AES 21th International Conference Preliminary Program

Architectural Acoustics and Sound Reinforcement

2002 June 1–3, St. Petersburg, Russia

Technical Sessions

Saturday, June 1

11:20 am–2:40 pm

SESSION 1: INVITED PAPERS

1-1 Architectural Acoustics in Russia
Michael Lannie
Research Institute for TV and Radio, Moscow, Russia

A historical review of the architectural acoustics in Russia is presented. Three periods are explored: 1930 to 1949, 1950 to 1989, and 1990 to the present. Three main topics are reviewed for each period: scientific studies, measurement techniques, and practical works on acoustic consulting. Attention is primarily paid to various theaters, concert halls, studios, cinemas, and sport halls which have been designed by the Russian acousticians. A detailed bibliography on the subject is given as well.

1-2 Some Rules and Methods for Creation of Surround Sound
Andrzej Czyzewski and *Piotr Odya*
Technical University of Gdansk, Gdansk, Poland

The problem of selecting adequate surround sound live recording and reproduction methods still exists. Alternative methods of organizing this process are discussed. Some experimental recording sessions employing the 5.1 format were made with the use of various miking techniques and the convolution-based multichannel audio processing algorithm. The results were submitted to some subjective assessments and then compared. Conclusions resulting from performed experiments are discussed.

1-3 Cinematographic Audio Equipment Development in Russia
K. G. Ershov
St. Petersburg State University of Cinema and Television, St. Petersburg, Russia

This paper discusses some issues pertaining to cinematographic sound engineering audio equipment development in Russia, particularly work being done at the Saint Petersburg State University of Cinema and Television. The domestic development of sound track recording and playing started in 1926 and was completed in two years. The first cinema featuring sound track equipment was opened in Leningrad in 1929. Later, many cinemas were equipped with sound equipment. This required special research devoted to the recording, throughout the country, of sound tracks (photographic) at film studios, design work, and industrial production of all requisite equipment (microphones, mixing panels, recorders, etc.), as well as sound playing systems installed at cinemas (sound units in projectors, electronic amplifiers, cinema high-power loudspeakers). Finally, many architectural acoustics issues had to be resolved to meet the required standards in cinema halls.

1-4 Stereo Boundary Microphone Technique for Live Staged Performances: Development of a New Mid/Side Boundary Stage Microphone
Ronald Streicher
Pacific Audio-Visual Enterprises, Pasadena, CA, USA

Given the inherent restrictions on microphone placement when producing stereophonic recordings and/or sound reinforcement of live staged performances, it is often difficult to achieve an accurate aural image and sense of movement across the stage. For many years, the use of spaced floor-mounted microphones has been the norm for these performances, resulting in acoustic interference anomalies that compromise the quality of these pickups. This presentation will demonstrate that by employing the mid/side microphone technique in a floor-mounted “boundary array,” an accurate and articulate stage image can be achieved without any of the comb-filtering or other phasing problems inherent to spaced microphones.

2:40 pm–3:40 pm

SESSION 2: ARCHITECTURAL ACOUSTICS, PART 1

2-1 Acoustic Design of the Great Philharmonic Hall in the Moscow International Music Dome (MIMD)
Wolfgang Ahnert, ADA Acoustic Design Ahnert, Berlin, Germany
Lev Bonsov and *Christofor Shirjetdk*i, Research Institute of Building Physics, Moscow, Russia (Invited)

In Moscow, the construction of an International Music Dome incorporating three halls is nearing completion. The Great Philharmonic Hall of the Dome seats 1800 listeners and excels in originality and outstanding architectural design. It is mainly meant for the performance of classical symphonic concerts without electronic support of the performers. A special feature of the hall is that it can be used for three different concert forms (organ, symphonic, and chamber) requiring different performance qualities.

2-2 The New Symphony Hall in Las Palmas, Gran Canaria
Jan Voetmann and *Lise-Lotte Tjellesen*
DELTA Acoustics & Vibration, Kgs. Lyngby, Denmark

One of the world's most beautiful new symphony halls is the Auditorio Alfredo Kraus in Las Palmas, Gran Canaria, Canary Islands, inaugurated in 1999. For serious reasons the hall was completed without the collaboration of the original acoustical consultant. Shortly after the opening of the hall, the acoustics had problems primarily in terms of too long a reverberation time. A team of Spanish and Danish consultants were brought in to work in close collaboration with the architects to find a solution. The solution included a number of measures: room acoustics, in order to bring the reverberation time down and increase the projection of sound to the audience; and special measures for the members of the orchestras, in order to make them feel more comfortable on the (large) stage. In the autumn of 2001, the alterations were completed, and the new acoustics were very favorably received by the public, the owners, and the orchestras. New measures to make the hall function for modern electronic amplified music are under preparation. Considerations and measurements are presented.

2-3 A New Criterion for Concert Hall Loudness Evaluation
Shuoxian Wu
South China University of Technology, Guangzhou, People's Republic of China

Loudness is one of the most essential parameters for assessing the acoustical quality of an auditorium. Because of the lack of an authentic criterion, how to evaluate the loudness in concert halls remains unresolved. In this paper LpF, the mean forte sound-pressure level of tutti sound, is suggested as a criterion to describe the loudness in a hall. The prediction procedure of the LpF value distribution in a hall is described. Comparison between predicted and measured LpF levels is given. Tentative optimum and allowable LpF values for concert halls are also discussed.

4:00 pm–5:20 pm

SESSION 3: SOUND REINFORCEMENT, PART 1

3-1 Dual-Range Horn with Acoustic Crossover
Marshall Buck
Gibson Labs, Los Angeles, CA, USA (Invited)

A new approach has been developed to combine midrange and high-frequency sound into the throat of a horn designed for sound reinforcement. An acoustic low-pass filter element is interposed between the lower frequency passage and the higher frequency passage, so that a smooth combination of the two frequency bands is achieved at the entrance to the horn bell. Thus each frequency band has nearly identical dispersion, and the two sources have equal delay.

