2014 4th Joint Workshop on Hands-free Speech Communication and Microphone Arrays

(HSCMA 2014)

Nancy, France
12-14 May 2014

IEEE Catalog Number: CFP14HSM-POD
ISBN: 978-1-4799-3110-1

Program

Monday, May 12

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Ad-Hoc Microphone Array Calibration from Partial Distance Measurements
Mohammad J. Taghizadeh (Idiap Research Institute & EPFL, Switzerland); Afsaneh Asaei (Idiap Research Institute, Switzerland); Philip N. Garner (Idiap Research Institute, Switzerland); Herve Bourlard (IDIAP, Switzerland)
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Kernel spectrogram models for source separation
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Supervised non-Euclidean sparse NMF via bilevel optimization with applications to speech enhancement
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Jort Gemmeke (KU Leuven, Belgium)
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Transient Noise Reduction Using Nonnegative Matrix Factorization
Nasser Mohammadiha (University of Oldenburg, Germany); Simon Doclo (University of Oldenburg, Germany)
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Hai Morgenstern (Ben-Gurion University of the Negev, Israel); Boaz Rafaely (Ben-Gurion University of the Negev, Israel)

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Jiri Malek (Technical University of Liberec, Czech Republic); David Botka (Technical University of Liberec, Czech Republic); Zbynek Koldovsky (Technical University of Liberec, Czech Republic); Sharon Gannot (Bar-Ilan University, Israel)
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Daniele Giacobello (Beats Electronics, USA); Jason Wung (Beats Electronics, LLC & Georgia Institute of Technology, USA); Ramin Pichevar (Beats Electronics LLC, USA); Joshua Atkins (Beats Electronics, USA)
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Lalan Kumar (Indian Institute of Technology Kanpur & TCS, India); Kushagra Singhal (IIT Kanpur, India); Rajesh M Hegde (Indian Institute of Technology Kanpur, India)
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Sebastian Stenzel (University of Applied Sciences Constance & Institute for System Dynamics (ISD), Germany); Juergen Freudenberger (University of Applied Sciences, Konstanz & Institute for System Dynamics (ISD), Germany); Gerhard Schmidt (CAU Kiel, Germany)
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Ankit Sohni (Indian Institute of Technology, Kanpur, India); Chaitanya Ahuja (Indian Institute of Technology Kanpur, India); Rajesh M Hegde (Indian Institute of Technology Kanpur, India)
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Panagiotis Giannoulis (National Technical University of Athens + ATHENA-RC, Greece); Antigoni Tsiami (National Technical University of Athens + ATHENA-RC, Greece); Isidoros Rodomagoulakis (National Technical University of Athens, Greece); Athanasios Katsamanis (National Technical University of Athens, Greece); Gerasimos Potamianos (University of Thessaly, Greece); Petros Maragos (National Technical University of Athens, Greece)
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Wednesday, May 14

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Armin Sehr (Beuth Hochschule für Technik Berlin, Germany); Hendrik Barfuss (University of Erlangen-Nuremberg, Germany); Christian Hofmann (University of Erlangen-Nuremberg, Germany); Roland Maas (University of Erlangen-Nuremberg, Germany); Walter Kellermann (University Erlangen-Nuremberg, Germany)
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Optimized Joint Noise Suppression and Dereverberation based on Blind Signal Extraction for Hands-Free Speech Recognition System
Fine Aprilyanti (Nara Institute of Science and Technology, Japan); Hiroshi Saruwatari (Graduate School of Information Science, Nara Institute of Science and Technology, Japan); Satoshi Nakamura (Nara Institute of Science and Technology, Japan); Tomoya Takatani (Toyota Motor Corporation, Japan)
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Cristina M. Guerrero (Fondazione Bruno Kessler & University of Trento, Italy); Maurizio Omologo (Fondazione Bruno Kessler - irst, Italy)
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M. Mimura (Kyoto University, Japan); Shinsuke Sakai (Kyoto University, Japan); Tatsuya Kawahara (Kyoto University, Japan)
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