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Automatic speech recognition

1-1-2-ASR: ASBERT: ASR-SPECIFIC SELF-SUPERVISED LEARNING WITH SELF-TRAINING 9 *Hyung Yong Kim, Byeong-Yeol Kim, Seung Woo Yu, Youshin Lim, Yunkyu Lim, Hanbin Lee, 42dot, Korea*

1-1-3-ASR: SUB-8-BIT QUANTIZATION FOR ON-DEVICE SPEECH RECOGNITION: A **15** REGULARIZATION-FREE APPROACH

Kai Zhen, Martin Radfar, Hieu D Nguyen, Grant Strimel, Athanasios Mouchtaris, Nathan Susanj, Amazon, United States

Yuan Wang, Bhuvana Ramabhadran, Pedro Moreno, Google, United States; Ekin D Cubuk, Quoc Le, Daniel S Park, Google Brain, United States; Andrew Rosenberg, Google LLC, United States; Shuyang Cheng, Waymo LLC, United States; Ron J Weiss, Google, Inc., United States

David Qiu, Tsendsuren Munkhdalai, Yanzhang He, Google, United States; Khe C Sim, Google Inc., United States

Antoine Bruguier, David Qiu, Trevor Strohman, Yanzhang He, Google, United States

1-2-3-ASR: A CONTEXT-AWARE KNOWLEDGE TRANSFERRING STRATEGY FOR CTC-BASED . **60** ASR

Ke-Han Lu, Kuan-Yu Chen, National Taiwan University of Science and Technology, Taiwan

Zhehuai Chen, Yu Zhang, Bhuvana Ramabhadran, Pedro Moreno, Nanxin Chen, Google, United States; Ankur Bapna, Google Research, United States; Andrew Rosenberg, Google LLC, United States

1-2-6-ASR: ALTERNATE INTERMEDIATE CONDITIONING WITH SYLLABLE-LEVEL AND76 CHARACTER-LEVEL TARGETS FOR JAPANESE ASR

Yusuke Fujita, Tatsuya Komatsu, LINE Corporation, Japan; Yusuke Kida, LINE Corp, Japan

Kwangyoun Kim, Felix Wu, Jing Pan, Prashant Sridhar, Kyu Jeong Han, ASAPP, United States; Yifan Peng, Shinji Watanabe, Carnegie Mellon University, United States

1-2-8-ASR: CONFORMER-BASED ON-DEVICE STREAMING SPEECH RECOGNITION WITH 92 KD COMPRESSION AND TWO-PASS ARCHITECTURE

Jinhwan Park, Junmo Park, Dhairya Sandhyana, Samsung Research, Korea; Sichen Jin, Samsung, Korea; Sungsoo Kim, Changheon Lee, Myoungji Han, Jungin Lee, Seokyeong Jung, Chanwoo Kim, Samsung Electronics, United States; Chang Woo Han, Samsung Reserch, Korea

1-2-9-ASR: ACCELERATOR-AWARE TRAINING FOR TRANSDUCER-BASED SPEECH100 RECOGNITION

Rupak Vignesh Swaminathan, Suhaila Mumtaj Shakiah, Hieu D Nguyen, Raviteja Chinta, Tariq Afzal, Nathan Susanj, Athanasios Mouchtaris, Grant Strimel, Amazon, United States; Ariya Rastrow, Amazon Alexa, United States

2-1-1-ASR: UNTIED POSITIONAL ENCODINGS FOR EFFICIENT TRANSFORMER-BASED108 SPEECH RECOGNITION

Lahiru T Samarakoon, Ivan Fung, Fano Labs, Hong Kong, Hong Kong SAR

2-1-2-ASR: MATCH TO WIN: ANALYSING SEQUENCES LENGTHS FOR EFFICIENT115 SELF-SUPERVISED LEARNING IN SPEECH AND AUDIO

Yan Gao, Pedro Gusmao, University of Cambridge, United Kingdom; Javier Fernandez-Marques, Samsung AI, Cambridge, United Kingdom; Titouan Parcollet, Avignon University, United Kingdom; Nicholas Lane, University of Cambridge and Samsung AI, United Kingdom

2-1-4-ASR: DAMAGE CONTROL DURING DOMAIN ADAPTATION FOR TRANSDUCER BASED 130 AUTOMATIC SPEECH RECOGNITION

Somshubra Majumdar, Shantanu Acharya, Vitaly Lavrukhin, Boris Ginsburg, NVIDIA, United States

2-1-5-ASR: PADA: PRUNING ASSISTED DOMAIN ADAPTATION FOR SELF-SUPERVISED136 SPEECH REPRESENTATIONS

Vasista Sai Lodagala, Indian Institute of Technology, Madras, India; Sreyan Ghosh, University of Maryland, College Park, United States; Srinivasan Umesh, IIT Chennai, India

2-1-6-ASR: MFCCA: MULTI-FRAME CROSS-CHANNEL ATTENTION FOR MULTI-SPEAKER144 ASR IN MULTI-PARTY MEETING SCENARIO

Fan Yu, Pengcheng Guo, Yuhao Liang, Lei Xie, Northwestern Polytechnical University, China; Shiliang Zhang, Alibaba Group, China; Zhihao Du, Speech Lab, Alibaba Group, China; Yuxiao Lin, Zhejiang University, China

2-1-9-ASR: RESIDUAL ADAPTERS FOR TARGETED UPDATES IN RNN-TRANSDUCER BASED .160 SPEECH RECOGNITION SYSTEM

Sungjun Han, University of Stuttgart, Germany; Deepak Baby, Valentin Mendelev, Amazon Alexa, Germany

2-2-1-ASR: IMPROVED NOISY ITERATIVE PSEUDO-LABELING FOR SEMI-SUPERVISED167 SPEECH RECOGNITION

Tian Li, Qingliang Meng, Yujian Sun, Shumei AI Research Institute, China

2-2-2-ASR: GUIDED CONTRASTIVE SELF-SUPERVISED PRE-TRAINING FOR AUTOMATIC174 SPEECH RECOGNITION

Aparna Khare, Amazon, United States; Minhua Wu, Jasha Droppo, Roland Maas, Amazon Inc., United States; Saurabhchand Bhati, Johns Hopkins University, United States