3-2 Modifying STI to Better Reflect Subjective Impression
Peter Mapp
Peter Mapp Associates, Colchester,Essex, UK

The Speech Transmission Index (STI) is becoming the universally accepted method for measuring the potential intelligibility of a sound system. However, a number of operating conditions and sound system characteristics seem not to be taken into account by current STI techniques. This paper highlights a number of these conditions and discusses possible modifications to the STI in order to improve its potential use and accuracy.

3-3 Loudspeaker Array Simulator with Coordinated Positioning of Elements
Arkady Gloukhov
Consultant, St. Petersburg, Russia

A high throughput simulator for loudspeaker array modeling and optimization has been developed. The array model simulates mechanical links between cabinets. The array baffle surface is approximated by two second-order equations. Splaying of an array is performed by variation of curvature coefficients. Displacement of the entire array and reorientation are performed by moving and aiming a single cabinet. The simulator automatically finds coordinates, splaying and aiming angles using direct sound coverage parameters as a target function. Interference pattern calculation is used in automatic optimization of delays for comb filtering reduction. The rigging simulation module calculates center of gravity location and mechanical loads.

3-4 Distributed Sound Reinforcement for Multiple Talker Locations
Michael Pincus
Acentech Inc., Cambridge, MA, USA

This paper describes techniques used for designing and implementing a distributed sound system for the historic renovation of a chapel in Concord, New Hampshire. The project is unusual because of the chapel's unique seating arrangement and multiple talker locations, several of which are used simultaneously during an event. The system is designed to help the audience localize the talkers. The paper compares the system with a typical distributed system and shows how today's digital signal processors allow complicated sound reinforcement techniques to be implemented easily and cost effectively, even for small-to medium-sized projects.

Sunday, June 2

9:00 am–10:40 am
SESSION 4: SOUND REINFORCEMENT, PART 2

4-1 Loudspeaker Placement for Enhanced Monitor Sound Field and Increased Performer Source Positioning
Thomas Lagö
Jönköping University, Jönköping, Sweden (Invited)

When handling the electroacoustics in a church, the reverberation time often is large enough to make the returned and delayed sound field irritating and confusing to the performer (typically a singer or talker). It can be compensated for by using monitor loudspeakers placed on stage, facing the performer. The sound field will be reflected in the wall behind the performer and will decrease intelligibility for the audience because of these reflections. If the wall behind the performer is soft or absorbent, it is not a problem, but in many churches and auditoriums the podium and the wall behind the performer are hard. By mounting loudspeakers on the wall facing the audience, the monitoring aspect can be resolved and other advantages can be achieved at the same time. Since the performer's sound field is not very loud, the position is often given by the loudspeaker system, thus negatively affecting the localization of the performer. This will result in a lower intelligibility, especially for people with decreased spatial hearing. By using loudspeakers behind the performer, a first and well-defined wave front is created. The next set of loudspeakers are then adjusted to stay within the 10-dB sound level that is stipulated by Haas (the so-called Haas Effect), which states that the second wave front will not contribute to direction given that the sound field stays within about 25 ms and 10 dB, measured at the listeners' position relative to the first wave front. This effect can be achieved because the sound level and time delay stays within certain boundaries at most positions in the auditorium. This idea has resulted in an approved patent and implementation in the Bankeryd's Missionskyrka Church in Sweden. A large measurement series, consisting of several hundred measurements, quantifying the effect of this innovative placement of the loudspeakers was performed. This paper gives background information about the sound challenges normally found in many churches and auditoriums and how they can be handled by a different loudspeaker placement. The paper also describes the results accomplished and other possible side effects that could occur.

4-2 Speech Reinforcement Inside Vehicles
Alfonso Ortega, Eduardo Lleida, and *Enrique Masgrau*
University of Zaragoza, Zaragoza, Spain

Improving oral communication inside vehicles is the goal of a cabin car communication system (CCCS). Communication can be difficult because of the distance among passengers, lack of visual contact between speakers, high level of noise, and many other factors. To achieve speech reinforcement, CCCS makes use of a set of microphones to pick up the speech of each passenger, then it amplifies these signals and plays them back through the car audio loudspeaker system. This system presents two main problems: electroacoustic coupling and noise amplification. To overcome these problems, CCCS makes use of an acoustic echo-cancellation system and a noise reduction stage. A brief description of the system and some results are provided.

4-3 Initial Investigation of an Air/Air Interface as an Acoustic Boundary for Sound Control at Outdoor Events
David Carugo
University of Limerick, Newbridge, Ireland

A basic ray model is used to examine refraction and reflection of sound from an air/air interface. The results of this treatment are presented and examined for the possibility of using a heated air mass, forming an air/air interface, as an acoustic boundary at outdoor events to reduce environmental noise pollution from sound leakage from the event.

4-4 Implementation of Intelligibility Algorithms into EASE 4.0
Wolfgang Ahnert, Stefan Feistel, and *Oliver Schmitz*
ADA Acoustic Design Ahnert, Berlin, Germany

In acoustic simulation programs very different algorithms are used to calculate the intelligibility of speech and music. To get results there are postprocessed fixed energy ratios as well as time-dependent impulse responses. In EASE 4.0 there now are all the usual intelligibility measures, derived by means of simulated high-resolution data or by applying statistical estimations. This paper compares all these measures and methods, such as STI, AL cons , clarity, definition, etc., by means of the results obtained within a common computer model. Finally, recommendations are given for applications which should be distinguished.

4-5 Practical Considerations for Field Deployment of Modular Line Array Systems
David Scheirman
JBL Professional, Northridge, CA, USA

This paper begins with a review of market trends leading to the availability and proliferation of modular multiway line arrays. Variously referred to as line arrays, curved arrays, line-source arrays or vertical arrays, such systems present opportunities to reliably predict cover age patterns and average level in the intended audience area. They can also present unique challenges for field deployment which are influenced by the mechanical design. Such systems provide relatively narrow vertical coverage patterns and increased apparent gain at distance. These acoustical characteristics can be used to great benefit when the system is properly configured. This paper reviews the various individual box design attributes that influence array performance. It then uses a case study approach to examine the practical aspects of deploying temporary systems in performance spaces and discusses various design tradeoffs encountered when using such systems in different types of venues.

