# Diamond Computer Sales Services Education
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## Audio speech & Language processing (2016)

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|---|---|
| 1 | A Doubly Hierarchical Dirichlet Process HiddenMarkov Model with a Non-Ergodic Structure |
| 2 | Superdirective Beamforming Based on the Krylov Matrix |
| 3 | A Hybrid Approach for Speech Enhancement Using MoG Model and Neural Network Phoneme Classifier |
| 4 | Sensitivity of Source–Filter Interaction to Specific Vocal Tract Shapes |
| 5 | Long-Term SNR Estimation of Speech Signals in Known and Unknown Channel Conditions |
| 6 | Extraction of Acoustic Sources Through the Processing of Sound Field Maps in the Ray Space |
| 7 | Single Snapshot Detection and Estimation of Reflections From Room Impulse Responses in the Spherical Harmonic Domain |
| 8 | Multichannel Audio Source Separation With Probabilistic Reverberation Priors |
| 9 | Integration of Optimized Modulation Filter Sets into Deep Neural Networks for Automatic Speech Recognition |
| 10 | An Approach to Score Following for Piano Performances with the Sustained Effect |
| 11 | Score-Aging Calibration for Speaker Verification |
| 12 | Estimating Speech Recognition Accuracy Based on Error Type Classification |
| 13 | Adaptive compensation of mis-equalization in narrowband active noise equalizer systems |
| 14 | Blind Separation of Audio Mixtures Through Nonnegative Tensor Factorization of Modulation Spectrograms |

| | |
|---|---|
| 15 | Robust Variable Step-Size Decorrelation Normalized Least-Mean-Square Algorithm and its Application to Acoustic Echo Cancellation |
| 16 | Instantaneous Fundamental Frequency Estimation with Optimal Segmentation for Non-Stationary Voiced Speech |
| 17 | Novel Unsupervised Auditory Filterbank Learning Using Convolutional RBM for Speech Recognition |
| 18 | Bayesian Networks to Model the Variability of Speaker Verification Scores in Adverse Environments |
| 19 | Kernel Method for Voice Activity Detection in the Presence of Transients |
| 20 | Composition of Deep and Spiking Neural Networks for Very Low Bit Rate Speech Coding |
| 21 | Scalable Audio Coding Using Trellis-Based Optimized Joint Entropy Coding and Quantization |
| 22 | Generation of Affective Accompaniment in Accordance With Emotion Flow |
| 23 | Very Deep Convolutional Neural Networks for Noise Robust Speech Recognition |

| | |
|---|---|
| 24 | On MMSE-Based Estimation of Amplitude and Complex Speech Spectral Coefficients Under Phase-Uncertainty |
| 25 | Factorized Hidden Layer Adaptation for Deep Neural Network Based Acoustic Modeling |
| 26 | Neural Network based Multi-Factor Aware Joint Training for Robust Speech Recognition |
| 27 | Context-Dependent Piano Music Transcription With Convolutional Sparse Coding |
| 28 | A Probabilistic Modeling Approach to Hearing Loss Compensation |
| 29 | Automatic Assessment of Speech Intelligibility for Individuals With Aphasia |
| 30 | Estimation of the Direct-Path Relative Transfer Function for Supervised Sound-Source Localization |
| 31 | Monaural Music Source Separation Using Convolutional Sparse Coding |
| 32 | Efficient Training and Evaluation of Recurrent Neural Network Language Models for Automatic Speech Recognition |

| | |
|---|---|
| 33 | Robust and Efficient Joint Alignment of Multiple Musical Performances |
| 34 | Compressive Sensing-Based Speech Enhancement |
| 35 | Using Agreement on Direction of Change to Build Rank-Based Emotion Classifiers |
| 36 | Automatic Environmental Sound Recognition: Performance Versus Computational Cost |
| 37 | Singing Voice Separation and Vocal F0 Estimation based on Mutual Combination of Robust Principal Component Analysis and Subharmonic Summation |
| 38 | A Loss-Augmented Approach to Training Syntactic Machine Translation Systems |
| 39 | Transition-Based Dependency Parsing Exploiting Supertags |
| 40 | Speaker Adaptation of SR-HPM for Speaking Rate-Controlled Mandarin TTS |
| 41 | Non-Parallel Training in Voice Conversion Using an Adaptive Restricted Boltzmann Machine |
| 42 | Maximum Likelihood Nonlinear Transformations Based On Deep Neural Networks |