2-2-4-ASR: NAM + : TOWARDS SCALABLE END-TO-END CONTEXTUAL BIASING FOR190 ADAPTIVE ASR

Zelin Wu, Jiayang Li, Pat Rondon, Google LLC, United States; Tsendsuren Munkhdalai, Golan Pundak, Tara Sainath, Google, United States; Khe C Sim, Google Inc., United States

2-2-7-ASR: HOW DOES PRE-TRAINED WAV2VEC 2.0 PERFORM ON DOMAIN-SHIFTED ASR? 205

AN EXTENSIVE BENCHMARK ON AIR TRAFFIC CONTROL COMMUNICATIONS Juan Pablo Zuluaga Gomez, Amrutha Prasad, Iuliia Nigmatulina, Seyyed Saeed Sarfjoo, Idiap Research Institute, Switzerland; Petr Motlicek, Idiap, Switzerland; Matthias Kleinert, Hartmut Helmke, Oliver Ohneiser, DLR, Germany; Qingran Zhan, Beijing Institute of Technology, China

2-2-9-ASR: INTERNAL LANGUAGE MODEL PERSONALIZATION OF E2E AUTOMATIC SPEECH ... 213

RECOGNITION USING RANDOM ENCODER FEATURES

Adam Stooke, Mason Chua, Tsendsuren Munkhdalai, Trevor Strohman, Google, United States; Khe C Sim, Google Inc., United States

3-1-1-ASR: TOWARDS END-TO-END UNSUPERVISED SPEECH RECOGNITION......221 Alexander H Liu, MIT, United States; Wei-Ning Hsu, Massachusetts Institute of Technology, United States; Michael Auli, Facebook, United States; Alexei Baevski, Facebook AI Research, United States

3-1-3-ASR: MONOTONIC SEGMENTAL ATTENTION FOR AUTOMATIC SPEECH229 RECOGNITION

Albert Zeyer, Robin Schmitt, Wei Zhou, Ralf Schlüter, Hermann Ney, RWTH Aachen University, Germany

3-1-4-ASR: STREAMING, FAST AND ACCURATE ON-DEVICE INVERSE TEXT237 NORMALIZATION FOR AUTOMATIC SPEECH RECOGNITION

Yashesh Gaur, Nick Kibre, Kangyuan Shu, Yuhui Wang, Issac Alphonso, Jinyu Li, Yifan Gong, Microsoft, United States; Jian Xue, Microsoft Corporation, United States

3-1-5-ASR: DUAL LEARNING FOR LARGE VOCABULARY ON-DEVICE ASR.......245 *Charles C Peyser, Google Inc., United States; W. Ronny Huang, Tara Sainath, Rohit Prabhavalkar, Google, United States; Michael Picheny, NYU, United States; Kyunghyun Cho, New York University, United States*

3-1-6-ASR: STREAMING BILINGUAL END TO END ASR MODEL USING ATTENTION OVER252 MULTIPLE SOFTMAX

Aditya R Patil, Vikas V Joshi, Purvi Agrawal, Rupesh Mehta, Microsoft, Australia

3-1-7-ASR: END-TO-END INTEGRATION OF SPEECH RECOGNITION,260 DEREVERBERATION, BEAMFORMING, AND SELF-SUPERVISED LEARNING REPRESENTATION

Yoshiki Masuyama, Nobutaka Ono, Tokyo Metropolitan University, Japan; Xuankai Chang, Shinji Watanabe, Carnegie Mellon University, United States; Samuele Cornell, Università Politecnica delle Marche, Italy

3-1-8-ASR: FULLY UNSUPERVISED TRAINING OF FEW-SHOT KEYWORD SPOTTING......266 *Minchan Kim, Dongjune Lee, Sung Hwan Mun, Min Hyun Han, Nam Soo Kim, Seoul National University, Korea*

3-1-9-ASR: LEARNING A DUAL-MODE SPEECH RECOGNITION MODEL VIA SELF-PRUNING ..273 *Chunxi Liu, Yuan Shangguan, Ozlem Kalinli, Meta AI, United States; Haichuan Yang, Meta, United States; Yangyang Shi, Raghuraman Krishnamoorthi, Facebook, United States*

4-1-1-ASR: INTER-KD: INTERMEDIATE KNOWLEDGE DISTILLATION FOR CTC-BASED**280** AUTOMATIC SPEECH RECOGNITION

Ji Won Yoon, Beom Jun Woo, Sunghwan Ahn, Hyeonseung Lee, Nam Soo Kim, Seoul National University, Korea

4-1-2-ASR: HMM VS. CTC FOR AUTOMATIC SPEECH RECOGNITION: COMPARISON BASED .287 ON FULL-SUM TRAINING FROM SCRATCH

Tina Raissi, Wei Zhou, Simon Berger, Ralf Schlüter, Hermann Ney, RWTH Aachen University, Germany

4-1-3-ASR: DOMAIN ADAPTATION OF LOW-RESOURCE TARGET-DOMAIN MODELS USING .295 WELL-TRAINED ASR CONFORMER MODELS

Vrunda N Sukhadia, Indian Institute Of Technology Madras, India; Srinivasan Umesh, IIT Chennai, India

4-1-6-ASR: UNIFIED END-TO-END SPEECH RECOGNITION AND ENDPOINTING FOR FAST ...310 AND EFFICIENT SPEECH SYSTEMS

Shaan Bijwadia, Shuo-yiin Chang, Tara Sainath, Bo Li, Chao Zhang, Yanzhang He, Google, United States

4-1-7-ASR: LEARNING MASK SCALARS FOR IMPROVED ROBUST AUTOMATIC SPEECH317 RECOGNITION

Arun Narayanan, Google Inc., United States; James Walker, Nathan Howard, Google Llc., United States; Sankaran Panchapagesan, Google, LLC, United States; Yuma Koizumi, Google, Japan