9:00 am–10:40 am
SESSION 5: POSTERS, PART 1

5-1 Measurements of Scattering Coefficient of Surfaces in a Reverberation Room
Jin Jeon, Byung Kwon Lee, and *Sung Chan Lee*
Hanyang University, Seoul, Republic of Korea

Scattering of surface materials is one of the most important aspects for evaluating the acoustics of concert halls. One of the methods that can reduce the errors in calculating the reverberation time and other acoustic parameters through computer modeling is to calculate the scattering coefficient of surface materials. However, so far, no objective and reliable method for measuring scattering coefficient has been suggested. In this situation, the ISO has suggested a method of measuring the random-incidence scattering coefficient of surfaces in a diffuse field; whereas the AES has introduced a method of directional incidence in a free field. In this study the scattering coefficients of different hemispheres were measured by using the ISO method in a 1:10 reverberation chamber.

5-2 Nonlinear Model of Condenser Microphone Capsule
Shakir Vakhitov
Mikrofon-M Ltd., St. Petersburg, Russia

Physical factors that cause nonlinear distortion at different parts of condenser microphones are analyzed. Detailed mathematical models of these phenomena determine dependence of distortions on acoustical, mechanical, constructive, and electrical parameters. The nonlinear models allow one to calculate fairly accurately frequency- and level-dependent sensitivity and harmonic distortion for capsules of different condenser microphones. The systematic nonlinear model of a capsule is obtained by incorporation and approximations of the most important factors. Comparison of the calculation and measurement results is presented. Recommendations for nonlinear distortion reduction in the process of microphone design are given.

5-3 Analysis of Deformation on Coated Paperboard during a Scratch Test by AE
Shigekazu Suzuki, Yasushi Fukuzawa, and *Shigeru Nagasawa,* Nagaoka University of Technology, Nagaoka, Japan; *Hideaki Sakayori*, Koutou Carving Co.
and *Isamu Katayama*, Katayama Steel Rule Die Co. Ltd., Shinjukuk, Tokyo, Japan

Coated paperboard, an anisotropic composite material, is used as packing material. It is important to machine test the deformation and fracture behavior of paperboard for practical work efficiency with little loss. In this study deformation and fracture behavior of coated paperboard during the scratch test were investigated with an acoustic emission (AE) test. Acoustic emission signals occurred mainly at each layer of exfoliation. The deformation and fracture behavior of several kinds of coated papers could be observed by an AE sensor during the scratch test.

5-4 Analysis of Sound Radiated by Paperboard Die Cutting
*Akira Sadamoto,*Tsukuba College of Technology, Tsukuba, Japan
Takashi Yamaguchi, Shigeru Nagasawa, Yasushi Fukuzawa, Nagaoka University of Technology, Nagaoka, Japan
Daishiro Yamaguchi, and *Isamu Katayama,* Katayama Steel Rule Die Co. Ltd., Shinjukuk, Tokyo, Japan

This paper reports on the radiated sound that occurs under paperboard die cutting. For increasing productivity and reducing the operator's task, any kind of automatic technique for detecting cutting conditions is required. To solve this problem, the sound radiated in the cutting process was analyzed. Several sounds were measured by varying several conditions: cutting force, paper thickness, paper direction, and blade tip width. Sound pressure level (SPL) denoted obvious differences in each condition. It was confirmed that the cutting condition could be diagnosed by seeing the SPL.

5-5 FSQ is for "Fast Sound Quality": New Techniques for Assessment of Sound Quality of a Car Audio System
D. Svoboda
Acoustics Center of the Broadcasting and Electroacoustics Department, Moscow Technical University of Communications and Informatics, Moscow, Russia

A new method for subjective-statistical expertise of sound quality, dubbed fast sound quality (FSQ), is primarily intended for sound quality judging at car audio competitions, providing reliable scores in shorter times compared with traditional IASCA-based techniques. Testing software is compiled into a CD with a total sounding time of less than 15 minutes. FSQ was used successfully in 12 Russian competitions in 2001. This new method can also be used for assessment of home high-fidelity and high-end sound systems. FSQ and its accompanying software were introduced at an AES Moscow Section meeting and then written up in *AvtoZvuk*, the leading Russian car audio magazine, and *Metrology and Metering Techniques Communications* magazine. The new FSQ method was developed at the Acoustics Center, Moscow Technical University of Communications and Informatics Department.

5-6 A Problem of Distortions at Electroacoustic Conversion
Alexander Gaidarov
Andreev Acoustical Institute, Moscow, Russia

The most general principle of not distorting transformation of information signals of the arbitrary shape is the scale copying by an output signal input with possible delay of an output signal on constant time for all signal components. However, a number of views and interpretations in basic concepts (fundamentals of physics in general and electroacoustics in particular and sufficiency used in the substantiation of amplitude-spectral representation of quality of conversion) demand major rethinking. In particular: a) Parameters of motion of any body— speed and acceleration are determined in mechanics, as a derivative from already given displacement. Substantially in the dynamics of Newton, the external force generates acceleration of mass. Speed and displacement are determined by a series integrating the acceleration of time. Thus, the constants of integration complementing an original signal are foregone. The values of constants depend on the condition of the oscillating system to the initial moment of arrival of the next signal, which upsets the invariance of a system concerning time. b) The definition of intermodulation distortions in a nomenclature of IEC, omissions of physical features of parametric distortions at Doppler intermodulation, is groundless for nonlinear distortions (sometimes referred to as subspecies). Presented is a revision and refinement of views along the path that requires veracity of sound reproduction.