| | |
|---|---|
| 43 | An Algorithm for Predicting the Intelligibility of Speech Masked by Modulated Noise Maskers |
| 44 | Alignment of Lyrics With Accompanied Singing Audio Based on Acoustic-Phonetic Vowel Likelihood Modeling |
| 45 | Piano Transcription in the Studio Using an Extensible Alternating Directions Framework |
| 46 | Robust Room Impulse Response Measurement Using Perfect Sequences for Legendre Nonlinear Filters |
| 47 | Community-Based Weighted Graph Model for Valence-Arousal Prediction of Affective Words |
| 48 | Graph-Based Semisupervised Learning for Acoustic Modeling in Automatic Speech Recognition |
| 49 | Speaker Diarization and Linking of Meeting Data |
| 50 | Distributed Marginalized Auxiliary Particle Filter for Speaker Tracking in Distributed Microphone Networks |
| 51 | Predicting the Intelligibility of Noisy and Non-Linearly Processed Binaural  Speech |
| 52 | Improving Mandarin Prosody Generation Using Alternative Smoothing Techniques |
| 53 | Towards  Multi-Purpose Spectral  Rhythm  Features: An Application  to Dance Style, Meter and Tempo Estimation |
| 54 | Parallel Reference Speaker Weighting for Kinematic-Independent Acoustic-to-Articulatory Inversion |

| 55 | A Morphological Model for Simulating Acoustic Scenes and Its Application to Sound Event Detection |
|----|---|
| 56 | Abstractive Cross-Language Summarization via Translation Model Enhanced Predicate Argument Structure Fusing |
| 57 | Summarizing Meeting Transcripts Based on Functional Segmentation |
| 58 | Similar Word Model for Unfrequent Word Enhancement in Speech Recognition |
| 59 | Near and Far Field Speech-in-Noise Intelligibility Improvements Based on a Time–Frequency Energy Reallocation Approach |
| 60 | Spherical Harmonic Signal Covariance and Sound Field Diffuseness |
| 61 | Optimal Microphone Array Observation for Clear Recording of Distant Sound Sources |
| 62 | Differentiable Pooling for Unsupervised Acoustic Model Adaptation |
| 63 | Auditory Model-Based Dynamic Compression Controlled by Subband Instantaneous Frequency and Speech Presence Probability Estimates |
| 64 | Variational Bayesian Inference for Source Separation and Robust Feature Extraction |
| 65 | Adaptive Filtered-x Algorithms for Room Equalization Based on Block-Based Combination Schemes |

| 66 | A Generalized Nonnegative Tensor Factorization Approach for Distant Speech Recognition  with Distributed Microphones |
|----|---|
| 67 | Generative Modeling of Pseudo-Whisper for Robust Whispered Speech Recognition |
| 68 | Efficient Implementation of Global Variance Compensation for Parametric Speech Synthesis |
| 69 | Estimation of room acoustic parameters: The ACE Challenge |
| 70 | On the Use of Acoustic Unit Discovery for Language Recognition |
| 71 | Multichannel audio source separation with deep neural networks |
| 72 | Similarity Search of Acted Voices for Automatic Voice Casting |
| 73 | Determined Blind Source Separation Unifying Independent Vector Analysis and Nonnegative Matrix Factorization |
| 74 | Computationally Efficient and Noise Robust DOA and Pitch Estimation |
| 75 | Maximum Likelihood PSD Estimation for Speech Enhancement in Reverberation and Noise |
| 76 | Relation Classification via Modeling Augmented Dependency Paths |
| 77 | Over-Determined Source Separation and Localization Using Distributed Microphones |
| 78 | Projection-based demixing of spatial audio |
| 79 | Reduced-Order Robust Superdirective Beamforming with Uniform Linear Microphone Arrays |