4-1-8-ASR: AN INVESTIGATION OF MONOTONIC TRANSDUCERS FOR LARGE-SCALE**324** AUTOMATIC SPEECH RECOGNITION

Niko Moritz, Frank Seide, Duc Le, Meta, United Kingdom; Jay Mahadeokar, Meta AI, United States; Christian Fuegen, Facebook, United Kingdom

4-1-9-ASR: MACRO-BLOCK DROPOUT FOR IMPROVED REGULARIZATION IN TRAINING331 END-TO-END SPEECH RECOGNITION MODELS

Chanwoo Kim, Samsung Electronics, Korea; Sathish Indurti, Jinhwan Park, Samsung Research, Korea; Wonyong Sung, Seoul national university, Korea

Spoken language processing

1-1-10-SLP: AUTOMATIC RATING OF SPONTANEOUS SPEECH FOR LOW-RESOURCE339 LANGUAGES

Yaroslav Getman, Ragheb Al-Ghezi, Ekaterina Voskoboinik, Mikko Kurimo, Aalto University, Finland; Mittul Singh, Silo AI, Finland

1-1-11-SLP: MIXTURE OF DOMAIN EXPERTS FOR LANGUAGE UNDERSTANDING: AN**346** ANALYSIS OF MODULARITY, TASK PERFORMANCE, AND MEMORY TRADEOFFS Benjamin Kleiner, AWS AI Labs, United States; Jack FitzGerald, Amazon Alexa Artificial Intelligence, United States; Haidar Khan, Gokhan Tur, Amazon Alexa AI, United States

Anupama Chingacham, Dietrich Klakow, Saarland University, Germany; Vera Demberg, Dept. of Mathematics and Computer Science, Saarland University, Germany

1-2-11-SLP: ON THE USE OF SEMANTICALLY-ALIGNED SPEECH REPRESENTATIONS FOR ...361 SPOKEN LANGUAGE UNDERSTANDING

Gaëlle Laperrière, Mickael Rouvier, Yannick Estève, LIA - Avignon University, France; Valentin Pelloin, LIUM, Le Mans Université, France; Themos Stafylakis, Omilia - Conversational Intelligence, Greece

2-1-10-SLP: RESPONSE TIMING ESTIMATION FOR SPOKEN DIALOG SYSTEMS BASED ON ...369 SYNTACTIC COMPLETENESS PREDICTION

Jin Sakuma, Tetsunori Kobayashi, Waseda University, Japan; Shinya Fujie, Chiba Institute of Technology, Japan

2-2-10-SLP: BUILDING MARKOVIAN GENERATIVE ARCHITECTURES OVER PRETRAINED ...382 LM BACKBONES FOR EFFICIENT TASK-ORIENTED DIALOG SYSTEMS Hong Liu, Yucheng Cai, Zhijian Ou, Tsinghua University, China; Yi Huang, Junlan Feng, China Mobile Research, China

2-2-11-SLP: NON-AUTOREGRESSIVE END-TO-END APPROACHES FOR JOINT AUTOMATIC ...390 SPEECH RECOGNITION AND SPOKEN LANGUAGE UNDERSTANDING

Mohan Li, Toshiba Europe Ltd, United Kingdom; Rama S Doddipatla, Toshiba Europe LTD, United Kingdom

3-1-11-SLP: A STUDY ON THE INTEGRATION OF PRE-TRAINED SSL, ASR, LM AND SLU406 MODELS FOR SPOKEN LANGUAGE UNDERSTANDING

Yifan Peng, Siddhant Arora, Yushi Ueda, Sujay Kumar, Karthik Ganesan, Siddharth Dalmia, Xuankai Chang, Shinji Watanabe, Carnegie Mellon University, United States; Yosuke Higuchi, Waseda University, Japan

4-1-10-SLP: ON THE EFFICIENCY OF INTEGRATING SELF-SUPERVISED LEARNING AND414 META-LEARNING FOR USER-DEFINED FEW-SHOT KEYWORD SPOTTING *Yuan-Kuei Wu, Wei-Tsung Kao, Hung-yi Lee, National Taiwan University, Taiwan; Chia-Ping Chen, Zhi-Sheng Chen, Yu-Pao Tsai, intelliGo Technology inc., Taiwan*

Speech enhancement and separation

1-1-12-SES: MULTI-STAGE PROGRESSIVE AUDIO BANDWIDTH EXTENSION422 Liang Wen, Samsung electronics, China; Lizhong Wang, Samsung, China; Ying Zhang, Kwang Pyo Choi, Samsung Electronics, China

1-2-12-SES: SPATIAL-DCCRN: DCCRN EQUIPPED WITH FRAME-LEVEL ANGLE FEATURE436 AND HYBRID FILTERING FOR MULTI-CHANNEL SPEECH ENHANCEMENT Shubo Lv, Shaanxi Provincial Key Laboratory of Speech and Image Information Processing, School of Computer Science, Northwestern Polytechnical University, China; Yihui Fu, Yukai Ju, Lei Xie, Northwestern Polytechnical University, China; Weixin Zhu, Wei Rao, Yannan Wang, Tencent, China

1-2-13-SES: IMPROVED NORMALIZING FLOW-BASED SPEECH ENHANCEMENT USING AN ...444 ALL-POLE GAMMATONE FILTERBANK FOR CONDITIONAL INPUT REPRESENTATION *Martin Strauss, Matteo Torcoli, Bernd Edler, International Audio Laboratories Erlangen, Germany*

2-1-13-SES: ADAPTIVE-FSN: INTEGRATING FULL-BAND EXTRACTION AND ADAPTIVE458 SUB-BAND ENCODING FOR MONAURAL SPEECH ENHANCEMENT Yu-Sheng Tsao, Berlin Chen, National Taiwan Normal University, Taiwan; Kuan-Hsun Ho, NTNU, Taiwan; Jeih-weih Hung, National Chi Nan University, Taiwan

2-1-26-SES: AVSE CHALLENGE: AUDIO-VISUAL SPEECH ENHANCEMENT CHALLENGE465

Andrea L Aldana, Edinburgh University, United Kingdom; Cassia Valentini, Ondrej Klejch, Peter Bell, University of Edinburgh, United Kingdom; Mandar Gogate, Kia K Dashtipour, Amir Hussain, Edinburgh Napier University, United Kingdom