11:00 am–12:40 pm

SESSION 6: ROOM AURALIZATION

6-1 Measurements of Church Impulse Responses Using a Circular Microphone Array for Natural Spatial Reproduction of a Choir Concert Recording

Diemer de Vries, Delft University of Technology, Delft, The Netherlands
Sandra Brix, Fraunhofer Institute IIS/AEMT, Erlangen, Germany
and *Edo Maria Hulsebos,* Delft University of Technology, Delft, The Netherlands

In a church in Weimar, Germany, a 12-track recording was made of a choir concert. Instead of trying to include the acoustics of the church in the recording, the impulse responses were recorded separately using a new measurement technique in which a microphone slowly moves along a circle. The microphone measures the pressure as well as the velocity response, enabling discrimination between wave field components from different directions and extrapolation of the data to other virtual microphone positions. This way the responses can be estimated at all listening places of interest and convolved with the "dry" recording of the singers' voices for reproduction by wave field synthesis. The measurements were done within the framework of the CAR-ROUSO project.

6-2 The Realisation of Ambisonics and Ambiophonics Listening Room "Arlecchino" for Car Sound Systems Evaluation

Lamberto Tronchin, Valerio Tarabusi, and *Alessandro Giusto,*
University of Bologna, Bologna, Italy

Ambisonics playback is a very promising technique in sound field reconstruction. Realistic acoustical environments can be reproduced supplying the listener with the feeling of being part of the reproduced acoustical field. The listener's brain can feel the illusion of the reproduced sound scene as if it is there. In this paper the technique is analyzed and the main issue concerning Ambiophonics and B format are revised. In order to test the fidelity of music and noise reconstruction, some B format tracks have been created in ancient churches and notable acoustical places along with the transfer function of the measured acoustical environments. A dedicated room, called *Arlecchino*, has been simulated, calibrated, designed, and finally realized for the purpose of Ambiophonics listening tests, especially for car audio systems improvements. Eight diffusers were implemented in the room in order to reproduce the eight-channel B format signal. The B format tracks were then tested in the listening room. Simulation of the measured acoustical fields was obtained through convolution of "dry" anechoic musical pieces recorded in B format. The main steps for designing the listening room are illustrated, and the Ambisonics–Ambiophonics listening test results are presented.

6-3 An Efficient Auralization of Edge Diffraction

Tapio Lokki, Helsinki University of Technology, Espoo, Finland
Peter Svensson, Norwegian University of Science and Technology, Trondheim, Norway
and *Lauri Savioja,* Helsinki University of Technology, Espoo, Finland

Principles and implementation of efficient auralization of edge diffraction are presented. The calculation principle for the impulse response from an edge is reviewed. The technique has been integrated into an acoustic modeling system which is based on the image-source method. For auralization purposes a low-order digital filter for each diffracting edge was designed, which efficiently implements the diffraction phenomenon and is suitable for parametric auralization. Finally, a comparison of auralized impulse responses with and without diffraction is presented. The case study was made in a simple room geometry containing occluders.

6-4 New Digital Filter Techniques for Room Response Modeling

Tuomas Paatero and *Matti Karjalainen*
Helsinki University of Technology, Espoo, Finland

Computationally efficient modeling of room responses is needed in many audio and acoustics applications, such as auralization, artificial reverberation, and equalization of loudspeaker–room responses for sound reproduction. Digital filtering is an efficient means for such modeling, particularly in real-time implementations. This paper discusses new DSP-based methods to model measured room responses. One technique is Kautz filtering, which is an attractive method, especially at low frequencies where the modal density is relatively low. Another approach is modeling dense modal patterns by filterbanks that approximate the response in a perceptually meaningful way. The optimization of filter parameters of the models is discussed; achieved performance is shown by example cases; and applications are briefly reviewed.

6-5 Multichannel Simulation and Reproduction of Virtual Acoustic Environments with Walls of Unequal Absorption

Kenji Suzuki and *William Martens*
University of Aizu, Aizuwakamatsu-shi, Japan

The goal of this research project has been to determine whether a simple 3-D model for multiloudspeaker simulation of room reverberation could produce identifiable differences in room geometry. This simple, image-model-based simulation was designed to produce distinctive-sounding results as the material was varied on each of the six walls of a modeled rectangular room. A realistic-sounding wall reflection simulation was developed and submitted to blind listening experiments designed to test whether listeners could determine which one of five walls had been eliminated from the simulation. Though listeners were not particularly good at this identification task, they were able to consistently distinguish between the spatial images associated with these five cases (five room geometries).

11:00 am–12:40 pm

SESSION 7: POSTERS, PART 2

7-1 Algorithms of Digital Audio Data Compression: Standards, Problems, and Perspectives of Development
*Yurii Kowalgui*n and *Dhammika Priyadarsana Yatagama Gamage*
St. Petersburg State University of Telecommunications, St. Petersburg, Russia

The main specifications of the algorithms of digital audio data compression in MPEG and ATSC standards, which include the newest hybrid methods that combine the advantages of parametric and subband coding, are considered. On the basis of analysis, a generalized structural diagram of a coder with digital audio data compression is given. The procedure of the audio data processing in the blocks of time–frequency segmentation, entropy coding, and psychoacoustic analysis is discussed. Special attention is given to the procedure of audio data processing in the psychoacoustic analysis block and directions for implementation.

7-2 Dynamic Behavior of the Nonholonomic Mechanical Systems in Changeable Working Regime
Miodrag Zlokolica, Bogdan Sovilj, and *Vladimir Miskov*
University of Novi Sad, Novi Sad, Yugoslavia

In many complex mechanical systems, the transmissions with nonholonomic characteristics as transmitters with changeable transmission ratios are found. This type of transmission has the connection of a differential character between transmission elements. A nonholonomic mechanical system can be recognized as variable-speed drives, contained in many complex systems of modern techniques. The aim of this paper is to give one approach to the dynamical description of a general example of transmissions with nonholonomic characteristics and to estimate the stability of the working system with a changeable working regime. For the dynamic description of the mechanical nonholonomic system, Appell's differential equations are used. By numerically solving the differential equations of movement, the answer concerning the working stability as well as the dynamical and kinematics behavior of the observed system is provided. The obtained results will serve as one of the constraints in choosing optimal parameters in the synthesis of power ansmissions.

