dept. of Electrical and Computer Eng., Queen's University (CAN)*

Text-To-Speech (TTS) technology has reached a level of maturity which seems to be sufficient for a number of telephony applications. In order to assess TTS quality, system developers need to carry out auditory tests where participants are asked to transcribe what they have heard, or to rate certain aspects of the auditory event. To overcome the temporal and financial effort involved in auditory testing, it is desirable to estimate the quality on the basis of the synthesized speech signal alone.

Two such approaches will be compared in this talk: The first has proven successful to estimate the impact of transmission degradations on telephone networks. It generates an artificial (high-quality) reference, compares the TTS signal to this reference, and combines the result with a noisiness, clipping and robotization detector in a parametric analysis to derive an overall quality estimate. The second approach is based on a Hidden Markov Model trained on natural (male or female) speech; features derived from TTS signals are compared to the model for natural speech and quality is derived from the normalized log-likelihood between the reference models and the extracted features. The performance of both approaches will be presented and proposals for improvements will be outlined.

Wed 17:40 Schadee Zaal Speech perception 2

Comparison of Spectrum-based Models for Speech and Audio Quality and Naturalness Estimation

Alexander Raake[a], Marcel Wältermann[a], Brian C. J. Moore[b] and Chin-Tuan Tan[b]
[a]*Deutsche Telekom Labs, TU Berlin;* [b]*Department of Exp. Psychology, University of Cambridge*

Spectral characteristics e.g. of electroacoustic interfaces in end-devices or network bandpass filters may degrade perceived naturalness and thus yield an impairment of transmitted speech quality. This effect is particularly prominent for wideband speech (50-7000 Hz) or speech with even wider bandwidth. However, the standardized models typically used to evaluate speech quality in telecommunications deliver accurate predictions only for narrowband speech (300-3400 Hz), and do not correctly predict the effects of linear distortion. In previous work (Raake, 2006; Möller et al., 2006; Wältermann and Raake, 2008), we suggested an extension of the E-model quality scale (ITU-T Rec. G.107) for application to wideband speech, a framework for the impairment due to wideband codecs, and a new impairment factor for quality under linear distortion. For a given spectral shape, this new impairment factor 'Ibw' is calculated from a rectangular filter whose effective bandwidth and centre frequency match those of the system under test. In this paper, the model is briefly outlined, and it is evaluated on a speech and audio database including ratings of auditory naturalness (Moore and Tan, 2003; 2004). The performance of the simple Ibw-model is compared to a more perception-oriented model of naturalness under spectral distortion (Moore and Tan, 2004).

Wed 18:00 Schadee Zaal Speech perception 2

Measurements of Sound Localization Performance and Speech Quality in the Context of 3D Audio Conference Calls

Mansoor Hyder, Christian Hoene and Michael Haun
University of Tübingen, WSI

The simulation of virtual acoustic environments helps to increase the quality of teleconferencing. Teleconferencing solutions supporting 3D audio have two advantages. First, the talkers can be identified easily by locating their position in the virtual acoustic environment. Second, if multiple persons talk at the same time, humans can focus on talkers of their choice by taking advantage of the cocktail party effect. However, 3D audio decrease the speech quality by decreasing loudness and adding reverberation and echoes. In this publication we study the trade off between speech quality and the ability of focusing and locating the talkers. We extend our initial work presented in the ITU Workshop in Lannion, Sep. 2008, in which we study the 3D audio rendering engine by Raine Kajastila et al. (AES 122th Convention, May 2007). We conduct subjective and objective listening-only tests using different head related transfer functions (HRTF), different algorithmic complexities, different positions of the conference call participants, and different geometries of the virtual environment. We present the statistical significant test results for sound localization performance and the speech quality. This study helps us to design our teleconference solution that we call 3D Telephony.

Wed 8:40 Ruys Zaal Railway noise 1

Assessment of track related noise mitigation measures

Manfred Kalivoda
psiA-Consult GmbH

Within the last decade methodologies for the measurement of railway noise generation have been strongly improved. This knowledge is used for type testing of railbound vehicles. However, a similar standard such as ISO 3095 is not available for track components, yet. There is a strong need for standardised assessment methods for track related noise abatement. This paper will present results from a recent measurement campaign in Austria dealing with changed track components. The discussion shows, that we have to control all the factors that might influence the results very carefully. Often we talk about effects of 1 - 2 dB(A) from the changed component while adverse influences can be of the order of 5 - 10 dB(A). Therefore it is necessary to collect enough data and thus get a sound statistical validation. It has been seen that only an unmanned automatic measuring system is able to collect sufficient acoustic and Meta information at recent costs.

Wed 9:00 Ruys Zaal Railway noise 1

Results of the UIC-project Noise Impact of Composite Brake Blocks (Nicobb)

Nicolas Meunier
Deutsche Bahn AG - DB Systemtechnik

Since the adoption of the TSI Noise by the European commission, all new freight wagons taken into operation have to fulfil stringent noise limits, making the use of composite brake shoes of the type K indispensable. The coefficient of friction of K-blocks is much higher than that of cast iron blocks. Therefore a retrofit of existing freight wagons would require an expansive modification of the brake system. UIC (International Union of Railways) supports the development of new composite brake blocks of type LL who combine the positive effect of K-blocks on wheel treads with the coefficient of friction of cast iron shoes. This opens the opportunity to substitute cast iron blocks by LL-blocks within regular maintenance, hence making the LL-blocks very attractive for freight operators. Acoustic assessment of brake blocks generally requires field tests, which are very time and money consuming. Thus there is a need of simplifying the acoustic validation procedure to encourage manufacturers to develop new products. The project Nicobb is a UIC project whose members are SNCF and Deutsche Bahn. It aims at developing a testing procedure for the validation of low roughness levels retained by new composite brake blocks as a low-cost alternative to field tests.

Wed 9:20 Ruys Zaal Physics of musical instruments 1

Reconstructing the Lituus: A Reassessment of Impedance, Harmonicity, and Playability

Adam Apostoli, Shona Logie, Arnold Myers, Jonathan Kemp, John Chick and Alistair Braden
University of Edinburgh

Bach's motet 'O Jesu Christ, meins Lebens Licht' (BWV no. 118), first performed in 1736-7, calls for a Lituus, believed to be a long trumpet-like instrument, with playing characteristics somewhere between a trumpet and an alphorn. Unfortunately, there are no surviving examples of Bach's instrument. Using software recently developed at the University of Edinburgh for brass instrument optimisation, a new bore profile with theoretically calculated input impedance curves has been proposed and submitted to a collaborating instrument maker with the aim of providing a natural instrument, with a distinct timbre, that responds well to the player and is in tune over a wide range of natural notes.
This paper describes the process, computational modelling, and experimental work undertaken in an attempt to ensure a satisfactory final instrument design. An initial validation of this process was achieved, in part, by investigating the playing properties of an instrument with a notionally similar bore profile, a 2.4m long Fiscorn in C (a type of Catalonian flugelhorn). This work compared the measured bore profile, input impedance, playability and intonation with those derived theoretically.

Wed 9:40 Ruys Zaal Physics of musical instruments 1

The Aubrapan: Revival of a lost invention

Soren Maloney
University of Cambridge

The Aubrapan is a new design of the melody (or soprano) pan developed in 1978 by Guyanese pan maker and panist Aubrey Bryan in the United Kingdom. In the Aubrapan the higher frequency notes are placed nearest the rim of the pan in contrast to the center of the pan where most high notes are generally located. The lower octaves are placed directly opposite each other in order to facilitate the pendulum-like action of the arms - these ascend in whole tones. This concept facilitates easier and rapid playing of the chromatic scale in single and double note form. In this paper we describe the development of the Aubrapan and an attempt is made to discover why the Aubrapan has not yet found a place among the steelpan family. To answer this and other questions, the experiences of the Aubrapan's inventor are noted and a comparison of the Aubrapan's playability is made with that of the conventional Trinidad and Tobago 4th's and 5th's soprano pan.

Wed 10:00 Ruys Zaal Physics of musical instruments 1

Embouchure Control of Brassiness at Constant Pitch and Dynamic Level in Orchestral Horn Playing

Lisa Norman, John Chick, Murray Campbell and Arnold Myers
University of Edinburgh

It is generally acknowledged that an increase in the dynamic level of a brass instrument brings about a change in timbre, often described as 'brassiness'. This phenomenon can be attributed to the generation of shockwaves at higher dynamic levels. However, the point at which this change occurs is not always clearly defined and some brass players have found that, by employing slight changes in their embouchure, they have a certain degree of control over the 'brassiness' at a constant dynamic level.

This paper focuses on how embouchure control influences wave propagation in the air column of a brass instrument during the onset of brassiness. A number of horn players were asked to play a long note at a loud dynamic level at the threshold of the onset of brassiness, and to move between a non-brassy and brassy timbre keeping dynamic level and pitch constant. Different horns were used to explore whether small changes in bore profile affected the ease of manipulation. The sound pressure in the mouthpiece and that of the radiated sound from the bell of the instrument were analysed and compared to gain a better understanding of the acoustic processes involved in player control of brassiness.

Wed 10:20 Ruys Zaal Physics of musical instruments 1

Effect of the structural dynamics of the bocal on the sound spectrum of a bassoon

Timo Grothe[a], Johannes Baumgart[b] and Roger Grundmann[c]
[a]*TU Dresden, Institut für Luft- und Raumfahrttechnik;* [b]*TU Dresden, Institute of Scientific Computing;* [c]*TU Dresden, Institute for Aerospace Engineering*

The bocal of a bassoon is a thin curved metal tube to which a double-reed mouthpiece is attached. When the instrument is played, the reed acts as a pressure controlled valve and due to the pulsating flow the bocal starts to vibrate. The outer contour of the bocal has a major influence on the modal shapes of the structural modes and the relation between the eigenfrequencies. The matter of interest in this investigation is the interaction of the structural dynamics of the bocal and the reed. We assume that modifications of the bocals geometry and mass distribution will lead to measurable changes in the sound spectrum. Therefore the modes of a bocal are computed by FEM and the frequencies are checked experimentally with a LDV. In a playing experiment mass was added locally on the bocal. Sound pressure measurements were performed, while a professional bassoonist played sustained notes on a modifiable bocal. Very small changes, less than 4 dB SPL, occur in the

overtones above 1 kHz and can be shown using principal components analysis of the sound spectra.

Wed 11:00 Ruys Zaal Physics of musical instruments 1

Mode coupling in the sound generation in wind instruments

Cornelis J. Nederveen
retired, Pijnacker

The excitation of a wind instrument by injection of air is a non-linear process; it generates Fourier components. These are coupled to the acoustic pipe modes, but in some cases also to wall and vocal tract resonances. Results can be unexpected. This appears from blowing experiments on a baroque oboe: applying a certain fork fingering can either cause the playing frequency to increase or decrease, depending on embouchure or blowing pressure. This phenomenon was studied by a simulation in the time domain employing a modified classic model for the reed action, coupled to the pipe impedance calculated with the transmission line method. The model confirmed the experiments quite well. However, the phenomena were difficult to understand from the spectrum; it was not clear what to look for. From varying the parameters it appeared that even very high frequencies could influence the playing frequency. This may mean that resonances in transverse direction, which can occur in flaring horns and bends, should also be considered. It corresponds to observations of a Dutch flute builder: blowing the flute changed when he introduced a bend after the mouthpiece to ease the handling of the flute.

Wed 11:20 Ruys Zaal Physics of musical instruments 1

Modal analysis of an alto recorder: Experiment and simulation

Franziska Möbius[a], Joachim Gier[a], Timo Grothe[b] and Steffen Marburg[a]
[a]*TU Dresden, Institut für Festkörpermechanik;* [b]*TU Dresden, Institut für Luft- und Raumfahrttechnik*

Herein, the authors present the investigation of an alto recorder. In a first step, a modal analysis of the structure of a recorder tunes the simulation model. It is possible to adjust material parameters to set up an eigenspectrum close to the one which has been measured in the experiment. However, some deviations remain. In a second step, the first resonance frequencies - which correspond to the played tone - are compared with the eigenfrequencies of a simulation model. Even without adjustment of parameters, the resonance frequencies and first eigenfrequencies of 16 tones are close together. Some unexpected effects of the tones of the particular recorder will be shown. Simulation reveals that the pressure magnitude in the duct does not form a half-wave as expected. It is one result of this project that it can be clearly shown that structural modes will hardly affect the tonal spectrum of the fluid.

Wed 11:40 Ruys Zaal Physics of musical instruments 1

Human Voice - a Sparse, Meaningful and Capable Representation of Sounds

Robert Mores
Hamburg University of Applied Sciences

This paper encourages the use of human voice features for mid-level representations of sounds. The human voice is a perfectly trained sound reference, combining two major classes of commonly used verbal sound descriptions, body tactile experience and every-day listening experience. Unlike pure technical representations, the human voice contains multidimensional physical and semantic features. It directly recalls cognitive patterns, and these, in return, bias perception. Examples of very sparse and capable representations are given by the extraction of vowel quality and nasality from short steady-state violin sounds. Such mid-level features can be identified with the help of automation, for instance, using psychoacoustic signal processing and feature extraction or using learning methods and classification. However, features can also be identified without automation, simply by listening tests and verification against a listeners own vocal tract. This general approach also supports manifold translation: backward to physical properties of a musical instrument and forward to other research fields such as cognitive musicology, or ethnological musicology, where researchers consider language-sensitive perception of sounds.

Wed 15:00 Ruys Zaal Physics of musical instruments 2

Nonlinearities in wind instruments at the example of organ pipes

Markus Abel
University of Potsdam

Nonlinear effects are typical for sound generation, let it be the aeroacoustical origin as in wind instruments, complicated stick-slip motion with string instruments or the dependence of body vibrations on the excitation amplitude. We demonstrate the intricate interaction of nonlinearities by the synchronisation properties of organ pipes by a high-accuracy experiment where a loudspeaker drives an organ pipe and find that even for minute speaker intensity the organ pipe still follows this control mechanism. Because the principle is very general, the consequences for other instruments and music has to be considered.

Wed 15:20 Ruys Zaal Physics of musical instruments 2

Frequency domain simulation of a bagpipe chanter with different tapers using the Harmonic Balance Method

Sandra Carral[a] and Christophe Vergez[b]
[a]*University of Music and performing Arts, Vienna;* [b]*Laboratoire de Mécanique et Acoustique - CNRS UPR7051*

The bore of conical bagpipe chanters, such as those found in Highland or Border bagpipes, usually consist of two cones of different tapers. The two cones meet approximately at the middle of the instrument, usually around the point of transition between right and left hands. This study is aimed at finding out what influence such change of taper has on the radiated sound of the instrument. For this purpose, three virtual chanters with no tone holes, a top taper of 2 degrees, and bottom tapers of 2, 2.5 and 3 degrees were modelled. The two cones meet at approximately half way from the top of the chanter. The input impedance and transfer function of each bore configuration were calculated using the program VIAS. In order to study the effect of the change in bottom taper only, the reed parameters were kept constant in all three bore configurations. With these data, the playing frequency and spectrum inside the mouthpiece were simulated using the Harmonic Balance Method. The radiated sound was calculated by multiplying the spectrum inside the mouthpiece by the transfer function of the instrument. The differences in pitch and spectral centroid found between the three instruments are presented and discussed.

Wed 15:40 Ruys Zaal Physics of musical instruments 2

Radiation Pattern and Characterization of Classical Guitars

Rolf Bader
Uni Hamburg, Musicology

Classical Guitars show difficult and complex radiation patterns when the strings are plucked. These patterns depend upon the geometries of the instruments. A listener perceives the integrated patterns binaurally at his listening position. Four different guitars were investigated, a high quality classical guitar, a flamenco guitar, a medium quality guitar and a guitar built like a bass reflex box. A microphone array consisting of 128 microphones with a sampling frequency of 48 kHz was used and the recorded sound fields was reconstructed at the top plate of the guitars. The measured radiation pattern showed similarities and dissimilarities according to the known geometry specifications of the instruments. As all six open strings were plucked and the first five partials were reconstructed according to their radiation patterns, statistical methods could be used to characterize the radiation. So i.e. the 'flexibility' of the guitar sounds judged highest with the high quality guitar and lowest with the bass reflex box guitar could be calculated by integrating the radiation binaurally. Next to other parameters, when just viewing the radiation patterns the basic character of the instrument clearly appeared where the high quality guitar shows the most clear, most differentiated and also the most beautiful patterns.

Wed 16:40 Ruys Zaal Physics of musical instruments 2

Characteristic Tones and Modes of a Church Bell

Helmut Fleischer

Universität der Bundeswehr München

A small church bell (37 kg, minor-third, strike note C6) was studied in terms of vibration (Laser Doppler vibrometry) and acoustic signal (spectral reduction). The bell sound was analysed using the aurally-related analysis software VIPER. The resulting auditory spectrograms were reduced by removing one or more partial tones. As well the original as the reduced auditory spectra were resynthesised and the sounds investigated in psychoacoustic experiments.
Two different problems were treated: The first was to compare the unmodified reference to sounds more or less reduced. In an A-B comparison, ten subjects had to indicate if they discriminated any difference. During a second series, signals resynthesised from reduced spectra were separately presented. Subjects had to judge whether the sound resembled a bell sound or not.
The psychoacoustic results reveal, which of the numerous partial tones are characteristic for the bell sound in a wider sense (”no difference audible”) or closer sense (”sounds like a bell”). Since each tone is assigned to a mechanical vibration, psychoacoustics gives the basis to focus on the vibration modes which are actually defining the sound of the bell. This way, the physical information can be reduced to its relevant core.

Wed 17:00 Ruys Zaal Physics of musical instruments 2

Transient behaviour in the motion of the brass player’s lips during a lip-slur.

Shona Logie, Samuel Stevenson, Adam Apostoli, John Chick and Murray Campbell

University of Edinburgh

When moving between notes on an orchestral brass instrument experienced players routinely change from one resonant mode of the instrument to another. Skilled players can ’lip-slur’ from one note to another making the transition between resonant modes sound very smooth as perceived by the listener, and do so with little thought or understanding about what is happening to the lip/air-column interaction during this intra-note transient.
This paper offers an investigation of the transient behaviour of the player’s lips when moving from a strong, fully developed coupling of the lips to one air column resonance, to the strong coupling of another resonant mode of the same air column. A high speed digital video camera is used to view the motion of the lips through a transparent mouthpiece for a range of orchestral brass instruments of different lengths. Image

processing techniques allow us to measure the opening area of the lips as a function of time, and this data is synchronised with pressure data measured in the mouthpiece together with the radiated sound from the bell of the instrument to provide us with greater insight to mechanics of the lip-slur transient.

Wed 17:20 Ruys Zaal Physics of musical instruments 2

Brass wind instruments: can the three dimensional motion of the lips account for the ‘brassy’ sound?

Samuel Stevenson, John Chick and Murray Campbell
University of Edinburgh

One distinctive feature of orchestral brass instruments is that there is a marked change in timbre with dynamic level. It is generally accepted that the explanation for this behaviour is caused by shock-wave generation within the body of the instrument. However, it has also been suggested that changes in the nature of the players’ lip vibration with increasing amplitude could play at least a small role in generating this distinctive ‘brassy’ sound. Previous studies have shown that the brassy sound cannot be accounted for by a dramatic change in the behaviour of the lip opening area during extremely loud playing. The experiments described here are designed to test the hypothesis that variations in the motion of the lip parallel to the airflow into the instrument may help produce the ‘brassy’ effect. A high speed digital camera and transparent mouthpieces are used to capture the motion of the lips in the plane perpendicular to the face of the player for different dynamic levels. Measurements have been taken over a variety of instruments and musicians and over a dynamic range from mezzoforte to fortissimo.

Wed 17:40 Ruys Zaal Physics of musical instruments 2

Resonance frequencies and sound radiation of musical woodwind instruments

Andreas Richter and Roger Grundmann
TU Dresden, Institute for Aerospace Engineering

Decoupling the acoustic resonator from the excitation mechanism allows to analyse the influence of the resonators geometry on the acoustics of the whole woodwind instrument. Such investigations are done typically in the frequency domain. Numerical investigations in the time domain offer some advantages. They allow to track single acoustic waves which gives an insight in the influence of the resonators geometry on the oscillation behaviour. This allows to study how connections, tone holes and bore perturbations may influence the wave propagation and act as sound sources. In comparsion to methods which are formulated in the frequency domain also transient effects can be studied.

We present results from numerical investigations of the acoustic resonator of the bassoon as one example for woodwind instruments. The

impulse reflectometry commonly used in experiments is modelled numerically. The results allow to analyse the resonance frequencies and also the acoustic radiation patterns.
Investigations in the time domain need to track acoustic waves over a long time period. We use a high-order discontinuous Galerkin formulation which offers the high accuracy required for this task.

Wed 8:40 Mees Zaal Tram noise

Reduction of Squeal Noise by Damping of Wheels

Udo Lenz
Ing.-Büro Uderstädt + Partner

Trams produce squeal noises in tight rail track curves. At the research project an anti-vibration mounting for the wheels was engineered, which enables a decrease in the loudness of the squeal noises. The later integration of that system in existing trams is possible without any problems. Results of comparable measurements will be presented.

Wed 9:00 Mees Zaal Tram noise

Squeal Noise Reduction in Urban Transport

André van Leuven
D2S International NV

Squeal noise reduction in urban transport by rail treatment The present paper summarises the results of the research carried out under the EC con-tract BRPR-CT97-0477, ”Squeal noise reduction in urban transport by rail treatment” and the results of recent findings about mitigation methods for squeal noise. Most rapid transit systems have tight short radius curves where squeal noise is generated, Curved track, and thus the areas where squeal noise is prevalent, cover only a minor fraction of the total track length. Individual curves are not longer than a few hundred meters. Therefore, the most cost effective way of solving the squeal noise problem is treating only those specific track sections, rather than treating the wheels. However, solutions at rolling stock level are evaluated.

Wed 9:20 Mees Zaal Tram noise

Is squeal noise really not to be predetermined

Helmut Venghaus
selbst. berat. Ing., Ingolstadt

Iron wheels running through a curve will create a lateral stip-slick movement between rail and wheel which very often cause a very loud squeal-noise. This stip-slick movement is caused by the friction coefficient. A high difference of the friction values for static or sliding situations will indicate a high ability for squealnoise. The static friction coefficient is in some way depending on vehicle speed, weight, weather conditions and some other circumstances and hence are not precisely to be defined. Due to this, the occurance of squealnoise seems to be stochastic.
Recent measurements of trams passing a curve showed, that the occurence of squealnoise seems to need only an certain condition of the rail head to initiate this stick-slick movement and the additional squealnoise. The same tram running the same curve several times with different speed produced a comparable signature of squealnoise. This is a contradiction to stochastic behaviour.

Wed 9:40 Mees Zaal Tram noise

Numerical Method for the Prediction and the Assessment of Rolling and Curving Noise

Arnold Groß-Thebing
SFE GmbH

The program SFE AKUSRAIL aims for the simulation of the whole chain leading to noise in railway operation. The tool generates models predicting the high frequency interaction between vehicle and track. It consists of modules regarding track and wheelset dynamics, W/R contact mechanics, rolling noise, curve squeal and interior noise prediction. The combination of track and wheelset dynamics, contact mechanics and vehicle structure dynamics represents their interaction taking into account physical properties of geometry, material and operational conditions. For the prediction of curve squeal two independent prediction methods filter the curve squeal frequency from all the possible wheel resonances. The methods show the sensitivity for variations of wheel geometry, for application of wheel absorbers and for material properties of the rubber in resilient wheels. An automatic parameter variation performs instability regimes for the linearized differential equations of the wheelset - track dynamics considering a contact point shift on wheel and rail tread. Wheel and rail amplitudes are calculated in dependency of the frequency and of the position in curved track using time step integration of the complete nonlinear model. The prediction method is validated in a research project funded by the Bundesministerium für Wirtschaft und Technologie (19U5001 B).

Wed 10:00 Mees Zaal Tram noise

Active Attenuation of Squeal Noise at Train Wheels

Hubertus Siebald
ERAS GmbH

The present work represents preliminary experimental investigation aimed at the possibility to prevent the squealing noise of a resilient train wheel under real operating conditions. The control effectiveness of two different kind of piezoelectric actuators have been tested in a non-rotating test stand designed with the aim of reproducing the wheel working conditions as closely as possible. It is not possible to fully simulate the rotating wheel behaviour, but promising results and important information have been obtained with collocated control. In the next step piezoelectric stack actuators have been adopted on a rotating wheel, on the condition that slip rings have been installed both for signals and power transmission. Tests with a prepared wheel set have been performed on a rotating test stand. A good vibration reduction at the wheel and also a noise reduction in the nearfield of the wheel have been achieved.

Wed 10:20 Mees Zaal Tram noise

Influence of impurities between train wheels and wheel damper on the effectiveness of the damper

Martin Fehndrich
Bochumer Verein Verkehrstechnik GmbH

Wheel vibration absorbers by Bochumer Verein Verkehrstechnik are successfully used with train wheels for vehicles of the light and heavy rail to avoid curve squeal and to optimise the level reduction of the rolling noise.

Anyhow, curve squeal occurred at individual vehicles in distinct cases. An investigation of the concerning wheel dampers did not lead to explanatory results. These results could not be reproduced by measurements at similar wheels on the test stand, either.

As working hypothesis, we assumed that impurities on the contact surface of wheel and damper had reduced the damping behaviour. Measurements with a layer of a model dirt are presented. They can explain the effect, as the damping of the wheel was reduced clearly.

Therefore, we developed new dampers, which are insensitive to such dirt layers. The effectiveness of the new damper can be shown on the basis of test results conducted during rides with trams in different cities.

Wed 11:00 Mees Zaal Tram noise

Squeal Noise - Long Time Examination of Stochastically Occurence on Tramways passing Curves

Peter Fürst
cdf Schallschutz Dresden

In the last years exactly parametrically examination on squeal noise has demonstrated, that many factors of influence are existing. In this paper we will demonstrate, that also stochasti-cally occurrence has an influence of the sound emission of a track curve. Observations in a track curve during one year show, that the frequency of noise emission under same condi-tions don't achieve exactly comparable results. So we say, that there is only a limited prob-ability depending on the different parameters that squeal noise is appearing or not. Other-wise we found, that the curve is squealing, but there is no typical situation for the appearance of squeal noise. We will present results within variable trams and weather parameters on the same curve. At the end we will compare our observations at another kind of tram curves.

Wed 11:20 Mees Zaal Tram noise

Curve squeal - Research without an End?

Friedrich Krüger
STUVA, Köln

Curve squeal is the by far the loudest noise of rail vehicles. Since this is already known for a long time numerous research work has been done (and is going on) worldwide to this subject. In this field the publications are so numerous that it is difficult to know all of them and to appreciate them correspondingly. There is agreement largely over the excitation mechanism. Due to a crossways moving of the wheel-set friction forces are produced. One wheel is running at the outer rail, the other one is sliding on top of the inner rail. These friction forces activate the natural frequencies of wheel and rail. Agreement also consists that gliding can be influenced by a change of the friction conditions and therefore the squealing can be avoided or reduced. Why is research and development work still going on? Some statements (strategies) are represented for this subject: "How solutions can be realized in praxis". These strategies can cause a squealing free future in small curves of trams, if the people responsible for this do a good job. This means that they realized the available measures.

Wed 15:00 Mees Zaal Flow acoustics

Modelling Sound Level and Pressure Loss in HVAC Duct Networks using a Framework in Dymola/Modelica

Helmut Kühnelt, Anton Haumer and Thomas Bäuml
arsenal research, Vienna

In the early design phase of large HVAC duct networks the joint calculation of the sound level and air flow rate and pressure loss as well as the prediction of the noise level in the passenger compartment gain more and more in importance. A unified framework for the one-dimensional flow acoustic simulation using Dymola/Modelica is introduced. The acoustic part is described by plane wave two-port theory, the fluid mechanic part by pressure loss models. Various approaches including analytic and (semi)empirical models based on experimental and simulation data can be implemented. Each component either represents the acoustic or the flow part of a duct element or can, itself, be composed of both types allowing the modelling of non-standard components. The object oriented approach provided by Modelica assures the extensibility of the models e.g. by transport and exchange of heat or the (de)humidification of the air. It also allows the integration into a complete system simulation of the HVAC network including the cooling circuit and the electronic control loop. The simulation is easily set up using a GUI; an external parameterization ensures persistent data management and a smooth workflow. The resulting system of equations is automatically pre-processed and solved by Dymola.

Wed 15:20 Mees Zaal Flow acoustics

Sound Generation and Propagation with the Nonlinear EIF-Approach

Otto von Estorff[a], Marian Markiewicz[b] and Thilo Michels[a]
[a]*Inst. of Modelling and Computation, TU Hamburg-Harburg;* [b]*Novicos GmbH*

The development of new methods of the numerical aeroacoustics is one of the governing research interests. For this kind of approaches, highly efficient methods are necessary. Aerodynamic noise predictions codes in use today are typically based on linearized wave equations combined with sound sources. In this paper a nonlinear approach is implemented into a three dimensional CFD code. The used method, the Expansion about Incompressible Flow (EIF) Method was proposed first by Hardin and Pope in 1994. Based on solutions of the incompressible flow, acoustic radiation is obtained in a compressible, inviscid domain. The acoustical equations are developed by subtracting the incompressible equations of the full Navier Stokes equations and neglecting the viscous effects. The advantage of this technique, as compared to the acoustic analogy theories, is that the source strength is obtained directly. It accounts for sound radiation as well as for scattering. In the current paper

a further development towards three dimensional simulation and numerical implementation are presented. To demonstrate the effectiveness and feasibility, two representative examples demonstrate the effectiveness of the proposed approach.

Wed 15:40 Mees Zaal Flow acoustics

Identification of a Low-Frequency Sound Source in an Aircraft Engine Test Stand

Wilhelm von Heesen
Müller-BBM GmbH

The tenants of an office building complained about strong low-frequency noise from a nearby aircraft engine test stand. The test stand resembles an open-circuit wind tunnel with the aircraft engine located at the end of a horizontal inlet duct. The engine discharges into a horizontal pipe with a conical insert at the inlet. The cross sectional area of the discharge pipe is substantially smaller than that of the inlet duct. Twelve microphones were installed inside the test stand, at the inlet and the outlet of the test stand, and at the office building. By analyzing the phase and the cross-correlation between the microphones the direction of sound propagation could be determined and the point of origin of the low-frequency sound could be identified. It turned out that the low-frequency sound is generated by the turbulent jet impinging on the conical inlet of the outlet pipe and propagates against the flow into the inlet duct. By decreasing the outer diameter of the conical inlet at the discharge pipe the sound level could be reduced. The main focus of the paper is the description of the data analysis which lead to the identification of the sound source.

Wed 16:00 Mees Zaal Flow acoustics

Retarded-time source field analysis of vortex-pairing noise

Florent Margnat
Arts et Métiers PARIS TECH - Sinumef

To date, the physical phenomenon that converts kinetic energy into acoustic waves escaping from the flow is not fully understood. Thanks to the increasing computational power, aeroacoustic prediction tools have become more and more fast and accurate. however, it is still challenging to link an acoustic emission pattern to the aerodynamic source field, in terms of causal events. Lighthill's acoustic analogy provides a way to extract the propagative motion from a flow through the expression of source term in an inhomogeneous wave equation. Unfortunately, when the flow is not compact, source field visualisations hardly exhibits flow locations where the acoustic energy could be produced.

In the present contribution, we study the source field considered at the retarded-time, which is the true radiating quantity. It differs from the fixed-time source field because it takes into account the solution of the wave equation, usually expressed by a Green function. This methodology is

applied to a 2D spatially evolving mixing-layer at Re=400 and 0.375 convective Mach number. Areas in the source domain are analysed by evaluating their net contribution to the aeroacoustic integral, and destructive interferences are noticed between non-radiating areas.

Wed 17:00 Mees Zaal Flow acoustics

Analysis of the Flow Field of Rotating Milling Cutters in Idle Mode

Lucia Groß[a], Uwe Heisel[a] and Winfried Keiper[b]
[a]*Universität Stuttgart, Institut für Werkzeugmaschinen;* [b]*Robert Bosch GmbH*

Rotating tools are generally assumed to be one of the major sources of noise in machines. Vortices shedding from the rotating tool may cause turbulent flow fields. This article concentrates on the flow field of milling cutters in idle mode. At first, measurements obtained by a scanning laser Doppler vibrometer are analyzed. The test object is a milling cutter for T-slots. The test reading shows turbulences caused by vortices only in the chip space of the milling cutter and the near field of the cog. Subsequently, the flow field of the same milling cutter is simulated by ANSYS CFX. The simulation confirms the test reading obtained by the scanning laser Doppler vibrometer. Additionally, a different type of milling cutter is tested in water. The turbulence caused by this milling cutter occurs again only in the chip space. Therefore different geometries are simulated to detect relevant parameters causing turbulences.

Wed 17:20 Mees Zaal Flow acoustics

Broadband noise investigation on rod-airfoil-configurations

Jens Giesler and Ennes Sarradj
BTU Cottbus, Aeroacoustics Group

One important broadband noise generating mechanism in axial fans is the interaction between the turbulent inflow or a turbulent wake of a rotor vane and the leading edge of the downstream stator vane. Additionally, the interaction of the turbulent vortices with the trailing edge of the airfoil generates noise as well. These mechanisms are the subject of an experimental research at BTU Cottbus.

The setup for measurements in the aeroacoustic wind tunnel consists of a rod and an airfoil mounted downstream. The geometric parameters of the rod-airfoil configuration, namely the cylinder diameter, the gap between rod and airfoil and the airfoil geometry are variable. For an improved flow quality in the measurement area of the wind tunnel the configuration is mounted between two porous sound absorbing side plates, directly attached to the nozzle exit. Acoustic measurements are done using an array of 38 flush-mounted microphones. The flow is characterized using constant temperature anemometry.

The presentation deals with the details of the experimental setup as well as with a first subset of experimental results. Moreover, the influence of the parameters on the leading and trailing edge noise generation is discussed.

Wed 17:40 Mees Zaal Flow acoustics

Silent owl flight: experiments in the aeroacoustic wind tunnel

Thomas Geyer, Ennes Sarradj and Christoph Fritzsche
BTU Cottbus, Aeroacoustics Group

It is common knowledge that most genera of owls fly silently in order to be able to catch their prey. To investigate the mechanisms leading to that quiet flight, several basic fly-over measurements were carried out by biologists in the past. But the very low noise that is produced when the owls are in gliding flight makes it very hard to do exact acoustic measurements, especially when using the measurement techniques available at that time.

The presentation describes the measurements of the noise generation at the wings of silent and nonsilent flying birds in the aeroacoustic wind tunnel of the BTU Cottbus. The measurement setup will be explained, especially regarding the usability of conventional measurement techniques for experiments on wing specimen. A 56 channel microphone array is used to measure the noise generated at the wings in the fluid flow. The acoustic data is processed with advanced beamforming algorithms to obtain the radiated sound pressure levels and the location of the noise sources on the surface of the wings.

First results will be given that indicate that the magnitude, the spectral shape and the location of the noise produced by owls wings is different than those of nonsilent flying birds.

Wed 18:00 Mees Zaal Flow acoustics

Acoustic Modal Analysis using CFD

Marco Oswald
Ansys Fluent Deutschland GmbH

One of the most important issues in aeroacoustics design and development is the optimization of noise and vibration in both interior and exterior design process. Especially the subjective comfort highly depends on the interior noise (e.g. sunroof buffeting). Besides the experimental investigations of this aeroacoustic phenomenon, numerical simulations become more and more an effective tool to shorten the development period. This paper proposes a new approach on finite volume method for determining the resonance frequencies for interior enclosures. The current formulation uses as input non-uniform mean flow obtained from a steady state solution together with prescribed boundary conditions. The method involves coupling the Linearized Navier-Stokes Equations with the Arnoldi algorithm. The Acoustic Modal Analysis, based on the iterative Implicitly - Restarted Arnoldi method, offers a fast and efficient solver for large industrial problems with inlets and outlets. Using the data from this model, the natural acoustic modes of the system and the according frequencies can be identified. These modes can resonate with interior acoustic source processes. The potential of the new method will be presented by means of several academic and industrial test cases ranging from thermo-acoustics to typical automotive and aerospace applications.

Wed 8:40 Van Rijckevorsel Zaal Transferpath analysis

Transfer Path Analysis - a Review of 15 years of Practical Application

Thomas Ahlersmeyer
Ford Werke GmbH Köln

Transfer Path Analysis (TPA) is a methodology to evaluate the contribution of individual transfer paths to a noise or vibration (NVH) problem. This is a first step to understand to root cause of a complex NVH problem. With the advent of affordable multi-channel digital (ADC) data acquisition systems the TPA has changed from simple detaching tests some 20 years ago to a structured process. However, in the last 15 years the inherent problems of the classical approach (exact force and transfer function determination) have not yet been overcome. Instead, shorter development cycles and shorter availability of less prototypes demand shorter test and analysis cycles and thus streamlining of the TPA approach. In the same time period the structural stiffness of passenger cars has increased significantly which results in higher crosstalk between the paths thus making a separation more difficult. The paper gives an overview how this has impacted the practical application of test-based TPA in the vehicle development of FORD in the last 15 years. It will also give an outlook on and first experiences with new approaches like TPA in time domain and estimation of the transfer matrix by using operational data only.

Wed 9:00 Van Rijckevorsel Zaal Transferpath analysis

Innovative Approaches to Fast Transfer Path Analysis

Herman van der Auweraer, Peter Gajdatsy, Karl Janssens, Peter Mas and Ludo Gielen
LMS International

Over the last decennia, Transfer Path Analysis (TPA) has become an accepted tool for NVH troubleshooting and internal load estimation. A major bottleneck preventing its even more widespread use in the actual vehicle development process is the test time to build the full data model, requiring extensive Frequency Response Function testing next to the obvious in-operation tests. Recent innovations on instrumentation and new procedures such as Fast and Multilevel TPA and Operational TPA address this limitation, however often introducing new constraints. For example, OTPA only requires operational response data at reference and target locations. But, being basically a transmissibility method, its application requires special care regarding the inherent ill-conditioning and reference-completeness problems. It essentially assesses a co-existence relationship between targets and references and does allow causality and contribution interpretations. Hence a new approach is proposed, using a parametric model for the estimation of loads. This makes the method scalable, enabling the engineer to use a simpler model based on a small amount of measurement data for quick troubleshooting or increase accuracy by using a more complex model together additional measurements. The method is applied to a customer problem and compared with the existing approaches, showing the advantages in real life.

Wed 9:20 Van Rijckevorsel Zaal Transferpath analysis

Path Analysis

Francesc Xavier Magrans
Ingenieria para el Control del Ruido, Barcelona

This paper will be devoted to explain the work done by the author in the Transmission Path analisys. The first part of this paper will explain the work in the frame of the historical evolution of the method, not only related with the theoretical-experimental concepts but also with the different theoretical frames like SEA or Room statistical Acoustics. We will focus in the concepts linked with the paths, a old intuitive idea in Vibroacoustics but without a mathematical definition for long time. The well known TPA and the less popular ATPA in both low frequency and middle-high frequency will be explained. In the second part the real aplications in the experimental field will be explained in relation with the Railways and also with a lot of mechanics vibrocoustic sources as in the Buildings field. A experience of more than 20 Years in the aplication of the methods has given a lot of useful data. The third part will be the explanation of the aplications of the equations in wich the method is based to carry on calculations of Vibroacoustics.

Wed 9:40 Van Rijckevorsel Zaal Transferpath analysis

Methods for the Transfer Path Analysis

Martin Quickert[a], Thomas Stichling[a] and Wolfgang Foken[b]
[a]*Fraunhofer-Institut für Werkzeugmaschinen und Umformtechnik;* [b]*Westsächsische Hochschule Zwickau (WHZ)*

Transfer Path Analysis (TPA) is a well known technique to evaluate and to separate the different transfer paths describing how noise gets from the sources into the compartment of a passenger car, i.e. to the driver's ear. It is not only a measurement technique but a prediction method as well. Based on previous measurements and assumptions TPA provides the opportunity to predict the interior sound for a virtual prototype. Due to the progress in computational technology and measurement techniques, TPA is being seen as an important tool in automotive acoustic design. The paper gives a brief survey of the mathematical background of different TPA methods like the Complex-Stiffness-Method or the Inertance-Matrix-Method. With the help of a 5:1 scaled simplified car model differences between the methods of calculation are verified and discussed. In conclusion the paper presents the measurement setup for a real car with a dominating transfer path of the structure borne engine noise into the passenger compartment.

Wed 10:00 Van Rijckevorsel Zaal Transferpath analysis

Application of Operational Transfer Path Analysis on a classic car

Dennis de Klerk[a], Martin Lohrmann[b], Martin Quickert[c] and Wolfgang Foken[d]
[a]*Müller-BBM VibroAkustik Systeme B.V., Zwolle;* [b]*Müller-BBM VibroAkustik Systeme GmbH;* [c]*Fraunhofer-Institut für Werkzeugmaschinen und Umformtechnik;* [d]*Westsächsische Hochschule Zwickau (WHZ)*

This paper discusses the analysis of a classic car with the Operational Transfer Path Analysis (OTPA). This method poses a fast and efficient way of identifying critical vibration paths and source contributions within vehicle design. Only measurement data of the operating vehicle are needed to perform this analysis. Real measurement situations for example are several vehicle run-ups, full load, halve load, run down. These kinds of measurements are commonly performed in the automotive industry and can therefore be easily adopted. The method OTPA is introduced and discussed based on a classic car Wartburg 311. This vehicle is described in the project sketch "method comparison of the transfer path analysis" (Fraunhofer IWU Dresden) and provided by Westsächsische Hochschule Zwickau, Institut für Kfz- Technik (IfK).

Wed 10:40 Van Rijckevorsel Zaal Transferpath analysis

Characterizing Tire and Wind Noise Using Operational Path Analysis

Roland Sottek[a] and Bernd Philippen[b]
[a]*HEAD acoustics GmbH;* [b]*RWTH Aachen University, Institut für Nachrichtentechnik*

An interactive driving simulation requires on-demand reproduction of vehicle sound components such as engine, wind and tire noises as separate sound contributions. For realistic sound perception, wind and tire noises are extracted from road measurements. The structure-borne excitation and airborne noise radiation of all four tires are measured during a coast-down from maximum vehicle speed to standstill. By evaluating the multiple coherence between the excitation signals and a simultaneous binaural recording of the interior sound, speed-dependent FIR filters can be calculated in order to predict, from the mixture in the cabin, the wind and tire noise shares.

In this paper an alternative approach will be presented describing the physical system as a **M**ultiple-**I**nput-**M**ultiple-**O**utput (MIMO) model. The tire noise can be synthesized using transfer functions estimated from road measurements using an **O**perational **P**ath **A**nalysis (OPA); the uncorrelated wind noise can be determined as the difference signal between actual interior noise and synthesized tire noise. In case of uncorrelated excitation signals OPA is very efficient and accurate; no additional laborious transfer function measurement is required. The advantages and possible drawbacks of OPA for characterizing tire and wind noise in comparison to coherence filtering will be discussed.

Wed 11:00 Van Rijckevorsel Zaal Transferpath analysis

Quantification of the Equivalent Sources of Different Tire Noise Models by means of the ASQ Technique

Dries Berckmans, Paul Sas and Wim Desmet
KU Leuven

Tire noise is a predominant noise contributor in many traffic situations nowadays. Accurate models are therefore needed to predict the spectrum that results at receiver locations due to the tire rolling under different operating conditions. Predicting the spectrum relies on a source - transmission path - receiver model which involves the modeling of all noise sources and the determination of all relevant transfer paths.
Locating and quantifying the different noise sources on an operating tire remains an inherent difficult task. The sources are thought to be numerous and complex, with their relative magnitudes changing with tire specifications, road specifications and operating conditions.
A rolling tire was experimentally characterized on a roller bench by means of the substitution monopole technique: the running tire is substituted by the non-operating tire covered by monopoles. The monopoles are quantified by means of the airborne source quantification (ASQ) technique, based on operating indicator pressure measurements in the neighborhood of the rolling tire. The role of regularization techniques, important in the quantification process, is highlighted and results for different tire models are presented. Moreover, the contributions of the different tire segments to the resulting spectrum at the receiver position are evaluated.

Wed 11:20 Van Rijckevorsel Zaal Transferpath analysis

Experimental Transfer Path Analysis on Helicopters

Antonio Vecchio[a], Emiliano Mucchi[b], Fausto Cenedese[c] and Elena Pierro[d]
[a]*LMS International;* [b]*Engineering Department, University of Ferrara;* [c]*AgustaWestland, Acoustics and Vibration Department (Italy);* [d]*Politecnico di Bari*

Noise levels recorded in helicopters’ cabin are severely affected by the strength and vicinity of noise sources. Jet engines, gearbox and rotors can be considered as separated sources - whose spectral content is strongly tonal and rpm dependant - exciting simultaneously the cabin’s acoustic cavity. Under the hypothesis of linear behaviour, the total sound pressure level measurable in the cabin is given by the summation of partial pressure contributions, each generated by one source acting separately. The mechanism responsible for transferring the mechanical energy from each sources to the target location can be structure borne - via the mechanical joints connecting the gearbox to the helicopter’s frame - or airborne - via the sound propagation in the air.

An experimental TPA approach is carried out to the helicopter Agusta A109, to assess noise source contribution to the cabin noise. Data collected through a number of experimental tests carried out both on the ground and in-flying conditions on the helicopter Agusta A109 are used to implement a numerical TPA model.
Synthesized FRF, computed with PoliMAX, are used as advanced tool to recover inaccuracies due to complex testing conditions and result in an overall improvement of data quality and reliability.

Wed 11:40 Van Rijckevorsel Zaal Transferpath analysis

Practical Measurement of Transfer Functions using Volume Velocity Sources

Andreas Schuhmacher
Brüel & Kjaer SVM A/S

Measurement of transfer functions is required for many automotive applications involving source-path-contribution or panel contribution techniques. For applications where the acoustic radiation from a complicated sound source is modelled, a series of sound pressure/volume velocity frequency response functions are measured as part of a method to find the airborne contribution of this sound source. In panel contribution analysis acoustic transfer functions are needed to relate sound field parameters on the interior cabin walls to desired target positions, eg driver's or passenger's ear positions. In this paper we will investigate several volume velocity sound sources for measuring acoustic transfer functions. The sources are based on the two-microphone method for in-situ measurement of volume velocity source strength, ie only the two microphones mounted in the source need to be calibrated. A typical setup making use of a vehicle environment will demonstrate the practical aspects of using such sources. Their validity will be investigated and some factors affecting the measurement quality will be discussed.

Wed 15:00 Van Rijckevorsel Zaal Active noise control 1

Active Exhaust Silencers for Internal Combustion Engines

Jan Krüger, Michael Pommerer and Rolf Jebasinski
Eberspächer GmbH

In the past years Eberspaecher has installed Active Exhaust Silencers on several personal vehicles with different engines on a prototype level. Meanwhile, a substantial reduction of the exhaust noise is regularly achieved in a broad frequency range covering all relevant engine orders. Results from roller test benches focus on the acoustic performance but as well demonstrate a marked improvement in backpressure which allows better engine efficiency and fuel economy. Further progress was made in the development of the durability and industrialization of all relevant components of the system. But instead of silencing also the shaping of different types of sound was implemented in some sports cars: Here, a variety of sounds are available from one and the same engine. Unlike with conventional passive exhaust systems the active silencer can thus be specifically adapted to the vehicle, to the driving situation, and on request even to the driver's preferences. Furthermore, in this paper current design trends and possible fields of application will be discussed.

Wed 15:20 Van Rijckevorsel Zaal Active noise control 1

Iterative learning control for an active exhaust noise attenuation valve for internal combustion engines.