2-2-12-SES: TEA-PSE 2.0: SUB-BAND NETWORK FOR REAL-TIME PERSONALIZED SPEECH ...472 ENHANCEMENT

Yukai Ju, Shimin Zhang, Lei Xie, Northwestern Polytechnical University, China; Wei Rao, Yannan Wang, Tao Yu, Shi-dong Shang, Tencent, China

2-2-13-SES: EEND-SS: JOINT END-TO-END NEURAL SPEAKER DIARIZATION AND SPEECH ...480 SEPARATION FOR FLEXIBLE NUMBER OF SPEAKERS

Soumi Maiti, CMU, United States; Yushi Ueda, Shinji Watanabe, Carnegie Mellon University, United States; Chunlei Zhang, Tencent AI Lab, United States; Meng Yu, Shixiong Zhang, Tencent, United States; Yong Xu, Tecent, United States

4-1-11-SES: END-TO-END MULTI-SPEAKER ASR WITH INDEPENDENT VECTOR ANALYSIS....496 *Robin Scheibler, LINE Corporation, Japan; Wangyou Zhang, Yanmin Qian, Shanghai Jiao Tong University, China; Xuankai Chang, Shinji Watanabe, Carnegie Mellon University, United States*

Speech analysis

1-1-14-ANA: LEARNING INVARIANT REPRESENTATION AND RISK MINIMIZED FOR509 UNSUPERVISED ACCENT DOMAIN ADAPTATION

Chendong Zhao, The Shenzhen International Graduate School, Tsinghua University, China, China; Jianzong Wang, Xiaoyang Qu, Ping An Technology (Shenzhen) Co., Ltd, China; Haoqian Wang, Tsinghua Shenzhen International Graduate School, Tsinghua University, China; Jing Xiao, Ping An Insurance (Group) Company of China, China

1-2-14-ANA: VSAMETER: EVALUATION OF A NEW OPEN-SOURCE TOOL TO MEASURE517 VOWEL SPACE AREA AND RELATED METRICS

Tianyu Cao, Laureano Moro-Velazquez, Jesús Villalba, Najim Dehak, Johns Hopkins University, United States; Piotr Żelasko, Meaning, United States

2-1-14-ANA: INVESTIGATING THE IMPORTANT TEMPORAL MODULATIONS FOR**525** DEEP-LEARNING-BASED SPEECH ACTIVITY DETECTION

Tyler Vuong, Nikhil Madaan, Rohan Panda, Richard M Stern, Carnegie Mellon University, United States

3-1-13-ANA: A MULTI-MODAL ARRAY OF INTERPRETABLE FEATURES TO EVALUATE532 LANGUAGE AND SPEECH PATTERNS IN DIFFERENT NEUROLOGICAL DISORDERS

Anna Favaro, Chelsie Motley, Tianyu Cao, Miguel Iglesias, Ankur Butala, Esther S. Oh, Jesús Villalba, Najim Dehak, Laureano Moro-Velazquez, Johns Hopkins University, United States; Robert Stevens, Johns Hopkins Hospital, United States

4-1-13-ANA: EFFICIENT DYNAMIC FILTER FOR ROBUST AND LOW COMPUTATIONAL540 FEATURE EXTRACTION

Donghyeon Kim, Korea university, Korea; Jeong-gi Kwak, Hanseok Ko, Korea University, Korea

Speaker and language recognition

1-2-15-SLR: FREQUENCY AND MULTI-SCALE SELECTIVE KERNEL ATTENTION FOR**548** SPEAKER VERIFICATION

Sung Hwan Mun, Min Hyun Han, Nam Soo Kim, Seoul National University, Korea; Jee-weon Jung, Naver Corporation, Korea

2-1-15-SLR: AN ATTENTION-BASED BACKEND ALLOWING EFFICIENT FINE-TUNING OF555 TRANSFORMER MODELS FOR SPEAKER VERIFICATION

Junyi Peng, Oldrich Plchot, Ladislav Mošner, Lukas Burget, Jan Cernocky, Brno University of Technology, Czechia; Themos Stafylakis, Omilia - Conversational Intelligence, Greece

2-2-14-SLR: FLOW-ER: A FLOW-BASED EMBEDDING REGULARIZATION STRATEGY FOR563 ROBUST SPEECH REPRESENTATION LEARNING

Woo Hyun Kang, Computer Research Institute of Montreal, Canada; Jahangir Alam, Computer Research Institute of Montreal (CRIM), Montreal (Quebec) Canada, Canada; Abderrahim Fathan, Computer Research Institute of Montreal (CRIM), Montreal, Quebec, Canada, Canada

2-2-15-SLR: UNSUPERVISED DOMAIN ADAPTATION OF NEURAL PLDA USING SEGMENT571 PAIRS FOR SPEAKER VERIFICATION

İsmail Rasim Ülgen, Sestek - Boğaziçi University, Turkey; Mustafa Levent Arslan, Sestek - Boğaziçi Üniversitesi, Turkey

3-1-14-SLR: THE CLEVER HANS EFFECT IN VOICE SPOOFING DETECTION**577** *Bhusan Chettri, Borac Solutions, India*

3-1-15-SLR: INVESTIGATING ACTIVE-LEARNING-BASED TRAINING DATA SELECTION FOR .585 SPEECH SPOOFING COUNTERMEASURE *Xin Wang, Junichi Yamagishi, National Institute of Informatics, Japan*

4-1-15-SLR: A COMPREHENSIVE STUDY ON SELF-SUPERVISED DISTILLATION FOR**599** SPEAKER REPRESENTATION LEARNING

Zhengyang Chen, Bing Han, Yanmin Qian, Shanghai Jiao Tong University, China; Yao Qian, Michael Zeng, Microsoft, United States

Speaker diarization

1-2-16-DIA: JOINT SPEAKER DIARISATION AND TRACKING IN SWITCHING STATE-SPACE ...605 MODEL

Jeremy H. M. Wong, Institute for Infocomm Research, Singapore; Yifan Gong, Microsoft, United States