7-3 Efficiency of Noise Barriers with Non-Straight Edge Profiles
Henrik Sandqvist
Royal Institute of Technology, Stockholm, Sweden

The straight edge of a noise barrier in some areas behind a screen causes noise levels to increase instead of decrease. Still, the noise barriers today are commonly built with straight edges. An exact analytical solution describing the sound field for straight-edge noise barriers as well as barriers with periodical-edge profiles has been previously derived. It has also been shown that for a given frequency, there is an optimum length of the period of the edge profile. Using these solutions, how to construct an efficient-edge profile for a broadband signal with a given spectrum was examined.

7-4 Acoustic Normal Mode Analysis for Coupled Rooms
Yuezhe Zhao and *Shuoxian Wu*
South China University of Technology, Guangzhou, People's Republic of China

A finite-element method is presented for studying the acoustic transmission function and acoustic impulse response of lightly damped rooms. It is shown that the computer model successfully predicts the effects of different source–receiver locations on the amplitude spectrum. Also, the model solution does capture the effects of direct sound and reflections. As an example, a fully three-dimensional rectangular room has been modeled with details.

7-5 Acoustical Design of an Electrical Emergency Plant to Reduce Outdoor Noise Level
Evgueni Podzharov, University of Guadalajara, Guadalajara, Mexico
Francisco de la Mora Galvez, University Panamericana, Guadalajara, Mexico
and *Lioudmila Oleinikova,* University of Guadalajara, Guadalajara, Mexico

An analysis of noise transmission in an electrical emergency plant was done using the statistical energy analysis method. This analysis permitted evaluation of different measures and materials to reduce noise level. A two-inch-thick layer of fiber glass was selected as coating for the walls and ceiling and silencers at the inlet of air and at the outlet of engine gases to reduce indoor and outdoor noise levels. The noise measurement showed that the noise level was considerably reduced after implementation of these measures. The reduction of noise was 7 to 8 dB(A) inside the plant, 19 dB(A) at 10-m distance from the plant, and 23 dB(A) at 15-m distance from the plant.

1:40 pm–3:40 pm

SESSION 8: PSYCHOACOUSTICS

8-1 A Special Form of Noise Reduction
Ronald Aarts, Philips Research, Eindhoven, The Netherland;
H. Greten, Greten Raadgevende Ingenieurs, Eindhoven, The Netherlands
Peter Swarte, P.A.S. Electroacoustics, Eindhoven, The Netherlands

Pop music reproduction or reinforcement in the entertainment world, such as at dance clubs or *poppodia* on a very high sound-pressure level, is highly appreciated by the so-called target group. However for the neighbors, it can be very annoying, especially when these music sessions take place during the night. Poor sound insulation creates an inadmissible sound emission level in, e.g., bedrooms. Noise reduction methods of a constructional nature are in most cases very expensive. Two methods of active noise reduction were tried out in the sound system of a pop platform in the Netherlands: one by anti-sound and the other based on the phenomenon of the missing fundamental. Both experiments and the results are discussed. The latter experiment is called dormant bass (DB).

8-2 Improving Perceptual Coding of Wideband Audio Signal when Taking into Consideration of Temporal Masking
Alexander Zakharenko and *Yurii Kowalguin*
St. Petersburg State University of Communications, St. Petersburg, Russia

All existing coding systems do not take into consideration the temporal masking phenomenon. However, it plays a vital part when decreasing bit rate. There are two kinds of temporal masking: backward and forward masking. Although many studies of backward masking have been published, the phenomenon is poorly understood; the highly practiced subjects often show little or no backward masking. Therefore in this paper forward masking applied to bitrate reduction is considered. This paper describes a new high-quality audio coding system based on the MPEG ISO/IEC 11172-3 layer 3 codec. This new coding system makes use of forward masking phenomenon to raise the coding efficiency and requires 11 to 20 percent less bits than the conventional MPEG layer 3.

8-3 Audio-Visual Perception of Video and Multimedia Programs
*Nina Dvork*o, St. Petersburg University of Humanities and Social Sciences, St. Petersburg, Russia
and *Konstantin Ershov,* St. Petersburg State University of Cinema and Television, St. Petersburg, Russia

This paper discusses the results of theoretical and experimental research of psychophysical and aesthetic aspects of sound and picture interaction. Perceptual experiments examine: 1) the influence of visual factors on threshold sensitivity of hearing, 2) the role of associative links in audio-visual perception, and 3) the correlation between sound and picture images in the perception of spatial localization in multichannel sound systems.

8-4 Subjective Validation of Perception Properties in Binaural Sound Reproduction Systems
Alois Sontacchi, Markus Noisternig, Piotr Majdak, and *Robert Höldrich*
University of Music and Dramatic Arts, Graz, Austria

A subjective validation of a mathematical model for characterizing binaural head-related impulse response (HRIR)-based reproduction systems is presented. The evaluated sound localization performance is validated by an informal listening test. The experimental setup is depicted, and the statistical evaluation of the results is given.

8-5 An Evaluation of Audio Warning Signals through Localization Behavior of the Eyes
Grigori Evreinov and *Darius Miniotas*
University of Tampere, Tampere, Finland.

This study focuses on establishing the type of envelope an auditory warning signal should have in order to minimize its distracting effect on attention. Listeners' ability to localize square wave and ramp spatial sounds was investigated. Their performance was evaluated using an SMI EyeLink Gaze Tracking system. Both of the auditory stimuli could be localized equally well. The reaction time was shorter for the square wave, but not significantly different from the ramp condition. The ramp stimulus, however, was reported by the participants to be more acceptable. The approach of using spatial sounds with a gradual onset may be a reasonable option to consider when selecting the most effective shape for a warning audio signal.