Rene Boonen[a], Gregory Pinte[b] and Paul Sas[c]
[a]*KU Leuven, PMA;* [b]*FMTC, Leuven;* [c]*KU Leuven*

An active silencer has been developed to attenuate combustion engine exhaust noise. The silencer consists of an electrically controlled valve connected to a buffer volume. The pulsating flow from the engine is buffered in the volume. The valve connected to it is controlled such that only the mean flow passes to the atmosphere. This flow is free from fluctuations and consequently free of sound. An iterative learning controller has been developed to control the active valve. In the control algorithm, non-causal filters are implemented. These filters allow to control systems with high dynamics over a broad frequency range. The active silencer has been experimentally validated on a cold engine simulator, which generates realistic exhaust noise and gas flow using compressed air. The exhaust noise has been reduced in a frequency range starting from 5 Hz until 800 Hz. Depending on the rotational speed of the engine, the typical reductions ranges from 13 dB to 24 dB.

Wed 15:40 Van Rijckevorsel Zaal Active noise control 1

Practical Approach for Application of Impedance Control for Noise Attenuation

Markus Sobon
Helmut-Schmidt-University Hamburg, Inst. Mechatronics

Active noise control (ANC) becomes more and more an important issue, because of increased restrictive regulations regarding noise emission, e.g. in employment protection. Instead of using passive noise reduction methods, which cannot always be adopted, active control applications can provide efficient sound pressure attenuation in the lower frequency range. The cancellation of a disturbing primary noise field by a destructively interfering secondary sound pressure at the area of interest is widely known and has been the subject of many publications. In the early 90's several paper described the theoretical work on other control mechanisms like power absorption and power output minimization. This paper deals with a practical approach for an ANC-system that uses the radiation characteristics of a loudspeaker, which is applied as secondary sound source. The method is based on measuring active sound intensity at the centre axis of the loudspeaker membrane. For the practical implementation the signals obtained from a phase-corrected pair of sensors consisting of an accelerometer and a microphone serve as input for an adaptive control algorithm.

Wed 16:40 Van Rijckevorsel Zaal Active noise control 1

Active absorption systems: Study and implementation of an adaptive control procedure.

Marco Norambuena[a], André Jakob[b] and Michael Möser[a]
[a]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics;* [b]*TU Berlin*

The present work follows the intensity measurement technique developed by Fahy to conduct a study into active absorption systems and its practical implementation. The method separates the total sound field into two parts; an incident (p_i) and a reflected (p_r) part. This is valid in simple cases, such as a source radiating sound over a rigid surface which in response generates a reflected wave. For cases in which the wave propagates in one dimension the separation is realized efficiently, however for cases in which the wave propagates in 2 or 3 dimensions the efficiency diminishes. Measurements of absorption coefficients for pure tones as well as broad band noise were made to evaluate the efficiency of the system. The practical implementation of the system was performed using an adaptive FX-NLMS filter programmed inside a DSP chip.

Wed 17:00 Van Rijckevorsel Zaal Active noise control 1

Reduction of Interior Noise in an Automobile Passenger Compartment by Means of Active Structural Acoustic Control (ASAC)

Malte Misol and Stephan Algermissen
German Aerospace Center (DLR), Braunschweig

Results of the latest research from the automotive industry corroborate the assumption that windshield vibrations occurring in normal driving mode are a crucial factor for interior noise in automobiles. Due to the lack of passive damping methods that are compliant with the objectives of automotive lightweight construction and the specific requirements of the system under consideration, only active methods can be used in order to appropriately address the low frequency noise. Thus the research presented here has been focused on the development of a structurally integrated and compliant ASAC-system for the active reduction of the sound pressure level (SPL) inside the car's passenger cabin. In this work both adaptive feedforward and optimal state-feedback controllers have been designed and experimentally validated on a medium-class car. The use of highly efficient and global system identification methods based on structural sensors and laser-scanning vibrometer measurements allowed the definition of a global performance index for the controller design. In accordance with the numerical predictions on control performance the experimental results showed global reductions in vibration level up to 7dB and reductions in SPL up to 15dB.

Wed 17:20 Van Rijckevorsel Zaal Active noise control 1

Optimum Placement of ANC Error Microphones in Vehicles

Rolf Schirmacher, Paul Geissler, Johannes Guggenberger, Frank Steinbach and Florian Walter
Müller-BBM GmbH

To obtain a good ANC system performance, the error microphone placement plays a decisive role. The error microphone placement has to reflect the physics of the underlying acoustical properties and thus requires a good understanding of the acoustical situation.

The presentation will demonstrate the application of different concepts for optimum error microphone placements (e.g. educated guess, simulation, acoustical modal analysis) for an in-vehicle ANC system. In-vehicle measurements showing ANC performance for different microphone positions will be presented showing the effectiveness of the different approaches.

Wed 17:40 Van Rijckevorsel Zaal Active noise control 1

A Three-Dimensional Acoustic Simulation for the Development and Evaluation of Active Noise Control Systems using the FDTD Method

Christian Kleinhenrich[a], Arndt Niepenberg[b] and Detlef Krahé[a]
[a]*University of Wuppertal;* [b]*WaveScape Technologies GmbH*

The development of Active Noise Control Systems to reduce noise in certain two- or three-dimensional areas or spaces, respectively, requires good knowledge of the desired installation location as well as the understanding of the systems properties and behavior. In this paper simulation software is presented which allows the user to build virtual rooms or open air test sites and to place noise sources and active systems in an arbitrary manner to reduce the noise. A FDTD (Finite Differences Time Domain) algorithm was implemented to compute sound field behavior, including reflection, absorption, transmission and diffraction in different media. The active systems can consist of an arbitrary number of sensors (microphones) and actuators (loudspeakers) which are combined and connected via different signal paths and signal tool boxes. This allows the creation of various system setups to find the best solution and to predict the effects of noise cancellation for a given problem.

Wed 18:00 Van Rijckevorsel Zaal Active noise control 1

Broadband Active Noise Control Around Human's Head - Determination and Achievement of Physical Limits

Thomas Kletschkowski and Delf Sachau
Helmut-Schmidt-University Hamburg, Inst. Mechatronics

In order to prevent serious diseases caused by continuous noise pollution and to guarantee a restorative sleep the concept of active noise control was applied to bedrooms. To determine the physical limits of this concept in a particular situation a transmission test rig was used to simulate real world conditions. A conventional bed - placed in the reverberation room - with a dummy head microphone was used to model the sleeping person. Two error microphones - placed inside the pillow - were used for multi-channel feedforward control. The reference microphone was placed in front of a tilted window that was mounted into the transmission path. At first the physical limits of the control profit were determined by coherence-analysis. Then a proper filter length for two adaptive controller was calculated by quadratic optimization in time domain. This procedure was based on measured impulse response functions. Finally the noise reductions at the error microphones were determined for broadband disturbances 80Hz$<$f$<$480Hz. It turned out that the control profit of up to 18dB was equivalent to the noise reduction predicted by coherence-analysis.

Wed 8:40 Plate Zaal Boundary Elements 1

Simulation of transient sound radiation using the Time Domain Boundary Element Method

Michael Stütz and Martin Ochmann
Technische Fachhochschule Berlin

Many transient processes, like tire-road interaction and squeal noise, are causing strong sound radiation. That is why the calculation of transient sound radiation has become a major point of interest in acoustical engineering. The Time Domain Boundary Element Method (TD-BEM) seems to be a promising tool to deal with such problems. Instable behavior, as reported in many publications, is a major problem of the method. The influence of different mathematical and numerical approaches, like numerical and analytical integration schemes and varied finite difference approaches (central / backward) will be discussed. As in the Frequency Domain BEM (FD-BEM) internal resonances of the structure may corrupt the solution for sound radiation. The CHIEF method, which is widely used in FD-BEM to overcome this problem, seems not to work well in time domain.

Wed 9:00 Plate Zaal Boundary Elements 1

TD-BEM for Sound Radiation in three Dimensions and the Numerical Evaluation of Retarded Potentials

Elke Ostermann[a], Ernst P. Stephan[a] and Matthias Maischak[b]
[a]*Leibniz-Universität Hannover, Institute of Appl. Mathematics;* [b]*Brunel University, Uxbridge (UK)*

For the transient modeling of sound radiation the time domain boundary element method (TD-BEM) is a powerful tool, but up to now a quite unpopular approach. We present a stable marching on in time (MOT) procedure in three dimensions using a Galerkin scheme for the underlying integral equations.

The special structure of the fundamental solution of the wave equation leads to a close interaction of space and time variables in a so-called retarded time argument. The corresponding retarded potentials lead naturally to sparse matrices in contrast to the dense matrices usually associated with the BEM, but in each time step the matrix has to be stored, creating a history of matrices. The sparsity of these matrices is a result of the intersection of the support of the discrete retarded potential of the trial functions (light cones) with the support of the test functions. Thus the actual number of interacting elements is rather small. We discuss non-near-field like singularities of the retarded potentials and their consequence for the numerical evaluation of the Galerkin integrals. Moreover, we present stable numerical simulations for the transient sound radiation in three dimensions using the developed quadrature scheme.

Wed 9:20 Plate Zaal Boundary Elements 1

Auralization of noise recordings behind a simulated noise barrier

Holger Waubke, Zhensheng Chen and Wolfgang Kreuzer
Austrian Academy of Sciences, Acoustics Research Institute

The insertion loss spectra of noise barriers are calculated using the boundary element method to test the psycho-acoustical performance of planned noise barriers. Recordings were made at the reference position 25m behind the planned noise barrier at a road and a railway in 2m height. The recordings are mixed in time to avoid audible pass by events. A boundary element method in two dimensions is used. A two parametric model for grassland developed by Keith Attenborough is implemented to simulate a partially reflecting surface. This model leads to smeared sources that do not allow for the application of the Fast Multipole Method. The insertion loss of the noise barrier is calculated using the difference of the spectrum with and without a noise barrier in 25m distance and 2m height behind the barrier. The assumption of a constant line source in the 2D simulation leads to strong interference effects in the spectra. The curvature of curved noise barriers is orientated away from the source allowing a better sight into the surrounding area. The test results of the psycho-acoustic tests for 3m height curved walls are as good as the results for a 4m height straight wall.

Wed 9:40 Plate Zaal Boundary Elements 1

Separation of acoustic and hydrodynamic components of the velocity for a CFD-BEM hybrid method

Rafael Piscoya and Martin Ochmann
Technische Fachhochschule Berlin

The acoustic far field from open turbulent flames can be determined by using the hybrid approaches of aeroacoustics. These methods obtain information of the near field which is delivered by a CFD code and transfered to an acoustic solver. In the present approach, the BEM is used to determine the radiated sound. If the input velocity has a high hydrodynamic component, the sound field calculated by the BEM will be overestimated, since local non-propagating velocity fluctuations will be assumed to propagate with sound speed. Techniques to eliminate the hydrodynamic component by employing a Helmholtz-Hodge vector decomposition are known and usually applied in the time domain. In this article, we propose a method which combines the Dual Reciprocity BEM in the frequency domain with a vector decomposition, in order to eliminate the hydrodynamic part of the particle velocity.

Wed 10:00 Plate Zaal Boundary Elements 1

Benchmarking for sound transmission and scattering from thin elastic structures using analytical, BE and FE coupling methods

Ralf Burgschweiger[a], Ingo Schäfer[b], Rafael Piscoya[a] and Martin Ochmann[a]
[a]*Technische Fachhochschule Berlin;* [b]*FWG, Kiel*

The calculation of the pressure scattered from elastic structures composed of thin elastic materials is one of the main purposes for the detection of underwater objects.
For this reason, the sound pressure scattered from spherically structures ("shells") placed in and filled with fluid will be calculated using different numerical and analytical coupling methods. Also, the sound transmitted through the shell into the interior of the structure will be investigated.
We will compare and benchmark analytical solutions based on spherical wave functions with results of an in-house developed BEM-package and commercial BEM/FEM applications. Especially, the minimum limit of the possible "wall thickness" depending on frequency and material parameters will be figured out by comparing the fully elastic model with a thin-walled shell.

Wed 10:20 Plate Zaal Boundary Elements 1

Delayed Potentials: variational theory and new developments

Toufic Abboud, Maxime Pallud and Xiaodong Zhou
IMACS, Ecole Polytechnique, Palaiseau (F)

Time marching schemes have been considered for a long time very delicate when applied to time domain integral equations arising from various wave problems. Stability issue has been and seems to remain at the centre of investigations. Attempts to reach stability include use of implicit schemes, spatial and temporal averaging procedures during the time stepping, or try to extrapolate the beginning of the response signals using different techniques as Prony's or autoregressive models. This type of approach only delays the onset of the instability and/or seriously compromise the precision of the results. Recent studies are dedicated to improve the precision by using high order approximation and leapfrogging, based on extrapolation techniques for band-limited signals. Thus complete time-space variational approximation leads to unconditionally stable and precise schemes. In the last decade a new rigorous formulation for the time domain BEM has been introduced that is unconditionally stable computation without any averaging trick. This approach is based on time and space finite element formulation that preserves the energy identity. The same approach is applicable to other wave problems as electromagnetic and elastodynamic waves. We present a unified presentation of the theory with applications to vibroacoustics problems.

Wed 11:00 Plate Zaal Boundary Elements 1

Point-source-scattering from tyre-like structures above an impedance plane

Haike Brick[a], Martin Ochmann[a] and Wolfgang Kropp[b]
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In preceding publications (DAGA'08, Acoustics'08), we discussed the possibility to incorporate an infinite impedance plane into the direct Boundary-Element-Method (BEM) by means of an appropriate Green's function. This approach does not require a discretization of the impedance plane, since the Green's function fully takes into account the effects of reflection and absorption of the sound field caused by the impedance plane. The implementation was verified by analytical test cases for radiation problems. Now we focus on the scattering problem in presence of an impedance plane. The incident sound field is included in the Helmholtz-Integral-Equation by an additional summand, which is also influenced by the boundary conditions of the half space problem. This extension of the BEM-half space approach makes it possible to investigate the influence of the ground impedance on the well known horn effect of a tyre/road interface by means of BEM-simulations. The horn effect, an amplification mechanism of the horn-like geometry of the tyre/road contact area, was experimentally investigated in previous papers. These measurements serve as validation data for the BEM solutions.

Wed 11:20 Plate Zaal Boundary Elements 1

Numerical study of bat echolocation using the Boundary Element Method

Peter Juhl and Vicente Cutanda Henriquez
University of Southern Denmark

ChiRoPing (Chiroptera, Robots, and Sonar) is an EU-funded research project aimed at understanding how bats use their echolocation perception ability and apply this knowledge to the design of new robotic senses. Four species of bats are selected for the study and models of their heads including minute details of ears, mouth and nose are obtained through CT scans. The project involves, among other things, the use of numerical methods on such scanned models to study the role of their features in the bat sensorial performance. As the bats operate at very high frequencies and as their ears and noses are very complicated and involve thin surfaces, the numerical modelling is challenging. The paper reports preliminary results of the numerical study and investigates the possibilities of reducing the computational complexity of the models.

Wed 11:40 Plate Zaal Boundary Elements 1

Brief Comparison of Kirchhoff-Helmholtz-BEM and Acoustical Energy BEM for Exterior Domains

Ennes Sarradj
BTU Cottbus, Aeroacoustics Group

The classical boundary element method (BEM) in acoustics bases on the Kirchhoff-Helmholtz integral equation and is especially well suited for the treatment of exterior domain radiation and scattering problems. However, for large structures and high frequencies the calculation requires high computational effort. In such cases, the results are often unnecessary detailed and need further reduction during the post processing.
The acoustical energy boundary element method is an approach that is based on an integral equation in terms of sound intensity and sound energy density. Inherent statistical assumptions limit its use to medium and high frequencies, but the computational effort needed is much less than for classical BEM.
Using some simple example problems for radiation and scattering in the exterior domain, drawbacks and advantages for both methods are shown. The methods are compared in terms of results and computational cost.

Wed 15:00 Plate Zaal Boundary Elements 2

Fluid structure interaction and non-local admittance boundary conditions: Setup of an analytical example

Steffen Marburg and Robert Anderssohn
TU Dresden, Institut für Festkörpermechanik

In systems with fluid and structure coupled at their interfaces, the system equations are formulated in terms of variables for the fluid, e.g. sound pressure, and variables for the structure, e.g. displacement. The variables for the structure can be substituted by evaluation of the Schur complement. In that case, part of the Schur complement can be understood as a non-local boundary admittance. Herein, the authors try to provide the reader with a simple example. For that, the one dimensional problem of a duct is chosen. Single degree of freedom systems are introducted at both ends of the duct. For the setup of an example with non-local boundary conditions, both ends are connected by two springs and a rigid block. For this system, the admittance matrix will be shown analytically and we will discuss certain cases, e.g. the case of a very heavy connecting block which makes the admittance condition a local one.

Wed 15:20 Plate Zaal Boundary Elements 2

Numerical Simulation of the Sound Emission from Supermarket Delivery Trucks

Kai Schirmer[a] and Sabine Langer[b]
[a]*Bonk-Maire-Hoppmann GbR;* [b]*TU Braunschweig, Institut für Angewandte Mechanik*

In the field of environmental noise protection, the detailed prediction of sound pressure level distributions based on calculations is one of the main topics of interest. The prediction depends on different parameters. Besides the knowledge of the emission parameters, the geometric shape of the situation affects the results significantly. Standard procedure computer models use simple sound sources, an abstract model of the environment and a ray trace-algorithm to analyze the geometric situation. The pure geometric analysis by ray trace-algorithms just regards the diffraction for freefield propagation in a general way. In complex situations a large number of reflections exist as well. This leads to uncertainness for the prediction of sound fields between the free-field and the diffuse sound field. One frequent example with a main relevance is the noise-immission by delivering processes for supermarkets. This paper discusses the uncertainties of the standard procedure. The quality of prediction is improved by modeling the problem with a wave-based description that is discretized using the Boundary Elements Method. Based on these presented studies an equivalent source model for the use in standard software will be developed.

Wed 15:40 Plate Zaal Boundary Elements 2

Vibroacoustic characterization of flexible hoses for air conditioning systems

Milena Martarelli
Università Politecnica delle Marche

Flexible tubes covered with rubber are the only way to damp vibrations and noise coming from compressor in air conditioning systems. Usually they have, as core, a corrugated steel pipe, for increasing the hose stiffness, this causing fluid turbulences generating vortex separation that gives a beginning of a audible and bothersome high frequency whistle (from 2500 Hz to 5000 Hz). The vibroacoustic analysis consists on the experimental and numerical (by Finite Element Methods) characterization of the hose vibrational behaviour and consequent acoustic characterization. The experimental measurement of the hose vibration has been performed in both controlled and operating conditions, the latter with the hose installed in the test bench including all the refrigeration cycle components. The acoustic characterization has been performed via propagation of the measured surface vibration in Boundary Element Codes and validated by means of acoustic intensity measurement. The acoustic measurement in operating conditions is very challenging, being the acoustic emission of the hose comparable to the total emission of the

refrigeration bench (in primis the compressor), and therefore the hose must be isolated from the rest of the bench. Consequently the possibility of predicting the acoustic emission via BEM allows to avoid this kind of complicated measurement.

Wed 16:00 Plate Zaal Boundary Elements 2

Numerical design and testing of a sound source for secondary calibration of microphones using the boundary Element Method

Vicente Cutanda Henriquez[a], Peter Juhl[a] and Salvador Barrera Figueroa[b]
[a]*University of Southern Denmark;* [b]*Danish Fundamental Metrology*

Secondary calibration of microphones in free field is performed by placing the microphone under calibration in an anechoic chamber with a sound source, and exposing it to a controlled sound field. A calibrated microphone is also measured as a reference. While the two measurements are usually made consecutively, a variation of this procedure, where the microphones are measured simultaneously, is considered more advantageous from the metrological point of view. However, it must be guaranteed that the two microphones receive the same excitation from the source, although their positions are some distance apart to avoid acoustic interaction.

As a part of the project Euromet-792, aiming to investigate and improve methods for secondary free-field calibration of microphones, a sound source suitable for simultaneous secondary free-field calibration has been designed using the Boundary Element Method. The source has a central plug with the effect of reducing the spatial variations of the sound pressure at the microphone positions within the frequency range of the calibration. After some preliminary measurements, the source has been tested thoroughly in a calibration setup at the acoustic laboratory of the Danish Fundamental Metrology Institute (DFM). The design and verification of the source are presented in this communication.

Wed 17:00 Plate Zaal Boundary Elements 2

Applications of BEM and FMM in Acoustic Simulation at EADS

Isabelle Terrasse and Guillaume Sylvand
EADS Innovation Works

Innovation Works is, inside EADS, an entity devoted to research and development for the usage of EADS Business Units (Airbus, Eurocopter, MBDA, etc.). The numerical analysis team has been working for now more than 20 years on integral equations and boundary element methods for wave propagation simulations, first in electromagnetism, later in acoustics. Since 2000, these BEM tools have received a multipole algorithm extension that allows to solve very large problems, with tens of millions of unknowns, in reasonable time on parallel machines. The resulting software, called actipole, is used on a daily basis for installation effects computation, aeroacoustic simulation (in a coupled scheme with other tools), and many other domains of application. The aim of this talk is to present a wide view of the realizations, to underline the most recent developments and to present the main perspectives and future directions of research.

Wed 17:20 Plate Zaal Boundary Elements 2

Ducted Fan Noise Propagation Using Boundary Element Methods

Christophe Schram and Hadrien Bériot
LMS International, Leuven

The noise produced by subsonic fans is a serious concern in a number of applications, including HVAC systems, cooling devices or domestic appliances. Many of these systems are characterized by the quite complex geometry of the ducting system. As a result, the acoustic scattering must be computed numerically. The present work deals with the development and validation of accurate numerical methods for the prediction of subsonic fan noise in complex enclosures.
A particular attention is brought to the role played by near-field terms in the acoustic scattering. Some existing formulations discard explicitly the near-field terms to derive an elegant formulation for tonal fan noise. However, Roger [Fan Noise 2007] showed that the near-field terms can account for important phase-shifting effects, which can play a significant role, even in the amplitude of the acoustic far field, if for example the edge of a semi-infinite plane is present in the near field of the fan. This paper extends this analysis for the case of a ducted fan with different duct inlet geometries. The near-field formulation is implemented in a numerical acoustic solver to highlight the effect that a duct inlet geometry may have on restructuring near-field effects into propagating waves.

Wed 17:40 Plate Zaal Boundary Elements 2

BEM-model to simulate the vibrations in a tunnel in layered orthotropic media

Georg Rieckh, Wolfgang Kreuzer and Holger Waubke
Austrian Academy of Sciences, Acoustics Research Institute

Due to an increase in heavy traffic and the construction of rail roads near or in residential areas, models for the prediction of vibrations in soil become more and more important. We would like to present a 3-dimensional BEM-model of a tunnel going through a horizontally layered orthotropic material.
Until now, there is no known analytical form of the Greens-function for this kind of material, thus a numerical approximation for this function has to be calculated.
This is done in the Fourier-domain, which has the advantage that the original 3-dimensional problem can be decoupled into smaller 2-dimensional problems and that the singularity of the Green's function can be avoided. Both the fundamental stresses and the fundamental deformations are approximated with exponential ansatz functions.

The Fourier-back-transformation and the solution of the boundary integral equation (BIE) are exchanged because the special form of the ansatz allows an analytical solution of the BIE in the Fourier domain. The back-transformation from the Fourier domain is then done numerically. Finally, a graphical representation of some results is given.

Wed 8:40 Van der Vorm Zaal Audio Technology

Classification of Reverberant Acoustic Situations

Jens Schröder, Thomas Rohdenburg, Volker Hohmann and Stephan D. Ewert
Universität Oldenburg

In daily communication, speech intelligibility depends on the acoustic surrounding or acoustic situation. Particularly for hearing impaired persons, speech understanding is often problematic if speech is distorted by (room) reverb, noise or competing talkers. Acoustic situations are characterized by different dominating types of distortion. Hearing aids might provide appropriate algorithms to enhance speech intelligibility in the different acoustic situations. A robust and fast automatic classification of the acoustic situation should therefore select the appropriate hearing aid algorithm without requiring an action of the hearing aid wearer. This study is concerned with the automatic estimation of the reverberation time (T60) in natural situations and with unknown excitation signal. Acoustic test situations were generated by convolving speech signals with artificial and real room impulse responses with T60-times ranging from 0.1 to 4 s. Three different features derived from the cepstral mean, the autocorrelation function and from the distribution of modulation energy were used to train and classify multiple groups of different reverb times with Gaussian mixture models (GMM). Additionally, information about the unknown excitation signal was gained by estimation of its (audio-) spectral content and used to improve the classification. The classifiers, features and results are presented and discussed.

Wed 9:00 Van der Vorm Zaal Audio Technology

Crosstalk Cancellation in audiology

Alexandra Winkler, Thomas Brand and Birger Kollmeier
Universität Oldenburg, Medizinische Physik

Sound reproduction via loudspeakers is influenced by the crosstalk that occurs between loudspeakers and listener. Crosstalk cancellation can be used to reproduce the acoustical room impression of a recording room in a different playback room using two loudspeakers. In order to achieve this, the contralateral pathways need to be cancelled and the ipsilateral pathways need to be equalized. In this study different algorithms for crosstalk cancellation were evaluated by assessing the localization performance in the horizontal plane and by speech intelligibility tests in spatial noise with 4 normal-hearing listeners. The experiments took place in two different playback rooms (acoustically damped and with reflections). The results showed that crosstalk cancellation yielded an

improvement up to 9dB of speech intelligibility in noise when the speech was spatial separated from the noise even in reverberant rooms. Using crosstalk cancellation, listeners were able to localize stimuli outside the triangle between them and the loudspeakers ($\pm 20^\circ$) which was not possible without using crosstalk cancellation. The use of crosstalk cancellation makes it possible to perform speech intelligibility tests and localization measurements in reflecting rooms in a similar way as in anechoic rooms or like using head phones.

Wed 9:20 Van der Vorm Zaal Audio Technology

Improvement of speech intelligibility by audio hearing systems

Hannes Seidler

TU Dresden, Dep. of Medicine

All hearing aids and cochlea implants have algorithm to improve the speech intelligibility. The idea is to share human speech and noise and to find different ratings. The result should be easier to understand by impaired people.

On the market there are big efforts to recognize speech and to separate it from noise. In this field we can notice a remarkable progress in the last years. But the reverberant sound in rooms or noise from same direction like the signal need a lot of work to suppress them effective.

The simplest way seems to be to gets the original speech signal direct from the source and to processes it individual in hearing devices. Induction loop systems for hearing aid users are well known in churches, cinemas, theatres and conference rooms. Other possibilities are wireless infrared or HF systems. Always it is easy to get an improvement of speech intelligibility. If these systems are used the hearing can be more relaxed and self-evident.

The presentation will show how audio hearing systems (Assistive Listening Devices) improve the signal to noise ratio or the STI in theory and measurements and what the benefit for hearing aid or cochlear implant users is.

Wed 9:40 Van der Vorm Zaal Audio Technology

Measurement, modelling and compensation of nonlinearities in hearing aid receivers

Timm Schaer[a], Stephan D. Ewert[a], Jörn Anemüller[b] and Birger Kollmeier[b]

[a]*Universität Oldenburg;* [b]*Universität Oldenburg, Medizinische Physik*

Hearing aid receivers operate over a large dynamic range and reach output levels of up to 130 dB. Electromagnetic receivers are commonly used in hearing aids since they offer a very high efficiency. Receivers are typically characterized by their linear transfer function. These linear distortions are easily measured with standard methods and if compensation is applied it is often limited to the magnitude transfer function. The current study addresses a more complete characterization of the

hearing aid receiver by measurement of the nonlinear transfer function. Nonlinear distortions of the receiver are output level dependent and might negatively influence measures like speech intelligibility of an aided hearing impaired person, particularly at high levels. Moreover, nonlinear distortions are problematic in the context of future "high-fidelity" hearing aids and compensation methods are of interest. Here, a fast and efficient sinesweep method proposed by Farina was employed to estimate the linear and nonlinear transfer function. The measurement was repeated at different output levels. From the data, nonlinear input-output functions can be constructed or alternatively, the diagonal elements of a Volterra-series based description of the nonlinear transfer function can be derived. A nonlinear receiver model and compensation methods are suggested and tested.

Wed 10:00 Van der Vorm Zaal Audio Technology

Audio Event Detection for In-Home Care

Peter van Hengel[a] and Jörn Anemüller[b]

[a]*Fraunhofer IDMT;* [b]*Universität Oldenburg, Medizinische Physik*

One in eight persons attending hospital following an accident in the home, are aged 65 and over. This number will increase with the demographic change and the desire of the elderly to remain in their own homes. Hence, the demand for unobtrusive monitoring systems based on autonomous sensor technologies will increase. Such systems can limit observation by a human operator to cases when there is evidence of an incident, by responding to events in a 'human-like' fashion. To do so, the system must pick up situations that also would attract human attention, including situations it has not encountered before. Hearing is an important sense in terms of steering the attention of a human observer. For this reason the detection of sound events that could indicate an incident is a key part of such a monitoring system. Formerly, an acoustic system was designed for the detection of verbal aggression. This technology is now expanded, combined with acoustic localization and integrated with video analysis and artificial intelligence. The resulting methods will be tested in the context of in-home care under real-life conditions. For this purpose a realistic test-home has been constructed.

Wed 10:40 Van der Vorm Zaal Bio-acoustics

A Three Dimensional Model of the Organ of Corti of the Guinea Pig

Johannes Baumgart[a], Mario Fleischer[b], Roland Gärtner[b], Anthony W. Gummer[a] and Axel Voigt[c]
[a]*TU Dresden, Institute of Scientific Computing;* [b]*TU Dresden, Institute of Solid Mechanics;* [c]*University of Tübingen, Physiological Acoustics and Commun.*

For the function of the inner ear the mechanical linkage of basilar membrane displacement to the deflection of the inner hair cell stereocilia is essential and investigated here. Besides the passive excitation by a pressure on the basilar membrane also the excitation by electromotile outer hair cells as well as by the hair bundles of the outer hair cells is studied.
Based on geometrical data of the guinea pig and experimental velocity measurements the organ of Corti is modelled by means of the finite element method. The fluid and the structure are modelled as linear material with small displacements. The interaction of the fluid and the structure is implemented as a matrix coupling. The geometry is extracted from laser scanning microscope stacks and material properties are taken from the literature and adjusted to match experimental results.
This model provides a helpful tool to understand experimental data more in detail. It is possible to look for the influence of geometrical variations and material properties. Different configurations and boundary conditions are modelled and the relation of excitation to inner hair cell stereocilia displacement due to bending is analysed.

Wed 11:00 Van der Vorm Zaal Bio-acoustics

Acoustic Streaming in a Viscous Fluid-Structure System

Frank Böhnke[a] and Christian Gerstenberger[b]
[a]*HNO-Klinik der TU München;* [b]*Leibniz-Universität Hannover, Inst. für Mensch-Maschine Komm.*

The numerical evaluation of fluid-structure coupled systems in complex geometries, e.g. the acoustical wave propagation in the inner ear of mammals, requires elaborate mathematical formulations. One of these is the consideration of the stimulating boundary condition by an acoustic radiator (JASA, Bradley C.E. 1996), which leads to linear behaviour in Lagrangian but is nonlinear in an Eulerian description. In coupled systems the interaction between the structure guided Rayleigh wave and the fluid causes an internal acoustic streaming (AS). AS effects are a result of the nonlinearity of the Navier-Stokes equations in combination with the viscosity. For a numerical evaluation different discretizations are used. A complete Finite Element (FEM) approach is computationally too extensive at least when three-dimensional formulations are used. The fast BEM-FEM mortar coupling for acoustic-structure interaction was recently formulated but the acoustic field is governed by the Helmholtz equation and therefore viscosity was not considered. Another approach which is mainly used for the simulation of AS effects in compressible, viscous gas filled two-dimensional enclosures is the Flux-Corrected Transport (FCT) algorithm to solve nonlinear, time-dependent continuity equations applying a Finite Volume Method (FVM). Special care has to be taken with the formulation of the density boundary condition in this case.

Wed 11:20 Van der Vorm Zaal Bio-acoustics

Call Directivity Modification by the rufous horseshoe bat, Rhinolophus Rouxi: Simulated Hypotheses.

Fons de Mey, Dieter Vanderelst and Herbert Peremans
Universiteit Antwerpen

In this paper we investigate a possible echolocation strategy for the rufous horseshoe bat (*Rhinolophus rouxi*). As this is a cf-fm bat, it concentrates most energy in the single frequency cf-component of its call. Therefore it cannot exploit a frequency dependent emission pattern focusing energy into different spatial regions, as is done by fm-bats using frequency modulated calls. This limits spatial information gathered from a single call. It has been conjectured that the typical pinna movements displayed by cf-bats would enhance their localization capacity. In this paper, we focus on two alternative means of generating additional spatial information, based on call directivity modification: on the one hand performing a controlled noseleaf movement, on the other, changing the phase between the emissions produced from the two nostrils. The latter mechanism would in effect constitute a phased array (2 elements approximately half a wavelength apart) allowing simple beam steering. We show how realistic noseleaf movements affect the call directivity. To this end, we simulate the acoustic field produced by the bat's emission system using a 3D boundary element method. In addition, we explore what phase shifts are required to produce useful beam steering when considering the emitter a phased array.

Wed 11:40 Van der Vorm Zaal Bio-acoustics

Auditory time resolution in dolphins in comparison with humans

Gennadi Zaslavski
Tel-Aviv University, Univ. authority for applied research

To emphasize more than tenfold difference in auditory time resolutions, electrophysiological estimates of dolphin's auditory integration time of about 0.3 ms (for example Supin et al. 2001) is usually compared to humans' minimum single detectable pause of 3-5 ms in continuous noise. However, there are many indications that the humans auditory time resolution can be as high as 0.5-1 ms (Schouten et al. 1962, Harris 1966, Viemeister et al. 1996, Purcell et al. 2004), or even 0.2 ms (Henning and Gaskell 1981) although the best estimates do not appear to be widely acknowledged. The general impression is that the shorter the stimuli used, the higher the humans' auditory time resolution. In this respect the time resolution of dolphins' broadband sonar system that uses very

short (0.02-0.05 ms) echolocation clicks could be even higher. Numerous behavioural results do indicate that bottlenose dolphins' auditory time resolution is as high as 0.02-0.03 ms. However, a bottlenose dolphin ability to analyse time domain features of pulsed signals appears to be even better than can be described by the auditory time resolution. In this paper we will discuss some differences in auditory discrimination of brief signals by human listeners and bottlenose dolphins.

Wed 15:00 Van der Vorm Zaal Aeroacoustics on vehicles

Complete Benchtest dedicated to Acoustic Array Measurement in S2A Wind-tunnel: 3D Maps and source listening.

Aurelie Chapelle[a], Nathalie Gorilliot[a] and Olivier Coste[b]
[a]*GIES2A Wind Tunnel;* [b]*Signal Developpement, Poitiers*

In order to keep a worldwide competitiveness and to guarantee the best acoustical comfort of their vehicles, the French car manufacturers invested and built new wind-tunnels in 2003: "les souffleries aéroacoustiques automobiles", S2A. There are several types of aerodynamical noise sources; they are generated for example by accessories across the flow, cavity or ruggedness on surface, and they can be observed all around the car. Today, in a wind-tunnel, Beamforming method is the best technique to locate those sources, quantify them and organize them into a hierarchy of noise level. To answer this need, new measurement and analysis systems are implemented in the wind-tunnel. Thus a side array (with optimized random geometry and 64 microphones) and a horizontal array (with multi arm logarithmic spiral geometry and 88 microphones) carried by a mobile support set up a complete benchtest dedicated to acoustic array measurement. The considerable volume of data is processed by quad-core calculators allowing a fast display of the aeroacoustic sources on 3D maps, a direct listening of the sources and the improvement of the system's performance with deconvolution's algorithm. This paper describes this new system.

Wed 15:20 Van der Vorm Zaal Aeroacoustics on vehicles

Investigations of Sunroof Buffeting

Jörg Ocker
Porsche AG

This paper describes an investigation of the feasibility of predicting sunroof buffeting by means of computational fluid dynamics (CFD), carried out by a consortium of the German automotive manufacturers Audi, BMW, Daimler, Porsche and Volkswagen in collaboration with the software vendors CD adapco and Exa.
The general physical mechanism of sun roof buffeting is well understood and pragmatic design solutions for suppressing buffeting are well known. However, experience in vehicle development has shown that making a priori predictions with the required degree of reliability and accuracy is not possible.

For a systematic analysis of these issues, the consortium devised a project to investigate only the issues related to fluid dynamics, and ensures that other factors play no role. Therefore an idealised generic vehicle model, based on the SAE Type 4 body was designed an built.
Aerodynamic and aeroacoustic investigations in various configurations are carried out in two different wind tunnels, applying both acoustic measurements with microphones as well as quantitative flow visualisation with high speed stereo PIV.
Two different commercial CFD codes were applied for the simulations: A finite volume code with a large eddy simulation (LES), and a lattice Boltzmann code with a very large eddy simulation (VLES) approach.

Wed 15:40 Van der Vorm Zaal Aeroacoustics on vehicles

A Procedure to Simulate the Turbulent Noise Interior of Cars

Susanne Krampol, Matthias Riegel and Jochen Wiedemann
FKFS Stuttgart

Vehicle development usually are carried out in windtunnel with smooth flow conditions, the velocity vector is nearly constant in time and space. On road the flow vector depends on the vehicle speed and the atmospheric wind velocity, the flow is more turbulent due to the climatic and local conditions (e.g. scenery, traffic). Because the turbulent flow simulation in the windtunnel is limited due to space limitation, an alternative way to simulate the turbulent noise interior of cars is to synthesize the noise by mixing sound samples which were measured in the windtunnel at different yaw angles and for different wind velocities regarding typical flow situations on road. By this procedure it should be possible to make objective evaluation of cars under turbulent flow situations as well as subjective ratings.

Wed 16:40 Van der Vorm Zaal Aeroacoustics on vehicles

Towards Industry-Standard Flow-Induced Noise Prediction

Christophe Schram[a], Paula Martinez[a] and Geraud Guilloud[b]
[a]*LMS International, Leuven;* [b]*TNO Science and Industry*

Engineers are nowadays predicting flow-induced sound in order to obtain quantitative assessment of complex devices in industries that now encompass air and ground transportation, energy sectors or domestic appliances. However, a fair amount of know-how still has to be captured to reach the decibel-accuracy sought by industrials for realistic configurations. A particular concern is the assessment of the numerical requirements of both the flow and acoustic modeling, in relation with the desired prediction accuracy over a pre-defined frequency range.
The present work is focused on the hybrid approach inspired from the acoustic analogy. A crucial issue regards the reliability of the flow data. In that respect, designing a mesh and selecting a numerical scheme for a given targeted noise frequency range can be complicated. A related diffculty lies in the data transfer that takes place from the CFD code to

the acoustic tool. The problem consists in transferring data from a fine mesh towards a considerably coarser mesh without significant loss of information. Hereto, conservative interpolation schemes are applied to optimize the effciency of the process. The optimized procedure is applied to the case of a rectangular diaphragm fitted in a square duct, for realistic Reynolds and Mach numbers.

Wed 17:00 Van der Vorm Zaal Aeroacoustics on vehicles

Evaluation of Aerodynamic Noise Propagation: Analyzing the Transmission for a Side Window into the Interior of a Car by Coupling CFD and AeroAcoustic Propagation Tools

Dirk Clasen[a], Thorsten Grahs[a], Romain Leneveu[b], Julien Manera[b] and Stéphane Caro[b]
[a]*Volkswagen AG;* [b]*Free Field Technologies*

The main objective of this study is to compute the sound field inside of the car cavity caused by the air flow surrounding a car. CFD computations alone can only give insight on the acoustics outside the car but such results are difficult to use for design purposes. In this paper a methodology is presented which gives access to the acoustic field inside of the car. A two-steps approach is used as first proposed by Lighthill. First, the CFD calculation is conducted using the OpenFOAM DES solver in compressible mode. This computation gives access to the Lighthill tensor in the volume downstream the mirror, and to the pressure fluctuations on the side window itself. Based upon these values the sound propagation into the car is calculated in a second step using the acoustic propagation code Actran/LA. The visco-elastic structure of the glass and its seal support is modeled in details using a standard, fully-coupled vibro-acoustics strategy. Two strategies are compared together, one relying on the wall pressure fluctuations as a source, and the other one relying on the Lighthill tensor. Thanks to existing measurement results, conclusions on the most appropriate strategy for this type of runs are then drawn.

Wed 17:20 Van der Vorm Zaal Aeroacoustics on vehicles

A hybrid method for Computational Aeroacoustic applied to internal flows

Mélanie Piellard[a] and Christophe Bailly[b]
[a]*Delphi Thermal, Bascharage (L);* [b]*LMFA, ECL & UMR CNRS 5509*

A hybrid method of aeroacoustic noise computation based on Lighthill's Acoustic Analogy is first assessed, and then applied to investigate the noise radiated by a low Mach number flow through a ducted diaphragm [1]. The simulation method is a two-step hybrid approach relying on Lighthill's acoustic analogy, assuming the decoupling of noise generation and propagation. The first step consists in an incompressible Large Eddy Simulation of the turbulent flow field, during which the Lighthill's source term is recorded. In the second step, a variational formulation of

Lighthill's Acoustic Analogy using a finite element discretization is solved in the Fourier space. The assessment of this method consists in validating the formal linking from the flow to the acoustic computations; the expression of the source term in Lighthill's Acoustic Analogy is also discussed and numerically demonstrated. The method is applied to a three-dimensional ducted diaphragm with low Mach number flow; a good agreement is found regarding aerodynamic results with Direct Noise Computation performed by Gloerfelt & Lafon [2]. However, the acoustic results prove the difficulties of applying a volume approach to practical flows, mainly due to the large computing resources required.

[1] Piellard & Bailly. 2008. AIAA Paper 2008-2873.
[2] Gloerfelt & Lafon. 2008. Computers & Fluids 37(4), 388-401.

Wed 17:40 Van der Vorm Zaal Cavitation 1

Bubble Motion and Jetting at Sonotrodes

Till Nowak and Robert Mettin
Drittes Physikalisches Institut, Universität Göttingen

The motion and oscillation of bubbles in water near sonotrode tips at 20 kHz and 60 kHz are observed by high-speed imaging. During collapse, some bubbles show a repetitive jetting and jumping behavior. The bubbles can undergo surface distortions and splitting during collapse, but they survive and partly recollect fragments in the subsequent expansion phase. Caused by the shape perturbations, some bubbles have been observed to re-expand fully only after one more acoustic cycle, thus exhibiting a period-two oscillation. Bubble oscillation and translation at both frequencies appear very similar, and they correspond reasonably well to numerical simulation by a spherical bubble model.

Wed 18:00 Van der Vorm Zaal Cavitation 1

Bubble-Shock Wave Interaction

Mohsen Alizadeh, Thomas Kurz, Dennis Kröninger and Werner Lauterborn
Drittes Physikalisches Institut, Universität Göttingen

The interaction of shock waves with a single bubble is studied by means of high speed photography and pressure measurements. The bubbles are produced by optical breakdown using a Q-switched Nd:YAG laser. A piezoelectric shock wave generator is applied as the source of the shock waves. The focus of the shock waves is located at the position of the bubble. In order to capture the very fast bubble dynamics with high resolution, a high-speed camera attached to a long distance microscope is used. The pressure field is measured with a Fiber Optic Hydrophone. Image sequences of the bubble dynamics are obtained at different delay times between the breakdown and the arrival time of the shock at the bubble. They show a reduction of collapse time by the shock wave which is more pronounced when the shock hits the bubble at a time when the bubble is already collapsing. The shock wave produced by the collapse itself is enhanced compared to the case of free oscillation. The shock leads to aspherical bubble collapse. A jet is generated in the direction opposite to the shock front which is stronger when the shock impinges the bubble in its collapse phase.

Wed 8:40 Hudig Zaal Source identification 1

Sound Source Identification and Noise Reduction of a Reciprocating Compressor

Jozèf van den Oetelaar[a] and Bert Roozen[b]
[a]*GEA Grasso B.V., 's-Hertogenbosch;* [b]*Eindhoven University of Technology*

Source identification is an important aspect of noise control engineering. A noise source can be very complex in nature. For control purposes it is rather important to identify the strongest contributing noise sources. This paper discusses the use of several structured methods on a reciprocating compressor in order to reduce the noise level significantly.
The measurement approach is quite logical, structured and practical, allowing an efficient identification of the contributing noise sources. Once the most contributing noise source is identified, the measurement technique also allows predicting the maximum achievable noise reduction of the machine as a whole and the way how to achieve this reduction. A BEM analyses shows areas for further optimizing the most significant noise source.
As a result, the sound power level of the reciprocating compressor is not only reduced with 7 dBA. Moreover, from a perceptual point of view the sound itself can be characterized as ”solid” as well. Another 2 dB reduction is achievable.

Wed 9:00 Hudig Zaal Source identification 1

Comparision of various Array Geometries with Respect to their Depth of Field for acoustic Mappings

Olaf Jaeckel, Ralf Schröder and Dirk Döbler
GFaI e.V.

The wide-spread use of planar microphone arrays for acoustic 2D-beamforming mappings does not consider the influence of actual three-dimensional object surface structures on the calculated maximum sound pressure levels. To quantify the effects of the very complex interactions between array- and objectgeometry in more detail, several investigations of common two- and threedimensional arrays with varying focus distances have been performed. All simulations were done using the correct spherical wave model for point sources for the distance calculations. In the simulations, ring- and spiral arrays as well as spherical 3D-arrays will be compared. Measurement examples for some of the arrays will also be shown and will be related to the simulations. The results show that most arrays have a remarkably strong dependence on the focal distance actually used for the beamforming calculation. Especially for the case of beamforming within 3D-interiours, which is very difficult to handle due to strong individual distance variations, it is therefore inevitable to take the actual object-geometry into account to get useful results at all. So the

use of 3D-CAD-models for acoustic 3D-mappings is a well-founded decision, even if it makes software-implementations much more complex than the easy but oversimplifying use of just planar beamforming maps.

Wed 9:20 Hudig Zaal Source identification 1

Source Identification in Practice, Two Cases with Tonal Noise

Theo Campmans
Lichtveld Buis & Partners

Source identification is of great relevance when annoyance is reported due to tonal noise. Tonal noise is in general extra annoying for the environment. In those cases there is an urge to find the cause of nuisance and solve it. Beside the aspect of the constant frequency, there can be other characteristics that identify the responsible installation that causes the reported annoyance.
In the first case, a source was not only identified by the frequency of the tonal sound, but also by the time pattern in which the installation under consideration is turned on and off. Such time patterns can help in finding a cause, but they can also be misleading. A proper handling of the data is therefore required.
In the second case it was not a real problem to find the responsible installation. The tonal noise started after a well-defined adaptation of the installation. This pointed directly towards the responsible installation. But in order to understand the causing mechanism, the known frequency of the source proved to be helpful in identifying the source mechanism. Such an analysis proves to be a proper basis for designing appropriate noise reducing measures.