2-1-16-DIA: DIARISATION USING LOCATION TRACKING WITH AGGLOMERATIVE613 CLUSTERING

Jeremy H. M. Wong, Institute for Infocomm Research, Singapore; Igor Abramovski, Xiong Xiao, Yifan Gong, Microsoft, United States

2-2-16-DIA: MUTUAL LEARNING OF SINGLE- AND MULTI-CHANNEL END-TO-END NEURAL 620 DIARIZATION

Shota Horiguchi, Yuki Takashima, Hitachi, Ltd., Japan; Shinji Watanabe, Carnegie Mellon University, United States; Paola Garcia, Johns Hopkins University, United States

2-2-5-DIA: CONTINUAL SELF-SUPERVISED DOMAIN ADAPTATION FOR END-TO-END626 SPEAKER DIARIZATION

Juan Manuel Coria, Sahar Ghannay, Université Paris-Saclay CNRS, LISN, France; Hervé Bredin, CNRS, France; Sophie Rosset, LISN, France

3-1-16-DIA: BERTRAFFIC: BERT-BASED JOINT SPEAKER ROLE AND SPEAKER CHANGE633 DETECTION FOR AIR TRAFFIC CONTROL COMMUNICATIONS

Juan Pablo Zuluaga Gomez, Seyyed Saeed Sarfjoo, Amrutha Prasad, Iuliia Nigmatulina, Idiap Research Institute, Switzerland; Petr Motlicek, Idiap, Switzerland; Karel Ondrej, BUT, Czechia; Oliver Ohneiser, Hartmut Helmke, DLR, Germany

4-1-16-DIA: LOW-LATENCY SPEECH SEPARATION GUIDED DIARIZATION FOR TELEPHONE 641 CONVERSATIONS

Giovanni Morrone, Samuele Cornell, Luca Serafini, Stefano Squartini, Università Politecnica delle Marche, Italy; Desh Raj, Johns Hopkins University, United States; Enrico Zovato, PerVoice S.p.A., Italy; Alessio Brutti, FBK, Italy

Text-only language processing

1-1-17-TLP: FINE GRAINED SPOKEN DOCUMENT SUMMARIZATION THROUGH TEXT647 SEGMENTATION

Samantha Kotey, Naomi Harte, Trinity College Dublin, Ireland; Rozenn Dahyot, Maynooth University, Ireland

1-2-17-TLP: AN ANALYSIS OF THE EFFECTS OF DECODING ALGORITHMS ON FAIRNESS655 IN OPEN-ENDED LANGUAGE GENERATION

Jwala Dhamala, Amazon Alexa AI, United States; Varun Kumar, Amazon Alexa, United States; Rahul Gupta, Amazon, United States; Kai-Wei Chang, UCLA, United States; Aram Galstyan, USC Information Sciences Institute, United States

3-1-17-TLP: EFFICIENT TEXT ANALYSIS WITH PRE-TRAINED NEURAL NETWORK671 MODELS

Jia Cui, Shiyin Kang, Liqiang He, Guangzhi Li, Tencent, United States; Heng Lu, Dong Yu, Tencent AI Lab, China; Wenjie Wang, Emory University, United States

3-1-24-TLP: FOUR-IN-ONE: A JOINT APPROACH TO INVERSE TEXT NORMALIZATION,677 PUNCTUATION, CAPITALIZATION, AND DISFLUENCY FOR AUTOMATIC SPEECH RECOGNITION Sharman W Tan, Piyush Behre, Nick Kibre, Issac Alphonso, Shawn Chang, Microsoft, United States

4-1-17-TLP: EMPIRICAL ANALYSIS OF TRAINING STRATEGIES OF TRANSFORMER-BASED ...685 JAPANESE CHIT-CHAT SYSTEMS

Hiroaki Sugiyama, Masahiro Mizukami, Tsunehiro Arimoto, Hiromi Narimatsu, Yuya Chiba, Hideharu Nakajima, Toyomi Meguro, NTT, Japan

Multimodal speech processing

1-1-18-MMP: PUSH-PULL: CHARACTERIZING THE ADVERSARIAL ROBUSTNESS FOR692 AUDIO-VISUAL ACTIVE SPEAKER DETECTION *Xuanjun Chen, Haibin Wu, Hung-yi Lee, Roger Jang, National Taiwan University, China; Helen Meng, The Chinese University of Hong Kong, Hong Kong SAR*

1-1-19-MMP: TOWARDS VISUALLY PROMPTED KEYWORD LOCALISATION FOR700 ZERO-RESOURCE SPOKEN LANGUAGES *Leanne Nortje, Herman Kamper, Stellenbosch University, South Africa*

1-2-18-MMP: EXPLOITING INFORMATION FROM NATIVE DATA FOR NON-NATIVE708 AUTOMATIC PRONUNCIATION ASSESSMENT

Binghuai Lin, MIG, Tencent Science and Technology Ltd., China; Liyuan Wang, Tencent Technology Co., Ltd, China

2-1-18-MMP: SPEECHCLIP: INTEGRATING SPEECH WITH PRE-TRAINED VISION AND715 LANGUAGE MODEL

Yi-Jen Shih, Hung-yi Lee, National Taiwan University, Taiwan; Hsuan-Fu Wang, Academia Sinica, Taiwan; Heng-Jui Chang, Massachusetts Institute of Technology, United States; Layne Berry, University of Texas at Austin, United States; David Harwath, The University of Texas at Austin, United States

2-1-8-MMP: TRANSFORMER-BASED LIP-READING WITH REGULARIZED DROPOUT AND723 RELAXED ATTENTION

Zhengyang Li, Timo Lohrenz, Matthias Dunkelberg, Tim Fingscheidt, Technische Universität Carolo-Wilhelmina Braunschweig, Germany

2-2-18-MMP: YFACC: A YORÙBÁ SPEECH-IMAGE DATASET FOR CROSS-LINGUAL KEYWORD ... 731

LOCALISATION THROUGH VISUAL GROUNDING

Kayode K Olaleye, University of Stellenbosch, South Africa; Dan Oneață, University Politehnica of Bucharest, Romania; Herman Kamper, Stellenbosch University, South Africa