8-6 The Newest Methods and Models of the Expert Quality Evaluation of Audio and Video Images
Nickolay Kolomensky
St. Petersburg State University of Cinema and Television, St. Petersburg, Russia

New integral and differential criterion and algorithms for evaluation of the quality of the image and sound of audio-visual systems based on the discovered psychophysical laws of single-line and nonlinear stochastic differential images of physical and touch spaces (instead of the known psychophysical Weber–Fehner and Stevens' laws) were studied and approved. The plural-probabilistic approach for axiomatic consideration of the theory of subjective evaluation of quality of the image and sound (in audio and video systems) was developed. This approach uses a ring of ensembles with ó-algebras and multivariate Gilbert's touch space structure from the probabilistic approach by Kolmogorov and uses the measure of ensembles from the undefined approach by Zade. Designed and approved universal integral and differential criterion (factor) of the expert evaluation of the quality of the image and sound using the multivariate indicative function was taken into account as deterministic and casual in nature of the perception touch–perceptional images of signals of the image and sound.

1:40 pm–4:20 pm
SESSION 9: TRANSDUCERS (FUNDAMENTALS, DESIGN, AND MEASUREMENT)

9-1 An Inexpensive Precise Passive Crossover System
Neville Thiele, University of Sydney, Epping, NSW, Australia

This paper describes a crossover system for a two-way loudspeaker in which the drivers are fed through conventional second-order passive filters, but the parameters of the high-pass filter take into account the parameters of the associated closed-back tweeter, to realize a desired overall fourth-order high-pass filtered acoustic output. When that output is combined with that of the second-order low-pass filtered woofer, the summed response is flat. With no impedance correction required, the system produces an inexpensive but precise crossover.

9-2 Sophisticated Tube Headphones for Spatial Sound Reproduction
Klaus Riederer Helsinki University of Technology, Espoo, Finland
and *Risto Niska, Unides Design Ay, Helsinki, Finland*

Custom tube headphones, fulfilling the high requirements of accurate spatial sound perception experiments, are presented. The UD-ADU1b headphones demonstrate a maximum ±5-dB deviation in the frequency band from 30 Hz to 9 kHz, one-third octave smoothed. The ear canal blocking attenuates background noise typically at 15 to 20 dB and allows a precise positioning of the sound source. The nonmagnetic tubes are used in neuro- and psychophysiological research.

9-3 The Influence of Losses on the Frequency Response of the Band-Pass Loudspeaker Systems
Andrzej Dobrucki, Wroclaw University of Technology, Wroclaw, Poland

The influence of acoustical losses upon the frequency response of fourth-order band-pass loudspeaker systems is examined. It has been proved that losses in enclosures and in the vent can usually be negated. However, the leakage losses between both chambers of the system very strongly influence the frequency response. The rules for the corrections avoiding the differences between frequency responses obtained for lossless and actual systems have been developed.

9-4 Low-Frequency Room Excitation Using Distributed Mode Loudspeakers
Bruno Fazenda, Mark Avis, and *W. J. Davies,* University of Salford, Salford, Greater Manchester, UK

Conventional pistonic loudspeakers excite the modes of an enclosed sound field in such a way that introduce modal artifacts, which may be problematic for listeners of high-quality reproduced sound. Their amelioration may involve the use of highly space-consumptive passive absorptive devices or active control techniques. Other approaches have concentrated on the design of the driver used to excite the room. Distributed sources ranging from the dipole to more complex configurations can be expected to interact with the room eigenvectors in a complicated manner, which may be optimized in terms of the spatial and frequency-domain variance of the sound field. Recent interest in distributed sources has centered on the distributed mode loudspeaker (DML). This paper reports on an investigation into the interaction of DMLs with modal sound fields. It is shown that large DMLs can be expected to modify the low-frequency sound field and that smaller panels may interact with the room in interesting ways at higher frequencies. Producing useful low-frequency control remains difficult but can be achieved in some circumstances.

9-5 Problems of Theory and Designing for Directional Interference Microphones
Shakir Vakhitov, Mikrofon-M Ltd., St. Petersburg, Russia

A mathematical model and theoretical analysis of a directional microphone that consists of an interference tube and a pressure-gradient capsule are presented. The analytical expressions for angular dependence of the geometrical path length and directional characteristic were received. Physical reasons of the differences between polar patterns of such microphones and separate capsules in the low-frequency range were studied. Dependence of the required rear aperture acoustic resistance on the acoustic antenna length is shown. The reasons of the polar pattern axis asymmetry were analyzed. Theoretical principles are illustrated with experimental data. Practical recommendations for such microphone designs are given.

9-6 A Problem of Efficiency of Loudspeakers
Alexander Gaidarov, Andreev Acoustical Institute, Moscow, Russia

Loudspeaker efficiency in a given frequency band is the major characteristic of any electroacoustic transducer. Until now, there was only an approximated analytical expression of this parameter suitable for use. The targeted synthesis and optimization of devices with given spectral properties was difficult to ascertain because efficiency was hindered by the absence of the conforming analytical software. The uniqueness and manifestive way of intercoupling the Thiele–Small parameters of drivers for loudspeakers with a flat-amplitude frequency response used in acoustic closed-box enclosures has allowed the use of this analytical expression for loudspeaker efficiency by the way products of dimensionless factors are implemented with an ideal limit. The obtained expression has a pictorial form and can be easily interpreted in a physical sense. It also allows analysis of the actual problem of optimization energy efficiency of loudspeakers. The primary analysis of technological problems of increased efficiency of loudspeakers and the development of a compromise between the degree of approximation to an ideal limit and capability of practical implementation are discussed.