Wed 9:40 Hudig Zaal Source identification 1

Estimating the velocity profile and acoustical quantities of a harmonically vibrating membrane from on-axis pressure data

Ronald Aarts and Guido Janssen
Philips Research, Eindhoven

Power series expansions are derived for acoustical quantities of a harmonically excited resilient, flat, circular radiator in an infinite baffle. These quantities are the sound pressure: on-axis, far-field, and at the edge of a radiator, as well as the reaction force on the radiator and the total radiated power. These expansions are obtained by expanding the velocity distributions in terms of orthogonal polynomials. For the rigid, simply supported and clamped radiators, this results in explicit, finite-series expressions involving (spherical) Bessel functions for both the on-axis and far-field pressure. In the reverse direction, a method of estimating velocity distributions from (measured) on-axis pressures by matching in terms of expansion coefficients is proposed. Together with the forward far-field computation scheme, this yields a method for assessment of loudspeakers in the far-field and of the total radiated power from (relatively near-field) on-axis data (generalized Keele schema). The relation between the radiated power as obtained from near-field measured sound pressure data via the theory on one hand and from a far-field measurement in a reverberation room on the other is highlighted.

Wed 10:00 Hudig Zaal Source identification 1

Spectral Analysis of Audio Signals for the Identification of Spam Over IP Telephony

Christoph Pörschmann and Heiko Knospe
Fachhochschule Köln

With modern computer and telecommunication systems voice calls can be automatically set up and prerecorded speech messages can then be played. Especially in IP-based networks the costs for voice calls are quite low, Spam over IP Telephony (SPIT) can become a serious problem in the near future. Most activities ongoing in order to identify SPIT and to protect telephone networks from being flooded with SPIT make use of black or white lists. In this talk, a method adapted from the area of music identification is proposed which can be used to identify SPIT and to automatically establish black-lists. From the audio speech data of voice calls an "acoustic fingerprint" is calculated. The "acoustic fingerprints" of the voice calls are compared to each other and those are identified which have a high degree of similarity. The method, which is resistant to various modifications of the audio signal, can be used to detect SPIT which is typically characterized by similar or identical voice data in a large number of calls. Privacy protection is assured since only a "fingerprint" but not the complete audio data is stored, which does not permit the reconstruction of the content or the identification of the speaker.

Wed 10:20 Hudig Zaal Source identification 1

Using the Fast Multipole Boundary Element Method to Update Loads in System Level SEA Models

Sebastian Müller[a], Vincent Cotoni[a] and Anders Wilson[b]
[a]*ESI US R&D;* [b]*ESI Group*

Automotive interior noise at mid and high frequencies is dominated by the airborne noise from acoustic sources spatially distributed around the vehicle. Given the vehicle size and the frequency range of interest (typically above 400 Hz), it is standard practice to use Statistical Energy Analysis (SEA) models of the entire vehicle to help designing for interior noise. Accurate description of the acoustic loads on the exterior SEA subsystems (windows, windshield, roof, etc...) is then required.
Exterior sound pressure fields can be efficiently modeled with the boundary element methods (BEM). However, because of the computational expense, standard BEM is limited to fairly low frequencies for full vehicle geometry. The Fast Multipole Method (FMM) allows to extend the BEM to reach the frequency range of interest for airborne noise applications.

This paper demonstrates and investigates the use of Fast Multipole BEM to calculate the automotive exterior acoustic loads at mid and high frequencies for inclusion in a system level SEA model. Numerical experiments are performed to define some guidelines regarding the sensitivity of the prediction to (i) geometrical details, (ii) mesh density, (iii) frequency resolution, (iv) surface impedances, and (v) details of the source description.

Wed 11:00 Hudig Zaal Source identification 1

Mechanisms of secondary sound source generation by air jet actuation in an axial Fan

Olaf Lemke[a], Philip Kausche[b], Wolfgang Neise[b], Lars Enghardt[b] and Michael Möser[c]
[a]*TU Berlin, Sonderforschungsbereich 557;* [b]*German Aerospace Center (DLR), Berlin;* [c]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics*

For subsonic blade tip speeds the radiated acoustic field of axial fans is typically dominated by discrete tones at the blade passage frequency (BPF) and its harmonics. In active noise control applications typically loudspeakers are used to generate the required anti-phase sound field. The space requirement and the weight of the loudspeakers can hinder practical application of this method. In this study, aerodynamic sound sources are used to cancel out tonal noise components of an axial fan. This is achieved by disturbing the flow field around the blade tips by compressed air injection via wall flush mounted nozzles. The resulting aerodynamic sound sources can be adjusted in both amplitude and phase by controlling the injected mass flow and the circumferential nozzle position. To gain a better understanding of the aeroacoustic secondary source generation mechanisms, pressure measurements were done between rotor and stator. Pressure transducers with high dynamic range were implemented in the fan casing wall in the rotor blade trailing edge region. First results show significant changes in the pressure filed on rotor blades tips if air jet actuation is applied. This leads to additional unsteady periodical forces on the rotor blades surfaces which can be interpreted as secondary sources.

Wed 11:20 Hudig Zaal Source identification 1

Methods for the identification of background noises and noise events

Dominic Hemmer, Christoph Pörschmann and Pascal Korte
Fachhochschule Köln

Noise is one of the main reasons for an increasing number of public complaints. This is why sound immissions caused by aircraft are supervised by noise long-term measurements. Yet, in many cases one is confronted with the problem that at the place of immission several noise sources, like noise from aircrafts and noise from street traffic, are mixed. In this case the contribution of aircraft noise needs to be detected from the recorded noise immissions. Within this presentation, methods are presented, which make it possible to separate the singular noise components by digital audio signal processing. Therefore, spectral and temporal characteristics of the audio signal are analysed and assigned to miscellaneous noise categories. One approach for this is to build a feature vector from several parameters and to compare it to vectors of reference signals. Different noise categories can thus be separated from one another. In a next step the noise contribution is automatically assigned to several emission sources. However, certain boundary conditions need to be considered, in which noise from different noise sources occurs at the same time. Furthermore, a high identification rate needs to be assured.

Wed 11:40 Hudig Zaal Source identification 1

Noise Source Identification in a Cross-Section of a Long-Range Airliner by Means of the Inverse Finite Element Method

Matthias Weber, Thomas Kletschkowski and Delf Sachau
Helmut-Schmidt-University Hamburg, Inst. Mechatronics

An inverse finite element method (IFEM) for noise source identification in closed cavities is presented. The procedure requires sound pressure measurements in the cavity first. If all sound sources are located on the boundary, the equation system resulting from a matching finite element (FE) model can be resorted in such a way that computation of the unknown boundary pressure and particle velocity is possible by solving the emerging inverse problem. Regularization techniques must be applied to account for the ill-posedness of the problem.
The method was first validated using a simplified three-dimensional FE model. The numerically calculated data inside an inner sub-domain were impinged with a stochastic error and used as simulated measurement data to re-calculate the boundary data.
In a next step, the IFEM was applied to an aircraft mock-up, consisting of a cross-section of a long-range airliner. The sound field attuning in the cavity with an interior noise source was mapped with a custom-built microphone array. A matching FE model was developed, and the sound pressure and particle velocity prevailing on the boundary were calculated from the measured data.

Wed 15:00 Hudig Zaal Source identification 2

Scalable Microphone Array System Optimized for Streaming Inverse Acoustics

Joost Kauffman
University of Twente

Sound source localization by using large microphone arrays commonly requires expensive hardware and an impractical large amount of cables and connectors. Our aim is to develop a compact and low-cost acquisition system that is scalable and expandable to hundreds of channels. Our system consists of a ring network of sensor modules, each equipped

with 16 input channels. There can be up to 16 modules linked into a single network, providing 256 input channels. One of the sensor modules interfaces with a PC through USB and functions as a master. Standard UTP cables are used to link the modules together for data transport and power supply. This principle reduces both weight and cost of cables. Multiple ring networks can be connected to a PC to enable further expansion of the array. As the raw data stream is proportional to the number of elements in the array, data rates can exceed the maximum bandwidth of standard computer interfaces and storage. To overcome this problem, each module has a buffer for 30 seconds of raw data storage. Additionally, each module has a DSP for optional signal processing. In the current configuration we preprocess each channel in order to reduce data rates and enable inverse acoustics as a streaming application.

Wed 15:20 Hudig Zaal Source identification 2

Source Localization Techniques with Acoustic Arrays

Henk van der Wal and Pieter Sijtsma
National Aerospace Laboratory NLR

Acoustic antennas are useful for source localization, both close to and away from the source. NAH (Nearfield Acoustic Holography) is an experimental method for source localization and quantification, measuring in the acoustic nearfield of the source. On the other hand, the beamforming technique is suited for the farfield. The presentation will address both methods. The NAH technique was applied on a composite fuselage barrel. At NLR and DNW, beamforming is applied on routine basis on stationary and rotating sources in wind tunnels. Outdoors, moving sources have been measured as well, both translating (aircraft flying over) and rotating sources (wind turbines). For these sources, moving and rotating focus was implemented in the post processing. In addition, techniques were developed to increase the accuracy of the source strength of the partial sound sources, determined from the array measurements (CLEAN-SC). A new development is in-duct beamforming, localizing and quantifying sources on the rotor and stator of an aero engine.

Wed 15:40 Hudig Zaal Source identification 2

Measurement results of a new slices wheel array design with detachable microphone sectors

Christoph Hundeck
Brüel & Kjaer GmbH

The SONAH (Statistical Optimized Nearfield Acoustic Holography) noise source identification technique is a well introduced technique to determine acoustic phenomena at low frequencies - therefore it is very suitable for combustion engines. Usually a planar array is placed as close as possible in front of the engine to get an optimal spatial resolution. Many products do not easily allow for direct access to large contiguous surfaces as required by traditional planar arrays. For example when measuring on engines some obstacles like fuel lines, electrical connections and even the exhaust can be modified to ease access, but the drive shafts represent and obstacle that cannot be removed. To accommodate for this a revised slice wheel array design allows parts of the array to be removed so the array can be placed at the optimal measurement. Some measurement results with this optimized array design will be discussed.

Wed 16:00 Hudig Zaal Source identification 2

Noise induced vibration of a thin projection screen?

Bert Roozen[a] and Luuk Yperlaan[b]
[a]*Eindhoven University of Technology;* [b]*Philips Healthcare, Best*

”Ambient experience” is a selling feature within Philips Healthcare to soothe down the patient during for instance an X-ray computer tomography (CT) scan. One aspect of ”Ambient experience” is to project a video on the walls and the ceiling of the examination room of the CT-scanner. The projection on the ceiling of the examination room is realized by means of a beamer mounted above a relatively thin projection screen, imaging the still picture or video through this thin projection screen.
Complaints were received because of vibrations of the projection screen mounted in the ceiling, resulting in visible waving of the video. This waving effect appeared to be related to the rotation of the gantry's rotating turn plate. Pragmatic tests are reported aiming at a better understanding of the physical phenomena involved, questioning ourselves; is this an acoustic effect?

Wed 17:00 Hudig Zaal Source identification 2

Contribution of stiffeners to the sound transmission of aircraft skin panels

Jelmer Wind, Arjan Bronsvoort and Andre de Boer
University of Twente

Traditionally, the sound radiation of aircraft skin panels is calculated using a structural model with stiffeners and an acoustical model where the stiffeners are omitted. We present a numerical study on the impact of this omission.
The numerical model consists of three parts. Firstly, the pressures in a diffuse sound field are modeled. Secondly, a Finite Element (FE) model calculates the structural response caused by the sound field. Thirdly, a Boundary Element (BE) model calculates the sound intensity. The impact of stiffeners is determined by comparing the results of a BE model with and without stiffeners.
The emphasis lies on the results of this model. Several stiffener configurations and frequencies are compared and a number of new analysis techniques are introduced to pinpoint the cause of these differences. Differences of upto 50% have been found for the geometries that have been tested.

Wed 17:20 Hudig Zaal Source identification 2

Inverse Determination of Ship Propeller Source Strength

Erik van Wijngaarden
MARIN, Wageningen

Ship trials aimed at the determination of propeller-induced hull excitation forces are often performed with a limited number of hull-mounted pressure transducers from which no accurate resultant force can be integrated. Instead, the result is usually given in terms of maximum pressure amplitude on the afterbody at several blade passage frequencies. These amplitudes are then compared with contract requirements stated in the same terms. This procedure may easily lead to a wrong qualification of the propeller cavitation as a source of inboard noise and vibration. It is proposed to solve this problem by qualifying the propeller on the basis of its source strength. A boundary element method is presented for the inverse determination of the propeller source strength and location, given a set of hull-pressure measurements. The method is validated against experimental data of cavitating and non-cavitating propellers. The same method is used in combination with another boundary element method for the analysis of the cavitating propeller to determine the hull-pressure field and the resulting hull excitation forces. Finally, the pressure distribution and integrated forces on the ship hull thus found are compared with the ones found from the inversely determined simplified propeller source description.

Wed 17:40 Hudig Zaal Source identification 2

Threedimensional Beamforming considering the Effects of covered Microphones

Olaf Jaeckel, Dirk Döbler, Ralf Schröder and Andy Meyer
GFaI e.V.

Beamforming onto complex geometric structures in three dimensions must take into account surface areas from which the emitted soundwaves can not reach the microphone array directly. Similar to optics, the beamformer is faced with the problems of ”full shade” (no microphone has a direct path to the acoustic source) and ”half shade”, i.e. the acoustic source is only visible partly by a certain subset of all microphones. The article presents two methods to geometrically take those areas into consideration in 3D-beamforming. The first method is a relatively simple one that can only determine either full shading or full visibility of some parts of the object geometry. It gives good results in most measurement scenarios when the microphone array is compact or has a comparatively small diameter, e.g. with a small spherical array in car interiours. The second solution presented is computationally far more intensive but also more general. This method does also consider the ”half shade” baffle effects and therefore it also allows measurements with distributed microphone arrays onto 3D-structures from the outside. The necessary modifications of the beamforming-algorithm for both cases are presented

and the resulting effects will be discussed using practical measurement examples.

Wed 18:00 Hudig Zaal Source identification 2

Near field sound source signal separation using a single acoustic particle velocity vector sensor

Erik Druyvesteyn[a], Jelmer Wind[b] and Michiel Ligtenberg[b]
[a]*Microflown Technologies;* [b]*University of Twente*

A method is presented to determine the contributions of incoherent sound sources based on measurements from a single acoustical vector sensor (vx,vy,vz). Well-known signal processing techniques such as Principal Component Analysis (PCA) can separate multiple incoherent contributions. In our method, however, we use the property that the acoustic particle velocity is a vector. It is an extension of a previously reported method [1], where several acoustical particle velocity sensors were necessary to a method where only one single acoustic particle velocity sensor is required. Source separation using a single sensor is achieved by applying the procedure in [1] to multiple frequency bands. When at one measuring point the distances to the various sources are different the frequency characteristics of the various sources in the near field will be different (for spherical waves this is the extra term (1+1/jkr), k wavenumber) and the procedure described in reference [1] is applied for different frequency bands. Experiments, using a three dimensional Microflown probe, and a simple model (with point sources) confirm that this method works well. W.F. Druyvesteyn, R. Raangs, Acta Acustica 91,932-935, 2005

Poster session Wednesday, 25 March 2009

Authors will be present: 16:00 - 17:00

Wednesday Poster session 2

Acoustic studies of phase transitions in Ti-Ni alloys

Andrew Abramovich[a], Elena Charnaya[b], Sergei Vasilkov[b], Sergei Belyaev[b] and Aleksandr Volkov[b]

[a]*St.Petersburg St.Technol.Univer.of Plant Polymers;* [b]*St.Petersburg State University*

We report results of acoustic studies of the martensite phase transitions in titan-nickel alloys. These ferroelastic alloys are of great importance for applied science, techniques, and medicine. Acoustic methods are very appropriate for studying the martensite phase transitions because of strong coupling of strains in elastic waves with the order parameters. However, early studies of the martensite phase transitions were limited to a temperature range above the transition due to strong scattering of the ultrasonic waves by ferroelastic domains. The phase transition temperatures for the samples under study were from 260 to 400 K. Pronounced anomalies of ultrasound velocity and attenuation caused by the phase transitions were observed. The influence of tempering the samples was also observed. For some compositions, anomalies corresponded to two successive phase transitions were found. The results obtained were treated within the framework of the Landau theory.

Wednesday Poster session 2

Estimation of spatially distributed temperature and flow fields in air using acoustic travel-time tomography

Manuela Barth, Michael Wilsdorf and Armin Raabe

University of Leipzig, Institute for Meteorology

Sound propagation in air mainly depends on temperature and flow properties along the propagation path of acoustic signals. Thus, measuring travel-time between a sound source and a receiver knowing the sound ray path length, averages of these meteorological quantities along the propagation path can be deduced. Using tomographic techniques as known from medical applications, spatially averaged distributions of temperature and flow can be calculated from a combined treatment of single-line measurements along different paths through the investigation area. In order to estimate three-dimensional fields of temperature and wind, hard- and software of an existing two-dimensional system had to be adapted. To enable a three-dimensional arrangement of sensors around the area under investigation loudspeakers were constructed that ensure a homogeneous sound emission into one half-space. Furthermore, the software was adapted to fit the requirements of a three-dimensional reconstruction of fields.

In this contribution, sensitivity studies with different temperature and flow fields are presented to demonstrate the ability of the system to estimate three-dimensional distributions of these quantities. Furthermore first results from wind tunnel experiments are shown which were carried out to estimate the sensitivity of the new system for the reconstruction of three-dimensional flow fields on meter scale.

Wednesday Poster session 2

Spectral Restoration of Narrowband Speech Recordings Supported by Phonetic Transcriptions

Patrick Bauer and Tim Fingscheidt
Braunschweig Technical University

Due to limitations of the acoustic bandwidth, historic and telephone recordings suffer from poor speech quality and intelligibility. Nevertheless they are widely found in radio, television, internet, as well as in storage media in the context of newscast reports, documentations, or simply in large databases with archived speech material. Spectral restoration of such narrowband speech recordings improves the auditory impression as well as intelligibility. As the recordings are available offline, phonetic transcriptions already exist or can be extracted from the speech data manually or automatically and support the spectral restoration process. Hence, subjective quality and intelligibility much closer to wideband speech can be expected. In this paper, an artificial bandwidth extension (ABWE) is presented to spectrally restore 8 kHz sampled speech signals by upsampling to 16 kHz and estimating further frequency regions of interest. It makes use of the phonetic transcriptions in order to improve the partially insufficient ABWE performance of critical phonemes, i.e., mainly fricatives /s/, /z/ and /f/. We found the spectrally restored speech significantly enhanced, particularly the typical lisping effect disappeared in many instances.

Wednesday Poster session 2

Perception tests with a replica of von Kempelen's speaking machine

Fabian Brackhane
Institute of Phonetics, Saarland University

We present experiments with a replica of the historic speaking machine constructed by von Kempelen around 1780.

Contemporaries reported that the machine sounded very realistic, like the voice of a child aged three to six. Despite the mentioned authenticity of the auditory impression it is impossible to generate most speech sounds in a human-like manner and quality. Notable exceptions are bilabial stops, bilabial nasals as well as open vowels which allow words like [mama] and [papa].

In a perception test 22 subjects verbally described various sounds (including giggle, breathing, scratching etc.). The results show that the words

[mama] and [papa] produced by our replica were recognized as the voice of a human child by 17 subjects.
In a further speech quality test we plan to explore possibilities to vary different prosodic features (word stress and intonation patterns) by modifying air pressure and segmental timing.
In addition, we report on several experiments generating different voice qualities by using different materials (ivory, brass, reed) and different measures of breadth and length for the reed pipe simulating the vocal chords.
This research helps to explain the fascinating possibilities of the probably oldest working speech synthesis to be presented with a live demonstration.

Wednesday Poster session 2

Schulergonomie - Von der subjektiven Empfindung bis zur objektiven Beeinflussbarkeit

Holger Brokmann
Saint Gobain Ecophon GmbH

In unserer immer schneller und lauter werdenden Welt wird zunehmend über Lärm in Bildungseinrichtungen geklagt. Umfangreiche Untersuchungen aus dem In- und Ausland kommen zu dem einheitlichen Ergebnis, Lärm in Bildungsstätten ist ein Problem (Oberdörster, Tiesler: ”Lärm in Bildungsstätten”, 2006; Hotter, Zollneritsch: ”Lärm in der Schule”, 2008). Geht es um Arbeitsplatzgestaltung erfährt das Thema Ergonomie durch Raumakustik, Belüftung oder Bestuhlung ein großes Interesse. Die gleichen Faktoren werden in Schulen großzügig übergangen. Dabei war es bereits im 19. Jahrhundert unter dem Begriff ”Schulhygiene” (Burgerstein, Netolitzky, 1902) ein wichtiges, wissenschaftlich bearbeitetes Thema, was im Laufe der Zeit aber in Vergessenheit geraten ist. Zwar wurden an verschiedenen Universitäten Lehrstühle für Schulhygiene eingerichtet, aber bereits um 1974/75 wurden diese in Deutschland wieder eingestellt. Am 31.10.2008 ist nach dem letzten Projekt ”Gesundheitsfördernde Einflüsse auf das Leistungsvermögen im schulischen Umfeld - Ein Beitrag zur Ergonomie der Schule” auch das Institut für Interdisziplinäre Schulforschung Bremen geschlossen worden. Das Thema hat allerdings nichts an Aktualität verloren, eher im Gegenteil zeigen diverse Veranstaltungen zum Thema ”Schulergonomie” von Unfallkassen, Umwelt- und Kultusministerien den Weg theoretischer Erkenntnisse zu den Endscheidern und Planern im Schulbau. Die Forschungen des ISF bestätigen zudem einen Zusammenhang zwischen objektiven Faktoren der Akustik, Lüftungsintervallen, und dem subjektivem Empfinden.

Transmission of noise through a pipe as structure-borne sound and fluid sound

Sandra Buss, Matthias Wildemann and Friedrich Hartmann
TKMS Blohm + Voss Nordseewerke GmbH

In shipbuilding different aggregates like electric motors, compressors and pumps are commonly mounted together on a platform. This design allows economic building. The platform as a whole is mounted double elastically in order to reduce vibrations from the aggregates transmitted to the ship foundation. The aggregates themselves are coupled rigidly to each other as well as to the platform. This rigid coupling is a cause of vibration transmission between the aggregates directly as well as via the platform. Noise is also transmitted via flanking paths like pipes and ducts. A special case is the vibration of a compressor transmitted via a pump into a pipe as structure-borne sound as well as fluid sound.

A pipe system including a pump as well as a fluid silencer and compensators is build in order to study this kind of vibration transmission. Structure-borne sound is generated at the pump e.g. by a hammer or a shaker. Measurements of structure-borne sound of the pipe and fluid sound in the pipe give indications of the transmission path of the vibration and show the influence of fluid silencers and compensators on vibration transmission in a pipe system.

PU nosecone intensity measurements in a windtunnel

Hans-Elias de Bree and Emiel Tijs
Microflown Technologies

Flow induced noise is known to depend on details in the geometry and surface of an object. Acoustic near field measurements in the flow might be helpful in analysing the noise sources. As accelerometers can not be used to measure flow induced noise, acoustic particle velocity sensors become a viable alternative worth while considering.

Although acoustic particle velocity sensors are susceptible to wind, they can be used in the presence of a flow if the sensors are mounted in a nosecone. Cross correlation techniques can be used to circumvent the issue of flow induced noise even further.

Experiments carried out up to 70m/s at the windtunnels at University of Twente and NLR will be presented and discussed.

Wednesday Poster session 2

Superdirective Beamforming Using an Extended Modal Subspace Decomposition

Martin Eichler and Arild Lacroix
J.W.Goethe-Universität Frankfurt, Inst. für Angewandte Physik

Beamforming techniques allow to control the characteristic of a microphone array in order to achieve a desired directivity. One of the most general formulations is the filter-and-sum beamformer which has readily been generalized by the concept of modal subspace decomposition. This approach finds optimum FIR filter coefficients for each sensor by solving an eigenvalue problem and projecting the desired beam pattern to the set of eigen-beam patterns found. However, the resulting beamformers fail to be directive at very low frequencies where the wavelength is large in comparison to the overall size of the sensor array. In this contribution, we propose an extension to the mentioned concept of modal subspace decomposition which helps to overcome this problem by making the algorithm superdirective.

Wednesday Poster session 2

Acoustic services at Philips Applied Technologies

Alexander Geerlings[a] and Bert Roozen[b]
[a]*Philips Applied Technologies;* [b]*Eindhoven University of Technology*

Within the Philips community and beyond, Philips Applied Technologies (formerly CFT) provides acoustic services under the umbrella of our mechatronics activities. Combined with Philips ATC in Drachten we service most divisions of Philips Consumer Lifestyle, Healthcare and Lighting, as well as various technology groups inside and outside Philips, worldwide. Acoustics knowhow and a fast access to related technologies e.g. flow, control and thermal management, for a broad range of application domains, is available within the grounds of the Philips High Tech Campus. The type of products we work on, in joint effort with application experts in production groups, range from MRI systems to small surface mounted components. The acoustics therefore range from over 90 dB(A) to below 20 dB(A) in soundpower level. This requires proper measurement facilities, such as the reverberant room, the anechoic room and an acoustic imaging tool. Since acoustic solutions are often restricted by the application we need innovative analysis and breakthrough solutions, to improve the quality of the products under consideration. Operating closely with the application experts we provide a way for acoustic solutions to be designed into the application. The poster presented addresses some cases from the Philips product range and explains how new technologies are used for challenging acoustic cases.

Wednesday Poster session 2

Speech intelligibility prediction for normal hearing and hearing impaired subjects - MCHI-S

Lorenz Goenner and Joerg Haubold
ciAD, Dortmund

Main sound quality dimensions are loudness, timbre, auditory comfort and speech understanding, especially if a hearing impairment exists. In addition to an existing prediction of the first named hearing dimensions (MCHI model) a new model for predicting speech understanding was developed. It unifies the relationship between presentation level, signal-to-noise ratio (SNR) and hearing loss information. The MCHI-S model estimates SNR values in critical bands based on a new Voice Activity Detector (VAD). An adapted implementation of Müsch's SRS model (Müsch 2000) computes resulting percentages of speech intelligibility. But, Müsch's model does not consider several additional parameters on which speech intelligibility also depends, such as speech presentation level and a possible hearing impairment. With the goal of predicting common speech intelligibility for both, normal hearing and hearing impaired listeners, MCHI-S enhances Müsch's SRS model by including relevant information and psychoacoustic processings. Multitudinous applications of the MCHI-S model are imaginable in the fields of telecommunication, car industry or hearing aid development.

Bibliography Müsch, H.: Predicting speech intelligibility: A new model based on statistical decision theory. Dissertation, Northeastern Univ., Boston, 2000.

Wednesday Poster session 2

Good Room Acoustic Comforttm (RACTM) can be achieved by using a selection of appropriate acoustic descriptors.

Guus Klamerek and Mariëlle Klijn
Saint-Gobain Ecophon B.V., Etten-Leur

In order to create a good Room Acoustic Comfortô (RACô) in rooms it is important to consider a number of different acoustic descriptors. These descriptors must match and facilitate for wanted human qualities such as ability to concentrate, reduced stress, clear speech etc. In this process it is important to consider the people, what they do (the activity) and what kind of room the activity will take place in. Today, when designing ordinary rooms from an acoustic perspective, mainly reverberation time (T20) is utilised - both in practice but also in building regulation and standards. Reverberation time (T20) only describe the later part of the decay curve, and therefore only partly mirror the wanted acoustic reality. Thus, based upon a large number of acoustic measurements, we suggest a good ”mix” of acoustic descriptors for ordinary rooms in buildings like schools, offices, health care premises etc. These descriptors have to cover both early and late decay, sound levels and speech quality. Our suggestions are Speech Clarity (C50), Speech Transmission Index (STI), Reverberation Times (EDT, T20) and Strength (G). Moreover, in open and long spaces we also suggest the acoustic descriptors Rate of Spatial Decay (DL2) and Excess of Sound Pressure Level (DLf).

Wednesday Poster session 2

Ferroelectret-film accelerometers with high sensitivities

Milos Kodejska, Joachim Hillenbrand and Gerhard M. Sessler
TU Darmstadt, Institute for Communications Technology

Ferroelectrets are piezoelectric films of charged cellular or porous polymeric electrets and have larger d_{33}-coefficients than lead zirconate titanate (PZT). Compared to PZT and other piezoelectric materials, ferroelectrets are flexible, lightweight, and cheap. These properties allow the construction of light, flat, and sensitive accelerometers of various shapes. Such accelerometers were built with polypropylene (PP) ferroelectrets in single- and in multi-layer arrangements and seismic masses from 1 to 20 g. Film area, number of ferroelectret films, and seismic mass determine sensitivity and resonance frequency of the accelerometer and these parameters were varied for the production of different accelerometers. The frequency responses of the sensitivity of the accelerometers were measured using an electrodynamic vibration exciter. A typical sensitivity of about 3 pC$\cdot$s^2/m was obtained for an accelerometer consisting of a single ferroelectret film with $d_{33} \approx 500$ pC/N in the audio frequency range, a seismic mass of 9 g, and an area of 3 cm^2. A theoretical description of the various accelerometers has been achieved using simple models. Experimental and theoretical data were compared and showed good agreement. The results demonstrate that ferroelectret-film accelerometers have advantageous properties, in particular when adapted to specific applications.

Wednesday Poster session 2

Eine Ära geht zu Ende - Schließung des Instituts für interdisziplinäre Schulforschung (ISF) der Universität Bremen - Ein Rückblick

Rainer Machner[a] and Gerhart Tiesler[b]
[a]*Saint Gobain Ecophon GmbH;* [b]*Institut für interdisziplinäre Schulforschung (ISF)*

Am 31. Oktober 2008 war es endgültig - die Türen des ISF der Universität in Bremen schlossen. Damit gehen nahezu 35 Jahre Schulforschung zu Ende. Doch blicken wir auf einen Fundus von unschätzbaren Erkenntnissen aus der interdisziplinären Schulforschung zurück. Dieser Beitrag soll Interessierten einen kurzen aber prägnanten Rückblick der einmalig geschickten Forschungsansätze des ISF ermöglichen. Zahlreiche Veröffentlichungen der vergangenen Jahre führten auch immer wieder auf die DAGA. Grund war nicht letztlich die Raumakustik, die als entscheidender Parameter, der die Pädagogen und Kinder an Ihrem Arbeitsplatz

Schule beeinflusst, registriert und mit anderen Disziplinen in einem Kontext erforscht wurde. Der Blickwinkel einer Fachgruppe wie z.B. Pädagogik oder Arbeitsmedizin allein reichte häufig nicht aus um die Hintergründe einer ”Schulergonomie” erfolgreich zu beleuchten. Das umfangreiche Instrumentarium der zahlreichen Arbeiten wurde daher seit den 90ger Jahren fortwährend durch akustische Aufzeichnungen komplettiert. Keine vergleichbare Einrichtung in Europa zeigt eine ähnliche Fähigkeit, aus der Perspektive so grundsätzlich verschiedener Fachdisziplinen eine Fragestellung im gemeinsamen Kontext zu überprüfen. Hier einige spannende Konzepte des ISF: ”Arbeitsplatz Schule”; ”Berufliche Belastung von Lehrerinnen und Lehrern”; ”Lärm in Schulen”; ”Akustische Ergonomie der Schule”; ”Ermüdung im Schulalltag”. Die Konzeptentwicklung der Saint Gobain Ecophon GmbH in Lübeck ist dankbar, dass Sie Etappen dieser Forschung begleiten durfte.

Wednesday Poster session 2

Improving transmission loss of light-weighted panels by increase of stiffness with an evacuated foil.

Volker Mellert[a], Roland Kruse[b] and Hermann Remmers[c]
[a]*Universität Oldenburg, Institut für Physik;* [b]*Universität Oldenburg;* [c]*itap GmbH*

Compartments within the cabin of transportation means like an airplane are separated by a wall-construction which should be light in weight as much as possible while providing sufficient stability. But one demand is in general not met by such a construction: Due to the mass-law it is difficult to achieve sufficient sound insulation, in particular for low frequencies,. The mass should not be increased for obvious reasons. One way to reduce sound transmission is to absorb the sound waves within the panel by appropriate absorbing material, e.g micro-perforated foils or structures with loss for bending waves. A more efficient way is to increase the stiffness of the material. The increase of stiffness behaves like an increase of mass for frequencies below the first mode of membrane resonance. If the stiffness is high enough, the resonances are shifted to higher frequencies, and higher frequencies are easier to be damped by classical methods. Results are reported for various panel constructions in which stiffness is increased by covering the panel completely with an air-tight evacuated membrane. The transmission loss of honeycomb panels is enhanced about 10 to 25 dB at low frequencies, depending on the construction of the original panel.

Wednesday Poster session 2

Smart Noise Reduction Based on Reliability of Direction-of-arrival Estimate

Mitsunori Mizumachi
Kyushu Institute of Technology

There are various approaches to reduce acoustical noises to satisfy the needs of noise reduction. Single-channel approaches rely on the prior knowledge of target signals and noises, and multi-channel approaches take advantage of spatial sparseness among sound sources. Those methods require more or less constraints, explicitly and inexplicitly. In this paper, the author proposes the use of the reliability of a spatial feature to achieve a less-constraint, smart noise reduction. Direction-of-arrival (DOA) of the target acoustic signal is one of the most important spatial features, and is often employed in noise reduction with beamforming. The author succeeds in estimating the confidence of DOA estimates by means of observing the state of a noisy cross-correlation function, which yields a DOA estimate, comparing with an ideal cross-correlation function with a single, monotonic, sharp main-lobe. In this paper, the confidence measure is used for selecting the suitable beamformer in between delay-and-sum and null-steering beamformers. In short, delay-and-sum and null-steering beamformers are expected to be selected out during less and more reliable periods in DOA estimation, respectively. In the conference, we will confirm the feasibility of the proposed scheme. This work was supported by New Energy and Industrial Technology Development Organization of Japan.

Wednesday Poster session 2

Generation of short ultrasonic pulses using active damping

Sergei Olfert, Jens Rautenberg and Bernd Henning
University of Paderborn, Measurement Engineering Group

In Non Destructive Testing with ultrasound the emission of short pulses is essentially if the echo signal should be separated from the emitted one. In this contribution a new approach is shown how to generate a short ultrasonic pulse using a 3-1 composite transducer. Therefore the ceramic bars, which are embedded in a polymer matrix, will be regarded as a kind of engine for the entire structure. After the pulse excitation the same bars can also be used for active damping to increase the bandwidth. The Mason model serves as a basis to the description of the ultrasonic transducer. This model is extended to the composite structure. The excitation signal is adaptively modified by a suitable signal preprocessing. Therefore the changes of boundary conditions, like the ambient temperature or acoustic impedance of adjacent media are considered. The modeling of the composite transducer, the experimental setup and first results will be presented.

Wednesday Poster session 2

Open Plan Office Acoustics: A Case Study in a Real Office

Carolina Reich Marcon Passero and Paulo Henrique Trombetta Zannin
Universidade Federal do Paraná

Ambient noise (LAeq), reverberation time (RT), and speech intelligibility (STI) have been measured in an open plan office. The average sound pressure level was of 61 dB(A), a value below the limit set by the pertinent Brazilian Standard, of 65 dB(A). RT values were of 0.49, 0.58, and 0.65 s, respectively for the frequencies 500, 1000, and 2000 Hz. According to the German Standard (VDI), a value of RT below 0.5 s is recommended for open plan offices, for those frequencies. Thus, only in the lowest frequency evaluated (500 Hz) does the open plan office fit the Standard. Given the existing background noise, only the workstation located in front of the source had STI values (0.48) considered fair by the IEC standard. In all other workstations the measured STI was considered poor or bad. In conclusion, the studied office displays reasonable acoustic absorbance (despite not meeting the German standards), and reasonable levels of background noise. Noise limits are thus not surpassed, providing privacy for neighbouring workstations.

Wednesday Poster session 2

A PU sound probe for high noise levels

Emiel Tijs
Microflown Technologies

For several high noise level applications, e.g. jet engines, the standard PU and USP probes can not be used. In order to meet these challenging acoustic requirements, both the particle velocity and the sound pressure transducers have been optimized. The novel design will be explained and its acoustic performance will be discussed. Attention will be paid to the calibration of such a probe.

Wednesday Poster session 2

Estimating physical properties of vocal fold paralysis from high-speed filimg data

Isao Tokuda[a], Miwako Kimura[b], Hiroshi Imagawa[c], Ken-Ichi Sakakibara[d] and Niro Tayama[e]
[a]*Japan Advanced Institute of Science and Technology;* [b]*University of Texas Southwestern;* [c]*University of Tokyo Hospital;* [d]*Health Sciences University of Hokkaido;* [e]*International Medical Center, Tokyo*

Pathological voice originates from complex nonlinear vibrations of the disordered vocal folds. Deep understanding of the voice disorders based on mathematical modeling of the vocal folds may provide an important insight into the abnormal vocal fold vibration as well as a good hint for improving the voice surgery. Up to date, various models have been developed to simulate complex vibratory patterns of the vocal folds, ranging from a simple two-mass model to complex multi-mass models. However,

only a few researches aim at construction of the vocal fold model, which realizes quantitative characteristics of the individual voice. Recently, Doellinger et al. proposed a method for estimating the parameters of asymmetric two-mass model for high-speed filming data recorded from human subjects. The present paper applies this technique to pathological data recorded from patients with vocal fold paralysis. Right and left vocal fold tensions, subglottal pressure and glottal area have been estimated. Whether the effect of the voice surgery is reflected in the estimated parameters is judged by using the data before and after the voice surgery. On the basis of the estimation results, applicability of the mathematical model as a simulator for the voice surgery is discussed.

Wednesday Poster session 2

Measurement of Head Related Impulse Responses for psychoacoustic research

Florian Völk, Martin Straubinger, Luis Roalter and Hugo Fastl
AG Technische Akustik, MMK, TU München

In recent years, the importance of Head Related Impulse Responses (HRIRs) for psychoacoustic experiments and especially for studies in directional hearing has been growing, mainly caused by the availability of fast computers and the resulting possibility of very realistic dynamic binaural synthesis. A necessary prerequisite for all these methods is the correct measurement of HRIRs.
In this contribution a setup will be presented which is capable of measuring HRIRs for psychoacoustic research by two different techniques: maximum length sequences and exponential sine sweep. To reduce the measurement time, an interpolation algorithm can be used. The resulting HRIR-sets are compared by typical physical parameters as well as judged in a localization experiment. Based on these data, parameters for high quality HRIR measurements for psychoacoustic research are proposed.

Wednesday Poster session 2

Coding of speech into nerve-action potentials

Huan Wang and Werner Hemmert
Technische Universität München

One of the most critical processing steps during encoding of sound signals for neuronal processing is when the analog pressure wave is coded into discrete nerve-action potentials. As any information lost during this process is no longer available for neuronal processing, it is important to understand and quantitatively model the underlying principles. We have developed a detailed model of auditory processing, which codes sound signals into spike-trains of the auditory nerve. We have also developed Hodgkin-Huxley models of cochlear nucleus neurons, which are driven by auditory nerve spike-trains. We analyze the quality of coding with the framework of automatic speech recognition and the temporal information processing capabilities with the methods of information theory.

Our latest improvements in speech coding by introducing the effect of offset-adaptation together with an improved matching of neuronal features to the speech recognizer using an artificial neuronal network has lead to significant improvements of recognition scores, now reaching the values of successful technical feature extraction methods. Offset adaptation is also required to drive onset neurons in the cochlear nucleus, which are able to code temporal information with sub-millisecond precision (< 0.02 ms). Our results provide quantitative insight into temporal processing strategies of neuronal processing and are highly relevant for cochlear implants.

Paper sessions Thursday, 26 March 2009

Thu 8:40 Willem Burger Zaal Psychoacoustics 2

Critique of Pure Psychoacoustics

Jens Blauert and Rainer Guski
Ruhr-University Bochum

Since product-sound quality has become a major engineering issue, psychoacoustics is widely applied in industrial R&D processes. It can. however, be observed that engineers which apply psychoacoustics sometimes have a rather naive understanding of the foundations and, thus, the potency of psychoacoustic methods. A common belief is, e.g., that psychoacoustic quantities can reliably be assessed with instrumental methods. The notion behind this belief is that, in laboratory experiments, "unbiased" versions of psychoacoustic quantities can be identified and assessed. It is the task of cognitive psychology to point to the risks that such an over-simplifying notion of psychoacoustics implies, and to provide knowledge and means to accordingly select and refine the psychoacoustic methods applied in engineering. It will be discussed that psychometric methods which are suitable for obtaining meaningful and valid information on product-sound quality vary substantially with the degree of abstraction of the references.

Thu 9:00 Willem Burger Zaal Psychoacoustics 2

Cortical processing of auditory motion cue information

Stephan Getzmann and Jörg Lewald
Institut für Arbeitsphysiologie, Dortmund

This study investigated the neural correlates of auditory motion processing by employing high-density electroencephalography. Sound motion was implemented by (a) gradual shifts in interaural time (ITD) or (b) level difference (ILD); (c) motion of virtual 3D sound generated using HRTF filters (derived from a set of KEMAR measurements); or (d) successive activation of 45 loudspeakers along the horizontal plane. In a subset of trials, listeners (N = 19) performed a 2A-FC motion discrimination task. Each trial began with a period of stationary acoustic stimulation from a central position, immediately followed by a period of motion starting from this location. The onset of motion elicited a specific cortical response that was dominated by large negative and positive deflections, the so-called change-N1 and change-P2. The temporal dynamics of these components depended on the auditory motion cues presented: Freefield motion and virtual 3D sound were associated with larger and earlier cortical responses and with shorter reaction times than shifts in ITD or ILD. Also, topographical analyses of the motion onset response indicated differences in the pattern of activation related to the different motion cues. The differences in cortical responses may allow conclusions on the processing of dynamic auditory information in the human brain.

Thu 9:20 Willem Burger Zaal Psychoacoustics 2

Algorithmic modelling of the disturbance impact of background sounds

Sabine Schlittmeier[a], Tobias Weissgerber[b], Stefan Kerber[b,c], Hugo Fastl[b] and Jürgen Hellbrück[a]
[a]*Environm. and Health Psychology, KU Eichstätt-Ingolstadt;* [b]*AG Technische Akustik, MMK, TU München;* [c]*MRC Institute of Hearing Research*

Mentally demanding tasks very rarely take place during silence. However background sound can reduce cognitive performance, even if it is irrelevant to the task and is intended to be ignored. This so-called Irrelevant Sound Effect (ISE) has been verified in a multitude of behavioral experiments for verbal short-term memory performance.
Although a multitude of cognitive psychological experiments have explored the ISE, no psychoacoustically based instrumental procedure existed to predict its occurrence. This poses problems in an applied context: In office environments, for example, the potential beneficial - or less advantageous - effects of noise abatement on cognitive performance could not be calculated before their realization.
The talk presents an algorithm which models performance data in ISE experiments on the basis of instrumental measurements of the hearing sensation fluctuation strength. It was verified with a database consisting of about 50 background sounds and corresponding performance data that have been collected in cognitive psychological experiments at the KU Eichstätt-Ingolstadt. The algorithm is able to reproduce the performance results in about 90 % of cases within the interquartile ranges. It will be discussed within the scope of cognitive short-term memory models, which claim to explain the ISE and with respect to practical implications.

Thu 9:40 Willem Burger Zaal Psychoacoustics 2

Application of Synesthetic Design as multi-sensory Approach on Sound Quality

Michael Haverkamp
Ford Werke GmbH Köln

During perception of sound events it appears to be obvious that auditory attributes transport information about objects of the world, which in most cases imply multi-sensory aspects. Hence, analysis of perceived sounds must include analysis of references which point to attributes of other modalities. Studies of the past have shown that an optimized, satisfying consideration of cross-sensory interaction is not possible if based on a single perceptual process.
This contribution provides a concept of synesthetic design which includes various strategies of cross-sensory coupling evident within the human perceptual system. Based on recent results of perception and cognition, this concept enables an extensive alignment of design rules to cerebral functions. Sound attributes can then be analysed with view of a

whole set of multi-sensory features. Those features can be of basic nature, show associative (iconic) content or semantic features (meaning). Research on individual (genuine) synesthesia offer further aspects.
It is also discussed to what extent algorithms and concepts used for coupling of cross-sensory features during the design process are capable to provide an integral figuration of products.

Thu 10:00 Willem Burger Zaal Psychoacoustics 2

Auditory capture in an auditory and visual spatial cueing task.

Thomas Koelewijn[a], Marieke van der Hoeven[b] and Adelbert Bronkhorst[b]
[a]*Vrije Universiteit Amsterdam;* [b]*TNO*

Previous cueing studies show that a sound coming from a particular location in space can capture auditory and visual attention. We conducted two cueing studies where a baseline condition in the form of an auditory spatially diffuse cue was included, using out-of-phase presentation through two loudspeakers. The first study examined how an auditory non-informative spatial cue influences detection and localization of auditory targets. Results show that, compared to the baseline, cues shortened reaction times at small cue-target angles (up to 7°) and increased them at larger angles. The reaction time difference was up to 80 ms. Furthermore, cueing seems to have no effect on sound localization: false alarms were normally distributed around the hits. These findings demonstrate that strong auditory capture effects occur that depend on cue-target distance. The second study shows how an auditory non-informative spatial cue influences detection of visual targets. Here the results show performance benefits when the target was preceded by a cue presented at target location and performance costs when a cue was presented elsewhere. Additionally, this study shows that even when attention is highly visually focused by means of a 100% valid endogenous visual cue, auditory stimuli still capture visual attention.

Thu 10:40 Willem Burger Zaal Psychoacoustics 2

Crying colours and their influence on loudness judgments

Daniel Menzel, Thomas Dauenhauer and Hugo Fastl
AG Technische Akustik, MMK, TU München

The German term ”schreiende Farbe” denotes a very salient colour which stands out among other colours. It can be translated as ”loud colour” or literally as ”crying colour”. As it is known from previous studies that the colour of objects like trains or cars can influence loudness judgments, experiments were performed to determine if such crying colours lead to higher loudness ratings compared to non-crying colours. As a first step, subjects were asked to synthesise colours which they perceived as ”crying” by adjusting values in the Lab colour-space. A representative sample of these colours was then combined with non-crying colours. The resulting group of 15 colours was rated on a five-point scale ranging from ”not-crying” to ”very crying”. In a final listening experiment, the same 15 colours were presented simultaneously with uniform exciting noise of different sound pressure levels to asses the influence of the visual stimuli on the subjective loudness ratings. Results indicate that some crying colours where able to elicit - at same SPL of the noise - higher loudness ratings compared to a neutral colour.

Thu 11:00 Willem Burger Zaal Psychoacoustics 2

Hearing and Cognitive Measures Predict Elderly Listeners' Difficulty Ignoring Competing Speech

Esther Janse
Utrecht institute of Linguistics OTS & MPI Nijmegen

Age-related hearing and cognitive factors may play a role in the difficulty elderly listeners have listening to one talker in the presence of a competing speaker, but it is not always easy to establish direct relations between speech performance and cognitive measures. The present study investigated how well (detection accuracy) and how fast (detection time) elderly listeners detected pre-assigned target sounds in the speech of a target talker, both in a single-talker condition and in a condition with a competing talker. If general cognitive decline contributes to problems with competing speech over and above hearing loss, a measure of visual selective attention might be related to performance in the present study. 39 Elderly listeners with varying degrees of hearing loss participated in the study. Apart from the effects of hearing loss, overall detection accuracy correlated with the selective attention measure: the more difficulty one had ignoring irrelevant visual information, the poorer one's speech processing performance. The response time analysis showed that individual hearing acuity and the attention measure predicted how fast listeners detected the pre-assigned sounds in both listening conditions. These results confirm that age-related cognitive factors contribute to the typical problems elderly adults report in daily speech communication.