3-1-18-MMP: ON THE USE OF MODALITY-SPECIFIC LARGE-SCALE PRE-TRAINED**739** ENCODERS FOR MULTIMODAL SENTIMENT ANALYSIS

Atsushi Ando, Ryo Masumura, Naoki Makishima, Keita Suzuki, Takafumi Moriya, Takanori Ashihara, Hiroshi Sato, NTT Corporation, Japan; Akihiko Takashima, NTT, Japan; Satoshi Suzuki, NTT Computer and Data Science Laboratories / The University of Electro-Communications, Japan

4-1-18-MMP: AN ANALYSIS OF SEMANTICALLY-ALIGNED SPEECH-TEXT EMBEDDINGS......747 *Muhammad Huzaifah, Ivan Kukanov, Institute for Infocomm Research, ASTAR, Singapore*

Multilingual processing

1-1-1-MLP: EXPLORATION OF LANGUAGE-SPECIFIC SELF-ATTENTION PARAMETERS FOR ..755 MULTILINGUAL END-TO-END SPEECH RECOGNITION *Brady Houston, AWS AI Labs, United States; Katrin Kirchhoff, Amazon, United States*

1-1-5-MLP: HOW DO PHONOLOGICAL PROPERTIES AFFECT BILINGUAL AUTOMATIC**763** SPEECH RECOGNITION?

Shelly Jain, Aditya Yadavalli, International Institute of Information Technology, Hyderabad, India; Sai Ganesh Mirishkar, IIIT Hyderabad, India; Anil Vuppala, International Institute of Information Technology Hyderabad, India

1-2-19-MLP: TEXTUAL DATA AUGMENTATION FOR ARABIC-ENGLISH CODE-SWITCHING ...777 SPEECH RECOGNITION

Amir Hussein, Najim Dehak, Sanjeev Khudanpur, Johns Hopkins University, United States; Shammur Chowdhury, Ahmed Abdelali, QCRI, Qatar; Ahmed Ali, Qatar Computing Research Institute, HBKU, Qatar

1-2-5-MLP: IMPROVING LUXEMBOURGISH SPEECH RECOGNITION WITH792 CROSS-LINGUAL SPEECH REPRESENTATIONS *Le Minh Nguyen, Shekhar Nayak, Matt Coler, University of Groningen, Luxembourg*

Alexis Conneau, FAIR, France; Min Ma, Ankur Bapna, Google Research, United States; Simran Khanuja, Yu Zhang, Jason Riesa, Clara Rivera, Google, United States; Vera Axelrod, Google, Inc, United States; Siddharth Dalmia, Carnegie Mellon University, United States

2-2-19-MLP: MULTILINGUAL SPEECH EMOTION RECOGNITION WITH MULTI-GATING806 MECHANISM AND NEURAL ARCHITECTURE SEARCH

Zihan Wang, Qi Meng, Haifeng Lan, Xinrui Zhang, Kehao Guo, Columbia University, United States; Akshat Gupta, JPMorgan, United States

2-2-22-MLP: DISENTANGLED SPEECH REPRESENTATION LEARNING FOR ONE-SHOT814 CROSS-LINGUAL VOICE CONVERSION USING B-VAE

Hui Lu, Disong Wang, Xixin Wu, Xunying Liu, Helen Meng, The Chinese University of Hong Kong, Hong Kong SAR; Zhiyong Wu, Tsinghua University, China

2-2-8-MLP: IMPROVING SEMI-SUPERVISED E2E ASR USING CYCLEGAN AND822 INTER-DOMAIN LOSSES

Chia-Yu Li, Institute for Natural Language Processing (IMS), University of Stuttgart, Germany; Ngoc Thang Vu, University of Stuttgart, Germany

3-1-2-MLP: EXPLORING A UNIFIED ASR FOR MULTIPLE SOUTH INDIAN LANGUAGES830 LEVERAGING MULTILINGUAL ACOUSTIC AND LANGUAGE MODELS *C. S. Anoop, Indian Institute of Science, Bengaluru, India; A G Ramakrishnan, INDIAN INSTITUTE OF SCIENCE, India*

4-1-5-MLP: A TRULY MULTILINGUAL FIRST PASS AND MONOLINGUAL SECOND PASS838 STREAMING ON-DEVICE ASR SYSTEM

Sepand Mavandadi, Bo Li, Chao Zhang, Brian Farris, Tara Sainath, Trevor Strohman, Google, United States

Emotion recognition and paralinguistics

1-1-20-EMR: SPEECH EMOTION RECOGNITION WITH COMPLEMENTARY ACOUSTIC846 REPRESENTATIONS

Xiaoming Zhang, Nanjing University of Technology, China; Fan Zhang, IBM Massachusetts Labratory, United States; Xiaodong Cui, IBM T. J. Watson Research Center, United States; Wei Zhang, Wayfair, United States

Amruta Saraf, Ganesh Sivaraman, Elie Khoury, Pindrop, United States

2-1-19-EMR: DISTRIBUTION-BASED EMOTION RECOGNITION IN CONVERSATION860 *Wen Wu, Chao Zhang, University of Cambridge, United Kingdom; Phil Woodland, Machine Intelligence Laboratory, Cambridge University Department of Engineering, United Kingdom*

3-1-19-EMR: EXPLORATION OF A SELF-SUPERVISED SPEECH MODEL: A STUDY ON868 EMOTIONAL CORPORA

Yuanchao Li, Yumnah Mohamied, Peter Bell, Catherine Lai, University of Edinburgh, United Kingdom

Speech synthesis and spoken language generation

Agriculture and Technology, Japan

1-1-21-TTS: WAVEFIT: AN ITERATIVE AND NON-AUTOREGRESSIVE NEURAL VOCODER884 BASED ON FIXED-POINT ITERATION *Yuma Koizumi, Heiga Zen, Michiel Bacchiani, Google, Japan; Kohei Yatabe, Tokyo University of*

