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Table of Contents

Lecture Session LM1: Audio and Music Signal Processing

Monday, 8:50, Conference House

Sine-wave Based PSOLA Pitch Scaling with Real-time Pitch Marking

Robert McAulay 1

A Probabilistic Line Spectrum Model for Musical Instrument Sounds and Its Application to Piano Tuning Estimation

François Rigaud; Angélique Drémeau; Bertrand David; Laurent Daudet 5

Modulation Filtering for Structured Time-Frequency Estimation of Audio Signals

Kai Siedenburg; Philippe Depalle 9

Rate-Distortion Optimization for Multichannel Audio Compression

Minyue Li; Jan Skoglund; W. Bastiaan Kleijn 13

Poster Session PM

Monday, 10:30, Sunset Lounge

Localizing Multiple Audio Sources From DOA Estimates in a Wireless Acoustic Sensor Network

Anthony Griffin; Athanasios Mouchtaris 17

Employing Moments of Multiple High Orders for High-Resolution Underdetermined DOA Estimation Based on MUSIC

Yuya Sugimoto; Shigeki Miyabe; Takeshi Yamada; Shoji Makino; Fred Juang 21

On the Misalignment of Stereophonic Acoustic Echo Cancellation with Decorrelation by Resampling

Jason Wung; Ted S. Wada; Mehrez Souden; Fred Juang 25

Wave-domain Echo-Path Model with Aliasing for Echo Cancellation

Satoru Emura; Yusuke Hiwasaki; Hitoshi Ohmuro 29

Sound Acquisition in Noisy and Reverberant Environments Using Virtual Microphones

Konrad Kowalczyk; Oliver Thiergart; Alexandra Craciun; Emanuel Habets 33

Adaptive Distance and Near-Field Compensation Applied to Microphones

Walter Etter 37

Multizone Near-End Speech Enhancement Under Optimal Second-Order Magnitude Distortion

João Crespo; Richard Hendriks 41

An LCMV Filter for Single-Channel Noise Cancellation and Reduction in the Time Domain

Jesper Rindom Jensen; Jacob Benesty; Mads Græsbøll Christensen; Jingdong Chen 45

Estimation of Room Dimensions From a Single Impulse Response

Dejan Marković; Fabio Antonacci; Augusto Sarti; Stefano Tubaro 49

Room Impulse Response Synthesis Based on a 2D Multi-plane Hybrid Acoustic Model

Stephen Oxnard; Damian Murphy 53

Gentle Acoustic Crosstalk Cancellation Using the Spectral Division Method and Ambiophonics

Jens Ahrens; Mark R. P. Thomas; Ivan Tashev 57

The REVERB Challenge: A Common Evaluation Framework for Dereverberation and Recognition of Reverberant Speech

Keisuke Kinoshita; Marc Delcroix; Takuya Yoshioka; Tomohiro Nakatani; Emanuël Habets; Reinhold Haeb-Umbach; Volker Leutnant; Armin Sehr; Walter Kellermann; Roland Maas; Sharon Gannot; Bhiksha Raj 61

A New Clustering Approach for Solving the Permutation Problem in Convolutional Blind Source Separation

Radoslaw Mazur; Jan Ole Jungmann; Alfred Mertins 65

Low-Artifact Source Separation Using Probabilistic Latent Component Analysis

Nasser Mohammadiha; Paris Smaragdis; Arne Leijon 69

Influence of Secondary Path Estimation Errors on the Performance of ANC-Motivated Noise Reduction Algorithms for Hearing Aids

Derya Dalga; Simon Doclo 73

Gaussian Process Data Fusion for Heterogeneous HRTF Datasets

Yuan Cheng Luo; Dmitry Zotkin; Ramani Duraiswami 77

Identifying Salient Sounds Using Dual-Task Experiments

Varinthira Duangudom; David Anderson 81

Evaluation of Spectral Transforms for Music Signal Analysis

Anil Nagathil; Rainer Martin 85

Music Self-Similarity Modeling Using Augmented Nonnegative Matrix Factorization of Block and Stripe Patterns

Joonas Kauppinen; Anssi Klapuri; Tuomas Virtanen 89

Modeling Nonlinear Circuits with Linearized Dynamical Models Via Kernel Regression

Daniel J. Gillespie; Daniel P W Ellis 93

Evaluating How Well Filtered White Noise Models the Residual From Sinusoidal Modeling of Musical Instrument Sounds

Marcelo Caetano; George Kafentzis; Gilles Degottex; Athanasios Mouchtaris; Yannis Stylianou 97

Sparse Representation and Epoch Estimation of Voiced Speech

Jake Gunther; Todd Moon 101

A Polynomial Interpolation-Based Scheme for Reducing Bandwidth in Distributed Speech Recognition System