Thu 11:20 Willem Burger Zaal Psychoacoustics 2

Acoustic Optimization of Rotary Switches

Alexander S. Treiber and Gerhard Gruhler
Heilbronn University

Due to the complexity of modern car's on-board systems car manufacturers tend to use menu based user interfaces which offer a relatively low number of control elements for a large number of functions.
Since cars nowadays are no longer sold simply as technical but as lifestyle products it is crucial for the success that the potential customer perceives every single aspect of the car to be valuable. Since the user interface of the car can be judged even before a test drive the feeling of buttons and switches is a key aspect. This feeling as affected by both haptical and acoustical feedback of operation.
This work describes an enhanced interactive simulator for the acoustic feedback of control elements. The simulator allows the subjective rating of recorded feedback sounds of control elements without visual or haptic

cues using semantic differentials or paired comparisons as well as the manipulation of stimuli by the test subjects. Furthermore, results obtained using this device are presented.

Thu 11:40 Willem Burger Zaal Psychoacoustics 2

Cognitive Factors in Speech Perception

Adriana Zekveld
dept. ENT / Audiology, VU medical center, Amsterdam
Authors: Adriana Zekveld, Sophia Kramer, Tammo Houtgast

Both auditory, modality-specific and cognitive, modality-aspecific abilities influence the ability to perceive and comprehend speech in noisy listening situations. Important cognitive factors contributing are information processing speed and working memory processes that allow the use of linguistic context to fill in incomprehensible ’gaps’ in the speech. When listening conditions are challenging (due to noise or hearing loss), listeners rely to a greater extent on effortful top-down processes to improve speech comprehension. Therefore, it is important to measure both auditory and cognitive abilities when examining the causes of speech comprehension problems. To prevent the performance on cognitive tests from being confounded by hearing loss, visual tests of cognitive functions need to be considered when testing hearing impaired participants. Therefore, our group recently developed the Text Reception Threshold (TRT) test, a visual analogue of the Speech Reception Threshold test. Several studies showed that speech comprehension in noise is associated with the TRT, indicating that both abilities rely on similar modality-aspecific cognitive processes. Future studies should focus on further unravelling the auditory and cognitive functions in speech comprehension and on the development of a clinically applicable test of relevant cognitive functions in speech comprehension in noise.

Thu 15:00 Willem Burger Zaal Psychoacoustics 3

Identification and detection of a tone in narrowband noise

Hans Hansen and Reinhard Weber
Universität Oldenburg, Institut für Physik

When comparing different pitch phenomena in salience, it is not clear whether the expression of pitch strength points towards a feature’s magnitude or the object’s salience within a background. As many studies concentrate on special phenomenona, the question, what is actually judged, seems implicitly answered. Kubovy and v.Valkenburg [Cognition, 2001, 80, 97-126] define a perceptual object as ”that what is susceptible to figure-ground segregation”. Pitch plays a major role in these grouping processes. The pitch strength’s judgment refers to two perceptual cases. The first one is the pitch strength of a tone-in-noise, i.e. its salience within a background, while the second is the pitch strength of tonal noise. Here, the noise evokes a pitch percept that is not related to a separate object. In order to explore the transition from case 1/ tone-in-noise to case 2/

tonal noise, the identification threshold hearing a separate tone centered on narrowband noise (NBN) is determined in an experiment. This is compared to the measured masking threshold for center frequencies at 250-4000 Hz octave-wise at 60 dB SPL noise level. The bandwidth of the noise is varied from 50-250 Hz accordingly. In a third experiment, the frequency difference limens (FDL) of the stimuli are measured. The FDLs are compared to the obtained thresholds.

Thu 15:20 Willem Burger Zaal Psychoacoustics 3

Noise reduction for media streams

Thomas Bisitz[a], Tobias Herzke[a], Melanie Zokoll[a], Anne-Marie Öster[b], Samer Al Moubayed[b], Björn Granström[b], Ellen Ormel[c], Nic van Son[c] and Richard Tanke[c]

[a]*HörTech gGmbH, Germany;* [b]*KTH, Sweden;* [c]*Viataal, Netherlands*

In the Hearing at Home project the central Home Information and Communication platform is developed to support hearing-impaired persons with easy to configure “Supportive Audio Signal Processing” and visual support on a TV screen in home environments. Amongst other, individually configurable algorithms, several algorithms can be used for noise reduction of the media streams without individual fitting to enhance the accessibility of the presented audio materials using the Master Hearing Aid as a real-time audio-signal-processing framework: (1) Single-channel noise reduction according to Ephraim-Malah together with a speech pause detector, (2) stereo noise reduction additionally removing the central signal in a control path, (3) frequency shaping noise reduction by emphasising a frequency range important for speech intelligibility, (4) time-varying attenuation and (5) noise reduction by selection of channels. It will be possible to classify the acoustic situation in the media stream automatically in real-time using six classes allowing to activate the appropriate noise reduction strategies. In user tests the performance of algorithm 1-3 is evaluated: Results of sound quality assessment, listening effort estimation and speech intelligibility measurements with hearing impaired subjects from groups with different types of hearing impairment will be presented.

Thu 15:40 Willem Burger Zaal Psychoacoustics 3

Sound Quality Evaluation of Power Seat Adjusters

Philipp Scheibner[a], Alfred Zeitler[a] and Andreas Wendemuth[b]

[a]*BMW AG - Munich;* [b]*Otto-von-Guericke University Magdeburg*

The sound quality of electric motors has become more and more important in the automotive industry. Increasing customer requirements and cost pressure have led to a strong demand for reliable and efficient testing methods. Noise-specific metrics are needed which allow for the prediction of customers’ quality perception throughout the development process. This paper reports about the psychoacoustic modelling of the sound quality of power seat adjusters. Clearly, loudness is not

sufficient to predict judgements of sound quality in this case but additional parameters reflecting tonal components and effects of modulation have to be accounted for. A listening experiment was performed to receive annoyance ratings of synthetically-produced test signals which were equally loud but varied in their temporal structure according to a fractional factorial design (design of experiments, DOE). In this method, a subset of the experimental runs of a full factorial design was used which is representative of a wide range of parameter configurations as it was found in the airborne noise of real sources. This way, experimental labour is saved and model attributes can be investigated more specifically. The results of the experiments are presented and the methodology of DOE is discussed.

Thu 16:00 Willem Burger Zaal Psychoacoustics 3

Psychoacoustics Without Psychology?

Rainer Guski and Jens Blauert
Ruhr-University Bochum

Today, several so-called psychoacoustic variables are measured or calculated by means of algorithmic procedures, e.g., loudness, sharpness, roughness, and tonality. The calculation procedures are primarily based on the results of listening experiments, using so-called ”context-free” sound samples in ”neutral” experimental situations. Problems arise when these variables are used to predict human reactions in real life contexts, because the local, temporary, situational, visual, and cognitive context of a sound often influences human reactions more strongly than the acoustical parameters of the sound itself. This is shown by means of examples from threshold measurements, loudness scaling, and auditory localisation. It will be discussed whether and how sophisticated experiments with human subjects in well-controlled settings can help to provide psychoacoustic data that can be generalised to certain classes of real-life settings.

Thu 16:40 Willem Burger Zaal Psychoacoustics 3

The tonality salience of pure tones as a function of frequency

Reinhard Weber, Deborah Brosig and Jesko L. Verhey
Universität Oldenburg, Institut für Physik

Tones with different frequencies do not only give rise to different pitches but they also may evoke sensations that differ in the prominence in their tonal character. Using magnitude estimation, Fastl found [Fastl, H.: pitch strength of pure tones. 13th ICA, Belgrade, 1989] that the salience of tonality, the pitch strength, of pure tones increases as a function of loudness and shows a band pass characteristic as a function of frequency. This frequency dependency of the tonal character is the matter of interest in the present study, where an adaptive method, similar to that for equal loudness contours, has been used to measure equal salience contours. 24 participants compared seven tones with frequencies between 125 Hz and 8 kHz with a 1 kHz tone. The levels were adjusted such that a point of subjective equality of pitch salience were reached. Additionally the equal loudness contours were determined for the seven tones with respect to loudness of the 1 kHz tone. The discrepancies between equal loudness results and the PSEs of pitch salience showed, that there is no clear relation between pitch salience and the loudness of the tone.

Thu 17:00 Willem Burger Zaal Psychoacoustics 3

Recruiting and Evaluation Process of an Expert Listening Panel

Alois Sontacchi, Hannes Pomberger and Robert Höldrich
Institute of Electronic Music and Acoustics, Graz

During the last few years an increased desire to assess and quantify acoustical properties of technical products can be recognized. Utilizing objective evaluation methods, which are commonly based on modelling the human sound perception and underlying auditory hearing process, can deliver qualitative and quantitative acoustical description of the investigated product. However, only a limited number of application areas with specific restricted requirements can be treated reliable within tolerable assessment results. The predominant crucial aspect is based on the complex interaction of acoustical stimulus and auditory perception. In general, overlapping and competitive acoustical stimuli can be instructively assessed only by a special group of listeners. To derivate the obtained answers to the majority of potential people how are involved with the regarded device under test specific demands result to select this special group of listeners. Beside the listeners selection process and trainings phases the design and accomplishment of the listening test has to guarantee that the subjective results are within defined limits and can be reproduced anytime. Within this work the constitution and rehearsal of an expert listening panel will be described and discussed on the basis of already established and published selection procedures and new presented methods.

Thu 17:20 Willem Burger Zaal Psychoacoustics 3

Spatial Sampling Artifacts of Focused Sources in Wave Field Synthesis

Sascha Spors and Jens Ahrens
Deutsche Telekom Labs, TU Berlin

Wave field synthesis (WFS) is a spatial sound reproduction technique that facilitates a high number of loudspeakers to create a virtual auditory scene. Amongst other interesting properties, WFS allows to reproduce virtual sources that can be positioned in the area between the loudspeakers and the listener. These are known as focussed sources. The physical properties of focussed sources have not been investigated in detail so far. Of special interest in the context of this contribution is the influence of spatial sampling. It will be shown that the resulting spatial sampling artifacts can be perceived as pre-echo artifacts. Their audibility depends on the source and listener position, and the size of the WFS system. This paper will discuss the physical foundations and will present

first subjective experiments to investigate the audability of the resulting artifacts.

Thu 8:40 Jurriaanse Zaal Building acoustics 2

Sound Insulation for practice rooms used by rock-bands

Franz Breuer and Jan Paprotny
Peutz Consult GmbH

During our work as acoustic consultant we had to judge the sound transmission of a cluster of more than 20 practice rooms in the neighbourhood. One of the basic question was to determine ”typical” sound levels in practice rooms. To answer this question measurements were made in other practice rooms with different kinds of bands and music. In addition to handmade music we analyzed recorded pop-music with very low-frequent parts. During the measuring at the location the sound insulation of all 20 practice rooms to the nearest flat was measured and calculated. Based on these data, in a further step, the sound pressure level in the flat could be calculated as if the bands actually would play in these rooms, or the pop music would be played there. The results of the 20 calculated sound insulations and the corresponding sound levels in the flat are presented. These results are compared to german standards and an unexpected rating of the different kinds of music is detected.

Thu 9:00 Jurriaanse Zaal Building acoustics 2

Acoustical characteristics of lightweight solid gypsum walls and their implementation into prediction models

Andreas Ruff and Heinz-Martin Fischer
Hochschule für Technik Stuttgart

In multi storey buildings gypsum blocks are often used to construct solid inner walls without static requirements. These gypsum walls with a mass per unit area of only about 90 kg/m^2 are not connected rigidly to the adjacent building elements. They are commonly decoupled from the adjacent construction elements with elastic interlayers made of bitumen, cork or polyethylene foam. The direct sound insulation of the gypsum walls is depending on the kind of the used elastic interlayers. The elastic interlayers have also a significant influence on the flanking sound transmission of the gypsum walls. Therefore the decoupled walls have to be treated different to rigidly connected walls.

Within a research project the direct and the flanking sound transmission of gypsum walls is investigated in different test facilities in the laboratory. For different kinds of elastic interlayers the sound reduction index of gypsum walls is measured in a test facility for direct sound transmission. Measurements of the vibration reduction index are carried out in two test facilities for flanking transmission (vertical and horizontal) and also in different building situations. These measurement results are used as input data for the prediction model according EN 12354-1.

Thu 9:20 Jurriaanse Zaal Building acoustics 2

Sound and vibration transmission through lightweight junctions: Numerical investigation and validation by laboratory tests

Sven Lentzen, Eddy Gerretsen, Carine van Bentum and Susanne Bron-van der Jagt
TNO

In this work the sound and vibration characteristics of several lightweight junctions are numerically investigated with FEM (for the low- and the mid-frequency range) and SEA (for the mid- and the high-frequency range). In order to validate these analyses, the junctions are tested in a laboratory setting. For sound insulation the investigations are focused on obtaining the vibration reduction indices K_{ij}. These are not only investigated for the floor-floor flanking transmission path, but also for all the other possible paths (e.g. floor-wall, floor-ceiling and wall-ceiling). Although applied in different frequency ranges, the results of the FEM and the SEA analyses agree fairly well with those obtained in the laboratory tests. Looking at the vibration behaviour and transmission, the main focus was put on obtaining the One-Step-RMS velocities of the floors. In these investigations the point- and transfermobilities are acquired either by numerical analyses or by measurements. Since these spectra are only required for frequencies up to 40Hz, the numerical study consists merely of FEM analysis. Here, both results agree fairly well. From the investigations in both areas, the appropriate conclusions concerning good building practices in lightweight buildings concerning sound and vibration can be drawn.

Thu 9:40 Jurriaanse Zaal Building acoustics 2

Sound and vibration transmission through lightweight junctions: Investigation of a practical example.

Arnold Koopman, Sven Lentzen and Flavio Galanti
TNO

Sound and vibration transmission through lightweight junctions is a relatively unexplored area. One of the major problems that have to be dealt with is the competitive requirements that should be ensured for a good sound insulation on the one hand, and for good vibration behaviour on the other. Due to the lack of experience and the aforementioned complexifying conditions it sometimes happens that the quality of the indoor acoustic climate is not as good as it was hoped. This contribution deals with a building which had to be constructed using lightweight concepts to be able to build it on an existing foundation. During construction it appeared that the wooden floors are susceptible to walking-induced vibrations. TNO was asked to investigate the situation, using on-site measurements and numerical analyses. Once the causes were found, counter measures could be taken. In order to obtain valuable predictions, tuned FEM- and SEA-models were used. As a result of this research, major vibration reductions could be achieved.

Thu 10:00 Jurriaanse Zaal Building acoustics 2

Experimental investigations on the sound transmission of drywall constructions at low frequencies

Raphael Völtl[a], Ulrich Schanda[b] and Thomas Franzen[b]
[a]*Hochschule Rosenheim / PMI GmbH;* [b]*FH-Rosenheim*

For a better understanding of sound transmission of drywall constructions in the frequency range below 100 Hz several measurements in a laboratory according to ISO 140-1 were carried out, the receiving part of the laboratory had been changed to semi free-field conditions. Sound intensity levels in the near field of the wall as well as acceleration levels were mapped by excitation either with airborne sound or with a shaker. With the mapped values the radiation efficiency of the wall was calculated. In the frequency levels below approx. 250 Hz a clear rise of the radiation efficiency was observed, contrary to different theoretical models. Furthermore the measured and mapped eigenmodes of the drywall show significant differences when exited by airbone sound or by the shaker in the frequency range below 50 Hz. In order to find out the reasons of these observations many experimental investigations have been carried out und willl be reported.

Thu 10:20 Jurriaanse Zaal Building acoustics 2

Sound insulation of triple insulating glass units

Bernd Saß
ift Schallschutzzentrum, Rosenheim

Mit steigenden Anforderungen an den Wärmeschutz von Außenbauteilen werden in Deutschland verstärkt Isolierglaseinheiten aus Dreifach-Isolierglas angeboten und in Fenster und Fassaden eingebaut. Der Beitrag beleuchten die schalldämmenden Eigenschaften von Dreifach-Isolierglas und gibt einen Überblick über besondere Merkmale von Dreifachglas, die bei der Planung von Außenbauteilen zu beachten sind.
With increasing requirements on thermal insulation of building elements in germany triple insulating glass units are reinforced as actual construction type of glazing in windows and facades. The article shows properties of the sound insulation from triple insulating glass units and highlights special characteristics to be followed in planning of building elements.

Thu 11:00 Jurriaanse Zaal Building acoustics 2

The Sound Insulation of Water

Remy Wenmaekers[a], Bart van der Aa[b], Arno Pronk[b], Agostinho Coutinho[c] and Renz van Luxemburg[a]
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Liquid filled multilayer membranes can be used as a building material for roofs [Pronk et all. 2007]. Between the membranes a fluid is injected for heat or cold transfer. The sound insulation of the fluid filled membranes is mainly determined by the fluid. Since fluids are unusual building materials, the sound insulation is unknown and hard to predict by common prediction models. Sound insulation measurements have been performed in a small laboratory setup [Rodrigues and Coutinho, 2008]. To investigate the sound insulation of water further, sound insulation measurements of a 26, 44 and 83 mm thick water layer have been performed according to ISO 140-3 in the Laboratorium voor Akoestiek of Eindhoven University of Technology. A water basin was created in the 10 m^2 floor opening, constructed on a wooden structure with steel net and foil. The resulting airborne sound insulation index Rw of the water layers are 35, 38 and 42 dB respectively. The graph of the sound insulation per frequency shows an average increase of about 3 dB/octave. The doubling of the water thickness also results in an increase of about 3 dB. It is shown that water is a useful material in terms of sound insulation properties.

Thu 11:20 Jurriaanse Zaal Building acoustics 2

Absorption and Thermal Capacity of Thermally Activated Concrete Slabs and Open Ceilings

Hanneke Peperkamp and Martijn Vercammen
Peutz bv, Mook

Thermally activated concrete slabs offer an interesting possibility to control the indoor climate. When applying this system, a thermal as well as acoustic comfortable indoor climate should be realised.
To control the sound pressure levels, sufficient sound absorbing material has to be applied. This absorbing material is, regarding the available surface, often placed as a closed, suspended ceiling. However, a proper sound absorbing material is often also thermally insulating, which negatively affects the thermally activated concrete slabs. Therefore open ceiling systems are applied.
In this publication, the acoustic quality of sound absorbing systems is discussed combined with the effect of the absorbing ceiling elements on the thermal capacity of the activated concrete slab.
Based on measurements accomplished in the Peutz Laboratory for Acoustics, the influence of geometric and material parameters of porous materials on the achievable absorption is discussed. Furthermore, an empirical model is given which estimates the achievable absorption

coefficient of an open ceiling, based on the applied materials and constructions. In addition, based on measurements at the Peutz Laboratory of Building Physics, the reduction of the thermal capacity is indicated for some ceiling geometries. The reduction is given relative to the capacity of concrete slabs without ceiling elements.

Thu 11:40 Jurriaanse Zaal Building acoustics 2

Reduction of Low-frequency Vibrations of Wooden Floors by Tuned Mass Dampers

Hendrik Reichelt[a], Ulrich Schanda[a] and Andreas Rabold[b]
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A steady increase of communal and industrial buildings in the wood building sector can be recognized. Due to their usage these buildings are built with wide spanned wooden floors. During normal usage these floors tend to be excited to unpleasant low-frequency (5-100 Hz) vibrations. Besides the conventional constructions it is feasible to implement supplemental assemblies to reduce these vibrations. Due to the strongly modal behavior of these light floors a significant reduction of the vibrations and the sound radiation can be achieved. One possibility to reduce vibrations of wide spanned constructions is the tuned mass damper (TMD). Within a study at the University of Applied Sciences Rosenheim the influence of TMDs in wooden floors was investigated. The contribution will show the mode of operation of a TMD and how to dimension the TMD. Subsequently the reduction of the vibration of two wooden floor systems by TMDs will be shown. This will be visualized by comparing the empirical modal analyses of the undamped and damped systems. Finally the experimental results will be compared with a FEM-simulation.

Thu 8:40 Fortis Bank Zaal Auditory processing 1

Headphone Reproduction via Loudspeakers using Inverse HRTF-Filters

Tobias Weissgerber[a], Klaus Laumann[b,a], Günther Theile[b] and Hugo Fastl[a]
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The ”Binaural Sky” is a loudspeaker array above the listener’s head serving as a ”virtual headphone” for binaural room synthesis. WFS focussed sources are rendered close to listener’s ears and used for HRTF inverse filtered signal reproduction (”HRTF inverse filtered sources”). Their locations are constant relative to the ears for every head direction by means of head tracking data. Hence, there is no need to change the inverse HRTF (”crosstalk cancelling”) filters due to head rotations, which could create audible artefacts and instabilities. The present study is concentrating on the stability of the virtual headphone characteristics due to the number and positions of the focussed sources. In a first step the optimum configuration of ”real” loudspeakers without using WFS is examined. This requires measurements of nearfield-HRTFs for different azimuth and distance of the speaker. HRTF inverse filters based on these data are applied to a number of speaker configurations. The speakers are placed in two circles of different diameter around the listener’s head for playback. Transfer functions of the virtual left and virtual right headphone channel as well as both channels are measured and the quality of the virtual headphone evaluated.

Thu 9:00 Fortis Bank Zaal Auditory processing 1

A loudspeaker-based room auralisation system for auditory perception research

Jörg M. Buchholz and Sylvain Favrot
Centre for Applied Hearing Research, TU of Denmark

Most research on basic auditory function has been conducted in anechoic or almost anechoic environments. The knowledge derived from these experiments cannot directly be transferred to reverberant environments. In order to investigate the auditory signal processing of reverberant sounds, a loudspeaker-based room auralisation (LoRA) system is proposed here. The LoRA system efficiently combines modern room acoustic modelling techniques with higher-order Ambisonic auralisation. Thereby, aspects of the auditory precedence effect are utilized to realise highly authentic room reverberation. This system provides a flexible research platform for conducting auditory experiments with normal-hearing, hearing-impaired, and aided hearing-impaired listeners in a fully controlled and realistic environment. This includes measures of basic auditory function (e.g., signal detection, distance perception) and measures of speech intelligibility. A battery of objective tests (e.g., reverberation time, clarity, interaural correlation coefficient) and subjective tests

(e.g., speech reception thresholds) is presented that demonstrates the applicability of the LoRA system.

Thu 9:20 Fortis Bank Zaal Auditory processing 1

Tone in noise detection: a review of spectral integration

Nicolas Le Goff[a], Armin Kohlrausch[b] and Jeroen Breebaart[b]
[a]*Eindhoven University of Technology;* [b]*Philips Research, Eindhoven*

An experiment is conducted to measure binaural (NoSπ) and monaural (NoSo) tone in noise detection thresholds for masker bandwidths that varied from 10 to 1000 Hz while the tone to detect had a frequency of 500 Hz. Thresholds are expressed as signal to masker spectrum level ratio (S/No). For bandwidths less than 1 ERB, thresholds in all measured conditions increase with increasing masker bandwidth. For bandwidths wider than 100 Hz three different behaviors are observed. Thresholds for NoSo conditions with a running noise masker remain essentially constant which is in line with the critical band paradigm. In contrast, thresholds for NoSo conditions with frozen noise masker and NoSπ conditions both show a slight increase with increasing masker bandwidth beyond 100 Hz. A third effect of bandwidth is observed in our data for NoSπ conditions with an overall ILD of 30 dB. In this case, thresholds show a decrease of about 5 dB when the masker bandwidth is increased from 100 to 1000 Hz. This unusual bandwidth dependence is also found for binaural detection with a reduced masker correlation and for monaural detection of a 6-kHz tone in noise. An overview of the spectral integration for these various conditions is given and potential explanations are discussed.

Thu 9:40 Fortis Bank Zaal Auditory processing 1

Lateralization and detection of interaurally time delayed tone complexes in diotic masking noise.

Steven van de Par[a], Armin Kohlrausch[a] and Nicolas Le Goff[b]
[a]*Philips Research, Eindhoven;* [b]*Eindhoven University of Technology*

We will report results on the lateralization and detection of interaurally time delayed tone complexes that were presented in diotic noise as a function of level of the tone complex. The tone-complexes had a component spacing of 20 Hz and a frequency range of 250-850 Hz, the noise was band limited to the same frequency range. It appears that, when the tone-complex-to-noise ratio is -5 dB or more, listeners can estimate the extent of lateralization fairly well in noise, but not for lower tone-complex-to-noise ratios. In a second experiment, the threshold level needed for detecting the presence of the same tone complex in noise was measured. Results show that thresholds decrease with increasing time delay and are as low as -19 dB for an interaural time delay of 0.5 msec. When no time delay is present, and detection depends only on monaural listening thresholds are around -6dB. Apparently, detectability of the tone complex based on binaural cues is not sufficient for being able to hear the lateralization of the tone complex. Results suggest that the tone-complex level should be high enough such that the monaural cues resulting from the presence of the tone-complex can support the lateralization of the tone-complex.

Thu 10:00 Fortis Bank Zaal Auditory processing 1

Effects of Binaural Jitter on Sensitivity to Interaural Time Differences: Hearing Impaired Listeners

Anna Katharina Könsgen, Bernhard Laback and Piotr Majdak
Austrian Academy of Sciences, Acoustics Research Institute

Interaural time differences (ITDs) provide important information for localizing sound sources and for understanding speech in noise. For pulse trains at higher rates, the sensitivity to ongo-ing envelope ITD is strongly reduced, an effect known as binaural adaptation. Recent studies showed that binaurally synchronized randomness (binaural jitter) of the timing of individual pulses improves ITD sensitivity at such high pulse rates. This has been shown in normal-hearing (NH) and in cochlear-implant (CI) listeners. This study tested the hypothesis that for such high pulse rates, binaural jitter improves ITD sensitivity also in hearing-impaired (HI) listeners with moderate sensorineural hearing loss. ITD sensitivity was measured in twelve HI listeners using a left/right discrimination task. Stimuli were narrow-band noise (NBN) and bandpass-filtered click trains with and without jitter. The pulse rates were 400 and 600 pulses per second (pps). The results support the hypothesis that introducing binaural jitter improves ITD sensitivity in HI listeners. ITD sensitivity improved with increasing amount of jitter. For NBN, ITD sen-sitivity was similar to that for 400-pps trains with moderate jitter and to that for 600-pps trains with large jitter. Overall, the importance of temporal randomness appears to generalize to normal and impaired acoustic hearing and electric hearing.

Thu 10:40 Fortis Bank Zaal Auditory processing 1

Azimuthal Localization of Concurrent Speakers Employing Interaural Coherence

Mathias Dietz, Stephan D. Ewert and Volker Hohmann
Universität Oldenburg

A computational model for the localization of concurrent speakers is presented that extends the binaural model presented by Dietz et al. (Fortschritte der Akustik - DAGA 2007, 107-108). The processing is motivated by mammalian physiology and human psychoacoustic data. A wide range of interaural parameters is exploited, namely the interaural level difference (ILD), the interaural phase difference (IPD) of the fine structure, the IPD of the fundamental modulation frequency and the interaural coherence (IC). The parameters are determined for each auditory filter which allows a simultaneous estimation of several sound directions. By focusing on instants with high interaural coherence the robustness of the estimation could be greatly improved (Faller and Merimaa, 2004, J.

Acoust. Soc. Am., vol. 116, 3075-3089). For two well-separated speakers in quiet the estimation error is generally less or equal 10°. In noisy environments a single speaker can be localized with an error of less than 20°, even for a signal-to-noise ratio of 0 dB.

Thu 11:00 Fortis Bank Zaal Auditory processing 1

A Probabilistic Model for Robust Acoustic Localization based on an Auditory Front-end

Tobias May[a], Steven van de Par[b] and Armin Kohlrausch[b]
[a]*Eindhoven University of Technology;* [b]*Philips Research, Eindhoven*

Although extensive research has been done in the field of localization, the degrading effect of reverberation and the presence of multiple sources on localization performance has remained a major issue. The classical approach to localize an acoustic source in the horizontal space is to search for the main peak in the cross-correlation function, which corresponds to the interaural time difference (ITD) between both ears. Apart from ITD, the interaural level difference (ILD) can contribute to localization, especially at higher frequencies where the wavelength becomes smaller than the diameter of the head, leading to ambiguous ITD information. Motivated by the robust localization performance of the human auditory system, its peripheral stage is used as a front-end for binaural cue extraction. The interdependency of ITD and ILD on azimuth is a complex pattern that depends also on the room acoustics and is therefore learned by azimuth-dependent Gaussian mixture models. Multi-conditional training is performed to incorporate the spread of the binaural features caused by multiple sources and the effect of reverberation. The trained localization model outperforms state-of-the-art localization techniques in simulated adverse acoustic conditions. Furthermore, the model is capable of generalizing to changes in the simulated room absorption and to unknown source/receiver combinations.

Thu 11:20 Fortis Bank Zaal Auditory processing 1

The effect of spectral density and bandwidth on the precedence effect

Bernhard Seeber
MRC Institute of Hearing Research

The precedence effect shows the ability of the auditory system to perceptually suppress reflections such that a single sound is heard at the direction of the leading sound. A low frequency tone is localized on the basis of its interaural phase, but the addition of its delayed copy alters the interaural phase and thus its location. A larger bandwidth is needed to stabilize localization at the lead either through integrating binaural information across frequency or through extracting information from the temporal envelope. Increased spectral density will give rise to faster envelope fluctuations within an auditory filter, thus facilitating the evaluation of binaural cues from the envelope.

The present study investigated the spectral density needed for the precedence effect for ongoing stimuli of various bandwidths. Stable precedence could not be obtained at or below 1 Bark bandwidth regardless of the number of tones per critical band (CB). Precedence emerges with at least two tones per CB over 2 Bark. No precedence was found with one tone per CB which suggests that envelope information within a CB is needed. The echo threshold increases with increasing bandwidth or spectral density, suggesting that within and across-channel information is combined.

Thu 11:40 Fortis Bank Zaal Auditory processing 1

Localisation dominance for long lead-lag stimuli in background noise

Stefan Kerber and Bernhard Seeber
MRC Institute of Hearing Research

The precedence effect describes the ability of the auditory system to integrate sounds reaching the ear directly with their various reflections into a single sound event. The first arriving sound determines the perceived location of a source regardless of present reflections which was named ”localization dominance”. In the laboratory this is studied by playing a leading sound from one location followed by its delayed copy, the lag, from a different location. Only few studies examined if the effect functions in the presence of background noise, particularly for stimuli other than clicks.

We studied the localization of long-duration lead-lag stimuli in the presence of diffuse background noise. Lead and lag were identical copies of noise played from loudspeakers at $\pm 30^\circ$ in an anechoic chamber. The signal-to-noise ratio was varied. Subjects pointed to the perceived location(s) of the lead-lag pairs with a light pointer. Preliminary results show that the maximum delay time for which lead and lag form one image at the lead position (echo threshold) increases through the presence of diffuse background noise. This is surprising since most models explain the precedence effect through the suppression of binaural cues in the lag which are made here less accessible by the noise.

Thu 15:00 Jurriaanse Zaal Auditory processing 2

Robust Vowel Detection

Bea Valkenier, Dirkjan Krijnders, Ronald van Elburg and Tjeerd Andringa
University of Groningen

Inspired by the robustness of human speech processing we aim to overcome the limitations of most modern automatic speech recognition methods by applying general knowledge on human speech processing. As a first step we present a vowel detection method that uses features of sound believed to be important for human auditory processing, such as fundamental frequency range, possible formant positions and minimal vowel duration. To achieve this we took the following steps: We identify high energy cochleogram regions of suitable shape and sufficient size, extract possible harmonic complexes, complement them with less reliable signal components, determine local formant positions through interpolating between peaks in the harmonic complex, and finally we keep formants of sufficient duration. We show the effectiveness of our method of formant detection by applying it to the Hillenbrand dataset both in clean conditions, as in noisy and reverberant conditions. In these three conditions the extracted formant positions agree well with Hillenbrand's findings. Thereby, we showed that, contrary to many modern automatic speech recognition methods, our results are robust to considerable levels of noise and reverberation.

Thu 15:20 Jurriaanse Zaal Auditory processing 2

Modelling binaural speech recognition

Thomas Brand

Universität Oldenburg, Medizinische Physik

In many cases speech has to be recognized in adverse listening conditions including noise sources from different directions and adverse room acoustics. Furthermore, hearing-impairment compromises intelligibility in such conditions. Binaural processing of the auditory system can cause a substantial improvement of speech recognition. A binaural speech intelligibility model (Beutelmann and Brand, JASA, 2006) yields accurate predictions of speech reception thresholds (SRTs) in presence of a stationary noise source at arbitrary azimuths and in different rooms. The model combines a multi-frequency channel equalization-cancellation (EC) process and the speech intelligibility index (SII). The effect of hearing loss is simulated by adding uncorrelated masking noises (according to the pure-tone audiogram) to the ear channels. A recent revision of this model is based on an analytical expression of binaural unmasking for arbitrary input signals and is computationally relatively efficient. An extension for non-stationary interferers was realized by applying the model to short-time frames of the input signals and averaging over the predicted SRT results. Model predictions are compared to binaural speech intelligibility data from normal-hearing and hearing-impaired listeners, incorporating different combinations of rooms, sound source setups and noise types.

Thu 15:40 Jurriaanse Zaal Auditory processing 2

Benefits and Risks of Hearing Instruments in a Noisy Environment

Wouter A. Dreschler

Academic Medical Center, Amsterdam

High levels of background noise and reverberation as found in many working environments increase the demands for auditory communication. This holds especially for employees with a hearing impairment that usually need a better signal-to-noise ratio. This paper describes the quantitative effects with respect to speech perception in noise and the detection and identification of warning signals. A related aspect is the

hindrance of background noise, as can be objectified by testing acceptable noise levels.
In some cases hearing instruments and additional technical options like remote microphones may be required for an adequate level of auditory communication. Modern hearing aids allow highly sophisticated features on noise reduction and directionality. In addition, automatic environment classification facilitates specific amplification strategies for special situations. However, the individual fitting of these advanced hearing aids is very critical and determines strongly the degree of success. The added value of interactive fitting methods will be discussed.
An important issue is that high levels of background noise may cause noise-induced hearing loss. For specific schemes of amplification even moderate levels of background noise may be damaging. The issue of ”aided” exposure deserves more attention.

Thu 16:00 Jurriaanse Zaal Auditory processing 2

Hearing Impairment: Benefit of Optimized Beamforming obtained with Hearing Glasses in a Multi-Source Noise Field

Uwe Baumann and Tobias Rader
Audiological Acoustics/ZHNO Univ. of Frankfurt

One of the main obstacles complained by persons suffering from hearing impairment is their highly degraded speech perception under adverse listing conditions. Even high end hearing aids with multi channel beamforming do not restore this ability sufficiently in complex noise. A recently introduced hearing aid embodied in a pair of glasses frame offers advanced directivity by means of four microphones mounted in line in each temple in order to obtain sophisticated beamforming (DI 9 dB). We were interested in the performance of this device in a complex noise field compared to behind the ear hearing aids equipped with custom directional microphones. Ten experienced users of different hearing aids with directional microphone technique were fitted with this type of hearing glasses. Speech perception performance was compared in quiet and in a multi-source noise field consisting of 4 independent sources with speech simulating noise according to Fastl (1987). Results showed for the hearing glasses significantly higher signal to noise ratios compared to the own BTE device condition. With the high end hearing glasses, some subjects were able to discriminate speech even at an SNR of -15 dB in a complex noise listening situation.

Thu 16:20 Jurriaanse Zaal Auditory processing 2

Sound localization with and without hearing aids

Tim van den Bogaert, Evelyne Carette and Jan Wouters
KU Leuven, ExpORL

Building a realistic spatial representation of our surrounding environment is an important task of our auditory system. This allows us to localize sounds and to understand speech in adverse scenarios. An inaccurate spatial map will degrade localization and speech perception and will lead to an unnatural perception of sounds. The question whether hearing aids transmit enough cues and whether hearing aid users are able to use these cues has been gaining a lot of interest recently. This is partly due to recent technology which enabled binaural hearing aid designs aiming at preserving binaural localization cues and flexible microphone configurations which aim at preserving the monaural spectral cues.
Normal hearing and hearing impaired subjects were evaluated in different localization experiments: elevation and front-back discrimination both focusing on monaural spectral cues and left-right localization focusing on binaural cues. Hearing impaired subjects were tested without and with different types of hearing aids. Most of the hearing impaired subjects were capable of localizing sounds relatively accurate if the test procedure included a frequency specific compensation for their hearing impairment in all three dimensions. A difference between the different hearing aid designs was found. This difference was especially prominent in the front-back and elevation experiments.

Thu 17:00 Jurriaanse Zaal Auditory processing 2

Localization of Sound Sources in Median-Plane with Channelized Head-Related Transfer Functions

Piotr Majdak, Bernhard Laback and Matthew J. Goupell
Austrian Academy of Sciences, Acoustics Research Institute

Head-related transfer functions (HRTFs) describe sound transmission from the free-field to the ear-drum. They contain spectral features, which vary according to the sound direction and differ among listeners. Median-plane localization with modified HRTFs mostly results in a degradation of localization ability.
Cochlear-implants (CIs) are auditory prostheses, which restore hearing in the profoundly deaf by electrically stimulating the cochlea. The transmission of spectral features of the HRTFs is very limited in CIs. Thus, CI listeners show a degraded localization ability for sounds along the median plane. Main limitations result from the small number of frequency channels, limited frequency range, and channel interactions.
This study investigated the effect of the number of channels on median-plane sound localization in normal-hearing subjects listening to a simulation of electric hearing where individual HRTFs were processed with a Gaussian-tone vocoder within the frequency range 0.3 to 16 kHz.

Results show that at least nine frequency channels are required to provide spectral features without a substantial degradation of localization ability. Four of those channels cover the frequency region of the most prominent median-plane localization features ($>$ 4 kHz).
This study is supported by FWF (P18401-B15).

Thu 17:20 Jurriaanse Zaal Auditory processing 2

BRTF - Body Related Transfer Functions for Whole-Body Vibration Reproduction Systems

Ercan Altinsoy and Sebastian Merchel
TU Dresden, Inst. f. Akustik und Sprachkommunikation

If binaural recorded signals are played back via headphones, the transfer characteristic of the reproduction system has to be compensated for. Unfortunately the transfer characteristic depends not only on the transducer itself, but also on mounting conditions and individual properties of the respective ear. This is similar with reproduction systems for whole body vibration (e.g. electro-dynamic or hydraulic exciters). The transfer characteristic depends to a great extend on the individual body properties, e.g. weight, body mass index, adipose. For fundamental scientific studies or comfort studies it is required to present the same stimulus (amplitude and frequency) to all subjects. This is only possible by taking into account individual equalization filters which are based on the individual body properties of each subject and the specifications of the reproduction system. In this study body related transfer functions of 40 subjects are measured using an electro-dynamic excitation system. In addition anthropometric data of the subjects are collected. The importance of the individual transfer functions for whole-body vibration perception experiments is discussed. One aspect is the definition and creation of an average body transfer function representing a ”best matched body”. Another topic is the influence of the body properties on the transfer function characteristic.

Thu 17:40 Jurriaanse Zaal Auditory processing 2

Audiotactile Feedback for Touch Screens

Sebastian Merchel and Ercan Altinsoy
TU Dresden, Inst. f. Akustik und Sprachkommunikation

Touch sensitive displays and touch surfaces are more and more replacing physical buttons. Ticket vending machines or telecommunication devices, such as cellular phones (e.g. the Apple iPhone), use this space and cost saving technology. If a physical button is pressed, audio and tactile feedback confirms the successful operation. The loss of audiotactile feedback in touch sensitive interfaces might create higher input error rates and user dissatisfaction. Audiotactile feedback is especially important if there is no optical confirmation (e.g. the finger hides the display). Therefore the design and evaluation of suitable signals is necessary. In this paper a touch sensitive system is presented that reproduces

event triggered audio tactile feedback. The tactile component is generated using an electro-dynamic exciter. The audio signal is played back via headphones. In a pilot study with six subjects the usability of such a system is investigated. Different synthetic signals are developed and compared with each other. In a dialing task the objective performance of the subjects (effectiveness, error rate) is measured. The perceptual quality of the designed signals is evaluated using a questionnaire.

Thu 15:00 Fortis Bank Zaal Modelling in room acoustics

Reflections At Room Boundaries In Computer Simulation Programs Based On Ray-Tracing

Margriet Lautenbach[a] and Martijn Vercammen[b]
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Room acoustic computer simulation programs based on ray-tracing use different simplification methods to model the reflections at the room boundaries. In due time, several methods have been proposed, discussed and implemented how to model the complex reality with diffusion and diffraction into specular and diffuse reflections in ray-tracing algorithms. For a user it is quite hard to find out what the differences are between the commercially available simulations programs on the given subject. Although there are three round robins and many articles that compare the output of several programs, the algorithms underneath are not always obvious. This makes the programs kind of ”black boxes” and it’s therefore difficult to make a well founded choice between the several programs or apply the right program in the right situation. This paper gives an overview of the used calculation methods in some well known software packages. A comparison is made to the wave based method for reflections at room boundaries to provide insight in the all-over results of the calculated impulse response and derived parameters.

Thu 15:20 Fortis Bank Zaal Modelling in room acoustics

Investigation of sound diffusion characteristics using scale models in concert halls

Jin Yong Jeon and Yong Hee Kim
Hanyang University, Dep. Architectural Engineering, Seoul

The diffuse characteristics of the surfaces in lateral walls were investigated to find out the appropriate diffuser profiles in concert halls. The design factors of diffusers in auditoria are geometrical shape, structural height and surface coverage. The hemisphere diffusers were evaluated by measuring the diffusion and scattering coefficients in a 1:10 scale model reverberation chamber, and then applied to scale model concert halls, where the acoustical parameters such as RT, EDT, C80, and SPL were calculated from the impulse responses. In addition, the numbers and amplitudes of the reflection rays within -20 dB after the direct sound were calculated to evaluate the diffuseness in the halls. It was found that the diffusive lateral surfaces close to the stage area are more effective in increasing both R_N and R_E.

Thu 15:40 Fortis Bank Zaal Modelling in room acoustics

Introducing diffraction into beam tracing - some new results

Uwe Stephenson
HafenCity Universität Hamburg

The problem of an efficient introduction of diffraction into ray tracing is still unsolved. The author's energetic approach to diffraction of 1986 is based on Heisenbergs un-certainty principle (' the closer the by-pass distance to edges the stronger the ray diffraction'). This model has been validated at the single screen and the slit as reference cases and in last years been extended to the more efficient beam tracing technique and tested for many additional configurations where the results are compared with Svensson's exact wave-theoretical secondary edge source model. Recent numerical experiments with a beam integration approach deal with improvements of the by-pass-distance-dependend diffraction functions and checking the fulfilling of the reciprocity principle, further more with the validation of the approach for diffraction of higher order at least double diffraction at the two edges of a "thick" obstacle. With some restrictions, it seems that even diffraction of sound - like light - may be handled as flow of particles. To avoid an explosion of computation time, it is later aimed at to combine beam diffraction with Quantized Pyramidal Beam Tracing (QPBT).

Thu 16:00 Fortis Bank Zaal Modelling in room acoustics

Optimization of Input Parameters for Real-Time Room Acoustics Simulations

Dirk Schröder
Institute of Technical Acoustics, RWTH Aachen University

Virtual Reality (VR) systems are today's state-of-the-art tools in many development and analysis processes, e.g., in the fields of architecture, engineering, psychology and medical science. The quality of a VR-system is assessable by the user's degree of immersion, which improves with the number of simulated coherent stimuli and level of interactivity. In this context, the visual perception is significantly augmented by matching sound stimuli, as auditory information helps to assign meaning to visual information, and vice versa. The user evaluates these events on attributes such as spaciousness and source localization, which enforces the feeling of actual presence especially in the case of indoor-scenarios. In this contribution user studies are presented which were carried out on a real-time VR-system with integrated hybrid room acoustics simulation (image sources/ stochastic ray tracing). Subjects had to evaluate situations with changing simulation parameterization, e.g., image source order and number of launched particles, to find a good balance between the accuracy of the room acoustic simulations, the user's subjective feeling of presence and the simulation's computational costs.

Thu 16:40 Fortis Bank Zaal Modelling in room acoustics

An interactive and real-time based auralization system for room acoustics, implementing directional impulse responses and multiple audio reproduction modules for spatialization (the AURALIAS project).

Lionel Bos and Jean-Jacques Embrechts
University of Liege

The AURALIAS project aims at developing an interactive and real-time auralization system for room acoustics. The user is immersed in a virtual sound field, looking at the virtual room and interacting with it through an appropriate interface. The real-time auralization system first imports pre-computed Directional Room Impulse Responses (DRIRs) according to the current situation in the virtual world. Those DRIRs, previously obtained by a ray tracing software, are then convolved with the anechoic signal using a frequency block segmented convolution. Filtered signals are then distributed either to a VBAP or an HRTF sound field reproduction module. The first prototype allows the auralization of multiple isotropic sound sources and the displacement of the virtual listener including head's orientation. Some other features like changing "on the fly" the anechoic signals and the current room are also implemented. Further developments will offer the support for moving sources, mobile sweet spot, user's own speech auralization and both real-time acoustical and geometrical modifications of the virtual room. Besides, the software's architecture should be compatible with several room acoustics algorithms, convolution modules or sound field reproduction systems.

Thu 17:00 Fortis Bank Zaal Modelling in room acoustics

Uncertainty in Acoustic Boundary Characterisation and its Influence on the Sound Field in Room Acoustic FE Simulations

Marc Aretz and Julian Knutzen
Institute of Technical Acoustics, RWTH Aachen University

The room acoustic design of the low frequency characteristics of small rooms still poses a challenging task to acoustic designers and consultants. In this field numerical methods such as the finite-element-method (FEM) are promising contenders for the realistic simulation of low frequency sound fields. While contemporary CAD and meshing tools facilitate the design of high quality geometric models with regular and sufficiently fine meshes, the Achilles' heel of room acoustic FEM simulations appears to be the realistic representation of the room boundaries. The presented study investigates the influence of different boundary representations on the simulated sound field in small rooms. Therefore measurements and FE simulations of the sound field in a scale model room with a well defined geometry and variable boundary conditions have been conducted. Two different approaches are followed to model the porous absorbers on the room boundaries. Firstly the absorber layers

are modelled with their exact 3D dimensions using an ’equivalent homogeneous fluid model’. Secondly the acoustic characteristics of the absorber layers are modelled by an appropriate impedance boundary condition, which is determined by measurement or model calculation based on measured flow resistivities. The audibility of differences between simulation and measurement results is assessed by subjective testing.

Thu 17:20 Fortis Bank Zaal Modelling in room acoustics

Auralization examples to discuss the reverberation time as a standard for sports facilities

Lau Nijs[a] and Monika Rychtarikova[b]

[a]*TU Delft;* [b]*KU Leuven, Afdeling Akoestiek en Thermische Fysica*

In sports facilities the height of the hall is mainly small when compared to the length of the hall. This non-cubic form is the first reason why deviations are found between the reverberation time from Sabine’s method on one side and measurements, ray-tracing predictions and calculations from standard IEC-12354-6 on the other. A second reason why reverberation times increase even more is that normally most absorption is positioned on the ceiling. Hence, to meet legal standards expressed in reverberation times, it seems that more absorption is needed than predicted by Sabine’s method. However, sound pressure levels are much less sensitive to the distribution of materials, which will be illustrated by examples from computer calculations and auralizations. Some differences between sound samples may be heard, but they vanish in noisy situations. In this paper the question will be discussed if there is a better acoustic variable than the reverberation time to express and measure the acoustical quality in a sports facility.