| | |
|---|---|
| 80 | Reverse Engineering  the Amen Break – <br> Score-informed Separation and Restoration applied to Drum Recordings |
| 81 | Semi-supervised Acoustic Model Training by <br> Discriminative Data Selection from Multiple <br> ASR Systems' Hypotheses |
| 82 | Signal-dependent spatial filtering based on <br> weighted-orthogonal beamformers in the spherical harmonic domain |
| 83 | An Expectation-Maximization Algorithm for <br> Multi-microphone Speech Dereverberation  and <br> Noise Reduction with Coherence Matrix Estimation |
| 84 | Combination of Language Models for Word <br> Prediction: An Exponential Approach |
| 85 | Listwise Ranking Functions for Statistical <br> Machine Translation |
| 86 | Learning Hidden Unit Contributions for <br> Unsupervised Acoustic Model Adaptation |
| 87 | Two Efficient Lattice Rescoring Methods Using <br> Recurrent Neural Network Language Models |
| 88 | Two Efficient Lattice Rescoring Methods Using <br> Recurrent Neural Network Language Models |

| | |
|---|---|
| 89 | A Regression Approach to Single-Channel Speech Separation Via High-Resolution Deep Neural Networks |
| 90 | A Variational EM Algorithm for the Separation of Time-Varying Convolutive Audio Mixtures |
| 91 | Semi-Supervised Sound Source Localization Based on Manifold Regularization |
| 92 | A Systematic ANSI S1.11 Filter Bank Specification Relaxation  and Its Efficient Multirate Architecture for Hearing-aid  Systems |
| 93 | Comparison of Loudspeaker Placement Methods for Sound Field Reproduction |
| 94 | Localization  and Characterization of Multiple Harmonic Sources |
| 95 | Least-Squares Estimation of the Common Pole-Zero Filter of Acoustic Feedback Paths in Hearing Aids |
| 96 | Power-Normalized Cepstral Coefficients (PNCC) for Robust Speech Recognition |
| 97 | Learning  the Multilingual Translation Representations for Question Retrieval  in Community  Question Answering  via Non-negative Matrix Factorization |
| 98 | Spotforming: Spatial Filtering with Distributed Arrays for Position-Selective Sound Acquisition |
| 99 | Improving the Performance of the PNLMS Algorithm Using $l1$  Norm Regularization |

| 100 | Fusion methods for speech enhancement and audio source separation |
|-----|-------------------------------------------------------------------|
| 101 | Improving Trajectory Modelling for DNN-Based Speech Synthesis by Using Stacked Bottleneck Features and Minimum Generation Error Training |
| 102 | Multi-Frame Amplitude Envelope Estimation for Modification of Singing Voice |
| 103 | A Novel Method of Glottal Inverse Filtering |
| 104 | Robust Estimation and Tracking of Pitch Period Using an Efficient Bayesian Filter |
| 105 | Foreground Speech Segmentation and Enhancement using Glottal Closure Instants and Mel Cepstral Coefficients |
| 106 | Text-dependent speaker recognition with random digit strings |
| 107 | Training Deep Bidirectional LSTM Acoustic Model for LVCSR by a Context-Sensitive-Chunk BPTT Approach |
| 108 | Multiple Non-negative Matrix Factorization for Many-to-many Voice Conversion |
| 109 | A Simplified Subband ANC Algorithm without Secondary Path Modeling |