1-1-22-TTS: ON GRANULARITY OF PROSODIC REPRESENTATIONS IN EXPRESSIVE892 TEXT-TO-SPEECH *Mikolaj Babianski, Kamil Pokora, Raahil Shah, Rafał Sienkiewicz, Daniel Korzekwa, Viacheslav Klimkov, Amazon, Poland*

1-1-23-TTS: CAN WE USE COMMON VOICE TO TRAIN A MULTI-SPEAKER TTS SYSTEM?900 Sewade O Ogun, Emmanuel Vincent, Inria, France; Vincent Colotte, LORIA, France

1-2-21-TTS: GAN YOU HEAR ME? RECLAIMING UNCONDITIONAL SPEECH SYNTHESIS906 FROM DIFFUSION MODELS *Matthew Baas, Herman Kamper, Stellenbosch University, South Africa*

1-2-22-TTS: ANONYMIZING SPEECH WITH GENERATIVE ADVERSARIAL NETWORKS TO912 PRESERVE SPEAKER PRIVACY Saring Meyer, Pascal Tilli, Pavel Denisov, Florian Lux, Julia Koch, Ngoc Thang Vu, University of

Sarina Meyer, Pascal Tilli, Pavel Denisov, Florian Lux, Julia Koch, Ngoc Thang Vu, University of Stuttgart, Germany

2-1-20-TTS: STYLETTS-VC: ONE-SHOT VOICE CONVERSION BY KNOWLEDGE920 TRANSFER FROM STYLE-BASED TTS MODELS

Yinghao A Li, Nima Mesgarani, Columbia University, United States; Cong Han, Columbia University, United States

2-1-21-TTS: LEARNING ACCENT REPRESENTATION WITH MULTI-LEVEL VAE TOWARDS928 CONTROLLABLE SPEECH SYNTHESIS

Jan Melechovsky, Dorien Herremans, Singapore University of Technology and Design, Singapore; Ambuj Mehrish, SUTD, Singapore; Berrak Sisman, Singapore University of Technology and Design (SUTD), Singapore

2-1-22-TTS: VTTS: VISUAL-TEXT TO SPEECH......**936** *Yoshifumi Nakano, Takaaki Saeki, Shinnosuke Takamichi, Hiroshi Saruwatari, The University of Tokyo, Japan; Katsuhito Sudoh, Nara Institute of Science and Techonology, Japan*

2-2-20-TTS: GENERATIVE MODELS FOR IMPROVED NATURALNESS, INTELLIGIBILITY,943 AND VOICING OF WHISPERED SPEECH

Dominik Wagner, Sebastian P Bayerl, Technische Hochschule Nürnberg Georg Simon Ohm, Germany; Hector Cordourier, Intel, Mexico; Tobias Bocklet, TH Nürnberg, Germany

2-2-21-TTS: TWO-STAGE TRAINING METHOD FOR JAPANESE ELECTROLARYNGEAL949 SPEECH ENHANCEMENT BASED ON SEQUENCE-TO-SEQUENCE VOICE CONVERSION *Ding Ma, Lester Phillip G Violeta, Kazuhiro Kobayashi, Tomoki Toda, Nagoya University, Japan*

3-1-20-TTS: SIMD-SIZE AWARE WEIGHT REGULARIZATION FOR FAST NEURAL VOCODING 955 ON CPU

Hiroki Kanagawa, Yusuke Ijima, NTT Corporation, Japan

Florian Lux, Julia Koch, Ngoc Thang Vu, University of Stuttgart, Germany

Rendi Chevi, Radityo Eko Prasojo, Kata.ai, Indonesia; Alham Fikri Aji, Amazon, United Kingdom; Andros Tjandra, Meta AI, US, United States; Sakriani Sakti, Japan Advanced Institute of Science and Technology, Japan

4-1-20-TTS: REGOTRON: REGULARIZING THE TACOTRON2 ARCHITECTURE VIA**977** MONOTONIC ALIGNMENT LOSS

Efthymios Georgiou, Georgios Paraskevopoulos, Alexandros Potamianos, National Technical University of Athens, Greece; Kosmas Kritsis, Vassilis Katsouros, Athena Research Center, Greece; Athanasios Katsamanis, ATHENA R.C., Behavioral Signal Technologies, Greece

4-1-21-TTS: REMAP, WARP AND ATTEND: NON-PARALLEL MANY-TO-MANY ACCENT984 CONVERSION WITH NORMALIZING FLOWS

Abdelhamid Ezzerg, Thomas Merritt, Kayoko Yanagisawa, Piotr Bilinski, Kamil Pokora, Renard Korzeniowski, Roberto Barra-Chicote, Daniel Korzekwa, Amazon, United Kingdom; Magdalena Proszewska, Jagiellonian University, Poland

Resources (new corpora, toolkits, evaluation metrics, etc.)

1-2-23-RES: STOP: A DATASET FOR SPOKEN TASK ORIENTED SEMANTIC PARSING......991 Paden Tomasello, Akshat Shrivastava, Daniel A Lazar, Po-chun Hsu, Duc Le, Ali Elkahky, Jade Copet, Robin Algayres, Tu Anh Nguyen, Meta, United States; Adithya Sagar, Facebook AI, United States; Wei-Ning Hsu, Massachusetts Institute of Technology, United States; Yossi Adi, Emmanuel Dupoux, Facebook AI Research, Israel; Luke Zettlemoyer, Facebook, United States; Abdel-rahman Mohamed, Facebook AI Research (FAIR), United States

2-2-23-RES: BENCHMARKING EVALUATION METRICS FOR CODE-SWITCHING**999** AUTOMATIC SPEECH RECOGNITION

Injy Hamed, New York University Abu Dhabi, Stuttgart University, United Arab Emirates; Amir Hussein, Johns Hopkins University, United States; Oumnia Chellah, Stanford University, United States; Shammur Chowdhury, QCRI, Qatar; Hamdy Mubarak, Ahmed Ali, Qatar Computing Research Institute, HBKU, Qatar; Sunayana Sitaram, Microsoft Research, India; Nizar Habash, New York University Abu Dhabi, United Arab Emirates