Thu 17:40 Fortis Bank Zaal Modelling in room acoustics

Measuring directivities of musical instruments for auralisation

Martin Pollow

Institute of Technical Acoustics, RWTH Aachen University

With the ever rising computational performance, auralisation becomes a more and more important tool in acoustics. It offers the possibility to make a sound source of known directivity audible at any arbitrary point in a spatial setting. As natural sources, like musical instruments, lack repeatability of excitation, their radiation has to be measured in all directions simultaneously. A spherical microphone array surrounding the musician can be used for this kind of measurements, whose geometry allows correct results for directivities that are band-limited to a certain degree of spatial variation.

Hereby, the exact locations of the physical sound sources can only be approximated and are in general not aligned to the center of the sphere. This yields to higher oscillations in the measured radiation pattern and can create errors by spatial aliasing. Furthermore, the ground reflections of a semi-anechoic chamber constitute a problem.

A comparison for different tones played in different styles allows to evaluate the similarity of their radiations for each instrument. These sets of directivity patterns are then simplified to gain a more general, static characteristic. The goal is the retrieval of compact directivity data resembling reality as close as possible to enhance the quality of auralisation of musical instruments.

Thu 8:40 Van Cappellen Zaal Noise control 2

Validation of a New Procedure for Rating Shooting Sounds with the Help of Field Survey Data

Joos Vos
TNO Human Factors

In laboratory studies on the annoyance caused by shooting sounds produced by small, medium-large and large firearms, the annoyance was well predicted from the outdoor A-weighted and C-weighted sound exposure levels (ASEL and CSEL). For single events, the rating sound level, Lr, was given by Lr = ASEL + β(CSEL - ASEL)(ASEL - α) + 12 dB. With this formula it is possible to express Lr in such a way that, overall, it numerically corresponds to the A-weighted day-evening-night level of equally annoying road-traffic sound. On the basis of the laboratory results the parameter values α = 47 dB and β = 0.015 were recommended. To validate these parameter values, subjective reactions to shooting and road-traffic sounds were determined for 400 respondents, the noise dose for road traffic was calculated, and the ammunition spent in the 12 months preceding the administration of the questionnaire was listed. Next, for all relevant impulsive sources, the required ASELs and CSELs received in the various residential areas were determined. On the basis of the data from the majority of the residential areas, the estimated β in combination with α = 45 dB was equal to 0.025 $\pm$ 0.012.

Thu 9:00 Van Cappellen Zaal Noise control 2

Noise Management at Civil Shooting Ranges

Dieter Knauss[a] and Karl-Wilhelm Hirsch[b]
[a]*deBAKOM GmbH;* [b]*Cervus Consult GmbH*

DIN ISO 17201-2 proposes a scheme to set up a daily noise management on civil shooting ranges. The scheme relies on rating levels based either on calculations or measurements of the LAFmax at relevant receptor points. In order to employ such a management system, detailed information have to be known. For civil shooting ranges however, it is sometimes difficult to gather all the details in advance and to put everything together in an appropriate computer based management system.

This paper reports on a system that follows the basic ideas of the DIN ISO 17201-5 but relies on the measurement of each single shot at given positions at close distance during normal operation. Using these single shot levels and the met. data, the system applies a transfer matrix to predict the contribution of each shot to the rating level at each relevant receiver point. The coefficients of the matrix are determined during a preceding measurement campaign and accounts for the wind speed and direction.

Thu 9:20 Van Cappellen Zaal Noise control 2

Auswirkungen des Lärmmanagements für das Schießen mit großkalibrigen Waffen

Frank Hammelmann[a], Bernd Wiedemann[b] and Eberhard Braun[b]
[a]*Cervus Consult GmbH;* [b]*Streitkräfteunterstüzungskommando, Köln*

Im Jahr 2007 ist vom Bundesministerium der Verteidigung (BMVg) die Lärmmanagementrichtlinie (LMR) für das Lärmmanagement auf Schießplätzen der Bundeswehr erlassen worden. Diese Richtlinie schreibt ein tägliches Lärmmanagement so vor, dass einerseits jeweils bestimmte Lärmkriterien einzuhalten sind (Schutz vor erheblichen Belästigungen) und andererseits in Umsetzung der zweiten Forderung des BImSchG der tägliche Schießbetrieb so zu planen ist, dass die Nachbarschaft so gering wie möglich belastet wird (Minimierung der Belastung). Das Lärmmanagement beruht also nicht auf einer statischen Genehmigung, sondern definiert einen dynamischen Prozess, der täglich mit dem Lärmproblem konfrontiert. Seit dem 1.1.2008 wird das Lärmmanagement auf allen Truppenübungsplätzen der Bundeswehr im Probebetrieb durchgeführt.
Der Beitrag berichtet über die ersten Auswirkungen des Lärmmanagements. Es wird aufgezeigt, dass das Lärmmanagement schon im Probebetrieb durch zunächst organisatorische Maßnahmen zu einer Minderung der Lärmbelästigung für die Anwohner geführt hat.

Thu 9:40 Van Cappellen Zaal Active noise control 2

Feedback Control for Active Noise Reduction in Headsets

Jens Graf
Leibniz-Univ. Hannover, Instit. for Meas. and Automatic Control

With regard to production costs, control strategies for commercial ANR-headsets (Active Noise Reduction) have to be realized simple and efficient. Therefore, the implementation of computationally expensive algorithms, which require fast and expensive digital signal processors, is avoided. Besides an expensive DSP, further production costs are caused by a reference microphone, which is necessary in case of a feedforward controller. However, in contrast to a feedforward controller, the feedback control approach does not require such a reference microphone. Thus, the realization of a feedback control strategy enables to save the reference microphone as well as the associated analog input circuit respectively. In order to design an efficient ANR-system, in this paper a time discrete feedback control approach is introduced. Basically, the proposed feedback control loop consists of two parts. One part is formed by the non-adaptive controller, while the other part consists of an adaptive algorithm. The non-adaptive controller guarantees steady-state noise reduction and the overall performance of the ANR-system is improved by the adaptive algorithm. The design of the control system is described and the active noise reduction performance is presented.

Thu 10:00 Van Cappellen Zaal Active noise control 2

Parameterization of an adaptive Multiple-Channel Active Noise Controller

Kay Kochan and Delf Sachau
Helmut-Schmidt-University Hamburg, Inst. Mechatronics

Noise induced by machines, vehicle engines or propellers can lead to high tonal sound pressure levels. Due to increasing comfort requirements and occupational health and safety regulations, noise protection is a mandatory component of each modern technical system. With advances in technology, this challenge can in some cases be solved by the active noise control method. As an application example, the active noise control method is applied to the acoustical problem of a loadmaster working area, which is in front of the cargo bay of a propeller driven transport aircraft. The present research is focused on the final design process of such an active noise control system. Here, the hardware and the positions of 8 secondary loudspeakers and 16 error microphones are already defined. But the controller software and parameters can still be adapted in order to improve the performance and robustness of the system. In the present paper the steepest descent frequency domain controller as well as different solutions of the robust controller parameterization will be discussed. Moreover, the active control results inside an acoustical mock-up of the loadmaster working area will be presented.

Thu 10:20 Van Cappellen Zaal Active noise control 2

Active Structural Control of Impacts with a View Towards Noise Control

Carsten Hoever[a], Wolfgang Kropp[b] and B. A. T. Petersson[c]
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Impacts between bodies of any sort often constitute an important source of noise and vibration. These impacts can be characterised as very short contacts between bodies leading to a sudden release of energy as audible noise and to vibrations of the involved structures. Resulting sound pressure levels often pose a serious health risk and structural vibrations lead to further noise emission, material fatigue and breakdown.
Due to a variety of source mechanisms and multiple possible ways of transmission, propagation and radiation, classical methods of noise and vibration control are often difficult to implement.
In this study it is theoretically investigated how noise and vibration generated by a sphere-plate impact can be affected by application of an active force acting on the plate at the impact location.

Two different numerical models are derived as a basis for simulations. A parameter study is conducted to investigate the possibilities of active structural control of impacts. With the obtained data, an active control configuration is developed which is applicable to a wide variety of plate-sphere impact situations and which leads to promising noise and vibration reduction.

Thu 11:00 Van Cappellen Zaal Active noise control 2

The Design and the Application of Fiber-Optic Microphones for Acoustic Measurements in Hot Environments

Holger Jochen Konle[a], Christian Oliver Paschereit[a] and Ingo Roehle[b]
[a]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics;* [b]*German Aerospace Center (DLR), Göttingen*

This paper presents the development of extrinsic fiber optic sensors for acoustic measurements in hot environments. Two interferometer designs were tested. The first designed prototype bases on the use of a commercial Mach-Zehnder interferometer and a stainless steel membrane (thickness 12.5μm) and was successfully applied for acoustic measurements in the exhaust gas duct of an atmospheric combustion chamber. All the studies with this prototype combination showed a high vibration sensitivity of the optical setup that arises from the fact that glass fiber vibrations change the optical length of the fiber and thus create a high noise level in the interference signal. Since the fiber type necessary for this interferometer is temperature resistant only up to 300°C, a new fiber based laser interferometer was designed basing on the etalon effect of a single mode optical fiber installed perpendicular in front of a highly reflecting membrane. This basic Fabry-Pérot interferometer was advanced by a Pockels-cell and an acoustooptical modulator to create a one-fiber interferometer creating two interferometer signals fulfilling the quadrature condition. Thus a highly vibration insensitive setup was created that showed good comparability to established measurement techniques for cold acoustic tests. Its application at an atmospheric combustion chamber is in preparation.

Thu 11:20 Van Cappellen Zaal Active noise control 2

Active control of the sound transmission through a double-glazing window

Joachim Bös[a], Sven Herold[b], Oliver Heuss[b], Michael Kauba[b] and Dirk Mayer[b]
[a]*TU Darmstadt, System Reliability and Machine Acoustics SzM;* [b]*Fraunhofer Institute LBF*

Within the framework of the European Integrated Project InMAR (Intelligent Materials for Active noise Reduction) a system for the active control of the sound transmission through a double-glazing window was developed. The window has a size of 1.2 m x 0.6 m and consists of two glass

panes (6 mm and 4 mm thick) with a gas-filled gap of 20 mm in between. The vibrational behavior of the window panes was analyzed by means of finite element simulations and experimental modal analyses using a scanning laser vibrometer. The results of these analyses were used to determine suitable locations for the application of piezoelectric patch transducers. If these patches are used as sensors they can serve as error sensors or reference sensors for an active control system, if they are used as actuators they can suppress the vibrations of the window panes and, therefore, reduce the sound transmission through the window. Several transducer configurations and control strategies were tested and compared with each other. The sound radiation of a broad-band shaker excitation was reduced by up to 20 dB at various frequencies, and the sound transmission through the window was reduced by up to 12 dB at various frequencies.

Thu 11:40 Van Cappellen Zaal Active noise control 2

LOEWE-Zentrum AdRIA - An important step towards the commercialization of adaptive systems

Joachim Bös[a], Thilo Bein[b] and Holger Hanselka[a,b]
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The LOEWE-Zentrum AdRIA (Adaptronik - Research, Innovation, Application) is a large interdisciplinary research project funded by the government of the German federal state Hessen with a duration of 3 years and a budget of 17.7 million euros for the time being. A second phase of three years with a tentative budget of 20 million euros is planned. It involves the Fraunhofer Institute for Structural Durability and System Reliability LBF, 22 institutes from 3 different departments of the Technische Universität Darmstadt as well as one department of the University of Applied Sciences, all located in Darmstadt, Germany. The goal of this project is the creation and sustainable implementation of an internationally leading adaptronics research institution in Darmstadt. Eventually, it will lead to the foundation of a new Fraunhofer Institute for Adaptronics and of four new university institutes. The project is organized in ten technology areas (e.g., materials, sensors and actuators, systems and control, embedded systems, simulation tools, reliability), which supply the necessary technologies for three application scenarios, i.e., adaptive car, quiet office, and adaptive tuned vibration absorber. An important step towards the commercialization of adaptive systems will be the close cooperation with companies and suppliers during the second phase of the project.

Thu 15:00 Van Cappellen Zaal Structural-acoustic optimization

Identification of the Optimal Lay-up for Fibre-reinforced Composite Plates with Minimized Sound Emission

Martin Dannemann and Werner Hufenbach
TU Dresden, Institute of Lightweight Eng. and Polymer Techn.

Increasing customers requirements and environmentally demands, the shortened development times as well as the necessary weight reduction and function-integration are pressing influence factors for the design of future lightweight structures. In the course of these challenging demands, composites with fibre- or textile-reinforcements offer new possibilities for a function-integrated design due to their versatile property profile. The big amount of adjustable parameters like layerwise fibre orientation, thickness and material requires the utilization of efficient optimization methods, which take into account not only the stiffness but also the vibration and sound behaviour. For this, an optimization tool based on a semi-analytical model for the calculation of the sound radiation of composite plates was developed. The design tool enables the calculation of optimal parameters to given requirements. Within this paper the results of selected optimization scenarios are presented including the influence of frequency/stiffness limits and mass restrictions on the optimal composite lay-up with minimized sound radiation. The developed tool and the performed studies allow the derivation of design guidelines giving the engineer the possibility of fully exploit the lightweight and vibro-acoustic potential of composite materials.

Thu 15:20 Van Cappellen Zaal Structural-acoustic optimization

Note on a Reliability Approach in Vehicle Acoustic Design

Leo W. Dunne[a] and Julian Dunne[b]
[a]*CDH AG;* [b]*University of Sussex*

In the acoustic design of passenger vehicles, simulation-based frequency response functions (FRFs) provide an important tool for noise, vibration and harshness (NVH) design assessment. Owing to the variability in actual vehicle components introduced by manufacturing, significant variability can exist between the measured acoustic responses of nominally identical vehicles. In the standardized NVH analysis processes currently used in the automotive industry, no attempt is made to account for variability introduced by manufacturing. The methods of reliability analysis, which include optimization algorithms such as used in FORM, provide tools for including the effects of the variability in model parameters in analysis. An interesting challenge is to predict upper bounds on the FRFs in the presence of probabilistically described uncertain model parameters. In this paper, an efficient acoustic FRF bounding method using extreme-value (EV) theory and a fast re-analysis known as the modal correction approach is described. In the method, relatively small samples of calculated (or measured) data are used to calibrate an asymptotic extreme-value model previously justified in an appropriate

hypothesis test. The paper shows that the method appears to provide a practical approach to acoustic response FRF bounding for probabilistically uncertain vehicle structures and thus may be used to calibrate the standard analysis processes.

Thu 15:40 Van Cappellen Zaal Structural-acoustic optimization

Optimization of Multilayered Porous Acoustic Absorbers

Heiko Andrä, Matthias Kabel, Hans Rieder, Martin Spies and Andreas Wiegmann
Fraunhofer ITWM

A new method for the optimal material and layer design of acoustic trims without the usage of prototypes is presented. As an example the acoustic behaviour of a two-layered ceiling panel is optimized without increasing its weight or its thickness.
The optimization method consists of two steps: (1) The possible varieties of porous materials are analysed by computer simulation on the microscale, i.e. a certain set of boundary value problems is solved on the fully resolved microstructure. In a subsequent homogenization procedure a characteristic set of effective acoustic material parameters is computed together with their dependency on microstructural parameters like porosity. (2) A data base of these pre-computed characteristic material parameters is used in the actual optimization algorithm, which determines both the optimal thickness of the layers and their optimal microstructure with respect to a desired acoustic absorption behaviour.
A wide variety of materials can be taken into consideration: fibrous absorbers, open and closed cell foams, viscoelastic solids, porous or viscoelastic composites.
Since the main difficulty in this procedure is the computation of the characteristic acoustic parameters, we present the verification of the numerical scheme used for the computation of these parameters for the case of cylindrical pores.

Thu 16:00 Van Cappellen Zaal Structural-acoustic optimization

Prediction of structure-borne sound and secondary airborne sound by three-dimensional finite element models

Mark Bless[a] and Ferry Koopmans[b]
[a]*Peutz Consult GmbH;* [b]*Peutz bv, Mook*

Nowadays sensitive utilisations as residential, cultural or office buildings often are located near traffic infrastructure. This situation leads to a rising noise and vibration exposure in urban environment. Typical vibration sources in urban environments are road and railway traffic, industrial sources or construction works. The prediction of structure-borne sound and secondary airborne sound in buildings can be difficult and uncertain due to the numerous characteristics of the source, transmission path and receiving structure. The two and three-dimensional finite element method (FEM) is a prediction procedure which is able to simulate high complex transfer paths and/or receiving structures. Models and methods

based on the FEM will be introduced to predict structure-borne and secondary airborne sound in complex buildings.

Thu 16:20 Van Cappellen Zaal Structural-acoustic optimization

Vibration analysis by means of structure-borne sound intensity (SSI) simulations

Thorsten Hering and Kai Wolf
TU Darmstadt, System Reliability and Machine Acoustics SzM

Measuring the airborne sound intensity (SI) is a well known and widely used method for the analysis of sound sources. By measuring sound intensities one can identify and locate sound sources and then compute the radiated acoustic power. In the case of real machine structures, the SI indicates the location of sound radiation but does not provide any information about the paths of the energy flow within the structure itself. Therefore the structure-borne sound intensity (SSI) is used to analyze energy flows and to evaluate the effects of structural modifications on the vibro-acoustic behavior.

Studies are carried out concerning the computation and measurement of the SSI to visualize the energy flow throughout test structures and real machine components that can be defined as plates and shells. The SSI is computed on FEM simulation data. Once the paths of energy flow are known it may be of interest to redirect energy flow into less critical areas instead of a mere global reduction of structure-borne sound. Based upon these SSI analyses, the effects of structural modifications, such as ribs and dampers, on the energy flow in plates or shells are demonstrated.

Thu 17:00 Van Cappellen Zaal Structural-acoustic optimization

Application of Transmissibility Matrix method to structure borne path contribution analysis

Dmitri Tcherniak
Brüel & Kjaer SVM A/S

Analysis of sound propagation paths from different noise sources is an important part of many automotive NVH evaluation processes. It is known that the classical transfer path analysis (TPA, NPA, SPC) methods addressing this problem are extremely time consuming and not considering practical by automotive engineers. Recently, a new technique based on a use of transmissibility matrix was suggested. Since it is relies on operational data only, the technique is very promising in terms of usability. However, lately there were some concerns expressed about the correctness of the results.

The current study continues investigation of the method accuracy and applicability. Comparing to the previous studies, where the method was used for air-borne scenarios, the current study applies it to the structure borne case. The method is applied to simulated data, which makes it possible to validate it against exact results and set the applicability regions of the method. A way to improve the method is suggested and validated.

Thu 17:20 Van Cappellen Zaal Structural-acoustic optimization

Structure-Borne Sound Excitation by Two-Phase Flow in Drainage Pipes

R. A. Alzugaray and B. A. T. Petersson
TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics

Cylindrical structures as drainage pipes are present in many domestic and industrial applications. These applications can be related to energy and fluid transfer. The coupling between the conveying fluid and the pipe walls generate dynamic excitations and potentially represent a structure-borne sound source, transforming part of the vibrational energy into radiated noise. The noise due to pipe works in apartment housing is of significant concern and can reduce the environmental comfort for occupants. This work deals with the analysis of velocity and force distribution on the surface of a vertical pipe in presence of a two-phase flow. This constitutes a first stage in flow and source characterization of drainage pipes. Some experimental results are presented.

Thu 17:40 Van Cappellen Zaal Structural-acoustic optimization

Modeling the Vibro-Acoustic Effect of Trim on Full Vehicle and Component Level Analysis.

Denis Blanchet
ESI GmbH, Munich

In the automotive industry, the influence of poro-elastic components on acoustic comfort has been mostly investigated for air borne noise at mid- and high frequency ranges; however, due to the lack of adequate theoretical formulations, the influence of poro-elastic in numerical vibro-acoustic simulation at lower frequency range has often been ignored or simplified by the use of distributed spring/mass on the BIW structure and impedance on the acoustic medium. Recent theoretical developments enabled to overcome this limitation by providing an efficient FEM formulation for poro-elastic material modelling. This FEM approach, implemented in VTM (Vehicle Trim Modeller) software developed by ESI-Group, enables the computation of the coupled response of a fully trimmed vehicle by taking into account the BIW structure, the acoustic cavity and the poro-elastic components (seats, carpet, dash insulator...).

This paper presents comparison between measurement and VTM simulation results for fully trimmed vehicle. It also presents the recently developed transmission loss (TL) module using FEM/BEM approach and comparisons between test and VTM prediction for academic and industrial cases.

Thu 8:40 Van Beuningen Zaal Environmental acoustics 1

Dose-Response-Relationship and the END

Kerstin Giering
FH Trier, Umwelt-Campus Birkenfeld

A Review of newer developments in dose-response relationships is given. It will be argued, that unique curves, applicable for all European countries are hardly to establish.

Thu 9:00 Van Beuningen Zaal Environmental acoustics 1

Techniques to Support Noise Rating and Action Planning in the Frame of the European Directive about Environmental Noise

Wolfgang Probst
DataKustik GmbH

The first round of Noise Mapping and Action Planning according to the European Directive about Environmental Noise should have been finalized now. Even with quite different approaches in some member states, there are some principles that are necessary in each case. Common is the calculation of noise levels on grids and around facades of residential buildings, even if different calculation methods have been applied. A further step was often the determination of areas where defined limiting levels are exceeded. In many cases a ”ranking method” has been applied to compare the possible improvement of a situation by planned mitigation measures. Possible methods how to apply such a ranking in a formal way with existing noise maps are demonstrated and practical examples are presented.

Thu 9:20 Van Beuningen Zaal Environmental acoustics 1

Cities cannot solve Noise Problems solitarily

Henk Wolfert
DCMR EPA Rijnmond

Noise Maps and Noise Action Plans according to the Environmental Noise Directive 2002/49/EC have been established by a lot of cities. Among those were members of the EUROCITIES network an European network of large cities. The Noise Maps have visualized the noise situation of a large part of Europe. Nowadays more than 70% of the European citizens are living in cities and it is expected to gain up to 80% in the next decades. Cities need to take action to reduce the noise in their territories and therefore a lot of measures are available. Traffic Management is a key solution but its effect is limited. Installing environmental zones has limited effect. Noise barriers give only a local effect on the spot. The most effective measure in urbanized areas will be the application of quiet road surfaces (quiet kinds of asphalts). After applying the quiet road surfaces all over the city where possible, a high number of annoyed people still remains. That’s why measures at European level are needed because if cars and lorries are provided with quiet tyres and manufacturers of vehicles are forced by European legislation to lower the noise, a reduction of annoyed people up to 50% can be achieved!

Thu 9:40 Van Beuningen Zaal Environmental acoustics 1

QCITY: providing cities validated mitigation measures for noise action plans

Geert Desanghere
Akron nv

QCITY (Quiet City Transport) is an integrated research project sponsored by the EC-sixth research framework - priority: sustainable surface transport. A major objective of the project is to provide municipalities with a guide with noise mitigation tools to establish action plans for compliance with Directive 2002/49/EC. This paper describes the development of a validated selection of mitigation measurers, fully documented with information such as the expected noise reduction, cost estimations, limitation in applicability. In addition to the selection list, datasheets with further documentation and references are provided. This guide focuses on practical and validated actions that can be undertaken by the municipalities themselves in their search for quiet city transportation. It involves mitigation measures such as traffic planning, noise reduction of road and rail surfaces, propagation paths and barriers, economic incentives, ...

Thu 10:00 Van Beuningen Zaal Environmental acoustics 1

Harmonized Strategies for the Development of Noise Action Plans

Sergio Luzzi and Raffaella Bellomini
Vie En.Ro.Se. Ingegeneria - Florence

In this paper an approach for developing Noise Action Plans, with special regard to hotspots and quiet areas detection and delimitation, is presented. Authors have been involved in the preparation of Noise Action Plan for the agglomeration of Florence and in the management of an Urban Matrix of targeted knowledge exchange for the Eurocities Network. A specific procedure for hotspot detection in areas annoyed by concurrent sources has been developed and criteria for selection of quiet areas in noisy city centres have been found.starting for their experiences the approach considers area characterisations and filters helping to achieve priorisation, such as noise level filter, minimum area filter, etc. The main actions of the Florence case study plan are then shown, together with a basic method for their harmonization with those of other noise reduction and noise control plans, required by National laws and to be applied to the same areas.

Thu 10:40 Van Beuningen Zaal Environmental acoustics 1

New Dutch legislation for motorways, where does it lead to?

Annemarie van Beek
Neth. Environmental Assesment Agency

The Dutch state is in a process to modernize the national legislation for noise from major roads and railroads. The objectives are to stop increase of noise and to reduce the amount of high exposed houses and other noise sensitive buildings. An important aspect is that the costs for noise measures to achieve those goals should be restrained. The legislation consists of both environmental as well as economical aspects. The new structure that is choosen consists of ”noise emission ceilings” to stimulate the most cost effective measures at noise sources. The Netherlands Environmental Assessment Agency has evaluated the plans for a new form of Dutch legislation by modelling the results of legislation in full perspective. The results of the study show the framework of costs, amounts of high exposed houses and expected noise annoyance due to highways within different scenarios of traffic growth. With these figures the choices in legislation are made visible and comparable with other options.

Thu 11:00 Van Beuningen Zaal Environmental acoustics 1

An estimate of the global exposure to environmental noise

Paul de Vos
DHV, Amersfoort

The data produced in the frame of the European Noise Directive, incomplete as it is, provides a useful basis to estimate (1) the relative importance of various sources of environmental noise, (2) the distribution of noise exposure in different classes relevant to health effects, and (3) the influence of demographic parameters to the noise exposure. Based on these relationships, an extrapolation for the world population’s exposure to noise has been made. The result supports the conclusion that environmental noise, in spite of ambitious legislative efforts, is a major and growing concern with respect to both public health and quality of life. This notice ought to be sufficient justification for a much higher political urgency, priority and interest than is expressed today.

Thu 11:20 Van Beuningen Zaal Environmental acoustics 1

Urban noise mapping in Romania - preliminary results for Iasi city

Daniel Condorachi

Univ 'AL.I.CUZA', Faculty of Geography and Geology, Iasi

Noise mapping is known to be a complex task. As a newcomer in EU, Romania has to comply the EU Directive regarding noise control especially in the urban areas. Iasi is the second largest city in Romania (340,000 inhabitants in 2002) and is in the first group of large cities that has to create noise strategic maps. Hybrid methods to collect data were used, required digital thematic layers were extracted in a GIS environment (TNTMips v.6.9 from Microimages Inc.) from cadastral maps and updated from orthorectified air imagery, numerous points for monitoring traffic at fixed hours were settled, classification of the roads regarding the traffic, database creation, points validation using a Quest sonometer, etc. Final integration in a software for noise mapping modeling (SoundPlan v.6.4) enabled us to compute the first city noise map for Iasi area using recommended EU noise calculation methods and standards.

Thu 11:40 Van Beuningen Zaal Environmental acoustics 1

A Sound Tool for Noise Control in urban planning.

Evert de Ruiter

Peutz bv, Zoetermeer

In the planning stage often studies (like EIA) are conducted to compare alternative locations for a new urban area or the alignment of a new road. Comparison of the noise load of the alternatives is part of the studies. In simple cases e.g. where a generic noise reduction is achieved the comparison is simple. In most cases however comparing the alternatives will be more complicated. How to weigh a small increase of the number of dwellings exposed to high noise loads, to a large decrease of the number of less noise loaded dwellings? This problem demands a metric for the total amount of noise annoyance to be expected in a certain context (planning of a new residential section or road). For this purpose a metric is proposed that is easy to calculate, called Population Annoyance Index (PAI). The PAI is based on dose-response relationships and the density of dwellings/people in the noise loaded areas; the method then results in a one-number metric for the "total annoyance" of the specific variant of the project. If minimising the amount of annoyance is the target of noise control, the PAI is a useful indicator in comparing the quality of the alternatives of the project.

Thu 15:00 Van Beuningen Zaal Environmental acoustics 2

A proposal for an European aircraft noise calculation procedure

Berthold Vogelsang[a] and Thomas Myck[b]

[a]*Niedersächsisches Ministerium für Umwelt und Klimaschutz;* [b]*Umweltbundesamt, Dessau*

In 2002 an Environmental Noise Directive was published which requires the production of noise mapping and action plans for major noise sources. Due to the fact that common methods are not available at present the Directive contains interim computation methods. For aircraft noise the European Commission recommends an interim computation method which is primarily based on ECAC Document 29 (2nd edition). Additionally, a proposal for a harmonised methodology for calculating environmental noise was developed in the project IMAGINE.

Recently a new Act for Protection against Aircraft Noise came into force in Germany. The act requires the establishment of noise protection zones at numerous airports. A modern methodology for calculating these zones is used which is called AzB. The calculation comprises aircraft noise emission data, the number of aircraft movements as well as the course of departure and approach tracks and airport circling. In addition taxiing aircraft are also taken into consideration. The AzB is compared with the interim computation method as well as with the results of IMAGINE. The comparison shows that the AzB has substantial advantages against the above-mentioned calculation procedures. Therefore, it is a candidate for a harmonized European computation method for aircraft noise in the future.

Thu 15:20 Van Beuningen Zaal Environmental acoustics 2

Calculation of Aircraft Noise Contours in Germany

Thomas Myck[a] and Berthold Vogelsang[b]

[a]*Umweltbundesamt, Dessau;* [b]*Niedersächsisches Ministerium für Umwelt und Klimaschutz*

In 2005 the European Directive on environmental noise has been transposed into national law in Germany. The act requires the development of noise mapping and action plans for major airports and other major noise sources. For this purpose a national interim computation method for environmental noise at airports is used. It is based on a closest point of approach model and provides information on the actual aircraft noise situation. Moreover, there exists a new methodology for calculating the noise protection zones according to the Act for Protection against Aircraft Noise of 2007. These zones will be established for numerous commercial airports and military airfields in the near future. The calculation algorithm differs from the computation method for environmental noise in essential points. In particular a segmentation procedure is applied which is based on an appropriate division of the three-dimensional flight path of an aircraft into linear segments. From each of these segments the aircraft contributes to the total sound exposure. Furthermore, there are regulations on land-use planning at regional level. For the determination of these zones a German standard is often used. This standard especially enables the calculation of aircraft noise exposure at small airports with mainly general aviation traffic.

Thu 15:40 Van Beuningen Zaal Environmental acoustics 2

Aircraft noise simulation for relevant daytime periods using a complete set of radar data

Beat Schäffer, Stefan Plüss and Georg Thomann
Empa, Laboratory of Acoustics / Noise Prevention (CH)

Aircraft noise contours are usually determined with model calculations. As the model results have important consequences for land use planning and payments of compensation, the model calculations need to be highly accurate. The hypothesis of this study was that modelling the daytime-specific changes in flight geometries determined from one year's radar dataset enhances the accuracy of the model results. To date, the model FLULA2 developed at Empa randomly chooses up to 100 flight paths per aircraft type and route in simulations to calculate 'footprints', i.e. daytime-unspecific averaged sound exposure levels scaled to one flight, which are then weighted with operational data to obtain daytime-specific noise contours. In this study, we extended the model to perform separate simulations for relevant daytime periods, using all available flight geometries of one year's flight operation to obtain daytime-specific footprints. Exemplarily for the year 2006, we determine the noise contours around Zurich Airport, Switzerland, with the common (∼27'000 simulated flights) and the extended simulation (∼235'000 flights). We compare the resulting daytime-specific noise contours with monitoring data to determine how much the extended simulation may improve the results and what possible consequences are, such as the changes in the number of persons above threshold values.

Thu 16:00 Van Beuningen Zaal Environmental acoustics 2

Health risks by nocturnal aircraft noise - do persons exposed get ill on average?

Christian Maschke
FBB-Maschke

Health risks by nocturnal aircraft noise must be minimised by adequate protection concepts in Germany. Such protection concepts can orientate themselves on epidemiological studies on selected health end-points or on "experimental" studies on noise induced sleep disturbances. New epidemiological studies already show an increased health risks at considerably lower continuous sound pressure levels as were able to be measured by sleep based examinations (at a usual flight volume). It has to be taken into account that in presenting sleep examinations, adverse sleep reactions were up until now measured as a group average (e.g. the averaged probability of waking up). Epidemiological parameters (RR, OR) on the other hand are based on the frequency of individual illnesses.

The risk of the exposed persons is described directly by the epidemiological parameters and therefore epidemiological findings are much more suitably for health protection. If health related protection concepts are based on the avoidance of sleep disturbances, however, it is necessary to deal with the individual sleep disturbances, because only individual people can become ill. For example the aircraft noise induced probability of waking up will be compared with epidemiological findings.

Thu 16:40 Van Beuningen Zaal Environmental acoustics 2

Continued experimental evaluation of a new approach to aircraft noise modelling

Derk-Jan Doeke Land, Gosse Oldenziel, Frits van der Eerden and Joris Sijs
TNO Science and Industry

A new approach to aircraft noise modelling was presented previously. This approach uses a directional emission model: each aircraft is represented by a hemisphere of emission spectra. These hemispheres are estimated from Noise Power Distance (NPD) tables. Next, the sound immission levels are obtained by applying a sound propagation model according to the Imagine project. For the previous experimental evaluation, with microphones in the vicinity of the runway, only the maximum power setting was taken into account. This paper presents the comparison of numerical and experimental results for 13 monitor positions up to 25 km distance for a total period of 14 days. Also, the effect of the thrust settings and the meteorology is demonstrated. Finally, the experimental data is used for a better estimation of the hemispheres for the sound emission spectra.

Thu 17:00 Van Beuningen Zaal Environmental acoustics 2

Helicopter detection - part 1 - Fast computational method for long range sound propagation

Marcel Janssens[a], Erik Salomons[b] and Frits van der Eerden[a]
[a]*TNO Science and industry;* [b]*TNO Built Environment and Geosciences*

A person can detect the presence of a nearby flying helicopter in various ways. One of these is aural detection. This paper, in combination with an accompanying paper, presents an approach to assess the probability of detection in varying circumstances of for example the terrain and the meteorology. Main parameters are: 1. the helicopter noise source strength and directivity, 2. the long range sound propagation as a function of terrain level and meteorology, 3. the presence of masking sound sources near the observer, 4. the probability of observer response to the exposed sound field. The focus of this paper is on long range sound propagation. A computational method is presented that allows a quick assessment (typically within a few seconds) of the long range sound propagation (typically up to 20 km) around a source or a receiver. The method allows for arbitrary terrain altitudes, so that shielding by (multiple) hills and mountains is taken into account. Moreover, the effects

of windspeed and temperature on sound ray curvature is included in a parametric way. Results of this fast computational method are compared with results from a numerical Parabolic Equation approach. Finally, results of the full approach (all four steps) are illustrated.

Thu 17:20 Van Beuningen Zaal Environmental acoustics 2

Helicopter detection - part 2 - Approach for the determination of the probability of detection.

Benoit Quesson[a], Marcel Janssens[b] and Frits van der Eerden[b]
[a]*TNO Defence, Security and Safety;* [b]*TNO Science and industry*

A flying helicopter can be detected, without any high tech material, by an observer hearing the characteristic sound emitted by the rotor of any helicopter. This paper, in combination with an accompanying paper, presents an approach to assess the probability of aural detection in various circumstances. The main parameters that influence this probability are: 1. the helicopter's noise source strength and directivity, 2. the long range (several km) sound propagation which is heavily influenced by e.g. terrain level and meteorological conditions, 3. the presence of masking sound sources near the observer, 4. the probability of the observer's response to the exposed sound field. This paper focuses on the two last points. Given a certain sound level (based on points 1. and 2.), it presents an approach to quantify the probability of aural detection. The calculation is based on studies found in literature on human aural response to helicopter sound and on an estimate of the masking sound level at the observer's location. The masking sound will vary from rural to urban areas and estimates for these are given. The paper presents the calculation of an auditory threshold but will address especially the estimate of the ambient wind induced noise levels.

Thu 17:40 Van Beuningen Zaal Environmental acoustics 2

Noise from elevated sound sources - Results of a dedicated measuring campaign

Mattias Trimpop
IfL Institut für Lärmschutz GmbH

For noise assessment purposes, outdoor sound propagation is mostly considered for situations having the source and the receiver close to the ground. Sound propagation from elevated sound sources is out of scope of many regulations like the ISO 9613 or "noise management directive" of the German MOD (Federal Ministry of Defence). In order to expand the given calculation schemes for these cases and to validate a 3D-ray tracing model developed by the Institute of noise control (IfL), reliable measuring data are needed to test the expansion and the 3D-Model.

This paper reports the results from a measurement campaign held in spring of 2008. Free explosions at different altitudes up to 300 m height were used to produce well known blasts. The blasts were recorded at 8 directions and at 6 distances up to 2 km. Weather data were acquired at 3 stations up to 400 m. The measurements were performed at three

different meteorological situations: at high, medium and low atmospheric turbulence conditions. The differences of the measured levels between high/medium/low turbulences, high and lower sources, up and downwind conditions are discussed in this paper.

Thu 8:40 Schadee Zaal Musical acoustics

An Experimental Musician-Based Study on Playability and Responsiveness of Violins

Friedrich von Türckheim, Thorsten Smit and Robert Mores
Hamburg University of Applied Sciences

While playing an instrument, musicians usually seek both, pleasant sound and perfect control. Focussing on the control part of string instruments, the mechanical response of the body translates into bow/string interaction and therefore into perceived responsiveness. How much do body admittances feed back through the bridge and thus influence playability? How significant is the player's conscious or unconscious adjustment? Investigations employ a specifically designed silent violin, where demountable bars under the bridge represent various admittance functions. In contrast to other research on violins, the method used here not only trusts in technical analyses but also in observations on musicians' subjective perceptions. One of the findings suggests that musicians expect an energy flux into the instrument, and zero effort playing aggravates bow and sound control. This work is part of a research project on desirable violin sound properties, where the investigated interaction between musicians and a parametric electronic violin must not be hindered by unfamiliar responsiveness.

Thu 9:00 Schadee Zaal Musical acoustics

Modal Testing of a Soprano Steelpan using a 3D Scanning Laser Doppler Vibrometer

Soren Maloney
University of Cambridge

Modal testing is a well established technique used to extract modal parameters such as natural frequencies, modal damping factors and vibration mode shapes from structures. Here, the technique is applied to a relatively new percussion instrument - the steelpan. The steelpan also referred to as the Caribbean steel drum or pan originated in Trinidad and Tobago during the Second World War. In this work modal testing is conducted on a soprano pan so as to obtain resonant frequencies and mode shapes as well as to provide a spatial illustration of modal tuning in the instrument. A soprano pan was excited with a burst chirp signal from a loudspeaker and its response recorded with a 3D Scanning Doppler Vibrometer. The results exhibit the harmonic relationship among tuned modes in each outer note. There is also significant modal coupling between adjacent notes that have harmonically related frequencies.

Thu 9:20 Schadee Zaal Musical acoustics

Reflection functions of pipes

Thomas Trommer, Hubert Ausserlechner, Judit Angster and Andras Miklos
Fraunhofer Institut für Bauphysik

The sound an organ pipe radiates is predominantly influenced by the resonator. In it sound waves travel back and forth between the lower end (consisting of upper lip, pipe mouth, languid) and the upper end being partially reflected as well as radiated from these ends. These reflections and herewith directly the pipe sound is dependent on different properties of which mainly contributing factors are the geometry of the resonator in size and shape (cylindrical, conical, diameter, length) as well as the attributes of the ends (open, closed, tuning geometries, mouth size). The first sound signal excites the stationary air column defined by the resonator. The reflected part of the sound signal from the upper end after being reflected from the lower end superimposes with the periodic sound emitted continuously by the sound source. In optimum this retroactivity leads to a steady state of the resonant circuit radiating the corresponding organ pipe sound. In order to model the resonant part of a pipe the reflection functions of both ends in dependency of the geometries are needed. These will be gathered by an especially developed method by means of time domain measurements on pipes with corresponding dimensions to in-situ employed organ pipes.

Thu 9:40 Schadee Zaal Musical acoustics

Measurements on an adjustable pipe foot model

Hubert Ausserlechner
Fraunhofer Institut für Bauphysik

The build-up of the sound in a flue organ pipe is mainly influenced by the edge tone. A small change of the flue parameters, such as positions of the lower and upper lip, the wind pressure in the pipe foot and the width of the flue, mostly results in big differences in the build-up and the spectra. In order to understand the basic physical processes a pipe foot model of a labial organ pipe with adjustable parameters has been developed. Furthermore, different types of languids and resonators (pipe bodies) can be attached to the model. A High value was set on comparisons and measurements with real metal organ pipes in order to give an indication for advisable ranges of the parameters. Flow velocity, edge tone, mouth tone and pipe sound measurements were carried out with the help of this model. From the results of the measurements theoretical models have been developed for flow velocity, velocity distribution, edge and mouth tone frequencies and for the ratios of the frequencies of edge tone modes. These models may facilitate the design and scaling of flue organ pipes.

Thu 10:00 Schadee Zaal Musical acoustics

Optimal scaling of the depth and width of wooden organ pipes

Judit Angster, Elena Esteve and Andras Miklos
Fraunhofer Institut für Bauphysik

In case of wooden pipes the reduction of pipe width would be desirable, because the space requirement of the pipe organ could then also be reduced. On the other hand, the sound quality could be worse for too narrow pipes. The aim is to find the narrowest scaling of wooden pipes with still appropriate sound quality. Optimal scaling of the depth and width of wooden pipes has been developed in order to help the design of the instrument to the space requirements of the room. A series of differently scaled wooden pipes has been acoustically investigated in the anechoic room of the IBP. On the basis of the results a design and scaling method has been developed that can take into account the effect of wall dimensions, material and finish on the stationary spectrum of the sound. With the help of the new method the organ builder can optimize the dimensions of the wooden pipes, but maintain an optimal sound quality in the same time.

Thu 10:20 Schadee Zaal Musical acoustics

String Vibrato in the Age of Recording: A Wavelet Study

Stijn Mattheij
Avans University Breda

String vibrato was observed from recordings of violin music, spanning the twentieth century. The use of spectral analysis, based on Morlet wavelets, resulted in a high resolution, simultaneously in time and frequency. Substantially more vibrato was observed in cantabile pieces than previous studies have indicated. This difference is attributed to partial masking effects present in the process of listening to old recordings of poor quality. The results confirm the common picture of a transition in vibrato during the early decades of the 1900s, but this change is primarily due to a change in width and to a lesser extent in continuity. The results provide supporting evidence for the so-called ’phonograph effect’ in connection with vibrato, as proposed by Katz.

Thu 11:00 Schadee Zaal Speech perception 3

Consonant Recognition of Listeners with Hearing Impairment and Comparison to Predictions Using an Auditory Model

Tim Jürgens, Thomas Brand and Birger Kollmeier
Universität Oldenburg, Medizinische Physik

Contrary to listeners with normal hearing, persons with high-frequency sensorineural hearing loss suffer especially from a poor consonant recognition. This is mainly attributed to the loss of audibility within high-frequency regions. But still it is not clear how suprathreshold factors like the loss of dynamic range and thus the modification of dynamic compression in the impaired auditory system contribute to the poor consonant recognition. In a former study we showed that consonant recognition for normal-hearing listeners can be predicted well using an auditory model combined to a speech recognizer. In this study we address the following question by modifying the auditory model to model the signal processing of listeners with hearing-impairment: Does a detailed modelling of hearing-impairment by including suprathreshold factors like a modified dynamic compression enhance the prediction of consonant recognition compared to a model using audibility only? Therefore, observed consonant recognition rates are compared to model predictions using different model approaches to involve hearing-impairment.

Thu 11:20 Schadee Zaal Speech perception 3

Prediction of Speech Intelligibility in Fluctuating Noise for Listeners with Normal and Impaired Hearing

Ralf Meyer[a], Thomas Brand[b] and Birger Kollmeier[b]
[a]*Universität Oldenburg;* [b]*Universität Oldenburg, Medizinische Physik*

Speech intelligibility prediction based exclusively on the pure-tone audiogram (e.g. SII-based models) shows some limitations. For listeners with similar pure-tone audiograms the same speech reception thresholds (SRT) are predicted even though the observed SRTs may differ. For SRTs measured in fluctuating noises, these differences are even larger than in stationary noises. One hypothesis is that cognitive parameters significantly influence the observed SRTs in fluctuating noise. In this study the speech reception thresholds for stationary and different fluctuating noises are measured for 12 listeners (4 with normal hearing and 8 with impaired hearing). For the listeners with impaired hearing three SRTs for different noises are measured using the master hearing aid (a computer program simulating an hearing aid), to evaluate the capabilities of the prediction models in predicting aided thresholds. Additionally, a pure-tone audiogram and a categorial loudness scaling are measured. For the measurement of cognitive parameters, the (visual) text reception threshold test and the lexical decision test are used. The results of the

cognitive tests are used to explain parts of the remaining deviations between the audiogram-based predictions and the measured values. Preliminary results show significant correlations between the deviations and some of the cognitive tests.

Thu 11:40 Schadee Zaal Speech perception 3

Optimising Patient Speech Acquisition at Magnetic Resonance Tomography Systems

Marco Friedrich
Universität Erlangen-Nürnberg

During operation of magnetic resonance (MR) tomography systems, switching noise is produced with high sound pressure levels. This noise is a disturbance for patient-to-operator communication, as well as for examinations that require synchronous recording of patient speech (fMRI). Signal processing algorithms can be applied for the removal of MR and other noise sources from patient speech. However, the disturbed speech signal should be obtained with the highest possible input SNR, which is complicated by limitations in practical application. On a typical MR installation, the room acoustics and sound field characteristics were examined, with the objective of optimising the microphone solution with regard to patient speech pick-up. Applying coherence based measures, the noise field is characterised on and around the MR system. General conclusions are drawn about possibilities to employ spatial information for optimising the speech level with regard to interfering noise sources, such as using microphones with directional characteristics, or multiple microphones and source separation algorithms. The choice of suitable microphones and their positioning on a MR installation are discussed. In addition, experiment results are presented for SNR improvement on a real MR installation, using microphones with different directional characteristics on suitable position.

Thu 15:00 Schadee Zaal Speech

Interactive Fitting-Wizard in a Home Environment

Hannah Baumgartner[a], Arne Schulz[a], Andreas Hein[a], Tobias Herzke[b] and Inga Holube[c]
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The European Hearing at Home (HaH) project aims at supporting hearing-impaired persons in their home environment. TV headsets with linear volume control allow people with hearing deficiencies good speech perception even in bad acoustic conditions. This approach attacks sensorineural hearing losses which imply a changed loudness perception combined with reduced perceivable dynamic range.
To help sensorineural users adapting the dynamic of their head phones according to individual hearing loss, three different interactive fitting procedures - based on broadcast audio material - have been evolved to fit parameters of a three band dynamic compressor performed in the master hearing aid (MHA). The MHA is the real-time audio-signal-processing framework in the background.
The fitting schemes can be used without any expertise and are intuitive and easy to use. A full walkthrough of the fitting wizard takes about 10 to 15 minutes. First results of user studies with 11 subjects (54-73 years old, 5 women, 6 men, slight to moderate hearing loss) show effectiveness of the approaches with respect to the improvement of the speech intelligibility and acceptance of sound quality. Further user studies will be performed.

Thu 15:20 Schadee Zaal Speech

Complex Wavelet Based Modulation Analysis of Speech

Jean-Marc Luneau[a], Jerome Lebrun[b] and Soeren Holdt Jensen[a]
[a]*Aalborg University - Dept. of Electronic Systems;* [b]*I3S-CNRS, Sophia Antipolis*

Low-frequency modulations of sound carry essential syllabic information for speech. The complex modulation spectrum is commonly obtained by spectral analysis of the sole temporal envelopes of the sub-bands out of a time/frequency analysis. Amplitudes and tones of speech or music tend to vary slowly over time. Thus the temporal envelopes are mostly of polynomial type. Filtering in this domain usually creates undesirable distortions because only the magnitudes of the temporal envelopes are processed and the phase data is often neglected. We remedy this problem with the use of a complex wavelet transform as a more appropriate envelope and phase processing tool. Complex wavelets carry both magnitude and phase explicitly with great sparsity and preserve well polynomials. Furthermore an analytic Hilbert-like transform is possible with complex wavelets implemented as an orthogonal filter bank. As a result, the transform delivers replicas of the time-frequency analysis at different modulation scales, namely ”Modulation Sub-Spaces”. By working in this

alternative domain the transform shows very promising denoising capabilities and suggests new approaches for joint spectrotemporal analytic processing of slow frequency and phase varying signals.