| No. | Title |
|-----|-------|
| 110 | A Frequency-domain Adaptive Line Enhancer with Step-size Control Based on Mutual Information for Harmonic Noise Reduction |
| 111 | Improving Short Utterance Speaker Recognition by Modeling Speech Unit Classes |
| 112 | Using Generic Summarization to Improve Music Information Retrieval Tasks |
| 113 | Fast Audio Fingerprinting System Using GPU and a Clustering-Based Technique |
| 114 | A MONTAGE APPROACH TO SOUND TEXTURE SYNTHESIS |
| 115 | An Iterative Approach to Source Counting and Localization Using Two Distant Microphones |
| 116 | A Pairwise Algorithm Using the Deep Stacking Network for Speech Separation and Pitch Estimation |
| 117 | Candidate Expansion and Prosody Adjustment for Natural Speech Synthesis Using a Small Corpus |
| 118 | A Framework for Speech Enhancement with Ad Hoc Microphone Arrays |
| 119 | Text-Independent Phoneme Segmentation Combining EGG and Speech Data |

| | |
|---|---|
| 120 | A Two-pass Framework of Mispronunciation Detection & Diagnosis for Computer-aided Pronunciation Training |
| 121 | Feature Adaptation Using Linear Spectro-Temporal Transform for Robust Speech Recognition |
| 122 | An Empirical Investigation of Word Class-Based Features for Natural Language Understanding |
| 123 | Scalable Discovery of Audio Fingerprint Motifs in Broadcast Streams with Determinantal Point Process based Motif Clustering |
| 124 | A Deep Ensemble Learning Method for Monaural Speech Separation |
| 125 | Binaural Noise Cue Preservation in a Binaural Noise Reduction System with a Remote Microphone Signal |
| 126 | Fundamental Frequency Informed Speech Enhancement in a Flexible Statistical Framework |
| 127 | Fundamental Frequency Informed Speech Enhancement in a Flexible Statistical Framework |
| 128 | An End-to-End Neural Network for Polyphonic Piano Music Transcription |

| 129 | On Analytic Methods for 2.5D Local Sound Field Synthesis Using Circular Distributions of Secondary Sources |
| 130 | Automatic Transcription of Flamenco Singing from Polyphonic Music Recordings |
| 131 | Fundamental Frequency Estimation In Speech Signals With Variable Rate Particle Filters |
| 132 | A Flexible Bio-inspired Hierarchical Model for Analyzing Musical Timbre |
| 133 | Unsupervised Incremental Online Learning and Prediction of Musical Audio Signals |
| 134 | Unsupervised Incremental Online Learning and Prediction of Musical Audio Signals |
| 135 | Turbo Automatic Speech Recognition |
| 136 | Robust and Efficient Multiple Alignment of Unsynchronized Meeting Recordings |
| 137 | An Empirical Investigation of Word Class-Based Features for Natural Language Understanding |
| 138 | Preserving Word-level Emphasis in Speech-to-speech Translation |
| 139 | Personalizing Recurrent Neural Network Based Language Model by Social Network |

| 140 | Sonority Measurement Using System, Source, and Suprasegmental Information |
|-----|---------------------------------------------------------------------------|
| 141 | Sonority Measurement Using System, Source, and Suprasegmental Information |
| 142 | Collaborative Joint Training with Multi-task Recurrent Model for Speech and Speaker Recognition |
| 143 | Feedback-Driven Sensory Mapping Adaptation for Robust Speech Activity Detection |
| 144 | Emotion Analysis in Code-Switching Text with Joint Factor Graph Model |
| 145 | Multiplicative Update of Auto- Regressive Gains for Codebook-Based Speech Enhancement |
| 146 | IEEE SIGNAL PROCESSING SOCIETY |
| 147 | On the Evaluation of Adaptive Machine Translation for Human Post-Editing |
| 148 | Packet Loss Concealment Based on Deep Neural Networks for Digital Speech Transmission |
| 149 | A Semidefinite Programming Approach to Min-max Estimation of the Common Part of Acoustic Feedback Paths in Hearing Aids |
| 150 | The Hearing-Aid Audio Quality Index (HAAQI) |