Azzedine Touazi; Debyeche Mohamed 105

Lecture Session LM2: Microphone Array Processing

Monday, 16:00, Conference House

A Recursive Generalized Sidelobe Canceler for Multichannel Blind Speech Dereverberation

Sarmad Malik; Jacob Benesty; Jingdong Chen 109

A Multichannel Wiener Filter with Partial Equalization for Distributed Microphones

Sebastian Stenzel; Toby Christian Lawin-Ore; Juergen Freudenberger; Simon Doclo 113

On the Accuracy of Open Spherical Microphone Arrays for Measuring Acoustic Intensity

Huseyin Hacihabiboglu 117

Spotforming Using Distributed Microphone Arrays

Maja Taseska; Emanuël Habets 121

Broadband Sensor Location Selection Using Convex Optimization in Very Large Scale Arrays

Yenming Lai; Radu Balan; Heiko Claussen; Justinian Rosca 125

Relative Transfer Function Modeling for Supervised Source Localization

Bracha Laufer; Ronen Talmon; Sharon Gannot 129

Lecture Session LT1: Loudspeaker Array Processing and Room Acoustics

Tuesday, 08:50, Conference House

MAP Estimation of Driving Signals of Loudspeakers for Sound Field Reproduction From Pressure Measurements

Shoichi Koyama; Ken'ichi Furuya; Yusuke Hiwasaki; Yoichi Haneda 133

Loudspeaker Placement for Sound Field Reproduction by Constrained Matching Pursuit

Hanieh Khalilian; Ivan V. Bajic; Rodney Vaughan 137

Deconvolution of Plenacoustic Images

Lucio Bianchi; Dejan Marković; Fabio Antonacci; Augusto Sarti; Stefano Tubaro 141

Detection and Classification of Acoustic Scenes and Events: An IEEE AASP Challenge

Dimitrios Giannoulis; Emmanouil Benetos; Dan Stowell; Mathias Rossignol; Mathieu Lagrange; Mark D. Plumbley 145

Poster Session PT

Tuesday, 10:30, Sunset Lounge

The Geometry of Sound-Source Localization Using Non-coplanar Microphone Arrays

Xavier Alameda-Pineda; Radu Horaud; Bernard Mourrain 149

A Simple Adaptive Cardioid Direction Finding Algorithm

Gary W Elko; Jens Meyer 153

Geometrically Constrained TRINICON-based Relative Transfer Function Estimation in Underdetermined Scenarios

Klaus Reindl; Shmulik Markovich-Golan; Hendrik Barfuss; Sharon Gannot; Walter Kellermann 157

MINTFormer: A Spatially Aware Channel Equalizer

Felicia Lim; Mark R. P. Thomas; Patrick A Naylor 161

Microphone Multiplexing with Diffuse Noise Model-based Principal Component Analysis

Sonia Badar; Nobutaka Ono; Laurent Daudet 165

On A Dual-Gain Approach to Noise Reduction in the STFT Domain

Chao Pan; Jingdong Chen; Jacob Benesty 169

Blind Low-Complexity Estimation of Reverberation Time

Christian Schüldt; Peter Händel 173

Efficient Implementation of the Spectral Division Method for Arbitrary Virtual Sound Fields

Jens Ahrens; Mark R. P. Thomas; Ivan Tashev 177

Optimizing Frame Analysis with Non-Integer Shift for Sampling Mismatch Compensation of Long Recording

Shigeki Miyabe; Nobutaka Ono; Shoji Makino 181

Perceptually Motivated ANC for Hearing-Impaired Listeners

Eric Durant; Jinjun Xiao; Buye Xu; Martin McKinney; Tao Zhang 185

Perceptual Cepstral Filters for Speech and Music Processing

Remi Mignot; Vesa Valimaki 189

Design of Arbitrary Delay Filterbank Having Arbitrary Order for Audio Applications

Asha Vijayakumar; Anamitra Makur 193

On the Use of Spectro-Temporal Features for the IEEE AASP Challenge 'Detection and Classification of Acoustic Scenes and Events'

Jens Schröder; Niko Moritz; Marc Schädler; Benjamin Cauchi; Kamil Adiloglu; Joern Anemueller; Simon Doclo; Birger Kollmeier; Stefan Goetze 197

Recurrence Quantification Analysis Features for Environmental Sound Recognition

Gerard Roma; Waldo Nogueira; Perfecto Herrera 201

Large-Scale Audio Feature Extraction and SVM for Acoustic Scene Classification

Jürgen Geiger; Björn Schuller; Gerhard Rigoll 205

Hierarchical Modeling Using Automated Sub-Clustering for Sound Event Detection

Maria E. Niessen; Tim L. M. Van Kasteren; Andreas Merentitis 209

Sound Event Detection Using Non-negative Dictionaries Learned From Annotated Overlapping Events

Onur Dikmen; Annamaria Mesaros 213

Acoustic Scene Classification Using Sparse Feature Learning and Event-Based Pooling