Thu 15:40 Schadee Zaal Speech

Speaker Verification Based on Formants Using Gaussian Mixture Models

Timo Becker[a], Michael Jessen[b] and Catalin Grigoras[c]
[a]*Austrian Academy of Sciences, Acoustics Research Institute;* [b]*KT 54 Bundeskriminalamt, Wiesbaden;* [c]*Ministry of Justice, Bucharest*

In a new method, the well known UBM-GMM framework for automatic speaker verification is applied to formant parameters replacing the common cepstral parameters (e.g. Mel frequency cepstral coefficients (MFCC)). By using full covariance matrices, both the static and variable vocal tract configurations are reflected in the speakers' models. Speaker comparisons are achieved by relating speaker similarity to speaker typicality, expressed as the likelihood ratio. The method reveals a low equal error rate while providing easy interpretability and low dimensionality.

Thu 16:00 Schadee Zaal Speech

Is the Fujisaki model a suitable (prosodic) model for the voice-conversion task?

Jan Schwarz, Martin Tran and Ulrich Heute
Christian-Albrechts-University of Kiel

The aim of voice conversion (VC) is to transform the voice of one speaker (source) in such a way that the converted voice sounds as if it was uttered by another speaker (target). The meaning and content of the speech are not changed. Nowadays VC-systems suffer from a poor naturalness and quality of the transformed voice. The transformed voice can only sound naturally, if it includes all characteristics relevant for the true target speaker. Within VC-systems, a main problem is the mapping of the prosody which is one of the essential features.
First results show that a statistical prosodic model can be used to describe prosodic parameters like the fundamental frequency and the timing of phonemes. However, the stress is not included within that model. From speech synthesis, the Fujisaki model is known to describe the fundamental frequency by parameters which also include accent commands. In addition the parameters are related to the physiology of speech production. Thus, this contribution analyses whether the Fujisaki model is suitable for the VC-task either as a single model or as an addition to the statistical prosodic model. Limitations, necessary extensions, and problems are pointed out.

Thu 16:20 Schadee Zaal Speech

Speech Analysis and Synthesis by Time-Varying Lattice Filters

Karl Schnell and Arild Lacroix
J.W.Goethe-Universität Frankfurt, Inst. für Angewandte Physik

Model-based analysis of speech is usually performed by time-invariant linear prediction. However, the articulatory speech production process is a continuous non-stationary process. In this contribution, time-varying estimations of autoregressive models are presented for speech analysis and synthesis. In comparison to the time-invariant estimation, the model parameters are time-varying within the frames. For that purpose, the coefficient trajectories are described by basis functions. The time-varying analysis is treated under the constraints of a continuous time evolution of the model parameters over frames. As model parameters the reflection coefficients of the lattice filter are considered, which are related to a simple vocal tract model of a lossless tube system. The estimation algorithm can be utilized for a variety of applications. Depending on the purpose of the estimation algorithm different adjustments and issues can be favourable such as the order of the model and the influence of the excitation.

Thu 17:00 Schadee Zaal Speech

Modeling the speaker-specific F0 Changes caused by raised Vocal Effort

Corinna Harwardt
FGAN, Wachtberg

The F0 behavior of a speaker is an important feature for forensic as well as for automatic speaker recognition tasks. Especially in forensic speaker recognition the two examined speech samples often contain speech of different degree of vocal effort. In such cases it is not recommendable to use the F0 measurements for speaker recognition because the F0 behavior changes with raising vocal effort. Hence an adequate estimation technique, that predicts a speaker's F0 changes for a given vocal effort, can improve speaker recognition tasks. The degree of raising the F0 caused by raised vocal effort is different for each speaker. To predict this speaker-specific degree of raising F0 is the goal of this investigation. Therefore we describe in this paper present estimation techniques for assigning F0 changes and introduce an estimation technique to predict the F0 changes affected by a raise of vocal effort. This prediction method and the problems in estimating speaker-specific F0 changes caused by raised vocal effort are presented in detail. Then the estimation technique is evaluated by using the Pool 2010 corpus. After discussion of the evaluation results we give an outline of future research.

Thu 17:20 Schadee Zaal Speech

Reverberation Time Estimation for Speech Processing Applications

Heinrich Loellmann and Peter Vary
RWTH Aachen University

The reverberation time (RT) is an important measure for the characterization of reverberant environments, which can be determined by different acoustical measurement methods. However, they use mostly a special measurement setup and dedicated excitation signals, which is inapplicable for most speech processing algorithms requiring knowledge about the RT. In this contribution, a new method is devised, which allows to estimate the RT from noisy observations. It is based on a maximum likelihood (ML) estimation which is derived from a statistical model of the sound decay in reverberant and noisy enclosures. This allows to determine the RT from a measured sound decay (or room impulse response) in the presence of noise. It is also shown, how the ML estimator can be used to determine the RT blindly from a noisy and reverberant speech signal. This blind RT estimation can be employed for the enhancement of noisy and reverberant speech signals.

Thu 17:40 Schadee Zaal Speech

A Study of Throat Microphone Performance in Automatic Speech Recognition on Motorcycles

Thomas Winkler, Serguei Pronkine, Rolf Bardeli and Joachim Köhler
Fraunhofer IAIS

Automatic speech recognition (ASR) for command and control and other applications in a car environment is very challenging due to the specific noise and environmental conditions. In the motorcycle domain, in some aspects like environmental noise the conditions are even more severe. In the EU-funded project MoveOn, a reliable command and control system for police motorcyclists is developed. The problem of decreasing ASR performance will be met considering the use of a more noise robust throat microphone instead of a standard close-talk microphone within the helmet. In this work an evaluation of the ASR performance for command and control on the motorcycle using a throat microphone for capturing speech is presented. In a first step the throat microphone signal and the close-talk microphone signal are analyzed. Then the results of the ASR performed on the throat microphone signal are compared to the results using the standard close-talk microphone. The evaluation is based on speech data synchronously recorded for both microphone channels. This training and evaluation data was recorded within the scope of the project under realistic noise and environmental conditions on the motorcycle.

Thu 8:40 Ruys Zaal Railway noise 2

Calculated Basic Sound Level as a Derivation of Measured Sound Levels of Freight Trains

Dirk Windelberg
Leibniz-Universität Hannover, Institut of Algebra

Railway rolling noise depends largely on wheel and rail roughness. Therefore the MEASURED sound level may differ at different locations along the rail. But also the wheels have different roughness, especially wheels of freight trains. However, the 'Grundwert' (basic sound level) defined in the German directive 'Schall03' may be valid for intercity coaches and 'average good rail conditions', and does not take into account these variations. In this paper, sound levels of freight trains measured at different locations are used to reverse calculate the actual basic sound level under the specific conditions of each measurement. Memorable illustrations show the shortcoming of the 'Grundwert' as defined in the directive. These illustrations form the basis for an open discussion about a lawful and generally accepted COMPUTABLE definition of the basic sound level of freight trains that is derived from many measurements. Such a definition has to be a solution of a 'number crunching' problem. We look for one value - the basic sound level for freight trains- which describes the noise for all species of freight trains with their individual and independent wheel roughness, and for all species of rails with their individual roughness. The aim is to find such a definition that is adequate for use in future directives and guidelines.

Thu 9:00 Ruys Zaal Railway noise 2

Railway noise - measurement and calculation

Dorothea Salz
Umweltbundesamt, Dessau

Die Akzeptanz und die weitere Entwicklung des Schienenverkehrs werden zunehmend durch seine ökologische Schwachstelle, den von ihm erzeugten Lärm, in Frage gestellt. Speziell der nächtliche Schienengüterverkehrslärm verursacht hohe Gesundheitsgefährdungen. Deshalb gibt es in einigen europäischen Regionen massiven Widerstand gegen mehr Verkehr auf der Schiene. Die vereinigten Bürgerinitiativen "Wucht an Main und Rhein" mit 240 000 Mitgliedern fordern beispielsweise Reduktionen der Geschwindigkeiten, Nachtfahrverbote und teilweise Führung der Strecken im Tunnel. Die Proteste zeigen, dass die Akzeptanz und die Wettbewerbsfähigkeit des klimafreundlichen Schienengüterverkehrs gefährdet sind.

Neben wichtigen Lärmminderungsmaßnahmen wie der Umrüstung von Güterwagen auf leisere Bremssysteme spielen auch die Beurteilungsmethoden von Schienenverkehrslärm eine große Rolle für eine effektive Lärmschutzstrategie. Vor- und Nachteile verschiedener Methoden werden gegenübergestellt und diskutiert.

Um beispielsweise die Synergieeffekte von fahrzeug- und fahrwegbezogenen Maßnahmen darstellen und damit für die Lärmminderung nutzen zu können, müssen die akustischen Fahrwegcharakteristika sowohl in die Messung als auch in die Berechnung hinreichend einbezogen werden.

Thu 9:20 Ruys Zaal Railway noise 2

Schall 03 1990 versus Schall 03 2006: A Comparison of the Calculation Methods for railway noise with Noise level Measurements

Ulrich Möhler[a] and Hans Onnich[b]
[a]*Möhler + Partner;* [b]*Deutsche Bahn AG - DB Systemtechnik*

A new method for the calculation of railway and tramway traffic noise was established by an expert team in the last few years. The results of this scope are described in the draft from December 2006. The estimated release date of the Schall 03 2006 will be the implementation by the german minister of transport during the year 2009 or 2010. Right now the Schall 1990 is the applicable regulation in the planning process for railway and tramway traffic noise. The implied differences between these two calculation procedures will be presented and compared by the results of noise level measurements. With respect on the transition period between the end of the appliance of Schall 03 1990 and use of the final paper)of the Schall 03 2006, some recommendations will be proposed for the practical use.

Thu 9:40 Ruys Zaal Railway noise 2

Calculation method for wheel dampers in railway applications

Randolf Arndt[a], Karoly Jalics[a], Gerald Schleinzer[b] and Günther Veit[c]
[a]*Virtual Vehicle Competence Center, Graz;* [b]*Siemens Transportation Systems, Graz;* [c]*Schrey & Veit GmbH*

Railway generated noise is a significant environmental problem in congested urban areas. Noise protection walls reduce the noise of the railway quite effectively, but impact the landscape and are costly in production and sustainment. Therefore, dampers on carriage wheels and rails have become a prosperous alternative for noise reduction and are already used by the railway industry. Presently, the optimization of these dampers is varied out by testing, mainly. Therefore, a calculation procedure for wheel dampers has been developed in a research project together with Siemens TS (production of trains) and Schrey&Veit (production of railway dampers).
This procedure is based on the Finite Element Method and requires detailed models of the damper parts and its connection to the wheel. The calculation results were compared and validated with modal analysis measurements. The article shows characteristic results and describes, how the new simulation method can support the design and optimization of rail wheel dampers in future applications.

Thu 10:00 Ruys Zaal Railway noise 2

Procedure and applications of combined wheel/rail roughness measurement

Michael Dittrich
TNO Science and Industry

Wheel-rail roughness is known to be the main excitation source of railway rolling noise. Besides the already standardised method for direct roughness measurement, it is also possible to measure combined wheel-rail roughness from vertical railhead vibration. This is a different quantity which offers its own particular applications. This measurement technique has significant practical benefits and has been in use for several years. In this paper the method is outlined and it is discussed how it could be implemented in a procedure. Applications are discussed and some illustrative data are presented.

Thu 10:40 Ruys Zaal Railway noise 2

Simulation of curve squeal considering the vehicle dynamics

Daniel Frese and Torsten Kohrs
Bombardier Transportation

When a railway vehicle traverses a tight curve an intense tonal noise can occur. This squealing noise is often the dominant sound in tight curves and emerges mainly in urban areas. The squeal mechanism is attributed to a stick-slip process which has been the topic of several investigations in the past. The occurrence of this instability depends on a large amount of parameters, e.g. wheel damping, friction characteristics and vehicle dynamics. Parametric studies using two different simulation tools (TWINS-SLYNX and SFE AKUSRAIL) are performed to identify the main influence parameters for the curve squeal phenomenon. The focus is on the investigation of vehicle dynamics parameters gained from multibody vehicle dynamic simulations on curve squeal. The aim is to identify vehicle dynamic potentials to inhibit the occurrence of squeal noise.

Thu 11:00 Ruys Zaal Railway noise 2

Statistical Analysis of Railway Noise: How long-term monitoring helps improving short-term measurements

Edwin Verheijen
dBvision & RIVM, Utrecht

Rolling noise of trains is a well-understood phenomenon for which very detailed source models have been developed (e.g. TWINS, Harmonoise). Although noise computations are preferred in many cases above noise measurements, long-term noise monitoring is becoming more important as a means to verify calculations. The noise monitoring stations developed by ProRail have yielded a large amount of statistical information in a few years time. How these measurements compare to calculations has been reported elsewhere. In this paper we will focus on

events and effects that influence the representativity of short-term measurements. This is important because many research results are based on just a couple of pass-by measurements. How reliable are they? As the monitoring stations are able to identify individual trains and even single vehicles, very refined variation analysis is possible. For example, the effects of wind, temperature, seasons, and state of maintenance have been sorted out. Besides this, the paper gives also answers to questions like: What mileage is required before the noise emission of new trains is stable and representative? What is the noise effect of wheel flats? Some of the results give reason to adjust the measurement conditions of the noise type testing standard (ISO 3095).

Thu 11:20 Ruys Zaal Railway noise 2

The project 'Low Noise Train on a Real Track (LZarG)' will help to halve the rail noise by 2020

Wolfgang Behr
Deutsche Bahn AG - DB Systemtechnik

Deutsche Bahn has the goal of halving the rail noise exposition of lineside residents by the year 2020 compared to 2000. Therefore the project "LZarG" investigates in technical improvements of boogies, wheelsets, wheel-absorbers, rail-absorbers, rail fastening systems and special under sleeper pads. Besides the German noise mitigation programme for existing lines and retrofitting the freight wagons with composite brake blocks, LZarG is the third column in the strategy of Deutsche Bahn to reduce railway noise by at least 10 dB(A) by 2020 compared to 2000. Deutsche Bahn will assure that each development within LZarG meets the relevant regulations and the special requirements of Deutsche Bahn. Also the potential impact on future maintenance procedures will be taken into account. At the end of the project in 2010, several field test measurements will evaluate the potential of the improvements developed or improved within LZarG, using a noise-optimized trainset on the various track systems.

Thu 11:40 Ruys Zaal Railway noise 2

TSI-Noise Standstill Measurements on Open Track.

Malte Tinter[a], Nicolas Meunier[b] and Christoph Eichenlaub[a]
[a]*Alstom LHB;* [b]*Deutsche Bahn AG - DB Systemtechnik*

Since the entry into force of the Technical Specification for Interoperability of Conventional Rolling Stock - Subsystem Noise (short TSI Noise), the homologation process of new vehicles includes a test of their noise emission. TSI Noise specifies in particular limits for stationary noise, starting noise, pass by noise and driver's cab noise. The conformity of rolling stock has to be assessed within type testing.
Modern vehicles developed under consideration of the TSI requirements may present a quite low noise emission, which involves that the measuring site has to be chosen carefully, since the exterior noise measurements are performed at a distance of 7.5 m from the track axis and the

background noise has to be at least 10 dB lower than the vehicle noise. The lecture demonstrates - using the example of the TSI certification of the electric trainset Coradia Continental - the advantages of carrying out standstill measurements at the test site for acceleration and pass by tests, resulting in benefits for precision, time and costs of the measurements.

Thu 15:00 Ruys Zaal Localisation of sound sources on vehicles

Capturing a noise source in an interior enclosure

Filip Deblauwe[a] and Maxime Robin[b]

[a]*LMS International;* [b]*MICROdB, Edully (Gambia)*

Sound source localization (SSL), such as NAH and beam-forming, have been around for the last decennia. SSL techniques have been mainly used for free field conditions and only the last couple of year they have found their way for interior acoustic applications where non-free-field conditions are met. In this paper a two step test procedure is proposed to perform a detailed interior SSL localization. In a first step, a solid spherical antenna is used to perform a SSL on the complete interior compartment. The uniqueness of this solution lays in the fact that the SSL propagation is not only based upon a beam- forming solution, but takes also into account the acoustic diffractions that happens around the solid sphere. The first step gives an overview of all sources present in the interior enclosure, but with a limited spatial resolution. In a second step this spatial resolution is improved by taking a second measurement focusing in at certain areas of the enclosure. For these measurements a cylindrical array is used and the data is processed using focalization. The combination of the two methods leads to an accurate SSL in an interior enclosure in a more efficient way that has been done till now using masking techniques.

Thu 15:20 Ruys Zaal Localisation of sources on vehicles

Sound Source Localisation with Acoustic Mirrors

Martin Helfer

FKFS Stuttgart

Acoustic mirrors, also known as sound mirrors, has been used for sound source detection and localisation for many years even though they were not very common. Probably their first application took place on the military field as early warning systems for adversary aircraft. In recent years many of them were used for sound source determination in rail and road transport. Especially in aeroacoustic wind tunnels sound mirrors have become one of the preferred measurement techniques for the investigation of extrior sound radiation of vehicles and planes. One of their favourable properties is the simple setup which allows easy identification and estimation of the main sources without requiring extensive data processing. On the other hand a complete documentation of the radiated sound pattern may be quite time-consuming.

Thu 15:40 Ruys Zaal Localisation of sound sources on vehicles

Advanced Source Localisation with Beamforming in Vehicle Acoustics

Sandro Guidati
HEAD acoustics GmbH

Beamforming using microphone arrays is a common used technique for the localization of sound sources. The first applications were mainly aeroacoustic measurements in wind tunnels. Here the measurement object can not be measured in the very near field with e.g. a single microphone without producing excessive noise, whereas the array can be placed outside the airflow. A problem for beamforming is a reverberant environment. The algorithm is based on the evaluation of phase relations between the array microphones. In a highly reverberant environment e.g. a vehicle cabin, the phase relations are distorted by the reflections. With the use of measured transfer functions between the array microphones and the points of interest in the cabin the accuracy of the results can be enhanced. Recent advantages in computer technologies allow for a real time processing of the microphone array data resulting in an online visualization of the sound sources. Computer vision technologies based on multiple video cameras detect the three dimensional distribution of the potential sound sources increasing again the accuracy of the evaluation. The paper describes these techniques in detail and shows applications in vehicle acoustics.

Thu 16:00 Ruys Zaal Localisation of sound sources on vehicles

Validation of a measurement based Area Contribution Analysis

Heiko Jakob[a], Arnaud Bocquillet[a] and Steffen Marburg[b]
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The Automotive Industry is in need of effective NVH measurement methods to optimize the design of isolation and damping concerning weight and cost constrains. In order to gain better knowledge on the contribution of each panel from the interior it is imperative to find a new and simple method of analysis in the early prototype stages of a vehicle. Using the technology of the Microflown-element it became possible to grasp the particle velocity close to any surface based on measurements. This information and the quantification of the transfer path allow for a prediction on contribution from subsystems in the cabin. The method is based on a combination of the Transfer Path Analysis principle and the quantification of the Emission of the surface. A simple example below shows the validation of such measurements as well as the comparison between a direct measurement and a simulation. This example depicts the critical range of the mid frequencies (200-3500Hz). The precision and usability of these simulated results will be discussed moving forward based on this specifically designed setup.

Thu 16:40 Ruys Zaal Noise

Breaking the Sound Barrier: Why Environmental Noise Still is Not Reduced Properly, and What To Do About It

Edwin Verheijen[a] and Jan Jabben[b]
[a]*dBvision & RIVM, Utrecht;* [b]*National Institute of Public Health and Environment (RIVM)*

Various international cost-benefit studies show that traffic noise can be reduced cost-effectively by noise measures, especially by measures taken at the source (i.e. silent tyres or silent road pavement). The benefits are obvious: a quieter environment leads to less annoyance and sleep disturbance, resulting in reduced health damage and production loss. Previous research pointed out that 7 billion euros can be spent cost-effectively on noise measures in the Netherlands. This does not only justify the present national noise policy (which aims at avoiding noise growth), but also suggests that much more ambitious goals are within reach: an overall noise reduction of 5 dB. How to convince decision makers and politicians that such a noise reduction is both urgent and cost-effective? Why do they still hesitate while environmental noise is annoying so many people and threatens their well-being and health? The present article analyzes the doubts and it provides actions that should be considered to take the noise issue higher on the political agenda.

Thu 17:00 Ruys Zaal Noise

Flow-induced Sound Radiation from Air-ducting Structures

Silja Beck[a] and Sabine Langer[b]
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In product development prediction of flow-induced noise becomes an increasing important issue. Examples include both liquid and gas flow such as heating and sanitary ducts, ventilation channels and air intakes. Here, a numerical approach is presented to determine this acoustic radiation. Once validated, it can reduce expensive prototyping and laboratory testing.

The model consists of the three interacting parts air flow, structure and surrounding space. Air flow within the structure is derived using the volume of fluid approach used by commercial software. The structure is represented by a shell model discretized by the finite element method (FEM). Acoustic radiation into the surrounding space is modelled accounting for the wave formulation and Sommerfeld's radiation condition employing the boundary element method (BEM).

Thu 17:20 Ruys Zaal Noise

An insitu non destructive measurement technique to assess the acoustic absorption of asphalts

Emiel Tijs
Microflown Technologies

Traditionally, the acoustic absorption properties of road surface asphalts are determined in a Kundt's tube, requiring a sample to be taken, thus destructing the surface. Furthermore, this method is time consuming and only limited to measure sound wave under a normal angle of sound wave incidence, which is not the case in practive. The PU surface impedance method, measuring both sound pressure and acoustic particle velocity,has been applied on asphalt road surfaces, measuring acoustic absorption under both normal and oblique angles of sound wave incidence. The method can also be applied while moving the probe along the test surface. Thus it becomes feasible to measure the acoustic properties of road surfaces whilst driving. As many mechanical properties are related to acoustic absorption, a road surface monitoring system based upon acoustic absorption measurements might be feasible. First results will be presented and an outlook will be discussed.

Thu 17:40 Ruys Zaal Noise

Noise Stage and Efficacy that the Workers who Work in Industrial Field are Exposed to During the Manufacture

Sezgin Ersoy
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Industrial noise might be described as a noise, because of using equipment and machinery in industry and it is getting to be a growing matter related with industrial development. The noise is a one of the main cause which threat staff health, especially such as manufactures and depends on technology of production capacity and kind of goods that manufactured might be different stages.

Measurements were made firm which produce a transformer. There were working 150 people, at the measurement time. This area which made a measurement, only produce a power transformer. Measurement areas are; coil production, nucleus production, coil montage and test laboratory. Measurements were made two minutes periods.

Measurements were carried out with a SL 4001 type 2 for suitable IEC 651 which has a ISO 9001, CE and IEC1010 certificated. This device can measure by ISO 1996-1.2003 standard. The result show that the highest noise level was nucleus production area, and the lowest noise level was test laboratory.

Thu 8:40 Mees Zaal Voice production 1

Numerical Modelling of Vocal Fold Dynamics by a 3D Multi-Mass-Model

Anxiong Yang[a], Jörg Lohscheller[a], Michael Stingl[b], Daniel Voigt[a], Ulrich Eysholdt[a] and Michael Döllinger[a]
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The emitted human speech signal originates due to oscillations of both vocal folds. So called functional dysphonia being only observable during phonation yield voice disorders, i.e. hoarseness. In historical voice research, the endoscopic vocal folds movements were mostly simulated via 1D Two- or 2D Multi-Mass-Models. However, by using these models only vocal fold movements in lateral and longitudinal directions are taken into account.

This work presents an enhancement of these mathematic biomechanical models: a Multi-Mass-Model expanded from 2D to 3D displacements. It consists of five mass planes arranged in vertical direction. Each plane contains five longitudinal coupled mass-spring oscillators. Via the new proposed model not only the vertical movements but also the diffusion of the mucosal wave from inferior to superior can be simulated.

In order to better simulate vocal fold vibrations, the time-invariant model parameters are adjusted in a developed hybrid optimization procedure. The evaluation of optimization procedure is performed by using lateral symmetric data sets synthetically generated by the suggested model. Different glottis closure-types are also considered.

The future goal is to describe the physiological relevant characteristics of 3D vocal fold dynamics, which can be extracted via high-speed imaging in combination with a new developed laser-spot projection system.

Thu 9:00 Mees Zaal Voice production 1

A novel approach for stable calculation of vocal fold oscillation using a multiple-mass model

Eva Loch[a], Malte Kob[a,c] and Sebastian Noelle[b]
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The modeling of continuous string-like systems is often performed by spatial and temporal discretization of a problem-dependent wave equation. For arbitrarily chosen discretizations instabilities may appear which in contrary to the expected increase of accuracy decrease the stability of the calculations. In this paper an approach to avoid such instabilities by adopting the temporal discretization to the spacial discretization is presented. Implementations of a simplified vocal fold model have been made in MATLAB and JAVA. The result is tested with a multi mass model of vocal fold oscillation. It is shown that for an arbitrary number of masses a stable calculation is obtained. The application of the method for more complex models of normal and pathologic voice production is planned.

Thu 9:20 Mees Zaal Voice production 1

Numerical study of the human phonation process by the Finite Element Method

Stefan Zörner[a], Manfred Kaltenbacher[a], Gerhard Link[b], Reinhard Lerch[c] and Michael Döllinger[d]
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The basis of the human phonation process is given by the complex interaction of air and flow in the larynx together with the structural mechanics of the vocal folds. Our goal is to study the influence of the geometric shapes of vocal folds on the quality of the generated sound. Since direct measurements on such variations are not feasible, we developed an enhanced numerical scheme based on 2d Finite Element Methods, capable to compute this coupled field problem. The geometric variations on the vocal folds change the main frequency component and, furthermore, influence the occurrence of the Coanda effect (the jet is attaching randomly to either side of the trachea). With our simulations, we can demonstrate, that this effect causes a broadband acoustic signal. Currently, we perform numerical simulations with asymmetric vocal folds to investigate different types of dysphonia.

Thu 9:40 Mees Zaal Voice production 1

An approach for numerical calculation of glottal flow during glottal closure

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Limitations of classical low-degree-of-freedom approaches (multiple-mass models) of modeling vocal fold (VF) oscillation have indicated the need for more advanced models. A couple of finite-element (FE) models have been presented in the last years, which include a number of numerically difficult solver algorithms. The presented model represents a coupled two-dimensional model of VF movement including fluid-structure interaction effects. A structural model of vocal fold tissues and a fluid model of the airflow are combined by the Arbitrary Lagrangian-Eulerian formulation. As solvers ANSYS was chosen for the structure and CFX for the fluid. A special problem is handling the stop of the airflow in the closed phase of the glottal cycle. Since the fluid elements in the glottal

gap channel would have to be zero-volume elements, a special modeling strategy is necessary. In this model a loss coefficient is introduced when the glottal gap falls below a certain control volume. As exemplary area of use, results of comparative simulations are shown, which differ in driving pressure values and VF shape. It is shown that typical convergent/divergent shapes are only obtained when a minimum length of a channel between the VF remains.

Thu 10:00 Mees Zaal Voice production 1

Sound Generation in a Flow-Induced Vibrating Human Vocal Folds Model

Stefan Becker, Stefan Kniesburges, Stefan Müller and Antonio Delgado
Universität Erlangen-Nürnberg, Lehrstuhl für Strömungsmechanik

For investigating the fluid-structure-acoustic coupled (FSA) process of human phonation, synthetic models of the vocal folds (VF) were designed which showed flow-induced oscillations and produced a tonal sound. Two different kinds of models were developed: one for observing the flow field vibrating at constant frequency and a more realistic which could be stretched in longitudinal direction to simulate the prestress within human VFs. Besides the investigation of the flow field using PIV the major aim was to observe the FSA process to determine the acoustic sources. Therefore two approaches were applied: Firstly the vibration of the VFs was recorded using a high-speed camera triggered by the subglottal pressure to determine phase shift between the two signals. Secondly synchronous measurements of the subglottal pressure, the flow velocity behind the VFs and the acoustic pressure were done to determine their coherence. The results support the existence of the Coanda-effect. Furthermore the vibration frequency and the phase shift between the subglottal pressure and the glottal width depended on the stiffness and the tension in the VFs. The synchronous measurements determined the pulsating flow rate as main acoustic source.

Thu 10:20 Mees Zaal Voice production 1

Human phonation analysis by 3d aero-acoustic computation

Stefan Zörner[a], Manfred Kaltenbacher[a], Reinhard Lerch[b], Willy Mattheus[c], Rüdiger Schwarze[c] and Christoph Brücker[c]
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A numerical model of the phonation process, which takes place inside the larynx, is presented. The air flow through the glottis has been replaced by water and the physical dimensions have been scaled accordingly. Therewith, the characteristic frequencies and velocities are reduced. The movement of the vocal folds are prescribed by geometric changes of the computational domain.

In utilising numerical simulations, the goal is to study the complex interaction of the modulated flow, due to the prescribed geometric variations.

The fluid flow is simulated by solving the incompressible Navier-Stokes equation with the help of a direct numerical simulation (DNS) based on the software OpenFOAM. The results are scaled to comply with a simulation in air. This data is then taken to calculate the acoustic sources inside the model by using Lighthill's analogy and the acoustic sound propagation by solving the inhomogeneous wave equation. Currently we prescribe a periodical flow on the inflow boundary and compute the flow field for fixed vocal fold geometries. The computed acoustic field inside the glottis shows the same dominant frequency as the prescribed fluid inflow.

Thu 11:00 Mees Zaal Voice production 1

A model of jet modulation in voiced fricatives

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The radiated sound during the production of voiced fricatives has harmonic and random components. The harmonic components are associated with the vibration of the larynx and the noise components are associated with the time-varying flow through a constricted region of the oral cavity. The two sources interact non-linearly to produce a noise component with an amplitude envelope modulated at the fundamental frequency of voicing. This paper describes pressure measurements made in a dynamic physical model of the larynx and vocal tract, which were used to study the aerodynamic and acoustic behaviour in a flow regime corresponding to that found during voiced fricative production. An orifice plate and a sharp-edged obstacle downstream model the geometry of the oral cavity, representing the tongue constriction and teeth respectively. The depth of modulation of the frication noise was found to be linearly related to the pressure upstream of the orifice plate. The phase of the noise bursts relative to the upstream pressure was found to vary systematically with the constriction-obstacle distance. These findings form the basis of a parametric model of voiced fricative production and give some insight into the likely aeroacoustic mechanism responsible for sound production.

Thu 11:20 Mees Zaal Voice production 1

PIV Measurement of Flow-Patterns in a Human Vocal Tract Model

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The contribution describes developed complex physical model of the voice production that consists of the trachea, the self-oscillating vocal folds and the vocal tract with acoustical spaces corresponding to the vowel /a:/. The vocal tract model was developed from the 3D FE models designed from MR images The vocal folds were made of a latex thin cover layer filled by a polyurethane rubber, joined to the subglottal spaces. The time-resolved PIV method was used for instantaneous velocity field evaluation. Preliminary results will be presented for measurements performed within a physiologically range of mean air flow rates and fundamental frequencies. The vibrating vocal folds during one oscillation period were recorded by the high-speed camera in the same instants as the airflow velocity patterns. Flow structures resembling large vortices with dimensions comparable with the channel cross-section is possible to see above the ventricular folds. The vortices disappear in the narrowest epilaryngeal part of the vocal tract where the flow is uniform. Large eddies of a size comparable with the channel height were observed in the model of the mouth cavity where the airflow was attached to one wall of the channel resembling a Coanda effect.

Thu 11:40 Mees Zaal Voice production 1

Three-Dimensional Unsteady Flow Nature in the Vocal Tract during Human Phonation

Michael Triep[a], Willy Mattheus[a], Michael Stingl[b] and Christoph Brücker[a]
[a]*University Freiberg, Mechanics and Fluid Mechanics;* [b]*Dept. Applied Mathematics II, University of Erlangen-Nuremberg*

The quality of the human voice during phonation is strongly related to the three-dimensional (3D) unsteady flow field in the vocal tract downstream of the vocal folds. A slight pathological change of the vocal fold's geometry and/or their movement may have already a strong effect on the flow field and thereby on the flow-induced noise (sources) which is a parameter that characterizes the voice quality. In this study the 3D unsteady flow field in a dynamical model of the vocal folds is simulated numerically and visualized with the same boundary conditions in the experiment. In addition to the 3:1 up-scaling of the model water is used as the working medium so that the time scale is best accessible in the experimental approach for the measurement of the flow field structures by time-resolved Particle-Image Velocimetry and of the time-dependent volume flow rate. A realistic pre-phonatory transglottal pressure difference and time-varying motion and profile of the glottis during the phonatory cycle are prescribed in our model, and the resulting characteristic volume waveform is used as input for the numerical simulation with the flow solver code OpenFOAM. In particular the comparison between the flow dynamics with and without model ventricular folds, which build a second constriction in the test channel, is analyzed.

Thu 15:00 Mees Zaal Voice production 2

Biomechanics and control of vocalisation in a non-songbird

Coen Elemans[a], Ricardo Zaccarelli[b] and Hanspeter Herzel[b]
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The neuromuscular control of vocalisation in birds requires complicated and precisely coordinated motor control of the vocal organ (i.e. the syrinx), the respiratory system and upper vocal tract. The biomechanics of the syrinx is not well understood. Here, we aim to unravel the contribution of different control parameters in the coo of the ring dove (Streptopelia risoria) at the syrinx level. We designed and implemented a quantitative biomechanical syrinx model that is driven by physiological control parameters and includes a muscle model. Our simple nonlinear model reproduces the coo, including the inspiratory note, with remarkable accuracy and suggests that harmonic content of song can be controlled by the geometry and rest position of the syrinx. Furthermore, by systematically switching off control parameters, we demonstrate how they affect amplitude and frequency modulation and we generate new experimentally testable hypotheses. Our model suggests that independent control of amplitude and frequency seems not possible with the simple syringeal morphology of the ring dove. We speculate that songbirds evolved a syrinx design that uncouples the control of different sound parameters and allows for independent control. This evolutionary key innovation provides an additional explanation for the rapid diversification and speciation of the songbirds.

Thu 15:20 Mees Zaal Voice production 2

Biomechanical modeling of chest-falsetto registers and their transition

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The exact definition of registers in the human voice is still under debate. Especially the quantitative analysis of transitions between the registers has not been investigated in much detail yet. Excised larynx experiments show different kinds of voice instabilities that appear close to the transition from chest to falsetto register. Similar phenomena are also observed in vocalization of untrained singers. Towards comprehensive understanding of such register transitions, this paper carries out a biomechanical modeling of the human voice. Our model has a body-cover structure, which is in total composed of fourth masses. A smooth geometry is realized by introducing a polygon shape to the vocal fold model. Acoustical coupling with the vocal tract as well as the trachea is also considered.

The simulation study shows that the model can reproduce many complex phenomena such as register jumps, hysteresis, subharmonics, and chaos, observed in excised larynx experiment as well as in vocalization of untrained singers. We study in detail the influence of the subglottal and supraglottal resonances on the transition point of the registers.

Thu 15:40 Mees Zaal Voice production 2

Investigation of voice production in death metal singers

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Singers in ”metal” style often use a unique form of voice production called ”growling”. The underlying mechanisms of voice production as well as possible potentially harmful influences on the singer’s voice have not yet been intensively studied. Supra glottal laryngeal constriction appears to play a major role in metal singing. This constriction results in difficulties to apply standard laryngeal endoscopy for visualization of vocal fold function during this particular phonation style. In this study we present an examination method using a questionnaire including the voice handicap index as well as a combined simultaneous endoscopy, electroglottography and acoustic analysis to investigate in particular the production of death metal singing in seven subjects. None of the subjects complained about persistent voice problems and none had signs of secondary organic voice disorders like vocal fold nodules or inflammation. However, most showed minor irregularities on stroboscopic examination. Further findings reveal that at least two forms of supra glottal laryngeal constriction are used to obtain the typical death metal sound which differ in use of vocal folds and ventricular folds as well as in supra-glottal adduction effort. The study shall help to answer the question whether death metal singing can be performed such that the risk of voice damage can be reduced or avoided.

Thu 16:00 Mees Zaal Voice production 2

A voice-producing prosthesis for laryngectomized patients

Bart Verkerke[a], Johan Tack[a], Henri Marres[b], Cees Meeuwis[c], Ward van der Houwen[a] and Gerhard Rakhorst[a]
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Introduction: Voice quality after laryngectomy is sometimes poor, due to a low pitch or even absent voice. A voice-producing prosthesis (VPP) is developed to create a better substitute voice. Materials/methods: The VPP is comprised of two elastic membranes inside a circular housing that can be inserted in the patient’s shunt valve. Prototypes were tested

in-vitro under physiological conditions and in 17 female laryngectomised patients. Results: Basic sound, containing multiple harmonics, was successfully produced under physiologic air pressure and airflow conditions. Fundamental frequency and sound pressure level were controlled by changing the driving pressure. The obtained frequency range is appropriate for producing a female voice. The noise-to-harmonics ratio was low (mean 0.15). The clinical study clearly showed that the pitch and sound intensity were increased with normal lung pressure. The flow rates were lower than normal, leading to significantly longer phonation times. Conclusion: The prototypes functioned under physiological conditions. Variation of the driving air pressure made intonation possible. An audible voice with sufficient intelligibility was created. Especially female laryngectomized patients with a weak, whispering TE shunt voice can benefit from the VPP voice. References: Tack JW, et al. Head Neck 29(7): 665-674, 2007; Head Neck 30(9): 1156-1166, 2008

Thu 16:20 Mees Zaal Voice production 2

Simulation of differences between male and female vocal fold configuration during phonation

Malte Kob[a,b], Philippe Dejonckere[c], Edwin Calderón[a] and Senay Kaynar[a]
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Statistically, some vocal fold lesions such as vocal fold nodules - considered as a localized tissue reaction to voice loading - are much more frequent in female (∼ 95%) than in male (∼5%) adult humans. This can only partially be explained by the about two times higher oscillation frequency during voice production in females. In previous studies, gender-related differences in the shape of the vocal fold edge have been observed that can be related to different geometries of the laryngeal framework. With a time-domain model of vocal fold function the oscillation patterns and resulting forces between and inside the vocal folds have been simulated, and the conditions for eliciting a hourglass-shaped vibration pattern (necessary for determining the localized tissue reaction of the nodules) could be clearly defined. Our simulations and measurements support the assumption that collision and tearing forces on the vocal fold edges during phonation are higher and more specifically located in females compared to males.

Thu 17:00 Mees Zaal Voice production 2

Optimizing Material Properties and Geometry of a Physical Multi-Layered Vocal Fold Model

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Understanding the physical fundamentals of human voice generation is a challenging and interdisciplinary task. The present work takes a closer look at a self-sustained physical model of the human larynx, in which the vocal folds are modeled by multi-layered elastic bodies, i.e. silicone layers. We search for optimal material and geometry parameters in the sense that the deformation of the multi-layer model at its surface is as similar to given deformation patterns arising from hemilarynx experiments as possible. For this purpose, we employ an optimization concept based on a 2D finite element simulation in which the vocal fold is modeled as linear elastic body. Keeping the layer geometry fixed, we use the material optimization approach in order to identify optimal material parameters for each layer by assuming isotropic or transversal-isotropic material properties. Additionally, we employ techniques from shape optimization, in order to optimize geometry of the boundaries between the different layers parameterized as spline curves, while keeping the material tensors fixed. We consider the simultaneous solution of both problems, discussing optimization of material and geometry concurrently. We conclude this presentation by giving some numerical results using a 2D vocal fold model.

Thu 17:20 Mees Zaal Voice production 2

Effects of Vocal Aging on Fundamental Frequency and Formants

Samuel Mwangi[a], Werner Spiegl[b], Florian Hönig[b], Tino Haderlein[b], Andreas Maier[b] and Elmar Nöth[c]
[a]*J. K. University of Agriculture and Technology, Nairobi;* [b]*University of Erlangen-Nuremberg, Chair of Pattern Recognition;* [c]*University Erlangen-Nuremberg*

In this paper we study changes of the articulatory organs from young adulthood to old age. In particular, we focus on how the larynx and vocal folds age along with the rest of the body. The objective of the paper is to automatically compute the effects of vocal aging on fundamental frequency and the formants.
Based on the work of Harrington et al. we compare trends of the fundamental, 1st and 2nd formant frequencies as the age progresses. We also investigate how different phonemes, vowels in particular, are affected by vocal aging. The data consists of Queen Elizabeth’s annual Christmas

speech since the age of 26 to 76. The idea is to have the same speaker over a long period of time.
The results indicate that vocal aging causes the decline of the fundamental and the 1st formant frequencies while the 2nd and 3rd formants are not closely correlated to vocal aging. In general, the correlation between the fundamental frequency and age is -0.9027, while the average correlation for the 1st formant and age is -0.8071. In particular, the correlation between the 1st formant frequencies of phoneme ‘o’ and age is -0.8946 while for the 2nd formant is 0.3351.

Thu 8:40 Van Rijckevorsel Zaal Aeroacoustics 1

Aeroacoustic Prediction and Measurements of Jet-Airfoil Interaction Noise

Christophe Schram[a] and Julien Christophe[b]
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The noise emitted by the landing gear the and high-lift devices is a significant component of the overall noise radiated by aircrafts. In these applications, unsteady turbulent interactions inside the flow, and interactions of the flow with solid boundaries play an important role.
These mechanisms are investigated in this work by considering the case of a turbulent jet that interacts with a large span airfoil. Our prediction method is based on the aeroacoustical analogy, which is well suited for applications with low-Mach numbers. A flow model is used to generate equivalent aeroacoustic sources, which noise may in turn be propagated towards the listener. Incompressible Large Eddy Simulation is used to provide the flow data, and two approaches are combined to yield the acoustic field over the frequency range of interest. Curle's analogy is employed at low Helmholtz numbers (based on the airfoil chord), and Amiet's theory is implemented to obtain the large-frequency spectrum including chordwise non-compactness effects. Both the flow field and the acoustic field are compared with hotwire and microphone measurements conducted in the anechoic facility of the von Karman Institute.

Thu 9:00 Van Rijckevorsel Zaal Aeroacoustics 1

Flow induced modal scattering in a slowly-varying engine duct

Alex Smith, Nick Ovenden and Robert Bowles
Dept of Mathematics, University College London

Cut-on cut-off transition of acoustic modes in hard-walled ducts with irrotational mean flow is well understood, where an analysis around the mode's axial turning point shows that the incident mode undergoes a total reflection with a phase shift of $\pi/2$. However, previous finite-element simulations of this phenomenon presented in a paper by Ovenden, Rienstra and Eversman 2004 reveal the possibility of energy scattering into neighbouring modes at large Helmholtz number and large radial wavenumber. In this paper, such scattering phenomena of sound modes in slowly varying aeroengine ducts are investigated using multiple-scales techniques. It is found that, for sufficiently high frequencies, there is a mechanism whereby the energy of a propagating incident mode can be scattered into neighbouring modes so long as there exists a mean flow inside the duct. The energy distribution of the incident and neighbouring scattered modes can be obtained via matched asymptotic expansions and comparisons are made to the finite-element solutions mentioned above. The analysis is presented first for a duct of rectangular cross-section but then extended to ducts of both circular and arbitrary cross-section. Finally, other extensions of the research to double turning points and lined ducts are also briefly discussed.

Thu 9:20 Van Rijckevorsel Zaal Aeroacoustics 1

Slat Noise Source Identification for a High-Lift Configuration

Daniel König, S. R. Koh, W. Schröder and M. Meinke
Institute of Aerodynamics

To investigate the noise sources of a high-lift configuration consisting of a slat and a main wing an efficient hybrid LES/CAA method is used. A freestream Mach number of M=0.16 and an angle of attack of 13 deg is assumed. The Reynolds number, based on the clean chord length and the freestream velocity is Re=1.4*106. The LES is done by a finite volume solver of second order accuracy. The used computational mesh consists of 55 million grid points. The acoustic analyses are performed by solving the acoustic perturbation equations (APE) whereas the perturbed Lamb vector is used as the only source, i.e., vortex sound dominates. The pressure distribution and the mean velocity field are in good agreement with experimental data. In the slat cove shear layer vortical structures similar to those in plane shear layers and impinging jets are observed. The acoustic simulation shows two pronounced areas where sound is generated, i.e., the main wing trailing edge and the slat region, the latter of which is dominant. The slat gap is identified to be responsible for the generated sound between 1 and 3 kHz via space and cross correlations, respectively, between the acoustic pressure and the perturbed Lamb vector.

Thu 9:40 Van Rijckevorsel Zaal Aeroacoustics 1

Aeroacoustics taking Fluid-Structure Interaction into Account

Manfred Kaltenbacher[a], Stefan Becker[b] and Frank Schäfer[c]
[a]*Alps-Adriatic University of Klagenfurt, Applied Mechatronics;* [b]*Universität Erlangen-Nürnberg, Lehrstuhl für Strömungsmechanik;* [c]*University Erlangen-Nuremberg*

In the last years, manufacturers have started to consider the aerodynamic noise level in many industrial applications as a relevant design parameter (e.g. airplanes, cars, air conditioning systems, etc.). Because of the growing demand for reducing noise levels and for fulfilling noise regulations, there is a great motivation to investigate the basic aeroacoustic phenomena and mechanisms of sound generation and propagation. In many cases, the occurring noise is not only generated by the turbulent flow but also due to vibrations resulting from the interaction between the flow and the mechanical structure.

In the present work, we investigate the three field coupling for a flow past a thin flexible structure. Our developed numerical algorithms are capable to simulate the complex interaction between fluid, structure and acoustics. The scheme is based on a partitioned approach employing a finite-volume flow solver and a finite-element structural mechanics and acoustics solver. An important feature of the computational scheme is that it allows for a separate prediction of flow-induced and vibration sound. We will present detailed numerical studies of the interaction of the fluid flow

with the flexible structure and the generated noise. Moreover, a comparison of computational results to our experimental investigations will be provided.

Thu 10:00 Van Rijckevorsel Zaal Aeroacoustics 1

Measurement and computation of jet noise at moderate Mach and Low Reynolds number

Bendiks Jan Boersma
TU Delft

Direct Numerical Simulation (DNS) is a powerful tool to study noise generation mechanisms in turbulent flows. In a DNS all relevant scales of motion are resolved on a computational grid, this gives a severe restriction on the Reynolds number of the flow. Experimental validation of DNS data for acoustics is complicated because flow velocities have to be rather high in order to have a noticeable acoustic component. This is combination with a low Reynolds numbers gives in general flows which very small spatial dimensions (less than 1 mm). A way to achieve flows with high Mach numbers, low Reynolds numbers and workable dimensions is to lower the ambient pressure in the system.

During the presentation results of jet measurements will be presented which are obtained in our low pressure (semi)anechoic pressure tank. The results will be compared with from a numerical model which solves the compressible equations and uses porous FWH formulation for the far field sound.