| 151 | Optimizing the Similarity of Loudspeaker–Room Responses in Multiple Listening Positions |
|-----|------------------------------------------------------------------------------------------|
| 152 | Real-Time Audio-to-Score Alignment of Music Performances Containing Errors and Arbitrary Repeats and Skips |
| 153 | A Fast Method for High-Resolution Voiced/Unvoiced Detection and Glottal Closure/Opening Instant Estimation of Speech |
| 154 | Directivity Factors of the First-Order Steerable Differential Array With Microphone Mismatches: Deterministic and Worst-Case Analysis |
| 155 | Two-Stage Supervised Learning-Based Method to Detect Screams and Cries in Urban Environments |
| 156 | Speech Dereverberation Using Non-Negative Convolutive Transfer Function and Spectro-Temporal Modeling |
| 157 | Source and Filter Estimation for Throat-Microphone Speech Enhancement |
| 158 | Boosting Contextual Information for Deep Neural Network Based Voice Activity Detection |
| 159 | Robust Speech Recognition via Enhancing the Complex-Valued Acoustic Spectrum in Modulation Domain |
| 160 | Design of Directivity Patterns with a Unique Null of Maximum Multiplicity |
| 161 | Regularization Approaches for Synthesizing HRTF Directivity Patterns |

| | |
|---|---|
| 162 | Beamforming with Optimal Aliasing Cancellation in Spherical Microphone Arrays |
| 163 | IEEE SIGNAL PROCESSING SOCIETY |
| 164 | Bayesian Factorization and Learning for Monaural Source Separation |
| 165 | Finite Volume Time Domain Room Acoustics Simulation under General Impedance Boundary Conditions |
| 166 | Novel Estimation Method for the Superpositional Intonation Model |
| 167 | Spatial Correlation of Radial Gaussian and Uniform Spherical Volume Near-Field Source Distributions |
| 168 | Mixture of PLDA for Noise Robust I-Vector Speaker Verification |
| 169 | Sparse Localization of Harmonic Audio Sources |
| 170 | Study of Senone-Based Deep Neural Network Approaches for Spoken Language Recognition |
| 171 | Unseen Noise Estimation Using Separable Deep Auto Encoder for Speech Enhancement |
| 172 | Using Pronunciation-Based Morphological Subword Units to Improve OOV Handling in Keyword Search |
| 173 | Speaker and Channel Factors in Text-Dependent Speaker Recognition |
| 174 | A Statistical Inverse Problem Approach to Online Secondary Path Modeling in Active Noise Control |
| 175 | Feature Extraction Using Power-Law Adjusted Linear Prediction With Application to Speaker Recognition Under Severe Vocal Effort Mismatch |

| 176 | i-Vector Modeling of Speech Attributes for Automatic Foreign Accent Recognition |
|-----|-----|
| 177 | Towards a Small Set of Robust Acoustic Features for Emotion Recognition: Challenges |
| 178 | HMM-based  Speech Segmentation: Improvements of Fully Automatic Approaches |
| 179 | Capacity-Approaching TQC-LDPC Convolutional Codes Enabling Power-Efficient Decoders |
| 180 | Noise Robust Formant Frequency Estimation Method Based on Spectral Model of Repeated Autocorrelation of Speech |
| 181 | Quadratic Programming Approach to Glottal Inverse Filtering by Joint Norm-1 and Norm-2 Optimization |
| 182 | Task Independent Fine Tuning for Word Embeddings |
| 183 | Background Noise Reduction Design for Dual Microphone Cellular Phones: Robust Approach |
| 184 | Window-Dominant Signal Subspace Methods for Multiple Short-Term Speech Source Localization |
| 185 | Single-channel Online Enhancement of Speech Corrupted by Reverberation and Noise |
| 186 | Preserving Word-level  Emphasis in Speech-to-speech Translation |