9-7 Measurement of Loudspeaker Large Signal Performance—Comparison of Different Testing Signals
Alexander Voishvillo, Eugene Czerwinski, and *Alexander Terekhov,* Cerwin-Vega Inc., Simi Valley, CA, USA

In this paper nonlinear reaction of low-frequency loud-speaker, horn driver, and free propagation to several different short-term signals has been investigated and compared. These signals are single tone burst, multi-tone burst, spectrally shaped pulse, and burst of Gaussian noise. The advantages and drawbacks of these signals are discussed. The relationship among the magnitude of the voice-coil excursion, distortion level, and variation of excursion-dependent parameters is discussed. Various situations with different behavior of excursion-dependent parameters are discussed. In measurement of maximum sound-pressure level produced by horn drivers, the distortion produced by the air propagation may affect the accuracy of measurement. Some examples of this effect are demonstrated. The difference between such aggregated criteria as reaction to multitone stimulus, incoherence function, and THD is discussed. Multitone burst and Gaussian noise bursts seem to be optimal signals to measure maximum SPL in loudspeakers because of the ability of these signals to excite a large number of intermodulation products.

4:00 pm–5:00 pm

SESSION 10: BINAURAL AND TRANSAURAL STEREOPHONY

10-1 Realisation of an Adaptive Cross-Talk Cancellation System for a Moving Listener

Tobias Lentz and *Oliver Schmitz*
DEGA, Aachen, Germany

The starting point of this paper is static crosstalk cancellation. The main task for an adaptive system is to update the crosstalk cancellation filter, depending on the listener's position. The required filter is calculated at run time of the program. Depending on the head position, the HRTFs required for the filter calculation will be selected from a database. The conclusion of the preliminary listening test is that the dynamic crosstalk cancellation produces impressive results. The listener can move in an area of about 1 sq. m. Head rotation is possible within the angle spanned by the loudspeakers.

10-2 Observed Effects of HRTF Measurement Signal Level

Agnieszka Jost, AuSIM Inc., Scotts Valley, CA, USA
and *Durand Begault,* NASA-Ames Research Center, Moffett Field, CA, USA

The effect of varying the signal level on the magnitude response of a head-related transfer function measurement was investigated. Measurement signals with levels ranging between 50- and 86-dB SPL were presented over a loudspeaker and recorded using blocked meatus microphones placed in a dummy head. Results indicate that relative to a 74-dB reference level for measurement signals below 62 dB and above 80-dB SPL: 1) the ipsilateral ear shows attenuated spectral notches; and 2) the contralateral ear demonstrates a 6- to 8-dB attenuation in bandwidths at 11.5-kHz and 16.5-kHz center frequencies.

10-3 An Objective Model of Localisation in Binaural Sound Reproduction Systems

Alois Sontacchi, Piotr Majdak, Markus Noisternig, and *Robert Höldrich*
University of Music and Dramatic Arts, Graz, Austria

A mathematical model is presented to objectively derive sound localization performance using head-related impulse responses (HRIR) based on binaural reproduction systems. Rendering a sound source via panning methods causes artifacts that will lead to errors in localization by human subjects. Studying the relationship between panning and perceived directions using listening tests entails an enormous effort of time. In addition, the presented mathematical model can be used to minimize the number of parameters to evaluate through listening tests. Furthermore, the localization performance of several HRIR-based panning methods were evaluated.

5:00 pm–6:0 pm

SESSION 11: WAVE FIELD SYNTHESIS

11-1 Distance Coding in 3D Sound Fields

Alois Sontacchi and *Robert Höldrich*
University of Music and Dramatic Arts, Graz, Austria

This investigation proposes a method to synthesize 3-D sound fields over loudspeakers taking distance coding into account. The system can be divided into two parts: combining the benefits by using both the wave field synthesis (WFS) and the Ambisonic approach. In order to code the virtual source distances, the driving functions using a derivative of the WFS approach were primarily calculated. In the second step, the apparent solid angle of the sources were coded.

11-2 Drawing Quality Maps of the Sweet Spot and Its Surroundings in Multichannel Reproduction and Coding

Aki Härmä, Tapio Lokki, and *Ville Pulkki,*
Helsinki University of Technology, Espoo, Finland

The sweet spot, or the optimal listening area in a room, is a central concept in multichannel audio reproduction. However, it is a difficult attribute to characterize in an objective way. Ways to measure the quality of sound within a wide listening area are discussed, and a map of the sweet spot and its surroundings in a simulated listening room setup is presented. The proposed technique can be used to evaluate and compare multichannel reproduction systems and audio coding algorithms.

11-3 Spatial Audio Reproduction Using Distributed Mode Loudspeaker Arrays

Ulrich Horbach, Studer Professional Audio AG, Regensdorf, Switzerland
*Diemer de Vrie*s, Delft University of Technology, Delft, The Netherlands
and *Etienne Cortee*l, IRCAM, Paris, France

True spatial reproduction of sound images over a large listening area can only be achieved by wave field synthesis, which requires a high number of individual loudspeaker channels. This paper describes a novel method to design such systems in a practical way using multiexciter distributed mode panels and digital filtering. Explained in detail are filter designs for the reproduction off-plane waves, which are required to efficiently transport and render a wave field in a perceptual sense, and filters for the creation of focused sound sources behind or in front of the panels. For MPEG-4 applications, the display of moving sound objects requires special algorithms to generate and interpolate long impulse responses.

Monday, June 3

9:00 am–10:40 am

SESSION 12: ARCHITECTURAL ACOUSTICS, PART2

12-1 Casa da Música, a New Concert Hall for Porto, Portugal
Laurentius van Luxemburg, Constant Hak, and *Heiko Martin,* Eindhoven University of Technology, Eindhoven,The Netherlands and *Ben Ko*k, and *Kjell Bijsterbosch,* Dorsserblesgraaf, Eindhoven, The Netherlands

In Porto a new concert hall is under construction. The new hall is shoebox-shaped with specific solutions to ensure sufficient strong lateral reflections. Acoustically, the main challenge is the front and back walls being made entirely of glass, giving the feeling that the rooms are open to the city. As well as keeping out noise from exterior sources, these transparent walls have to be designed such that they contribute to the sound distribution within the hall. With these considerations in mind, the glass walls of the Casa da Música's concert hall will have horizontally waved structures. The acoustical quality of the hall has been studied by using a simulation model and a scale model.