Thu 10:40 Van Rijckevorsel Zaal Aeroacoustics 1

New generation of super low noise fans extends application potential

Henk van der Spek
Howden, Hengelo

For many air-cooled installations, like air-cooled heat exchangers and cooling towers in the chemical industry and air-cooled steam condensers of power stations, the air cooling fans are dominating noise sources. This results to restrictions in operation permits in particular for the operation during the night. That is why there is a growing need for more quiet fans. Already more than 10 years ago the first generation of so called super low noise fans has come to the market. Such impellers with diameters from 1 to 10 meter, have found successful applications all over the world from Australia in far east up to Hawaii in the far west. From the beginning of this millennium a new development has broken through. In this concept the hub section has been fully integrated into the composite construction of the impeller and that allows a more effective aerodynamic shaping. This has resulted into a 10 percent lower power consumption of the fans while keeping its superior acoustic features. The lower power consumption brings a new application potential in particular for diameters around 1 meter and pressure drops up to 500 Pa. Technical shaping, working and applications of the new fans will be explained.

Thu 11:00 Van Rijckevorsel Zaal Aeroacoustics 1

Experimental Investigation of Mode Scattering at C-Shaped Bypass-Duct Sections of Aero-Engines

Roland Bauers[a], Christian Stöhr[b], Ulf Tapken[a] and Lars Enghardt[a]
[a]*German Aerospace Center (DLR), Berlin;* [b]*Rolls-Royce Deutschland*

Modern aero-engines feature high bypass ratios to reduce fuel consumption and jet noise. As a consequence, fan noise has become one of the most annoying noise sources. Over a wide frequency range, fan tones excited by the rotor-stator interac-tion propagate in form of higher order modes through the bypass duct, which is in mosta cases radially divided by a pylon. The pylon is often paired with a supporting strut at perpendicular circumferential position. Both bifurcations form two independent c-shaped ducts, respectively, the boundary conditions of which substantially differ from the regular case of a cylindrical bypass-duct with hub. At DLR in Berlin, a di-ploma thesis was undertaken to experimentally investigate the impact of the py-lon/strut combination on the propagation of modes in a bypass-duct. The experimen-tal setup comprised different geometrical bypass duct configurations. Modes were excited by means of a loudspeaker ring in a cylindrical duct section, which bifurcated downstream to a c-shaped duct. Measurements of the modal field were accom-plished in both the cylindrical and the c-shaped duct sections. The paper will present the experimental results with emphasize on mode scattering and transmission loss at the c-shaped duct section.

Thu 11:20 Van Rijckevorsel Zaal Aeroacoustics 1

Extended Multi-plane Mode Matching for CFD/CAA Coupling

Christian Weckmueller, Attila Fritzsch and Sebastien Guerin
German Aerospace Center (DLR), Berlin

The objective is to extend the pressure mode matching method proposed by Ovenden and Rienstra in order to improve the aerodynamic/acoustic splitting at the interface between CFD and CAA computational domains. The numerical calculation and propagation of BPF tones in bypass ducts is the aimed application. CFD/CAA hybrid approaches have been developed for reducing the costs of aeroacoustic computations. Within the sound source region (rotor/stator stage), the unsteady Reynolds Averaged Navier-Stokes equations are solved, whereas in the rest of the computational domain, the sound propagation is computed with CAA methods solving the Euler equations. At the CFD/CAA interface, only the acoustic fluctuations should be transfered into the CAA domain to avoid stability issues in the CAA simulation. Ovenden and Rienstra proposed a so-called multiplane pressure mode matching method to remove the vortical part of the fluctuations. As suggested by De Roeck, this method can be extended with an aerodynamic model to improve the splitting. Here we have studied an aerodynamic model consisting of the same radial eigenfunctions as the ones taken for the acoustic modal basis, while the axial phase-velocity is the mean flow convection velocity. This is analytically investigated and applied to some numerical data.

Thu 11:40 Van Rijckevorsel Zaal Aeroacoustics 1

A method based on the ray structure of acoustic modes for predicting the liner performance in annular ducts with flow

Antoine Moreau[a], Sebastien Guerin[a] and Stefan Busse[b]
[a]*German Aerospace Center (DLR), Berlin;* [b]*TU Berlin*

The objective is to develop a fast method for predicting the sound attenuation by acoustic liners mounted in the inlet and bypass duct of turbofan engines. The method will be integrated into a prediction tool dedicated to design-to-noise. It works as follows: 1) The sound field generated by the fan stage is decomposed into acoustic modes (m,n) for a hard-walled annular duct with flow. 2) The ray structure of each acoustic mode is determined using an extension of an existing theory. 3) Finally the sound attenuation is calculated for each mode in function of the polar and azimuthal angles of propagation (angles at which the rays impinge on the wall) and the number of bounces; it is assumed that liners do not significantly modify the propagation angles and generate no mode scattering. The wall impedance is predicted using a model based on the geometrical parameters of practical liners. The results presented include the impact of acoustic liners on the modal distribution, the sound spectra, and the overall sound power. Both broadband and tonal components of fan noise are considered for a realistic configuration.

Thu 15:00 Van Rijckevorsel Zaal Aeroacoustics 2

Broadband Simulation Of Flow-Induced Noise Generation in Air Conditioning Ducts

Carsten Spehr, Roland Ewert and Jan Delfs
German Aerospace Center (DLR), Braunschweig

Air conditioning systems in cabins are possible noise sources influencing the passenger comfort in civil aircrafts. The current study continues an initiative to predict the generated noise on components of air distribution systems. The approach presented here is based on the numerical inexpensive RANS-simulation. Using a perturbation technique the turbulent flow fluctuations are represented by artificial turbulences injected into the mean flow field. In the first part of the study the time dependent perturbation was induced by injecting individual artificial vortices which allow qualitative comparison of the sound generation of different arrangements on the basis of the overall sound power level. In the current work a broadband turbulence is induced in the source region. The turbulence is thereby generated by filtering white noise to reproduce the autocorrelations and integral length-scales provided by a RANS-simulation. The interaction of the turbulence with AC-components generates pressure fluctuations with broadband character which are compared with corresponding measurements. Arrangements of two consecutive orifices were investigated as typical components in the air conditioning system. The sound generation on the downstream orifices depends therefore not only on its shape, but also on the flow disturbance, which in turn depends on the distance to the upstream orifice.

Thu 15:20 Van Rijckevorsel Zaal Aeroacoustics 2

Multiple side branch system as a model for corrugated pipes

Devis Tonon[a], Baptiste Landry[a], Stefan Belfroid[b], Johannes Willems[a], Gerardus Hofmans[a] and Avraham Hirschberg[a]
[a]*Eindhoven University of Technology;* [b]*TNO Science and Industry*

Corrugated pipes are widely used because they combine local rigidity with global flexibility. Whistling induced by flow through such pipes can lead to serious environmental and structural problems. The multiple side branch system is proposed as a model for corrugated pipes. The study has been restricted to cavities with sharp edges which are convenient for theoretical modeling. The side branch depth is chosen to be equal to the side branch diameter which corresponds to cavity geometries in typical corrugated pipes. Experiments have been carried out to characterize both the passive acoustic response of the system and its whistling behavior. These experiments show that the multiple side branch system is a reasonable model for corrugated pipes. A prediction model for the whistling behavior based on the vortex sound theory is proposed.

Thu 15:40 Van Rijckevorsel Zaal Aeroacoustics 2

Numerical Simulation of Rotor/Air-Jet Interaction with Active Control of Tonal Noise in Turbomachinery

Mathias Steger[a], Ulf Michel[b] and Frank Thiele[c]
[a]*TU Berlin, Inst. of Fluid Mechanics and Eng. Acoustics;* [b]*German Aerospace Center (DLR), Berlin;* [c]*TU Berlin*

A major contributor to the overall noise generated by modern high-bypass-ratio turbofan engines is fan noise. This noise source comprises both tonal and broadband components and arises due to a variety of mechanisms. For subsonic blade tip speeds, however, the radiated acoustic field is dominated by discrete tones at the blade passing frequency (BPF) and harmonics thereof. Primarily the interaction of the rotor-wakes with the downstream stators is the dominant noise source mechanism. The focus is to investigate the active control of tonal noise generated by rotor/stator interactions (primary sound field) by perturbing the flow near the outer casing using high-speed air jets. In the strategy followed here air nozzles are mounted in the rotor casing between the rotor and stator blades. For such a configuration a secondary sound field is generated by the periodic interaction of the rotor field with the high-speed air jets. This strategy is shown to be very effective in controlling the amplitude of higher-order mode sound fields at the BPF and harmonics thereof. The test case used comprises a scaled state-of-the-art axial fan with outlet guide vanes in a ducted inlet/outlet configuration. Numerical simulations are performed with a three-dimensional Unsteady-Reynolds-Averaged-Navier-Stokes flow solver.

Thu 16:00 Van Rijckevorsel Zaal Aeroacoustics 2

Experimental and Numerical Characterisation of a Non-Locally Reacting Liner

Stefan Busse[a], Christoph Richter[a], Claus Kückens[b], Ulrich Müller[c], Lars Enghardt[d] and Frank Thiele[a]
[a]*TU Berlin;* [b]*PFW Aerospace AG;* [c]*MeliCon GmbH;* [d]*German Aerospace Center (DLR), Berlin*

A new type of aero-engine liner was investigated. It is manufactured by combining a cell structure of assembled metal strips with a perforated face sheet and a solid backing plate by using a point welding process. The structure is completed with an additional face sheet layer of meshed metal baffle. Since neighbouring cells are not air tight the damper is non-locally reacting and inter-cellular communication is possible. The acoustic properties in terms of the insertion loss were experimentally analysed by integrating a liner sample in a side wall of a grazing flow acoustic test facility. Measurements with microphones and plane wave excitation were conducted in order to calculate the reflection and transmission coefficients. The influence of parameters like the flow Mach number and the liner length on the damping behaviour were investigated. By using a numerical method which incorporates the Extended Helmholtz Resonator model (EHR) the impedance was educed from the measurement values utilising an optimisation algorithm. It is shown that the method provides reasonable results even for this special type of liner.

Thu 16:40 Van Rijckevorsel Zaal Aeroacoustics 2

Separation of Aero- and Vibroacoustic Noise from a Flexible Structure

Jens Grabinger[a], Stefan Müller[b] and Stefan Becker[b]
[a]*Universität Erlangen-Nürnberg, Lehrstuhl für Sensorik;* [b]*Universität Erlangen-Nürnberg, Lehrstuhl für Strömungsmechanik*

The reduction of noise due to turbulent flow over a car's underbody is an important — yet complicated — issue in automotive engineering. Our objective is to study the mechanisms of noise generation at a simplified model of a car's underbody. For this purpose we mount a flexible rectangular plate into a rigid surrounding plate, which is then placed in a wind tunnel in parallel to the main flow direction. A square cylinder is mounted upstream of the flexible plate in order to model a strut in an underbody. This assembly generates noise by means of two mechanisms, namely the vibrations of the flexible plate and the turbulent flow.

These two mechanisms require different counter measures for noise reduction. Therefore, one has to separate the aeroacoustic and vibroacoustic contributions in the radiated sound. We measure the sound pressure with a microphone and, simultaneously, the normal deflections of the plate with a laser scanning vibrometer. On the basis of these data, we conduct FEM simulations to compute the vibrational sound at the position of the microphone. Subsequently, we estimate the contribution of the

vibrational sound by comparing it to measured data. We find that the vibration generates dominant tonal components at the plate's resonance frequencies, while the flow-induced noise gives a broadband contribution.

Thu 17:00 Van Rijckevorsel Zaal Aeroacoustics 2

Conception of a Mechatronik Test Rig for Acoustic Ground Test of Aircraft

Norbert Hoevelmann, Thomas Kletschkowski and Delf Sachau
Helmut-Schmidt-University Hamburg, Inst. Mechatronics

In an advanced - especially propeller driven - aircraft the sound pressure level may reach up to high values caused by propeller blades producing disturbances when passing the fuselage. This excitation induces high tonal noise levels in the low frequency range in the aircraft interior. Due to mass reduction as a major aspect in aircraft design research has been focused on active noise control (ANC). In order to test the performance of ANC-systems test facilities are needed for different technology readiness levels. To analyze the expected acoustic behaviour of future aircrafts under realistic conditions a test rig for acoustic ground test is needed. The present paper will outline the design process of a mechatronic mock-up including both airborne and structure borne noise sources as well as the associated transfer paths. It is planned to use data taken from numerical simulations to adjust a specific noise field inside the mock-up by a proper control scheme. To simulate relevant transfer paths it is planned to integrate modern light weight structures - such as CFK - into the test rig.

Thu 8:40 Plate Zaal Signal processing

Acoustic Object Detection in Adverse Conditions

Jörg-Hendrik Bach[a] and Jörn Anemüller[b]
[a]*Universität Oldenburg;* [b]*Universität Oldenburg, Medizinische Physik*

In the current work, the problem of acoustic object detection in adverse and realistic background noise is tackled based on a recently proposed method for speech detection in noise. The method is based on the decomposition of the signals into amplitude modulation spectrograms. These perceptually motivated features provide a signal description in terms of power density at combinations of acoustic frequencies and modulation frequencies within meso-scale temporal windows of about 1s length. The target object sounds -ranging from speech to animal voices and characteristic elements of office activities- are embedded in recordings from noisy scenes at various SNRs. These acoustic background scenes include, e.g., recordings from heavy traffic streets, pedestrian zones, landscape. Numerous combinations of objects and environments have been analysed to provide some typical real-world acoustic scenes. These rather adverse conditions are compared to the performance in white noise as a reference. The results from our perceptually motivated

features are compared to those obtained from standard of mel frequency cepstral coefficients (MFCC). Classification is based on the discriminative model of support vector machines (SVM) that are particularly suited for high-dimensional discrimination tasks.

Thu 9:00 Plate Zaal Signal processing

On the Classification of Acoustic Intervention Sequences for Essential Hypertension

Petra Friedrich, Tom Kohler and Bernhard Wolf
TU München, Heinz Nixdorf-Lehrstuhl f. Medizin. Elektronik

Acoustic signals can modulate the human metabolic and central-nervous functions and evoke physiological effects. Especially the anti-hypertensive effect of certain iterative sound-patterns as possible intervention for essential hypertension is examined in many recent studies as well as our own research work. It was shown that the acoustic sequences and sound clusters used for intervention can decrease blood pressure significantly. For a therapy, however, it is necessary to identify the musical active ingredients of the sounds in the sense of active pharmaceutical ingredients. This contribution discusses the systematic analysis of the musical features that are responsible for the anti-hypertensive effect. More than 400 features were extracted and investigated in terms of their relevance concerning a blood pressure lowering effect. The 17 most significant characteristics were used to develop a classifier based on the discriminant analysis that decides whether a sound pattern is sedative or stimulating. This will be clarified in acoustic demonstrations. With this tool it is possible to filter from a large selection of music sequences the most suitable therapeutical sound patterns. We now have the starting basis to provide individualized and personalized therapies while respecting personal preferences of users.

Thu 9:20 Plate Zaal Signal processing

On the Structure of the Phase around the Zeros of the Short-Time Fourier Transform

Florent Jaillet[a], Monika Dörfler[a], Peter Balazs[a] and Nina Engelputzeder[b]
[a]*Austrian Academy of Sciences, Acoustics Research Institute;* [b]*University of Vienna, Faculty of Mathematics*

The short-time Fourier transform is a time-frequency representation widely used in audio signal processing. At first sight, the phase of the transform may appear to be difficult to interpret, a fact that led to the development of many processing schemes in audio processing relying on the modulus of the transform only. However, it has been shown, that the phase information can be successfully exploited for improved analysis and processing. This is illustrated for example by the reassignment method or the phase vocoder algorithm.

In order to better exploit the phase information in audio processing, a better understanding of its structure is desirable. This study describes a particular behavior of the phase which can be observed when looking at the phase derivative over time or frequency around the points where the transform equals zero. This recurrent behavior is illustrated using numerical simulations on synthesized and real audio signals. The performed simulations show the influence of the parameters of the transform on the observed structures. Some consequences of this phase structure for audio processing are also presented.

Thu 9:40 Plate Zaal Signal processing

Signal Processing for Structure-Borne Sound of Rotating Processes

Jan O. A. Köhler
Institute of Technical Acoustics, RWTH Aachen University

For the acoustical monitoring of precise and ultra precise production processes, acoustic emission (AE) sensors are being used. Rotating machine tools like grinding, drilling, milling and especially turning machines are the objects in focus. The first cut and tool crashes as well as tool abrasion are meant to be recognised in order to minimise the non-productive time. For the laboratory experiments two AE Sensors can be used. The correlation between the two signals is used to gain the signal to noise ratio. However, for the final application the mounting of two sensors is technically not always possible. So the signal processing will be focused on only one AE sensor. Algorithms in the time as well as the frequency domain will be presented, inter alia to identify changes in the structure-borne sound caused by different rotational speeds and infeeds. Finally Wiener Filters are considered to de-noise the signals.

Thu 10:00 Plate Zaal Signal processing

Signal Component Estimation in Background Noise

Renante Violanda, Hedde van de Vooren, Ronald van Elburg and Tjeerd Andringa
University of Groningen

The selection of coherent regions in the time-frequency representation of sound signals is an essential step in computational auditory scene analysis. These regions, referred to here as signal components (SC), ideally represent all information about the sound sources present in the scene. However, due to the limitation imposed by the uncertainty relation between frequency and time, the corresponding energy distribution in the time-frequency representation is often smeared. This poses a difficulty in extracting SCs belonging to a specific sound source especially in the case where multiple sources are present. In this paper we present a method of estimating the SCs of a source using the instantaneous frequency (IF) and grouped-delay corrected time (GDCT) representation. The spread of the IF and GDCT in every time-frequency bin is used as a measure of the local signal-to-noise ratio, this ratio is high where a possible single sound source is present. We show examples of the applicability of this technique in the analysis of tonal sounds ranging from speech, music to tonal environmental sounds. With proper grouping of

the SCs, we show that our method is an important step towards speech extraction from noisy backgrounds.

Thu 10:20 Plate Zaal Signal processing

Real-time filtering structures for interactive geometry modification in acoustic virtual reality

Frank Wefers and Dirk Schröder
Institute of Technical Acoustics, RWTH Aachen University

Interactive modification of scene geometry is an important factor to improve the user's immersion in a virtual acoustic environment. However, modifications such as changes of surface properties and the scene geometry itself can result in strong sound field variations within very small time intervals. The opening and closing of a door to a neighbouring room which contains a high volume sound source illustrates such an event. In the sense of physical consistency, it is not sufficient to simulate these events by just adapting the volume of the sound source as the process of opening and closing of a door usually results in a significant change of reverberation time, sound coloration and source localization.
Real-time auralization by means of FIR filters is usually performed by non-uniform partitioned frequency-domain convolution. Slight changes in room acoustics can easily be incorporated into these algorithms, but fast changes require special filter exchange strategies. The low-pass characteristic of a door being closed will immediately influence the whole room impulse response (RIR). Therefore, the entire filter must be exchanged immediately to maintain physical consistency. In this contribution, efficient strategies for fast exchange of filters are proposed for a non-uniform partitioned convolution approach. Their relations to filter partitions are discussed and the computational costs are analyzed.

Thu 11:00 Plate Zaal Signal processing

Removing Components from a Time-Frequency Representation

Monika Dörfler, Peter Balazs and Florent Jaillet
Austrian Academy of Sciences, Acoustics Research Institute

Time-frequency representations such as the spectrogram or short-time Fourier transform seem to be well suited to the task of removing certain components with approximately disjoint support in the time-frequency domain. For example, one might be interested in suppressing a certain instrument's contribution from a music signal. Such approaches are used in Computational Auditory Scene Analysis by the name of Time-Frequency masks. However, the trivial approach of just deleting the corresponding component in the time-frequency representation leads to artefacts such as ”ghost-tones”, i.e. a modified version of the deleted component is still audible, or to the perception of unnatural sound coloration comparable to a ”phasing effect”. We compare different approaches to tackle this problem: a soft-thresholding procedure is applied and compared to a method using statistical models. On the other hand, we draw conclusions from considering the optimal separating mask for known signal components. In this situation, a model promoting sparsity in the representation can lead to favourable results by yielding a feasible approximation of the optimal mask. We perform informal listening experiments to evaluate the resulting removal of certain components in a complex audio-signal.

Thu 11:20 Plate Zaal Signal processing

Multi-channel noise reduction for binaural hearing aids by using short-time spectral attenuation combined with noise estimators for non stationary noise

Sven Franz, Jörg Bitzer and Uwe Simmer
FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie

Multi-channel noise reduction (NR) with binaural output is an ongoing research topic for future hearing aids. Especially the preservation of interaural level and time differences is important in order to keep the spatial impression intact. In this contribution we present an extended noise reduction algorithm based on the two-channel NR scheme from Kim et al. The algorithm has a structure similar to the generalized sidelobe canceler (GSC) structure, however, instead of reducing the noise components by cancellation, short-time spectral amplitude filtering is used. We extended and tested this algorithm with different compensation filters to reduce the estimation error as well as musical noise. As a new approach a compensation filter based on minimum statistics will be presented. This method has the advantage that no voice activity detector is necessary. The evaluation will compare different filter estimation techniques on a realtime developing platform for hearing aid algorithms. Finally, we will discuss the advantages and drawbacks of the compared methods.

Thu 11:40 Plate Zaal Signal processing

How to obtain high quality input data for auralization?

Pascal Dietrich and Matthias Lievens
Institute of Technical Acoustics, RWTH Aachen University

Auralization of structure-borne and airborne noise problems contributes to understanding of sound transmission in a significant way. One expects the auralization results to be as close to real world result as possible. In hybrid models where measurement and simulation data are both used to generate the final audible result, the measurement data obviously has to be very precise. Even highly accurate numeric simulations require measured material data, e.g. structure-borne impedances. Therefore, the measurement of input data is of high interest regarding the simulation quality.

Measuring structure borne impedances up to high frequencies for ongoing coupling calculations often lacks of insufficient signal to noise ratio and phase errors. Mostly measured force and acceleration signals are directly used for impedance calculations without exploiting the advantage of impulse response.

This contribution aims to clarify the influences of special post-processing of the measurement data on the obtained impedance by means of deconvolution, bandpass filtering and time-windowing.
Based on a measurement setup to study the prediction of the sound-radiation of a small structure-borne sound source, the signal processing for the measured impedances will be presented and discussed. Nevertheless, this methodology can be applied for airborne impulse responses and other transfer paths as well.

Thu 15:00 Plate Zaal Audiological acoustics

Predicting the acoustics of individual ears for hearing aid and audio applications - model framework and future work

Matthias Blau, Tobias Sankowsky, Philipp Roeske and Sven Fischer
FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie

Applications involving sound delivery to the ear, such as hearing aids, mobile phones, ear- and headphones, will always have to cope with the problem of varying acoustic loads, due to interindividual differences in shape of the outer ear and mobility of the ear drum. It has been known for a long time that this variability can lead to considerably different sound pressure levels at the ear drum for frequencies above a few kHz. An additional challenge is the more and more frequent use of ear molds which do not acoustically seal the ear canal ("open fittings" in hearing aid terminology). In this paper, a simple 1-dimensional model framework is presented, which may be used to predict individually different acoustics (including the pressure at the ear drum and acoustical feedback), or for real-time simulations of hearing instruments. The discussion will include known model sub-components, as well as open questions concerning the needed "individuality" parameters, or resulting for instance from violations of the 1D assumption in some parts of the model. Some of the open questions will be dealt with in the companion papers by Sankowsky et al. and by Roeske et al.

Thu 15:20 Plate Zaal Audiological acoustics

Estimation of the Sound Pressure at the Ear Drum for Hearing Aid Applications

Tobias Sankowsky[a], Matthias Blau[a], Philipp Roeske[a], Hamidreza Mojallal[b], Jens Schroeder[b], Magnus Teschner[b] and Martin Bokemeyer[b]
[a]*FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie;* [b]*Medizinische Hochschule Hannover*

The sound pressure at the ear drum is a reference quantity not only in hearing aids but in virtually all applications involving sound delivery to the ear, such as mobile phones, ear- and headphones. As the direct measurement of the ear drum pressure is too elaborate, it must be predicted in some way. One approach to do so is to establish a 1D model of the residual ear canal, based on the measured acoustic impedance at the interface between the ear shell and the residual ear canal.

In a study on 10 human temporal bones with closed ear shells, 1D models established this way were used to predict the drum pressure in the 100Hz..10kHz frequency range. The models were all based on the same measured impedances, but several variants of algorithms to derive the ear canal model were employed. The model predictions were then compared to direct measurements of the drum pressure. It turned out that while the robustness of the different algorithms varies, their mean predictions do not differ substantially. The agreement with the direct measurement will be discussed.

Thu 15:40 Plate Zaal Audiological acoustics

Modeling the sound field in front of the human ear drum for hearing aid applications

Philipp Roeske[a], Matthias Blau[a], Tobias Sankowsky[a], Hamidreza Mojallal[b], Jens Schroeder[b], Magnus Teschner[b] and Martin Bokemeyer[b]
[a]*FH Oldenburg/O/W, Institut für Hörtechnik und Audiologie;* [b]*Medizinische Hochschule Hannover*

The sound pressure at the ear drum is a reference quantity not only in hearing aids but in virtually all applications involving sound delivery to the ear, such as mobile phones, ear- and headphones. As the direct measurement of the ear drum pressure is too elaborate, it must be predicted in some way. Most attempts to do so exclude the influence of the vibrating drum from their analyses, although it is known that the ear drum vibrates in a complicated manner. In this paper, a finite element (FE) model comprising a detailed representation of the vibrating drum while preserving enough flexibility to be applied to a number of dead and live human specimen is presented and validated. The proposed method involves the representation of the vibrating drum by velocity sources whose source quantities are in turn based on laser doppler vibrometry (LDV) data measured on the respective specimen. The sensitivity of this method to calibration errors, to the effect of missing drum velocity data (towards the edges) and to deviations between the direction of measurement and the predominant direction of drum vibration is analyzed via comparison to a complete FE model (comprising the middle ear and the cochlea load).

Thu 16:00 Plate Zaal Audiological acoustics

Comparative simulations of adaptive psychometric procedures to estimate perceptual thresholds

Stefanie Otto and Stefan Weinzierl
TU Berlin, Fachgebiet Audiokommunikation

Various psychometric procedures have been proposed to measure perceptual thresholds as points on a subject's psychometric function. Adaptive procedures are generally preferred due to their precision and efficiency. However, different approaches exist with respect to the stimulus adaptation rule and the final estimation of the resulting threshold. In the

presented study, four widely-used methods were evaluated in computer simulations to analyse their performance over a large sample size in the case of known truth. An adaptive staircase method, Taylor and Creelman's PEST procedure, Pentland's Best PEST and the ZEST method (King-Smith et al.) were chosen. The presented methods differ in the amount of a priori information about the assumed threshold, the stimulus selection rule and the calculation of the final estimate (average of turnaround points, Maximum Likelihood or Bayesian approach). Each method was combined with a yes-no, 2AFC, and 3AFC stimulus presentation paradigm. The subject's true threshold was randomly distributed within a certain range of the stimulus set. Testing was terminated after a fixed number of simulated judgements, ranging from 10 to 50 trials. One thousand subjects were simulated for each condition reported. Bias, accuracy and efficiency of each method were analysed. The obtained results provide criteria for choosing the appropriate method for a particular application.

Thu 16:20 Plate Zaal Audiological acoustics

Fundamental sound fields in the core region of curved ear canals

Herbert Hudde and Sebastian Schmidt
Ruhr University Bochum, Institute of Communication Acoustics

The core region of ear canals denotes the section between the ”entrance” and the beginning of the ”drum coupling region” (DCR). The extension of the core region is specified by the absence of higher order modes. Within the core region usually pure fundamental modes are assumed which follow a middle axis of the ear canal. Such modes are characterized by regular surfaces of constant sound pressure (”isosurfaces”). Regular isosurfaces completely cover the aperture of the ear canal, and the middle axis is normal to the surface. However, Finite Element computations reveal that in the core region irregular surfaces of constant sound pressure occur. Such ”one-sided isosurfaces” are not oriented to the middle axis, but form dome-shaped surfaces oriented to a point on one side of the ear canal wall. In the talk the origin of ”one-sided isosurfaces” will be elucidated. Although one-sided isosurfaces cannot be described as fundamental modes, it is reasonable to consider these local disturbances as being part of a ”fundamental sound field”. It can be shown that one-sided isosurfaces are widely governed by the principle of energy minimization which also applies to regular fundamental modes.

Thu 17:00 Plate Zaal Audiological acoustics

Features of sound fields within and outside the ear canal

Sebastian Schmidt and Herbert Hudde
Ruhr University Bochum, Institute of Communication Acoustics

In the sound field between pinna and eardrum, three regions with different characteristics can be distinguished: (a) The external field is determined by the pinna geometry and the source. Surfaces of equal pressure

(”isosurfaces”) have distinctly three-dimensional structure. (b) In the core region, the isosurface shape does not depend on the source and is fairly regular. (c) Near the drum, higher order modes are excited by its vibration, thus the local field does not belong to the core region. A finite element model of a natural pinna and ear canal terminated with a vibrating eardrum was used to study the sound field features in detail and to check several conditions usually assumed when using one-dimensional models. It turned out that an entrance surface of the core region can actually be specified although it depends to some extent on the wave incidence direction and on frequency. Isosurfaces of pressure magnitude and phase are not always coinciding. Close to the drum, the field is composed of the expected higher order modes and of isosurfaces that are oriented orthogonal to the membrane.

Thu 17:20 Plate Zaal Audiological acoustics

Comparison of Speech Intelligibility by EAS, Bimodal, Uni- and Bilateral Cochlear Implant Patients in a ’Multi-Source Noise Field’ (MSNF)

Tobias Rader[a], Uwe Baumann[a] and Hugo Fastl[b]
[a]*Audiological Acoustics/ZHNO Univ. of Frankfurt;* [b]*AG Technische Akustik, MMK, TU München*

Speech perception in noise is one of the most difficult tasks for people suffering from hearing impairment. The Oldenburg Sentence Test (OLSA) is a useful tool to investigate speech intelligibility threshold in noise. In the present study, a multi-source noise field introduced by Rader et al. (DAGA2008) consisting of a four loudspeaker array with independent noise sources was combined with the OLSA. The MSNF allows to present a more realistic noise environment and shows a higher effect of binaural interaction regarding the separation of signal and noise from different directions.

Three different noise characteristics were applied in two conditions (S0N0: Signal and noise from the front; MSNF) to investigate speech reception threshold: Oldenburg noise (generated by superposed sentences of OLSA), CCITT noise, and speech simulating noise according to Fastl (1987).

Four different groups of cochlear implant patients separated into listening conditions unilateral, bimodal, bilateral and EAS (electric-acoustic stimulation) served as subjects in the present study.

Results showed a clear discrepancy between the examined patient groups for Fastl noise in MSNF (ΔSNR=11.0dB). For OLSA noise in S0N0-condition, the results differ less (ΔSNR=4.7dB). The MSNF with Fastl-noise is particularly suitable to measure the performance of various aided CI-patients in realistic sound environments.

Thu 8:40 Van der Vorm Zaal Cavitation 2

About the Threshold of the Transient Ultrasonic Cavitation in the Cavitational Noise at Different Frequencies

Christoph Jung and Reinhard Sobotta
Elma Hans Schmidbauer GmbH & Co KG

Results are Reported of Measurements of the Threshold Behaviour for the Transient Cavitation in Water at Different Ultrasonic Frequencies and Dependent on the Active Input Power.

Thu 9:00 Van der Vorm Zaal Cavitation 2

Comparison of cavitation bubble arrays at different frequencies

Andrea Otto[a], Till Nowak[a], Robert Mettin[a], Frank Holsteyns[b] and Alexander Lippert[b]
[a]*Drittes Physikalisches Institut, Universität Göttingen;* [c]*SEZ AG, Villach*

Arrays of bubbles are formed in gassy water under sonication at different frequencies from 20 kHz to 1 MHz. We investigate details of the structures with respect to similarities and frequency scaling. The experimental methods include high-speed imaging, acoustic pressure and luminescence measurements. Visually the structures appear rather similar: bigger ”degassing” bubbles gather in arrays close to the pressure nodes, and smaller bubbles show a fast motion inbetween. Bubble sizes, velocities, and densities are measured, and we try to highlight also differences between the structures at different acoustic frequencies from an experimental and theoretical point of view. One goal of the study is to reveal and understand qualitative transitions in acoustic cavitation bubble structure formation for more and more increased driving frequency.

Thu 9:20 Van der Vorm Zaal Cavitation 2

Parameters of sonochemistry in cleaning vessels in comparison to sound field and mechanical effects

Christoph Kling, Christian Koch and Klaus-Vitold Jenderka
Physikalisch-Technische Bundesanstalt, Braunschweig

There are many applications of ultrasound in medical science and engineering which are based on the physical effect of cavitation in liquid media such as cleaning and sonochemistry. Unfortunately, cavitation is a stochastic process and is strongly influenced by environmental conditions. This makes it difficult to control or even optimise a certain application process. Up to now, empirical assessment methods are used to account for the influence of parameters like temperature, frequency, vessel geometry, surfactants, input power and others because it lacks of an objective description of the different cavitation processes. Therefore, a project at PTB aims to relate different cavitation effects to each other and to measurement parameters obtained from the underlying sound field.
In this context, sonochemical reaction in a cleaning vessel was studied. Simple oxidation of iodide ions was used to investigate the influence of

different parameters. The reaction was restricted to small closed volumes inside the filled vessel to obtain spatial resolution of the cavitation action. The chemical efficiency of ultrasonic irradiation was analysed spectrometrically. The spatially resolved data were compared to quantitative results from mechanical cavitation effects.

Thu 9:40 Van der Vorm Zaal Cavitation 2

Acoustic Cavitation with Laser-generated Bubbles

Thomas Kurz, Dennis Kröninger, Tobias Wilken, Laurens Wißmann, Alexander Rohr, Hendrik Söhnholz and Werner Lauterborn
Drittes Physikalisches Institut, Universität Göttingen

Optical generation of bubbles in an insonated liquid by laser-induced breakdown is a versatile tool to study the dynamics of acoustically driven single bubbles or bubble groups. Such bubbles have well-defined, reproducible parameters, and the technique allows to investigate the radial dynamics, translatory motion and bubble interaction e.g. by high-speed imaging.

We report on measurements of bubbles produced with (femtosecond and nanosecond) laser pulses at the pressure anti-node of a standing wave field in a water-fillled cuvette driven by an ultrasonic transducer. With appropriate timing of the laser pulses spherical bubbles can be produced that emit light upon collapse. The luminescence is enhanced upon increase of the acoustic pressure and decrease of water temperature. Bubbles generated off the anti-node collapse aspherically, and jet formation is observed. The jet is aligned with the direction of the pressure gradient and can be used to map the sound field qualitatively. By using a spatial light modulator as a digital holographic device the focus position can be controlled electronically, and bubble pairs or bubble groups in the bulk of the liquid can be created to study their mutual interaction. First results of this method are presented.

Thu 10:00 Van der Vorm Zaal Cavitation 2

Translation of Contrast Agent Microbubbles Under the Effect of Ultrasound Radiation Pressure: the Role of History Force

Valeria Garbin
Physics of Fluids, University of Twente, Enschede

The acoustic coupling between bubbles in a sound field produces a net attraction between bubbles pulsating in phase. This modifies their resonance frequency, which depends on the distance from neighboring bubbles. Quantifying this effect is crucial for optimizing medical imaging protocols. We study experimentally the attraction between microbubbles under the effect of the ultrasound field. In order to prevent sliding friction at the wall, and to decouple the acoustic effect due to the wall vicinity, we use optical tweezers to position a bubble pair away from the sample chamber wall. The ultra-high speed imaging facility Brandaris 128 is used to optically record the bubble dynamics at 15 million frames per second. The bubbles execute small translatory oscillations around a position that slowly drifts, resulting in the net attraction observed for bubbles pulsating in phase. We model this behavior by writing the force balance for each bubble. Since, to prevent dissolution or coalescence, the bubbles are coated with a layer of lipid molecules, a condition of no-slip is assumed at the boundary. We find that history force effects are of paramount importance in determining the viscous dissipation on coated microbubbles pulsating and translating under the effect of ultrasound radiation pressure.

Thu 10:40 Van der Vorm Zaal Cavitation 2

Density Threshold for Acoustic Cavitation in Water

Kristina Davitt, Arnaud Arvengas and Frédéric Caupin
Ecole Normale Supérieure

We use a focused 1 MHz acoustic burst to generate large mechanical tensions in bulk water. We describe how this technique allows us to generate cavitation and to study its statistics. We have recently built a fiber optic probe hydrophone, to measure the density threshold for cavitation. We report the temperature dependence of this quantity, and compare it with other experiments.

Thu 11:00 Van der Vorm Zaal Cavitation 2

Ultrasound contrast agents: Why shell buckling matters!

Michel Versluis
Physics of Fluids, University of Twente, Enschede

The working principle of ultrasound contrast agents is based on the non-linear scattering of the coated microbubbles. Pulse-inversion and power modulation imaging rely on the harmonic response of the bubbles. The bubble dynamics is described by an extended Rayleigh-Plesset-type equation, and harmonic imaging using contrast agents has always been attributed to the non-linear properties of this equation. Here we show that the coating material leads to an increased non-linear bubble response even at low acoustic pressures where the traditional models for coated bubbles would only predict linear behavior. It was observed that the radial bubble dynamics can be highly non-linear and observations of so-called thresholding and 'compression-only' behavior were found to lead to an increased acoustic harmonic response. The non-linear behavior is captured by the model introduced by Marmottant et al., taking into account the elastic regime of the shell, and in addition, buckling and rupture of the lipid shell. Optical recordings of the resonance curves of the bubbles revealed that the shell viscosity appears to decrease with increasing dilatation rate. Secondly it was observed that the resonance curves become asymmetrical with increasing driving pressure, and the skewness of the resonance curves can be fully explained by the buckling model.

Thu 11:20 Van der Vorm Zaal Cavitation 2

Properties of Bubbles at Pressure Antinodes in a Standing Sound Field

T. Davaadorj, P. Koch, A. Pluta, U. Parlitz and Werner Lauterborn
Drittes Physikalisches Institut, Universität Göttingen

The parameter space for trapping bubbles in an acoustic resonator is limited by various factors, notably dissolution of the bubble by diffusion of the gas inside the bubble into the surrounding liquid, the primary Bjerknes force driving strongly oscillating bubbles away from the pressure antinode and parametric instabilities leading to surface oscillations of large amplitude. The region of stability in the parameter plane made up of the bubble radius at rest and the driving sound pressure amplitude at a frequency of 1 MHz is determined for different static pressures. The temperatures at bubble collapse in this region are calculated. The maximum attainable temperatures appear along the upper stability boundary in bubble resonance conditions.

Thu 11:40 Van der Vorm Zaal Cavitation 2

What are Contents of Spark-Generated Bubbles: Gas or Vapour?

Karel Vokurka[a] and Silvano Buogo[b]
[a]*Technical University of Liberec;* [b]*CNR-Institute of Acoustics, Roma*

According to their contents, two basic kinds of bubbles can be distinguished from a theoretical point of view: gas bubbles, which contain a noncondensable gas in their interior, and vapour bubbles, which contain a vapour of the surrounding liquid in their interior. As far as real bubbles are concerned, such as those generated by spark discharges, little is known about their actual contents. In this presentation, data obtained recently during experiments with spark generated bubbles will be examined with an aim at throwing more light on this question. It will be shown that in some intervals of their life the spark-generated bubbles behave predominantly as gas bubbles and in other intervals predominantly as vapour bubbles. For example, vapour bubble behavior is prevailing during the first compression phase and gas bubble behavior is prevailing during all other phases. These conclusions are based on comparison of theoretical bubble models with measured pressure waves radiated by spark bubbles. And in this respect a possibility to generate experimental bubbles oscillating with different oscillation intensities has proved to be essential. Experimental evidences supporting the above conclusions will be presented at the conference.

Thu 15:00 Van der Vorm Zaal Ultrasound

The effect of ultrasonic parameters on pressure fields in a membrane cleaning application

Fabian Reuter, Robert Mettin and Werner Lauterborn
Drittes Physikalisches Institut, Universität Göttingen

Successful ultrasonic cleaning of biofilms from polymer membranes has been repeatedly reported. Responsable for the cleaning are cavitation processes induced by the intense ultrasound. However, systematic investigations of ultrasound parameters and their effects on the acoustic field have not yet been conducted. In this work, the pressure field in a water filtration test plant, working with a polymer membrane, was measured with high spatial and temporal resolution for different conditions such as frequency or intensity. The measurements were carried out under realistic running conditions of the water filtration. During the experiments the membrane integrity was ensured by online particle and turbidity measurements of the permeate. A spectral evaluation of the pressure field together with high speed videometry of cavitation bubble structures reveals the influence of parameters for the generation of certain bubble structures. Furthermore, the membrane surface has been inspected microscopically afterwards to identify the role of bubble structures in the cleaning process.

Thu 15:20 Van der Vorm Zaal Ultrasound

Ultrasound field of a cylindrically-focused transducer

Arik Funke[a], Laure Bossy[b], Emmanuel Bossy[a], Marie-Françoise Cugnet[c], Patrick Chauvin[d] and Didier Cassereau[b]
[a]*ESPCI, Laboratoire d'Optique;* [b]*ESPCI, Laboratoire Ondes et Acoustique;* [c]*AREVA - CEZUS Research Center;* [d]*AREVA - CEZUS*

In this work, we investigate the ultrasound field generated by a cylindrically focused transducer with a circular aperture, used in non-destructive testing of metallic tubes. Both experimental and simulation studies were performed. Different from usual diffraction side lobes, additional significant side lobes are observed at a much larger scale in the focal pattern, along the direction normal to the transducer axis and the curvature axis. These additional side lobes appear to be caused by the particular three dimensional geometry of the aperture.

Based on a spatial impulse response approach, we show that they result from constructive interferences between edge waves originating from specific points of the aperture boundary. Such interferences occur only under specific aperture geometries such as circular or elliptic aperture, and cannot be observed for cylindrically focused transducer with rectangular apertures.

Thu 15:40 Van der Vorm Zaal Ultrasound

Advances in Ultrasonic Measurements Using Laser Doppler Vibrometer

Heinrich Steger and Reinhard Behrendt
Polytec GmbH

Laser Doppler Vibrometry is a widely used technique to exermine ultrasonic waves in solids , liquids and gases. The main advantages using the Laser Doppler Vibrometry in the field of ultrasonic measurements are the constant sensitivity over a very wide frequency range, the non contact measurement and the high spatial resolution. Different application examples prove the high perfomance of this measuring technique. The sensitivity and the resolution of different ultrasonic Laser Doppler Vibrometer will be discussed among thmeselves and in comparison with standard ultrasonic detectors. Suggestions for the improvement of the signal-to-noise ratio are given and verificated experimentally.. Finally the properties of new high-performance Ultrasonic-Laser-Doppler-Vibrometer systems will be discussed and new appülications will be pronounced.

Thu 16:00 Van der Vorm Zaal Ultrasound

In-line monitoring of polymer extrusion processes with ultrasonic attenuation spectroscopy

Jan Müller, Sven Kummer, Karin Sahre and Dieter Fischer
Leibniz-Institut für Polymerforschung Dresden e.V.

Extrusion is one of the most applied technologies for mixing and modifying of polymers in the melt. We use extrusion to process polymer nanocomposites. For a real time characterization of the material it is necessary to make in-line measurements to get information about concentration, dispersion and distribution of particles in the final material. An ultrasonic measuring system for in-line monitoring using two transmission probes to determine ultrasonic velocity and attenuation was developed. In particular, the analysis of attenuation spectra in the range from 3 to 10 MHz has proved as a useable method for in-line monitoring in an extruder. By using multivariate analysis, especially multivariate regression, different morphologic properties like particle size, distribution and dispersion and concentration of nanofillers can be determined.

Thu 16:40 Van der Vorm Zaal Physical acoustics

Application of Atomic Force Acoustic Microscopy for High Resolution Quantitative Elastic Imaging

Sigrun Hirsekorn[a], Ute Rabe[a], Anish Kumar[b], Kai Geng[a] and Walter Arnold[c]
[a]*Fraunhofer Institut für zerstörungsfreie Prüfverfahren (IZFP);* [b]*Indira Gandhi Centre for Atomic Research (India);* [c]*Universität des Saarlandes*

Atomic force acoustic microscopy (AFAM) is a dynamic operating mode of an atomic force microscope (AFM). The vibration resonances of an AFM cantilever in the ultrasonic frequency range when contacting a sample surface are exploited for a contact resonance spectroscopy technique. As imaging quantity local contact resonance frequencies or cantilever vibration amplitudes or phases can be used. The contact stiffness distribution in polycrystalline metals (nickel base alloy 625 and 9Cr-1Mo ferritic steel) was imaged with spatial resolution in the nm range. Taking advantage of the fact that the anisotropic indentation moduli of metals with cubic single crystal symmetry are very close to its isotropic value following from the macroscopic elastic constants of the polycrystal, the indentation moduli of precipitates were determined using the matrix as a reference. For quantitative evaluation of AFAM data flexural and torsional resonances of silicon cantilevers were calculated by finite element methods (FEM) taking into account the elastic anisotropy of a silicon single crystal. The results of the FEM simulations were compared to measured resonance frequencies of real cantilevers and to values calculated by analytical models. The influence of different geometrical parameters of the cantilevers on the results obtained for the local elastic stiffness was discussed.

Thu 17:00 Van der Vorm Zaal Physical acoustics

Surface and guided waves in porous materials

Jan Descheemaeker, Naima Sebaa, Ponnam Khurana and Walter Lauriks
KU Leuven

The last decades, a lot of research has been done about the propagation of sound in heterogeneous and porous materials. Especially the two-phase nature of the poro-elastic material leads to interesting physical phenomena. During recent years, several measuring methods have been developed to determine the material parameters. However, a lot of problems aren't solved yet. Because of the visco-elastic behavior of a lot of these materials, the elastic moduli will become frequency dependent (rubberlike behavior at low frequencies and glasslike behavior at high frequencies). Experimental data about the frequency- and temperature dependence of the elastic moduli are scarce because of experimental difficulties. Some experimental and numerical results about this subject will be presented . Porous materials still have to be studied in a lot of configurations. One of them is a porous material in an elastic cylinder. Some numerical results about the dispersion curves of the wave propagating in the system will be presented.

Thu 17:20 Van der Vorm Zaal Physical acoustics

A source path contribution analysis on tire noise using particle velocity sensors

Hans-Elias de Bree[a] and Dirk Bekke[b]
[a]*Microflown Technologies;* [b]*Vredestein Banden B.V., Enschede*

Road tire noise is an important topic of research where acoustic particle velocity based testing techniques can be expected to bring new insights. Modal analysis can be carried out using non contact particle velocity sensors, and PU sound probes can be used to measure the radiated sound without a need to use anechoic testing conditions. A further breakdown of the overall sound pressure levels measured in to its various sources can be made by applying a source path contribution analysis, using the PU probes to measure velocity, sound intensity and, for determining the reciprocal transfer path, the sound pressure. The concept of using this type of transfer path analysis will be outlined and illustrated by a tire noise case.

Thu 17:40 Van der Vorm Zaal Physical acoustics

Optimization of lightweight floors in the low frequency range with a FEM based prediction mode

Andreas Rabold[a], Alexander Düster[b], Joachim Hessinger[a] and Ernst Rank[b]
[a]*ift Schallschutzzentrum, Rosenheim;* [b]*Lehrstuhl für Computation in Engineering, TU München*

The impact noise transmission at low frequencies is a well known problem of lightweight floors, which is treated in many publications. A satisfying solution, considering the different construction principles of lightweight floors, could not be found so far. To overcome this problem a FEM based prediction model for the optimization of the floor construction and the improvement of the impact sound insulation has been developed and applied in a current research project at the TU München. The overall approach of the prediction model consists of the three-dimensional modelling of the structure and the excitation source (standard tapping machine), the subsequent modal- and spectral analysis and the computation of radiated sound from the ceiling. The validation of the prediction model has been carried out by comparing the evaluated impact sound pressure levels with results from measurements on 25 different floor constructions. In the next step the prediction model was used for the improvement of established lightweight timber floors. Finally these constructions were tested in a laboratory test stand according to ISO 140-6 at ift Rosenheim centre for acoustics and the ibp Stuttgart. This contribution shows the results of the computations and the construction rules developed for optimized lightweight floors.