Machine learning for speech applications

1-1-15-MLS: SPEED-ROBUST KEYWORD SPOTTING VIA SOFT SELF-ATTENTION ON1014 MULTI-SCALE FEATURES *Chaoyue Ding, Jiakui Li, Martin Zong, Baoxiang Li, SenseTime Group Limited, China*

1-1-24-MLS: DISTILLING SEQUENCE-TO-SEQUENCE VOICE CONVERSION MODELS**1022** FOR STREAMING CONVERSION APPLICATIONS

Kou Tanaka, NTT corpration, Japan; Hirokazu Kameoka, NTT Communication Science Laboratories, NTT Corporation, Japan; Takuhiro Kaneko, Shogo Seki, NTT Corporation, Japan

1-1-25-MLS: AUTOMATIC PREDICTION OF INTELLIGIBILITY OF WORDS AND PHONEMES 1029 PRODUCED ORALLY BY JAPANESE LEARNERS OF ENGLISH

Nobuaki Minematsu, Chuanbo Zhu, Takuya Kunihara, Daisuke Saito, The University of Tokyo, Japan; Noriko Nakanishi, Kobe Gakuin University, Japan

1-2-24-MLS: SVLDL: IMPROVED SPEAKER AGE ESTIMATION USING SELECTIVE1037 VARIANCE LABEL DISTRIBUTION LEARNING

Zuheng Kang, Jianzong Wang, Junqing Peng, Ping An Technology (Shenzhen) Co., Ltd, China; Jing Xiao, Ping An Insurance (Group) Company of China, China

1-2-25-MLS: PEPPANET: EFFECTIVE MISPRONUNCIATION DETECTION AND DIAGNOSIS ...1045 LEVERAGING PHONETIC, PHONOLOGICAL, AND ACOUSTIC CUES *Bi-Cheng Yan, Hsin-Wei Wang, Berlin Chen, National Taiwan Normal University, Taiwan*

2-1-24-MLS: IMPLICIT ACOUSTIC ECHO CANCELLATION FOR KEYWORD SPOTTING AND .1052 DEVICE-DIRECTED SPEECH DETECTION

Samuele Cornell, Università Politecnica delle Marche, Italy; Thomas Balestri, Thibaud Senechal, Amazon, United States

Suliang Bu, Tuo Zhao, Yunxin Zhao, University of Missouri, United States

2-2-24-MLS: PHONEME SEGMENTATION USING SELF-SUPERVISED SPEECH MODELS1067 Luke Strgar, University of Texas, Austin, United States; David Harwath, The University of Texas at Austin, United States

3-1-23-MLS: AN EXPERIMENTAL STUDY ON PRIVATE AGGREGATION OF TEACHER1074 ENSEMBLE LEARNING FOR END-TO-END SPEECH RECOGNITION

Chao-Han Huck Yang, Chin-hui Lee, Georgia Institute of Technology, United States; I-Fan Chen, Amazon Inc., United States; Andreas Stolcke, Amazon, United States; Sabato M Siniscalchi, Kore University of Enna, Italy

4-1-23-MLS: PHONE-LEVEL PRONUNCIATION SCORING FOR L1 USING1081 WEIGHTED-DYNAMIC TIME WARPING

Aghilas Sini, Antoine Perquin, Damien Lolive, Univ Rennes, CNRS, IRISA, France; Arnaud Delhay, IRISA, France

4-1-24-MLS: PROFICIENCY ASSESSMENT OF L2 SPOKEN ENGLISH USING WAV2VEC 2.0....1088 *Stefano Bannò, University of Trento, Italy; Marco Matassoni, Fondazione Bruno Kessler, Italy*

SUPERB challenge

4-2-1-SUP: SUPERB @ SLT 2022: CHALLENGE ON GENERALIZATION AND EFFICIENCY OF 1096 SELF-SUPERVISED SPEECH REPRESENTATION LEARNING

Tzu-hsun Feng, Shu-wen Yang, Tzu-Quan Lin, Kai-Wei Chang, Haibin Wu, Hung-yi Lee, National Taiwan University, Taiwan; Annie Dong, Shang-Wen Li, Meta, United States; Ching-Feng Yeh, Facebook, United States; Jiatong Shi, Xuankai Chang, Shinji Watanabe, Carnegie Mellon University, United States; Zili Huang, Johns Hopkins University, United States; Abdel-rahman Mohamed, Facebook AI Research (FAIR), United States

1-1-26-SUP: ON THE UTILITY OF SELF-SUPERVISED MODELS FOR PROSODY-RELATED1104 TASKS

Guan-Ting Lin, Chi Luen Feng, Wei-Ping Huang, Yuan Tseng, Chen An Li, Tzu-Han Lin, Hung-yi Lee, National Taiwan University, Taiwan; Nigel Ward, UTEP, United States

2-1-25-SUP: IMPROVING GENERALIZABILITY OF DISTILLED SELF-SUPERVISED SPEECH ...1112 PROCESSING MODELS UNDER DISTORTED SETTINGS

Kuan-Po Huang, Tsu-Yuan Hsu, Liang-Hsuan Tseng, Hung-yi Lee, National Taiwan University, Taiwan; Yu-kuan Fu, NTU, Taiwan; Fabian Alejandro Ritter Gutierrez, National University of Singapore, Singapore; Fan-Lin Wang, Academia Sinica, Taiwan; Yu Zhang, Google, United States

2-2-25-SUP: EXPLORING EFFICIENT-TUNING METHODS IN SELF-SUPERVISED SPEECH1120 MODELS

Zih-Ching Chen, Chin-Lun Fu, Chih Ying Liu, Hung-yi Lee, National Taiwan University, Taiwan; Shang-Wen Li, AWS AI, United States

3-1-25-SUP: ON COMPRESSING SEQUENCES FOR SELF-SUPERVISED SPEECH MODELS1128

Yen Meng, Hsuan-Jui Chen, Hung-yi Lee, National Taiwan University, Taiwan; Jiatong Shi, Shinji Watanabe, Carnegie Mellon University, United States; Paola Garcia, Johns Hopkins University, United States; Hao Tang, The University of Edinburgh, United Kingdom