| | |
|---|---|
| 187 | Personalizing Recurrent Neural Network Based Language Model by Social Network |
| 188 | Sonority Measurement Using System, Source, and Suprasegmental Information |
| 189 | Collaborative Joint Training with Multi-task Recurrent Model for Speech and Speaker Recognition |
| 190 | Feedback-Driven Sensory Mapping Adaptation for Robust Speech Activity Detection |
| 191 | Emotion Analysis in Code-Switching Text with Joint Factor Graph Model |
| 192 | Multiplicative Update of Auto- Regressive Gains for Codebook-Based Speech Enhancement |
| 193 | GPU-Based Dynamic Wave Field Synthesis Using Fractional Delay Filters and Room Compensation |
| 194 | A New Strategy for Online Secondary-Path Modeling of Narrowband Active Noise Control |
| 195 | Multimodal Crowdsourcing for Transcribing Handwritten Documents |
| 196 | An Analysis of Adaptive Recursive Smoothing with Applications to Noise PSD Estimation |
| 197 | An Instrumental Quality Measure for Artificially Bandwidth-Extended Speech Signals |
| 198 | Feedback Delay Networks: Echo Density and Mixing Time |
| 199 | Estimating the Structural Segmentation of Popular Music Pieces Under Regularity Constraints |

| | |
|---|---|
| 200 | Combined LCMV-TRINICON beamforming for separating multiple speech sources in noisy  and reverberant environments |
| 201 | Gain Scheduling of Auxiliary Noise and Variable Step-Size for Online Acoustic Feedback Cancellation in Narrow-Band Active Noise Control Systems |
| 202 | Estimating the Direct-to-Reverberant Energy Ratio Using a Spherical Harmonics-Based Spatial Correlation Model |
| 203 | Acoustic Reflector Localization: Novel Image Source Reversion and Direct Localization Methods |
| 204 | Weakly Supervised Learning of Hidden Markov Models for Spoken Language Acquisition |
| 205 | Online Estimation of Multiple Harmonic Signals |
| 206 | Online Estimation of Multiple Harmonic Signals |
| 207 | Online Estimation of Multiple Harmonic Signals |
| 208 | Online Estimation of Multiple Harmonic Signals |
| 209 | On-the-Fly Audio Source Separation—A Novel User-Friendly Framework |
| 210 | Multitask learning of context-dependent  targets in deep neural network acoustic models |
| 211 | Topic-Aware  Deep Compositional Models for Sentence Classification |
| 212 | Direct-to-Reverberant Energy Ratio Estimation Using a First-Order Microphone |

| | |
|---|---|
| 213 | Direct-to-Reverberant Energy Ratio Estimation Using a First-Order Microphone |
| 214 | Deep Convolutional Neural Networks for Predominant Instrument Recognition in Polyphonic Music |
| 215 | Speech Intelligibility Potential of General and Specialized Deep Neural Network  based Speech Enhancement Systems |
| 216 | Score-Informed Analysis of Tuning, Intonation, Pitch Modulation, and Dynamics in Jazz Solos |
| 217 | Direction of Arrival Estimation in the Spherical Harmonic  Domain  using Subspace Pseudo-Intensity Vectors |
| 218 | Mispronunciation  Detection and Diagnosis in L2 English Speech Using Multi-Distribution Deep Neural Networks |
| 219 | Relaxed Binaural LCMV Beamforming |
| 220 | Score-Informed Analysis of Tuning, Intonation, Pitch Modulation, and Dynamics in Jazz Solos |
| 221 | Speech Intelligibility Potential of General and Specialized Deep Neural Network  based Speech Enhancement Systems |
| 222 | Relaxed Binaural LCMV Beamforming |
| 223 | Relaxed Binaural LCMV Beamforming |
| 224 | Relaxed Binaural LCMV Beamforming |

| 225 | Single Sideband Frequency Offset Estimation and Correction for Quality Enhancement and Speaker Recognition |
|-----|------------------------------------------------------------|
| 226 | Modeling Latent Topics and Temporal Distance for Story Segmentation of Broadcast News |
| 227 | A Reverberation-Time-Aware Approach to Speech Dereverberation Based on Deep Neural Networks |
| 228 | A Reverberation-Time-Aware Approach to Speech Dereverberation  Based on Deep Neural Networks |
| 229 | Phone Synchronous Speech Recognition With CTC Lattices |
| 230 | Robust Downbeat Tracking Using an Ensemble of Convolutional Networks |