Thu 8:40 Hudig Zaal Source identification 3

Comparison of Sound-Source Localization Methods for Vibrating Structures

Markus Müller-Trapet and Pascal Dietrich
Institute of Technical Acoustics, RWTH Aachen University

A thorough comparison of sound-source localization methods for vibrating structures, exploring their theoretical and practical limitations, will be presented. The beamforming technique will be focused on as this method has become more and more popular and various approaches and algorithms exist that have to be evaluated for their accuracy and usability. This work concentrates on the planar case, which is assumed to be the most common one.
As a practical example, velocity measurements on vibrating plates will be compared to the output of different beamforming algorithms. To thoroughly investigate the signal processing, a virtual measurement setup based on the measured velocity data will be used. The microphone arrays and their input will be modeled using Boundary-Element-Method (BEM) simulations in order to be independent of microphone directivity and measurement uncertainties. The different results of the beamforming algorithms, the velocity measurements and the simulated sound intensity will be compared and discussed.

Thu 9:00 Hudig Zaal Source identification 3

Acoustic Ground Tests in a Cross-Section of a Long-Range Airliner for Validation of the Inverse Finite Element Method

Kai Simanowski[a], Thomas Kletschkowski[a], Delf Sachau[a] and Bernhard Samtleben[b]
[a]*Helmut-Schmidt-University Hamburg, Inst. Mechatronics;* [b]*Airbus Deutschland*

Identification of noise sources in aircraft cabins proves to be difficult particularly at low frequencies. A new approach, based on the Inverse Finite Element Method (IFEM), reconstructs the spatial distribution of sound pressure and particle velocity at the interior wall. This procedure requires measurements in the cavity first. If all sound sources are located on the boundary, the equation system resulting from a matching FE model can be resorted in such a way that computation of the unknown boundary impedance is possible. The paper presents the results of acoustic ground tests that were performed to prepare a validation of this noise source localization technique. An aircraft mock-up (cross section of a long-range airliner) was excited by both interior and exterior noise sources for tonal as well as broadband disturbances. In both cases the resulting SPL and the frequency response were mapped with a custom-built microphone array at 7000 measurement points. The measured data was used to analyze the operational modes and the acoustic properties (such as reverberation time) of the cavity. Furthermore, the impedance at selected

boundaries was calculated. The experimental results were then compared with numerical results of forward (FEM) and backward (IFEM) calculations.

Thu 9:20 Hudig Zaal Source identification 3

Fingerprinting rotation machinery

Hedde van de Vooren, Renante Violanda, Ronald van Elburg and Tjeerd Andringa
University of Groningen

Malfunctioning of a complex rotation machine is often preceded by (subtle) changes in its vibrational behavior on top of the non-stationarities induced by the variation in rotation frequency. Although the analysis of these vibrations is fairly complex, these changes can be exploited to detect possible defects. An actual implementation in an early-warning system can be used to prevent machinery from severe damage. All the expected defect related vibrations , especially bearing damage, show the same dependency on rotation frequency as the vibrations associated with normal operation. For that reason, the defect-related-vibration frequency component can be found at a fixed position after normalizing the vibration signal. Our normalization technique, based on instantaneous frequency analysis and harmonic grouping, extracts the rotation frequency from the vibration signal itself rather than using a separate measurement . Nederlandse Gasunie - a gas transport company - provided us with 25 vibration recordings gathered from the main axes of 5 operating gascompressors. Each recording was normalized providing us with a characteristic fingerprint for every machine. First results show that these fingerprints show a remarkable robustness under changes in operating conditions and strong stability over longer periods of time ($>$1 year) making them suitable for condition monitoring purposes.

Thu 9:40 Hudig Zaal Sound propagation

A meteorological-acoustical model: comparison with measurements

Ando Randrianoelina and Frits van der Eerden
TNO Science and Industry

The propagation of sound over long distances, typically between 1 and 10 km, can be largely affected by the meteorology. For areas with a combination of land and water, the vertical wind and temperature gradients can vary along the propagation path. A detailed meteorological-acoustical model, using large scale weather forecast data, has been developed and presented before. This paper presents the comparison of the model results with the results from a detailed measurement campaign in the Netherlands. Within the Dutch project ”Geluid in Beeld” (A view on sound), the propagation of industrial sound from Europort/Maasvlakte area towards the city of Oostvoorne is investigated. This project was initiated by The Port of Rotterdam, DCMR, EMO and ECT, and has the objective to obtain a better understanding of the relationship between the emitted industrial sounds, the sound propagation and the reported complaints at the surrounding community. For the measurements, a set of 16 loudspeakers was used as a sound source. For two source positions and a total of three measurements lines, sound measurements were done in the afternoon, evening and night. Meteorological measurements at several heights were performed at two positions.

Thu 10:00 Hudig Zaal Sound propagation

General classification scheme for outdoor sound propagation situations

Karl-Wilhelm Hirsch[a] and Berthold Vogelsang[b]
[a]*Cervus Consult GmbH;* [b]*Niedersächsisches Ministerium für Umwelt und Klimaschutz*

Noise assessment normally considers a long term energy equivalent sound pressure level and a maximum level to indicate the noise for correlation with noise impairment, e. g. annoyance. This concept is found throughout many national and local regulations. The information on noise situations from these two indicators is not sufficient to make reliable assessments and, in particular, to make progress in the field of environmental noise control. There are several reasons to go beyond. For instance, in complex noise situation the perception of noise depends on audibility and background noise. And modern noise management wants to take advantage of weather conditions on a daily base. Progress is possible because the knowledge on sound propagation outdoors has strongly improved. Also the increasing computer power and the facilities of modern measurement systems add new options to overcome the constraints of a two-level-indicator concept.

This paper suggests the prediction of a level distribution and its indicators to broaden the information on noise exposure. It recommends a general scheme of classification of all decisive input parameters of the source, of the sound propagation situation and of the receiver site in order to obtain this distribution through a variation of these parameters within their classes.

Thu 10:20 Hudig Zaal Sound propagation

Inverse reconstruction techniques and their applicability for acoustic travel-time tomography

Gabi Fischer, Manuela Barth and Astrid Ziemann
University of Leipzig, Institute for Meteorology

Remote sensing by acoustic travel-time tomography enables one to determine area-averaged sound speed fields in an area of interest, e.g. the atmospheric boundary layer. Due to the dependency of the sound speed on temperature, the distribution of this meteorological quantity can be reconstructed by inverting the given travel times of sound signals along defined sound ray paths through a scanned medium. Over the past years, different inverse approaches were used for acoustic tomography. In this study, the results achieved with matrix inversion, simultaneous iterative reconstruction technique, and stochastic inversion are compared. To identify the algorithm which yields the optimal results by reconstructing two-dimensional temperature fields, both synthetically generated and experimental travel times were considered. On the one hand, the temperature fields were compared concerning their reconstruction accuracy and spatial resolution. The results show that the stochastic approach yields spatially highly resolved and accurate temperature fields. Furthermore, to estimate the applicability of the different methods during operational studies, the computational effort was analysed. Here, the stochastic inversion offers some deficits because of the relatively long computational time. Oppositely, the matrix inversion and the iterative reconstruction technique are suitable for online interpretations by providing temperature fields with acceptable accuracy.

Thu 11:00 Hudig Zaal Sound propagation

Adapting a sound ray model of the atmosphere to simulate sound propagation in urban environment

Astrid Ziemann and Gabi Fischer
University of Leipzig, Institute for Meteorology

Noise exposure in urban areas is an important factor which influences the well-being and quality of life of urban residents. According to recent studies, the population is most frequently disturbed by road traffic noise. Particularly in residential areas near main roads this problem is essential, because prescribed noise levels are often exceeded here. When analysing this problem, the influence of the atmosphere on sound propagation under specific urban environmental conditions also has to be investigated. Therefore, to operationally answer questions of noise protection, the sound propagation model of the atmosphere using ray-tracing (SMART) has been developed. In the present study, this model, which considers the influence of the vertical atmospheric structure on sound refraction and absorption, was enhanced to allow simulations of sound propagation in urban areas. Thereby, investigations of the dependency of the sound propagation on temporally variable weather conditions were performed. Furthermore, as a consequence of noise exposure near residential areas, noise barriers are often installed along busy roads. To quantify the effect of this active noise protection, a barrier was implemented in SMART. The results of first studies illustrate that special weather conditions might reduce the effect of such noise barriers.

Thu 11:20 Hudig Zaal Sound propagation

The Harmonoise sound propagation model: further developments and comparison with other models

Dirk van Maercke[a] and Erik Salomons[b]
[a]*CSTB Grenoble;* [b]*TNO Built Environment and Geosciences*

The Harmonoise sound propagation model ('the Harmonoise engineering model') was developed in the European project Harmonoise (2001-2004) for road and rail traffic noise. Further developments of the model were performed in the European project Imagine (2004-2006), including extensions of the model to aircraft noise and industrial noise. In 2008, CSTB Grenoble and TNO Delft have prepared a detailed description of the various steps involved in a calculation with the Harmonoise model. In the course of this joint project, some elements of the model were further improved. In 2009, test calculation were performed with the model, and results were compared to results of other models. In this presentation, an overview will be given of the Harmonoise model, including salient features such as the convex hull approach, Fresnel weighting for irregular terrain, and ground curvature to account for the effect of atmospheric refraction. Results of the comparisons with other models will also be presented.

Thu 15:00 Hudig Zaal Electro-acoustics

Investigations on measurement control and signal processing for the pressure reciprocity calibration of microphones

Thomas Fedtke and Thomas Rewig
Physikalisch-Technische Bundesanstalt, Braunschweig

The realization of the unit of sound pressure is done by determining the open-circuit pressure sensitivity of laboratory standard microphones using the reciprocity technique with microphones coupled in pairs by means of cylindrical couplers. The metrological centrepiece of this process is the measurement of the ratio between the open-circuit output voltage of the receiver microphone and the current through the transmitter microphone: the electrical transfer impedance. For its determination, which is based on a computer controlled reference resistor, four alternating voltages are measured, and their quotient is used to calculate the transfer impedance from the resistance value of the reference resistor.

Therefore, the measurement uncertainty of the electrical transfer impedance is influenced by the uncertainties of the reference resistor and of the voltage ratio. In order to obtain an optimal stability and to suppress disturbances, signal analysis and signal processing are used for the voltage measurements, such as averaging procedures and signal stability criteria. Measurement control and signal processing were optimized with regard to the time required for the measurement and the uncertainty. For this purpose, techniques derived from statistical process control, such as control charts, were used and adapted to provide objective criteria for the quality of measurement signals.

Thu 15:20 Hudig Zaal Electro-acoustics

Fast distortion measurements in relation to frequency and level

Christian Budde[a] and Swen Müller[b]
[a]*AIXcoustic creations;* [b]*Instituto Nacional de Metrologia (INMETRO), Brazil*

Distortion measurements vs. frequency and/or signal level are normally performed with a series of pure tones whose frequency or level is incremented in discrete steps, turning this type of test rather tedious. This work presents two different methods in which one of the two parameters is changed continuously to speed up the overall process and to deliver data for a 3D plot easily interpretable. The pros and cons of the two methods (log sweeps and level sweeps) are discussed and some practical examples shown.

Thu 15:40 Hudig Zaal Electro-acoustics

Simplification of subjective loudspeaker evaluation by means of dummy head recordings

Immo Ress[a], Rainer Huber[b] and Markus Meis[a]
[a]*Hoerzentrum Oldenburg GmbH;* [b]*HörTech gGmbH, Germany*

The perceptual evaluation of loudspeaker sound quality is time-consuming and cost-intensive. Therefore, simplified, less time-consuming evaluation methods are highly desirable. This presentation compares two subjective methods for relative sound quality evaluation of loudspeakers. In both methods, the perceived overall sound quality of a test loudspeaker is assessed in direct comparison to a reference loudspeaker, which had been identified as the best of a set of loudspeakers in a prior complete paired-comparison experiment. The two methods differ with respect to the way the sounds are presented to the listeners: directly by the test/reference loudspeakers, or indirectly by dummy head recordings of the loudspeakers, played back over headphones. The latter method represents a much faster procedure, since the test loudspeakers do not have to be replaced physically after each rating for each listener, but only once for the dummy head recordings. It has been found that subjective quality ratings obtained from 9 experienced listeners do not significantly differ between both methods more than between test and re-test measurements using the conventional direct loudspeaker presentation method. It is therefore concluded that subjective loudspeaker evaluation can be simplified using dummy head recordings without a loss of accuracy.

Thu 16:00 Hudig Zaal Electro-acoustics

Application of Calibrated Noise Recordings in Simulation and Auralisation of Sound Systems

Thomas Steinbrecher
Bose Professional Systems

The accurate assessment of sound systems with regards to various performance criteria, such as the Speech Transmission Index STI, requires the consideration of frequency-dependant background noise spectra. Background noise can have a significant influence on the available ratios of speech and noise levels and thus speech intelligibility. Another well-known effect of background noise is the masking of sound field components with lower level, such as echoes or even reverberation. A realistic auralisation of a sound system must therefore include background noise that is reproduced faithfully in level, spectrum and spatial attributes. The paper will describe the functionality and design of two stand-alone software applications developed in Matlab and implemented as stand-alone graphical user interfaces. One of the tools converts a binaural recording of background noise into a format suitable for a calibrated transaural playback device, the Bose Auditioner system. The second tool converts either standardized noise criteria such as NCB or RC curves or any arbitrary third-octave band noise spectra into calibrated diffuse noise for the same playback system. Both applications create the corresponding audio and text files for use in the simulation software Bose Modeler Design Program.

Thu 16:20 Hudig Zaal Electro-acoustics

Design of public address systems for larger sport stadiums

Alfred Schmitz[a] and Anselm Goertz[b]
[a]*TAC -Technische Akustik;* [b]*Audio & Acoustics Consulting Aachen*

In the last years the design of public address systems (PA-systems) for larger sport stadiums got increasing relevance. First of all the revised legal requirements concerning a minimum speech intelligibility on voice alarm systems has to be met. Furthermore a good quality of music performance and commercials must be given. This paper describes how PA-systems can be designed using modern computer simulation techniques considering the strong influence of room acoustical conditions. It will be shown which concepts can be applied dependent on technical boundaries and financial aspects. This paper will also focus on evaluation of STI-simulations with regard to national and international standards (EN 60489, VDE 0833-4, IEC 60268-16).

Thu 17:00 Hudig Zaal Electro-acoustics

The Interaction between Ultrasonic and Audio Waves in Air - an Experimental Study

Tobias Merkel
Technische Fachhochschule Berlin

When an ultrasonic sound beam is propagating in air a second audio wave will influence its sound velocity. This is caused by two effects: Firstly, the audio wave produces a motion of air particles. Because the same particles also transport the ultrasonic wave, an addition of particle motion can be observed. Secondly, there is a dependence between sound velocity and air density. Both effects have nearly the same magnitude even though the angle between the direction of propagation of both waves will only influence the first effect. The wave fields of an ultrasonic and an audio sound source were brought to the same volume. The ultrasonic wave was received after a short distance by a microphone. The phase shift of the received signal represents the variation in sound velocity. Although the expected effects are very small the measured results come very close to the findings of theory. The described effects will lead to a new kind of "virtualmicrophone: The ultrasonic wave will be sent over a long distance through the room and can be modulated by any kind of audio sources. No technical equipment is required at the place of sound reception through the ultrasonic carrier wave.

Thu 17:20 Hudig Zaal Electro-acoustics

Flat Panel Transducers Based on Electroactive Materials

Lutz Ehrig and Daniel Beer
Fraunhofer IDMT

Electroactive materials are a class of materials that change their geometry due to the presence of an electrical field. A well known effect that occurs in crystals, polarized ceramics and plastic materials is piezoelectricity. Since there is no center of symmetry and a preferred direction in such materials respectively an electrical voltage applied on the materials surface yields a deformation. The same macroscopical behaviour can be observed in relatively new materials called ferroelectrets. Voids in these cellular polymer foams are internally charged and form giant dipoles. The dipoles are deformed by an electrical field likewise in piezoelectric materials. Electroacoustic transducers based on electroactive materials may have a unique feature: driver and membrane are one part. Thus it is possible to build extremely simple, compact and flat transducers. Using the material by means of a thickness oscillator even the rearward air spring, causing a major part of problems for thin transducer constructions, can be avoided. In this work several approaches for transducers using electroactive materials are presented.

Orientation plans

Access and parking De Doelen

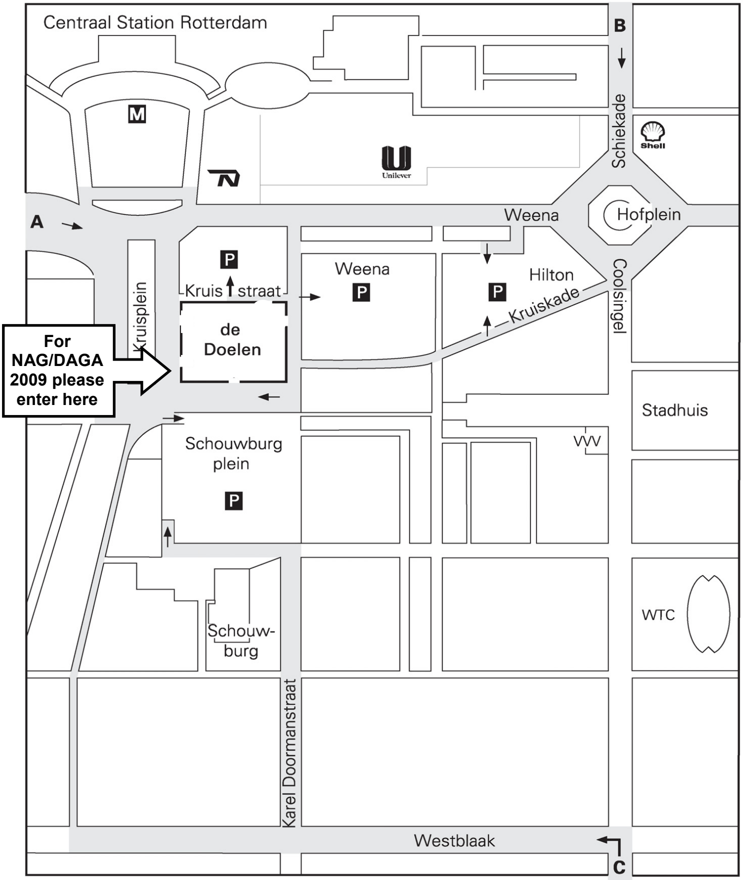
BY ROAD
From Amsterdam / Den Haag: A13 in the direction of Rotterdam, at Kleinpolderplein follow the signs in the direction of Centrum, at the second traffic light follow the signs for Euromast/Maastunnel. Take the right tunnel after the traffic light and go left at the next traffic light. You will now be on the Weena Boulevard. For further details, consult the street map next page (route A).

From Utrecht: A20 in the direction of Den Haag / Hoek van Holland, take the Rotterdam Centrum / Schiebroek / Hillegersberg exit, at the end of the exit follow the signs to Centrum (Schieweg / Schiekade). You will now be on Hofplein square. For further details, consult the street map next page (route B).

From Breda / Dordrecht: Take the right lane on the A16 motorway (follow Kralingen / Rotterdam Centrum). Cross the Van Brienenoordbrug and take the first exit (Rotterdam Centrum). Turn left at the end of the exit lane (roundabout in the direction of Centrum), and turn onto the Maasboulevard. Continue straight ahead (along de Boompjes). At Hotel Inntel (on your right-hand side), turn right onto the Schiedamsedijk. Continue until you have reached the Coolsingel-Westblaak intersection. Turn left onto the Westblaak and continue in the direction of the Hofplein. For further details, consult the street map next page (route C).

PARKING
There are several Parkings on walking distance from de Doelen:
1. Parking Schouwburgplein (consult the map on next page)
2. Parking Groothandelsgebouw
3. Parking Weena (consult the map on next page)
4. Parking Plaza
5. Parking Stad Rotterdam
6. Parking Bijenkorf

TRAIN
When you arrive by train at Rotterdam Central Station De Doelen is at 5 minutes’ walking distance (consult the street map on next page).



ROTTERDAM AIRPORT
Rotterdam is easily accessible by air. For more information please visit http://www.rotterdam-airport.nl. From the airport you can take bus number 50 towards Rotterdam Central Station, this takes about 20 minutes. From Rotterdam Central Station De Doelen is at a few minutes' walking distance. By taxi the distance is about 8 kilometres.

AMSTERDAM AIRPORT SCHIPHOL
Rotterdam also has a good accessibility via Amsterdam Airport Schiphol. For more information please visit http://www.schiphol.nl/Transport/Transport.htm. From the train station Schiphol (at the airport) you can take the train (about every 15 minutes), platform 5 or 6, to train station Rotterdam Central. This will take about 50 minutes. From Rotterdam Central Station De Doelen is at a few minutes' walking distance.

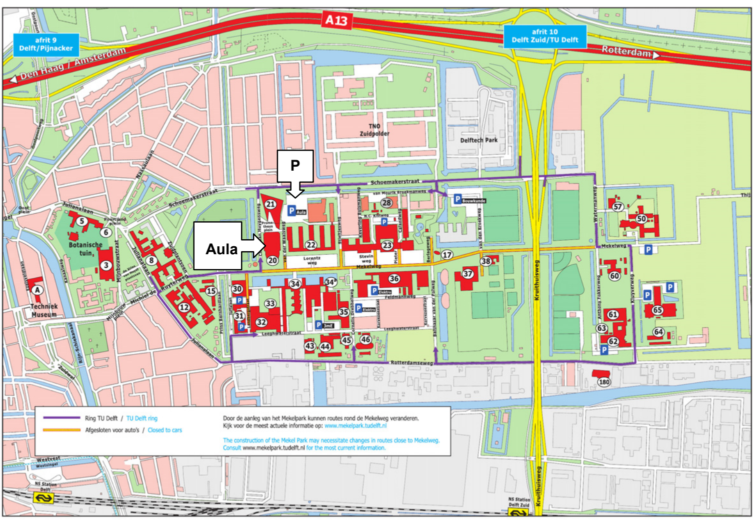
OTHER PUBLIC TRANSPORT
Metro: Central Station (starting point and terminus).
Tram: Kruisplein 4, 7, 8, 20, 21, 23 and 25.
Bus: Central Station, bus numbers 33, 38, 44, 48 and 49.
Tickets: In Rotterdam the public transport chipcard is the only means of payment for the subway. These chipcard can be bought at the VVV (tourist promotion office, near city hall) or RET sales and information points (Stationsplein Central Station) or Primera shops. Before you use the chipcard you have to upgrade the card. For tickets for trolley cars or busses the ticket strips are still used. These strips can also be bought at the above mentioned locations.

Note: Both the railway station and the station square are presently undergoing reconstruction. This will also affect visitors to de Doelen. Select "Stations Redevelopment and Facelift" on the left side of the page for more information on the following website http://www.dedoelen.nl/congresgebouw/pagina/166/practical_information

Access and parking TU Delft

(for precolloquium p. 41: Aula Congress Centre, Mekelweg 4, Delft, lecture room C)

BY ROAD
From the A13 you will enter the TU-area by taking the exit Delft/Pijnacker (9) or Delft Zuid/TU-wijk (10) - see also the streetmap below. The pre-colloquium will be in the Aula Congress Centre.

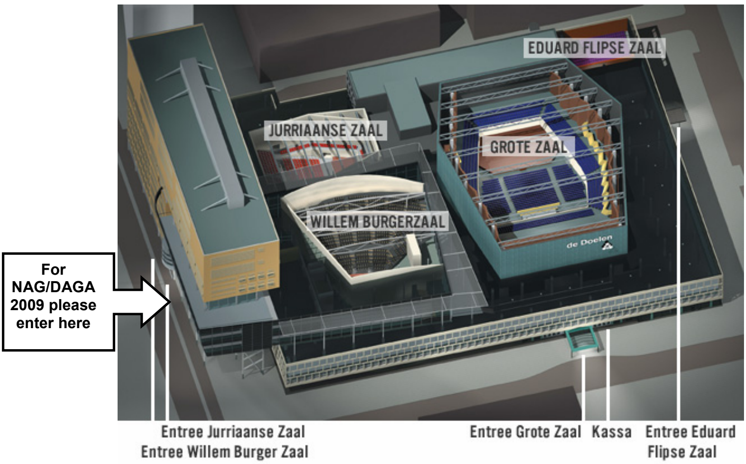


PARKING
For parking space see the 'P' on the streetmap.

TRAIN
From Rotterdam Central Station, which is a few minutes' walking distance from De Doelen, you can go by train, platform 8 or 9, to Central Station Delft. From there you can bus number 40 to Delft, to stop Balthasar van der Polweg. Then it is a 5 minutes' walk to the Aula Congress Centre of TU Delft. The travelling time in total in about 35 to 45 minutes. Also busses 69, 121 and 129 will bring you from Central Station Delft to the TU-area.

BUS
From De Doelen it is a few minutes' walk to Rotterdam Central Station. From there you can bus number 40 to Delft, to stop Balthasar van der Polweg. Then it is a 5 minutes' walk to the Aula Congress Centre of TU Delft. The travelling time in total is about 30 minutes.

Venue De Doelen



De Doelen comprises three groups of halls or complexes: the Grote Zaal (Main Hall, max. 1850 p.), the Willem Burger Zaal (max. 700 p.) and the Jurriaanse Zaal (max. 465 p.). Each complex has its own entrance, hall, foyer, and a number of smaller meeting spaces.

For the conference NAG/DAGA 2009 the complex of the Willem Burger Zaal and the Jurriaanse Zaal will be used.

WILLEM BURGER ZAAL COMPLEX

De Doelen Congress Centre (the Willem Burger Zaal complex) was opened less than ten years ago and specially designed for conferences and corporate events.

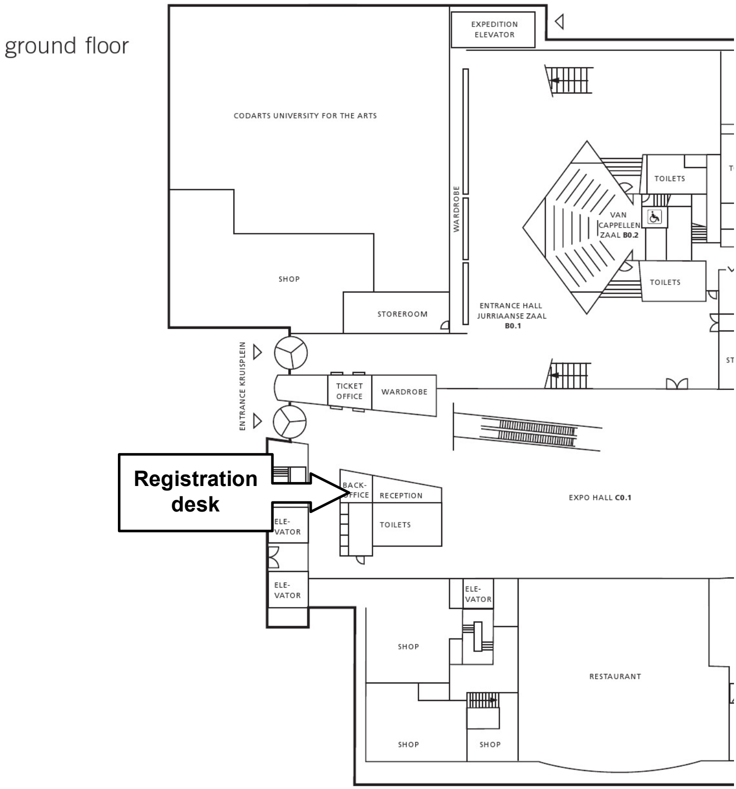
The centre has its own entrance (from the Kruisplein, consult the street map on page 350) as well as a central hall, an exhibition lobby, a foyer with a roof garden and ten smaller meeting rooms. At its heart is the Willem Burger Zaal, seating a maximum of 700 delegates (535 not including the balcony) in a fixed-rake, amphitheatre arrangement.

JURRIAANSE ZAAL COMPLEX

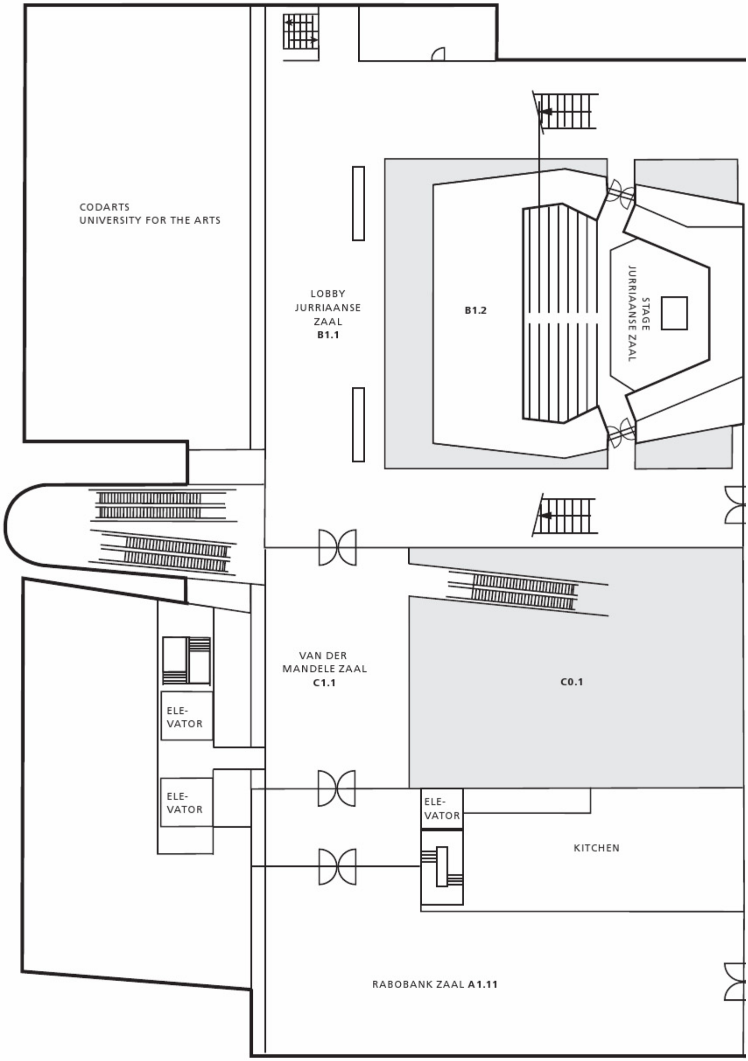
The Jurriaanse Zaal complex has a spacious foyer and a number of smaller spaces. The centre of this complex is the Jurriaanse Zaal itself.

Floor plans of De Doelen complex

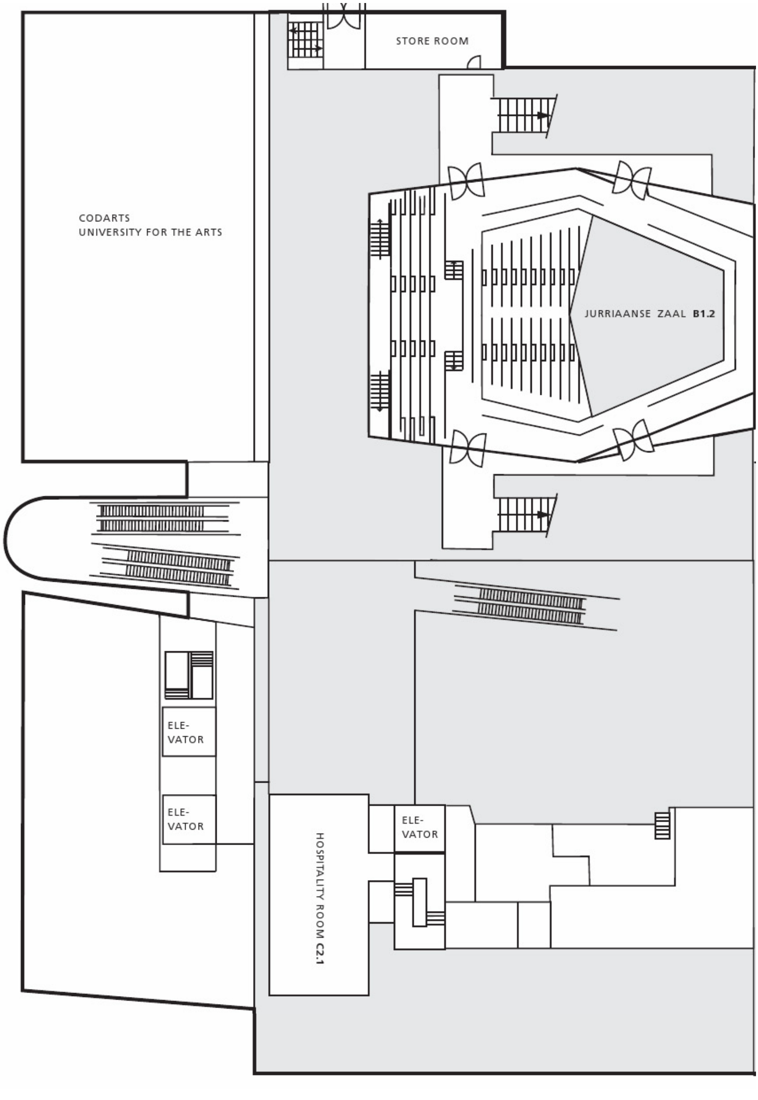
Plan of ground level:



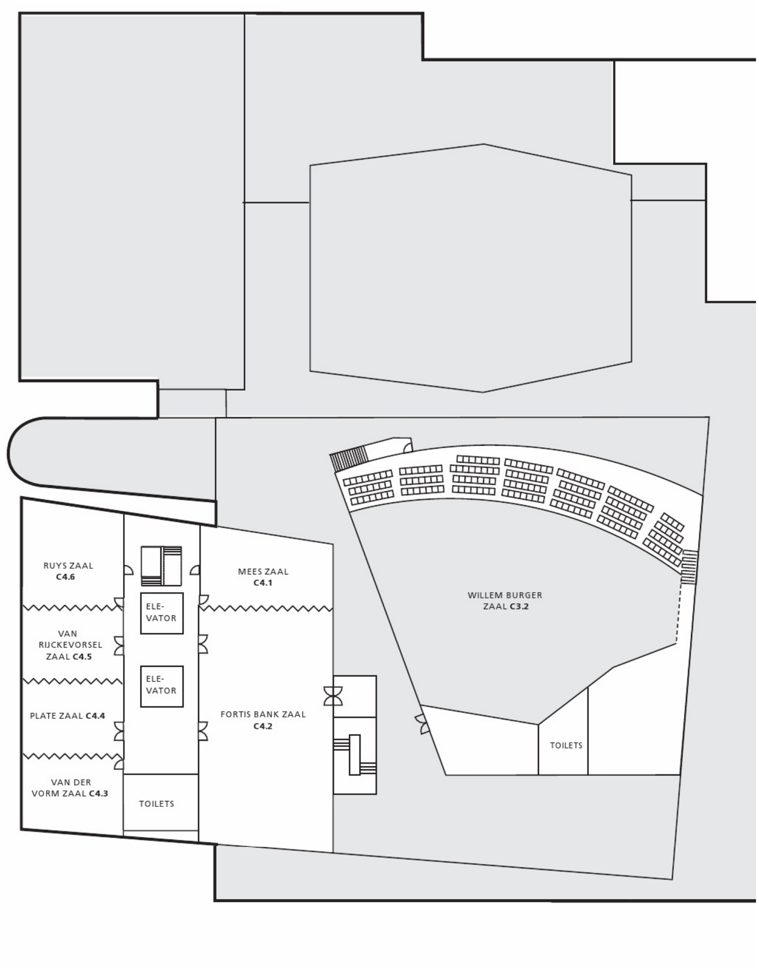
Plan of 1st floor



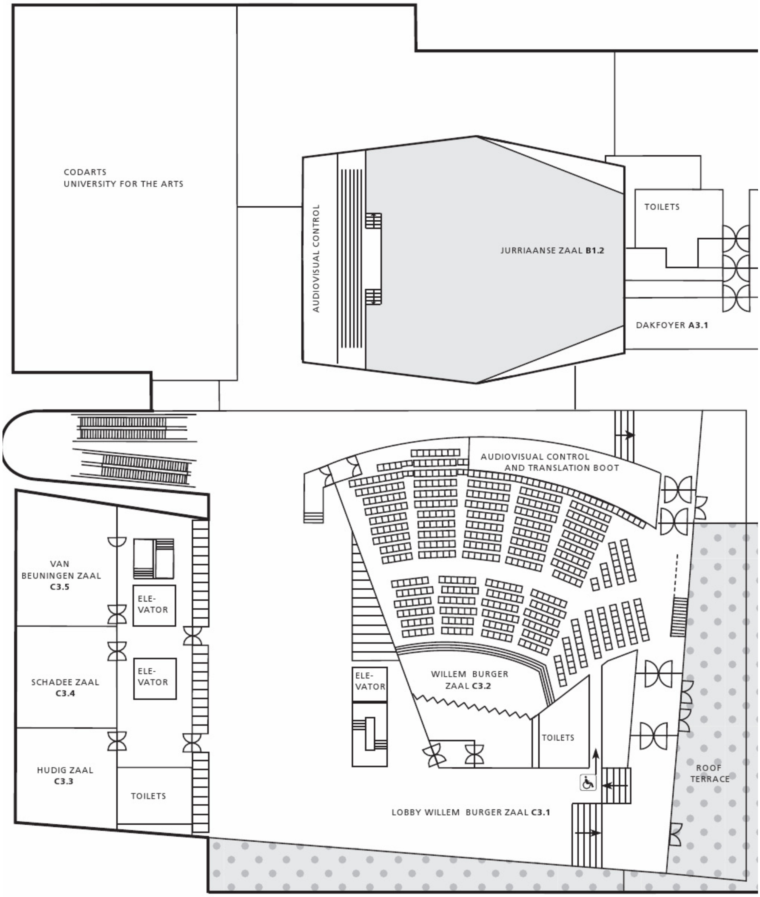
Plan of 2nd floor



Plan of 3rd floor



Plan of 4h floor



Lunchrooms, restaurants and bars

Street map of Rotterdam, vicinity of De Doelen (with lunchrooms, restaurants and bars):

LUNCHROOMS
You can find the location of these lunchrooms on the street map above:

1. Engels, Stationsplein 45, C.S. Kwartier, Rotterdam, grandeur of the past mixed with surprising modern elements
2. Floor, Schouwburgplein 28, 3012 CL, Rotterdam, modern style and a fairly chic atmosphere
3. DoelenCafé, Schouwburgplein 52, 3012 CL, Rotterdam, various dishes
4. Toaster, Pannekoekstraat 38a, 3011 LH, Rotterdam, simple lunchroom, specialised in toasted sandwiches
5. Dudok, Meent 88, Rotterdam, popular place for breakfast, lunch, dinner, drinks and socialising
6. Euromast, Parkhaven 20, 3016 GM, Rotterdam, great view, tasteful lunch

RESTAURANTS
You can find the location of these restaurants on the street map above.

near De Doelen:

1. Engels, Stationsplein 45, C.S. Kwartier, Rotterdam, grandeur of the past mixed with surprising modern elements

2. Floor, Schouwburgplein 28, 3012 CL, Rotterdam, modern style and a fairly chic atmosphere

7. Grand Palace, Schouwburgplein 54, 3012 CL, Rotterdam, Chinese restaurant

8. Sumo Japans Sushihouse, Westersingel 1, 3014 GM, Rotterdam, all you can eat sushi and grill

9. Maria, Mauritsweg 52, Rotterdam, Mediterranean, very good and surprising tapas

10. Restaurant 70 (Bilderberg Parkhotel), Westersingel 70, Rotterdam, culinary first-class for an affordable price

10 minutes' walking distance from De Doelen:

5. Dudok, Meent 88, Rotterdam, wide choice of Mediterranean dishes

11. Amarone, Meent 72a, 3011 JN Rotterdam (Michelinstar), Amarone's cuisine is French-Mediterranean with the emphasis on French cuisine

12. Napoli, Meent 81a, 3011 JG, Rotterdam, real Italian kitchen

13. Boudoir, Meent 65a, 3011 JD, Rotterdam, restaurant with a glamorous atmosphere, a large menu card and entertainment

14. Cinéma, Rodezand 36, 3011 AN, Rotterdam, restaurant and club in the heart of Rotterdam

15. O'Pazzo, Mariniersweg 90, 3011 NV, Rotterdam, original Italian (Napoli) kitchen

16. Coopvaert, Blaak 776, Rotterdam, Dutch restaurant

Westelijk Handelsterrein:

17. Rosso, Van Vollenhovenstraat 15, 3016 BE, Rotterdam, trendy French-Mediterranean kitchen, tasteful dishes

18. Smaak, Van Vollenhovenstraat 15, pakhuis 32, 3016 BE, Rotterdam, international kitchen

19. Biblio, Van Vollenhovenstraat 15c unit 26/27, 3016 BE, Rotterdam, perfect location for a nice dinner A lot of restaurants are to find in "Het Westelijk Handelsterrein", a recent renovated trade building. Entrance to restaurants is an indoor walking road.

Kop van Zuid:

20. Hotel New York, Koninginnenhoofd 1, Zuid, Rotterdam, a la carte restaurant, nice view

21. Humphrey's restaurant Rotterdam, Otto Reuchlinweg 1004, 3072 MD, Rotterdam, pleasant restaurant, principles: three courses, one price

22. Las Palmas, Wilhelminakade 330, 3072 AR, Rotterdam, characteristic fish restaurant

23. Rotterdam, Café Restaurant Club, Wilhelminakade 699, Zuid, Rotterdam, European oriented kitchen with characteristic view

Binnenweg-area:
24. Louis, Nieuwe Binnenweg 151, Rotterdam, very good French kitchen
25. Ari, Café, Nieuwe Binnenweg 142, Rotterdam, bar to eat and drink, good comfort, low price
26. Dizzy, ’s-Gravendijkwal 127-129, Rotterdam, bar to eat and drink (low price) / jazz café, life music at Tuesday 10.00pm
At the foot of the Erasmusbrug: 27. La Stanza, Van Vollenhovenstraat 19, Rotterdam, very good International kitchen
28. Kip, Van Vollenhovenstraat 25, Rotterdam, very good French kitchen
29. La Pizza, Scheepstimmermanslaan 21, Rotterdam, Italian, very good price/quality ratio

Centre:
30. Oliva, Witte de Withstraat 15a, Rotterdam, very good Italian food (no pizza)
31. Bazar, Wereld Eethuis, Witte de Withstraat 16, Rotterdam, African food, low price

BARS
You can find the location of these bars on the street map by number on page 359.

2. Floor, Schouwburgplein 28, Rotterdam, theatre cafe
3. DoelenCafé, Schouwburgplein 52, Rotterdam, every Thursday evening jam sessions
5. Dudok, Meent 88, 3011JP, Rotterdam, nice meeting point 13. Boudoir, Meent 65a, 3011 JD, Rotterdam, city bar, jazzy tunes and easy listening
14. Cinema, Rodezand 36, 3011 AN, Rotterdam, trendy bar/club with DJ’s
25. Ari, Nieuwe Binnenweg 142a, Rotterdam, nice atmosphere, diverse public
26. Jazz café Dizzy, ’s Gravendijkwal 127, 3021 EK, Rotterdam, live jazz bands perform here frequently
32. Melief Bender (Nieuwe Binnenweg), Oude Binnenweg 134b, Rotterdam, one of the oldest pubs in Rotterdam centre
33. Club Vie, Maasboulevard 300, Rotterdam, trendy club with glamorous touch, mix of jazz, Latin and house
34. Elit Rotterdam, Wijnhaven 3 a, 3011 WG, Rotterdam, cocktail bar with perfect jazz and soul tunes
35. De witte aap, Witte de Withstraat 78, 3012 BE, Rotterdam, always crowded, diverse music

Index of authors

Meetings during the conference

Please note: From Tuesday to Thursday, the meetings are scheduled for the lunch breaks. Particularly for participants in committee meetings, tickets for lunch boxes can be purchased at the registration desk.

Monday, March 23

09:00 - 12:00	DEGA-Vorstandsrat	Van Beuningen Zaal
17:00 - 19:00	DEGA-Mitgliederversammlung (DEGA members general assembly)	Willem Burger Zaal

Tuesday, March 24

12:40 - 14:00	EAA/DEGA student meeting	Van Beuningen Zaal
12:40 - 14:00	DEGA-Hochschulbeirat	Hudig Zaal

Wednesday, March 25

12:40 - 14:00	NAG members general assembly	Van Beuningen Zaal
12:40 - 14:00	DEGA-Fachausschuss Bau- und Raumakustik	Fortis Bank Zaal
12:40 - 14:00	DEGA-Fachausschuss Lärm: Wirkungen und Schutz	Hudig Zaal
12:40 - 14:00	DEGA-Fachausschuss Lehre der Akustik	Plate Zaal
12:40 - 14:00	DEGA-Fachausschuss Musikalische Akustik	Mees Zaal
12:40 - 14:00	DEGA-Fachausschuss Sprachakustik	Schadee Zaal
12:40 - 14:00	DEGA-Fachausschuss Ultraschall	Ruys Zaal

Thursday, March 26

12:40 - 14:00	DEGA-Fachausschuss Elektroakustik	Hudig Zaal
12:40 - 14:00	DEGA-Fachausschuss Fahrzeugakustik	Van Beuningen Zaal
12:40 - 14:00	DEGA-Fachausschuss Hörakustik	Plate Zaal
12:40 - 14:00	DEGA-Fachausschuss Physikalische Akustik	Van der Vorm Zaal
12:40 - 14:00	DAGA-Beirat	Van Rijckevorsel Zaal

NAG/DAGA 2009 - registration form (without presentation)

Title, first and family name:______________________________

Address:______________________________

____

E-mail:______________________________

Registration fee
Please mark the category according to table of fees on page 33:

Category: ❒ 1 ❒ 2 ❒ 3 ❒ 4

❒ Dinner ❒ Concert ❒ Pre-coll. 'Ports'" ❒ Pre-coll. "Array'"

Excursions: ❒ Harbour ❒ Architecture ❒ Delft Blue

(Fees for social events are listed on www.nag-daga.nl)

Payment

❒ **Credit card**

❒ Eurocard/Mastercard ❒ Visa ❒ American Express

Card owner:____________________ Expiry date:____________________

Card number:______________________________

❒ **Bank transfer** to the account of Stichting NAG/DAGA 2009 in the Netherlands, number: 535560958, ABN AMRO. For payments from foreign countries please use IBAN: NL15ABNA0535560958 and BIC: ABNANL2A. Please indicate the name of the participant.

❒ **innerhalb von Deutschland:** Ich ermächtige die DEGA e.V., den von mir zu entrichtenden Beitrag zu Lasten meines Kontos

Konto-Nummer:______________________________

bei:____________________ Bankleitzahl:____________________

mittels Lastschrift einzuziehen.

*Date and Signature:*______________________________

Fill in this page and send to:
DEGA e.V., Headquarter
Voltastrasse 5, Building 10-6
13355 Berlin, Germany Fax: +49 (0)30 46 06 94-70