Kyogu Lee; Ziwon Hyung; Juhan Nam 217

An Exemplar-Based NMF Approach to Audio Event Detection

Jort Gemmeke; Lode Vuegen; Peter Karsmakers; Bart Vanrumste; Hugo Van hamme 221

Lecture Session LT2: Speech Enhancement

Tuesday, 16:50, Conference House

Ensemble Learning for Speech Enhancement

Jonathan Le Roux; Shinji Watanabe; John Hershey 225

Speech Enhancement by Sparse, Low-rank, and Dictionary Spectrogram Decomposition

Zhuo Chen; Daniel P W Ellis 229

Tracking Pitch Period Using Particle Filters

Geliang Zhang; Simon Godsill 233

Spectral Feature-based Nonlinear Residual Echo Suppression

Andreas Schwarz; Christian Hofmann; Walter Kellermann 237

Lecture Session LW1: Hearing & Perception

Wednesday, 08:50, Conference House

The Influence of Informational Masking in Complex Real-World Environments

Adam Westermann; Jörg Buchholz 241

Using Articulation Index Band Correlations to Objectively Estimate Speech Intelligibility Consistent with the Modified Rhyme Test

Stephen D Voran 245

Virtual Autoencoder Based Recommendation System for Individualizing Head-Related Transfer Functions

Yuancheng Luo; Dmitry Zotkin; Ramani Duraiswami 249

Environment-aware Ideal Binary Mask Estimation Using Monaural Cues

Tobias May; Torsten Dau 253

Poster Session PW

Wednesday, 10:30, Sunset Lounge

Closed-Form Solutions for Robust Acoustic Sensor Localization

Diego B. Haddad; Leonardo Nunes; Wallace A. Martins; Luiz W. P. Biscainho; Bowon Lee 257

Robust DOA Estimation of Speech Signals Via Sparsity Models Using Microphone Arrays

Eleonora Cagli; Diego Carrera; Giacomo Aletti; Giovanni Naldi; Beatrice Rossi 261

A Sparse Nonuniformly Partitioned Multidelay Filter for Acoustic Echo Cancellation

Daniele Giacobello; Joshua Atkins 265

A Probabilistic Approach to Acoustic Echo Clustering and Suppression

Mehrez Souden; Jason Wung; Fred Juang 269

Average Output SNR of the Multichannel Wiener Filter Using Statistical Room Acoustics

Toby Christian Lawin-Ore; Simon Doclo 273

Under-determined Source Separation Based on Power Spectral Density Estimated Using Cylindrical Mode Beamforming

Effect of Higher-Order Ambisonics on Evaluating Beamformer Benefit in Realistic Acoustic Environments

Chris Oreinos; Jörg Buchholz; Jorge Mejia 281

Functional Analysis Guided Approach for Sound Field Reproduction with Flexible Loudspeaker Layouts

Wen Zhang; Thushara D. Abhayapala; Filippo M Fazi 285

Modeling Early Reflections of Room Impulse Responses Using a Radiance Transfer Method

Hequn Bai; Gaël Richard; Laurent Daudet 289

Roomprints for Forensic Audio Applications

Alastair H Moore; Mike Brookes; Patrick A Naylor 293

Multichannel HR-NMF for Modelling Convolutional Mixtures of Non-Stationary Signals in the Time-Frequency Domain

Roland Badeau; Mark D. Plumbley 297

Frequency Domain Multi-channel Expectation Maximization Algorithm for Audio Background Noise Reduction

Jichi Deng; Simon Godsill 301

Tapping-noise Suppression with Magnitude-Weighted Phase-Based Detection

Akihiko K. Sugiyama 305

A Fast Griffin-Lim Algorithm

Nathanaël Perraudin; Peter Balazs; Peter Soendergaard 309

Speech Understanding in Noise Provided by a Simulated Cochlear Implant Processor Based on Matching Pursuit

Abigail Kressner; Christopher Rozell 313

Learning an Intelligibility Map of Individual Utterances

Michael I Mandel 317

An Efficient Time-varying Loudness Model

Dominic Ward; Cham Athwal; Munevver Kokuer 321

Hierarchical and Coupled Non-Negative Dynamical Systems with Application to Audio Modeling

Umut Şimşekli; Jonathan Le Roux; John Hershey 325

The Shift-ACF: Detecting Multiply Repeated Signal Components

Frank Kurth 329

Non-Negative Matrix Factorization for Irregularly-Spaced Transforms

Paris Smaragdis; Minje Kim 333

Bayesian Non-parametric Matrix Factorization for Discovering Words in Spoken Utterances

Sayeh Mirzaei; Hugo Van hamme; Yaser Norouzi 337

Comparison of Windowing in Speech and Audio Coding

Tom Bäckström 341

A Hybrid LF-Rosenberg Frequency-Domain Model of the Glottal Pulse

Aníbal J Ferreira; Sandra Dias 345