12-2 Acoustical Measurements of Courtyard-Type Traditional Chinese Theater in East China
YenKun Hsu, Weihwa Chiang, and *Jinjaw Tsai,* NTUST, Taipei, Taiwan
and *Jiqing Wan*g, Tongji University, Shanghai, People's Republic of China

Acoustical measurements were taken at six courtyard-type traditional Chinese theaters in east China. The theaters were generally rectangular in shape and some of them were built inside Chinese gardens. All measurements were taken in unoccupied conditions and in some cases with no seats. The theaters consist of a pavilion-like stage attached to a courtyard surrounded by covered corridors or buildings. Preliminary analysis showed an average strength (G) of 3.1 dB, an average early decay time (EDT) of 0.8 s, and an average early support (ST1) of –9.4 dB. This research was the first step in a three-year ongoing project about traditional Chinese theater. Subjective assessments will be undertaken later.

12-3 Acoustics Design of the Music Suite in Taipei National Architectural University of Arts
Weihwa Chiang, Liangkuang Yang, and *Wenling Jih*
NTUST, Taipei, Taiwan

An acoustical study of the music suite in Taipei National University of Arts based on computer modeling and 1:20-scale modeling was conducted. This paper reports on the design progress of a 7000-cub. m concert hall and a 1500-cub. m orchestra rehearsal room. The shoebox-shaped concert hall (600 seats) was designed mainly for the university orchestra, and the high ceiling (13.1 m above the stage floor) was kept to prevent the hall from being overpowered. The stage was restricted to 155 sq. m to provide good support for the musicians and good balance and blend for the audience. The midfrequency reverberation time (RT) was set to 1.7 s with a fully occupied audience and a sixty-member orchestra. With disconnected overhead reflectors 9 m above the stage floor, preliminary tests in the scale model yielded an early support (ST1) of –13.1 dB at the solo location. The orchestra rehearsal room was designed to provide both adequate reverberation and strength of coplayers. This objective was achieved by implementing a coupled room effect partitioned by an array of overhead reflectors hanging 5 m above the floor. Preliminary measurements with 55 upholstered seats yielded a ST1 of –12.9 dB and a RT of 1.6 s. Future study will be conducted on the field tuning and measurements after the completion of the building.

12-4 Room Acoustic Quality of a Multipurpose Hall: A Case Study. How to Dress a Cinema for Music Performances
Maria Ribeiro
FEUP/CEDEC, Porto, Portugal

The paper describes the acoustic solutions defined to adjust room acoustic quality to the aesthetic demands of architecture and the reduced budget available for construction of a high-standard, multipurpose hall. The paper presents the values predicted by computer simulation and the measured values of acoustical parameters—such as reverberation times (RTs), early decay times (EDTs), central time (Ts), loudness (G10), clarity (C80), and definition (D50)—for the main uses defined for the hall, i.e., cinema, conferences, and music presentations of very different styles ranging from classical music, jazz, and ethnical music to percussion groups.

12-5 Operational Methods of Forming Sound Field in the Room
Y. P. Shchevyev
St. Petersburg State University of Cinema and Television, St. Petersburg, Russia

A frequency reverberation method in a room with absorbers, where the given characteristic of the silencers is known in advance, is discussed. Results of the analytical investigation of the multilayer absorbers are given. A method of synthesis construction, which has a heterogeneous material basis, is described. The wave resistance changes toward the sound as the sound wave expands. A sound absorption system was developed to provide a calculation method.

11:00 am–12:20 pm

SESSION 13: LINEAR AND NONLINEAR DIGITAL PROCESSING OF MUSICAL AND SPEECH SIGNALS

13-1 Echo Compensation by Equalizer with Precise Spectrum Estimation

Andrey Barabanov, St. Petersburg State University, St. Petersburg, Russia
*Konstantin Putyako*v, Children School of Art, St. Petersburg, Russia
*Sergey Salische*v and *Vasilij Sitnikov,* St. Petersburg State University, St. Petersburg, Russia

This approach is based on measurement of the audio signal by two microphones that are posed at different distances from the source of the signal. A new adaptive equalizer was developed for echo compensation and optimal signal reinforcement. This technique uses the noise attenuation algorithm developed for the linear filtering model. Identification of the model parameters becomes the main problem of the approach.

13-2 Q Factor Modification for Low-Frequency Room Modes

Mark Avis
University of Salford, Salford, Greater Manchester, UK

Low-frequency normal modes of an enclosed sound field introduce unwanted frequency, spatial, and temporal artifacts to reproduced electroacoustic signals. A novel control approach has been reported based on an analytical modal decomposition, using a low-frequency sound field model in a one-dimensional environment formed from the sum of a number of second-order IIR filter sections. In this paper these techniques are applied to the low-frequency resonances of a three-dimensional test room. It is shown that significant reductions in modal Q and corresponding reductions in modal decay times can be achieved, leading to smaller low-frequency sound field variance and decreasing audibility of time-domain modal artifacts.

13-3 Further Developments of Methods for Searching Optimum Musical and Rhythmic Feature Vectors

Bozena Kostek, Marek Dziubinski, and *Pawel Zwan*
Technical University of Gdansk, Gdansk, Poland

The aim of this paper is first to review recent developments in the domain of musical information retrieval and then to present some methods developed at the Sound and Vision Engineering Department of the Technical University of Gdansk, Poland. Especially important for music retrieval systems is to find an optimum musical and rhythmic representation. This was done using both statistical evaluation and soft-computing methods. Results of the performed experiments are shown, and conclusions to the content of the feature vectors are discussed.

13-4 Non-Stationary Filtering Methods for Audio Signals

*John Sarris, Sotirios Daliani*s, and *George Cambouraki*s
National Technical University of Athens, Athens, Greece

This paper deals with filtering methods for audio signals using time–frequency analysis. The concept of time–frequency filtering is vital for enhancement of nonstationary signals and systems. Time–frequency filtering is performed as masking or convolution in the time–frequency domain and is based on nonparametric modeling using direct convolution or multiplication. Applications of filtering in the time–frequency domain, including artificial reverberation and sound source motion simulation, are presented.