

Bibliography

- Abdel-Hamid, O., Mohamed, A. R., Jiang, H., & Penn, G. (2012). Applying convolutional neural networks concepts to hybrid NN-HMM model for speech recognition. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 4277–4280.
- Abdelaziz, A. H., & Kolossa, D. (2016). General hybrid framework for uncertainty-decoding-based automatic speech recognition systems. *Speech Communication*, 79, 1–13.
- Aggarwal, R. K., & Dave, M. (2011). Acoustic modeling problem for automatic speech recognition system: advances and refinements (Part II). *International Journal of Speech Technology*, 14(4), 309–320.
- Aggarwal, R. K., & Dave, M. (2012). Integration of multiple acoustic and language models for improved Hindi speech recognition system. *International Journal of Speech Technology*, 15(2), 165–180.
- Aggarwal, R. K., & Dave, M. (2015). An Empirical Approach for Optimization of Acoustic Models in Hindi Speech Recognition Systems. *8th International Conference on Natural Language Processing (ICON)*, IIT Kharagpur, 1–9.
- Aggarwal, R. K. & Dave, M. (2013). Performance evaluation of sequentially combined heterogeneous feature streams for Hindi speech recognition system. *Telecommunication Systems*, 52(3), 1457–1466.
- Aggarwal, R. K. (2012). Improving Hindi speech recognition using filter bank optimization and acoustic model refinement. PhD Thesis, National Institute of Technology, Kurukshetra, Haryana, India.
- Agrawal, S. S., Sinha, S., Singh, P., & Olsen, J. O. (2012). Development of Text and Speech database for Hindi and Indian English specific to Mobile Communication environment. *Proceeding of International Conference on the Language Resources and Evaluation Conference*, 3415–3421.
- Alam, M. J., Kenny, P., & O’Shaughnessy, D. (2013). Low-variance Multitaper Mel-frequency Cepstral Coefficient Features for Speech and Speaker Recognition Systems. *Cognitive Computation*, 5(4), 533–544.
- Alam, M. J., Kinnunen, T., Kenny, P., & Ouellet, P. (2013). Multitaper MFCC and PLP features for speaker verification using i-vectors. *Speech Communication*, 55(2), 237–251.
- AlHanai, T., Hsu, W. N., & Glass, J. (2016). Development of the MIT ASR system for the 2016 Arabic multi-genre broadcast challenge. *IEEE Spoken Language Technology Workshop (SLT)*, 299–304.
- Allauzen, C., Riley, M., Schalkwyk, J., Skut, W., & Mohri, M. (2007). OpenFst: A general and efficient weighted finite-state transducer library. *International Conference on Implementation and Application of Automata*, 11–23.

- Arora, S., Saxena, B., Arora, K., & Agrawal, S. S. (2010). Hindi ASR for Travel Domain. *Proceedings of O-COCOSDA*, 7(6), 233-241.
- Atal, B. S. (1974). Effectiveness of linear prediction characteristics of the speech wave for automatic speaker identification and verification. *The Journal of the Acoustical Society of America*, 55(6), 1304–1312.
- Aubert, X. L. (2002). An overview of decoding techniques for large vocabulary continuous speech recognition. *Computer Speech & Language*, 16(1), 89–114.
- Baggenstoss, P. M. (2001). A modified Baum-Welch algorithm for hidden Markov models with multiple observation spaces. *IEEE Transactions of Speech and Audio Processing*, 9(4), 411–416.
- Bansal, S., Sharan, S., & Agrawal, S. S. (2015). Corpus design and development of an annotated speech database for Punjabi. *International Conference on Oriental COCOSDA held jointly with Conference on Asian Spoken Language Research and Evaluation (O-COCOSDA/CASLRE)*, 32–37.
- Beaufays, F., Sankar, A., Williams, S., & Weintraub, M. (2003). Learning linguistically valid pronunciations from acoustic data. *Eighth European Conference on Speech Communication and Technology*, 2593–2596.
- Bengio, Y., & Grandvalet, Y. (2004). No unbiased estimator of the variance of k-fold cross-validation. *Journal of Machine Learning Research*, 1089–1105.
- Bharali, S. S., & Kalita, S. K. (2015). A comparative study of different features for isolated spoken word recognition using HMM with reference to Assamese language. *International Journal of Speech Technology*, 18(4), 673–684.
- Bhowmik, T., Mukherjee, S., & Mandal, S. K. D. (2015). Detection of attributes for bengali phoneme in continuous speech using deep neural network. *2nd International Conference on Signal Processing and Integrated Networks (SPIN)*, 103–108.
- Bhuriyakorn, P., Punyabukkana, P., & Suchato, A. (2008). A genetic algorithm-aided hidden markov model topology estimation for phoneme recognition of thai continuous speech. *Ninth ACIS International Conference on Software Engineering, Artificial Intelligence, Networking, and Parallel/Distributed Computing, (SNPD)*, 475–480.
- Boe, O., Krstulovic, S., Charlet, D., Fohr, D., & Mella, O. (2006). Optimizing the coverage of a speech database through a selection of representative speaker recordings, 48, 1319–1348.
- Campbell, J. P. (1997). Speaker recognition: A tutorial. *Proceedings of the IEEE*, 85(9), 1437–1462.
- Chen, T. Y., Mei, X. D., Pan, J. S., & Sun, S. H. (2004). Optimization of HMM by the tabu search algorithm. *Journal of Information Science and Engineering*, 20(5), 949–957.
- Chen, X., & Cheng, J. (2014). Deep neural network acoustic modeling for native and non-native Mandarin speech recognition. *9th International Symposium on Chinese*

- Spoken Language Processing (ISCSLP), 6–9.
- Cheng, J., Chen, X., & Metallinou, A. (2015). Deep neural network acoustic models for spoken assessment applications. *Speech Communication*, 73, 14–27.
- Cheshomi, S., Rahati-Q, S., & Akbarzadeh-T, M. R. (2010). Hybrid of Chaos Optimization and Baum-Welch Algorithms for HMM Training in Continuous Speech Recognition. *International Conference on Intelligent Control and Information Processing (ICICIP)*, 83–87.
- Chiu, Y. H. B., & Stern, R. M. (2014). Minimum variance modulation filter for robust speech recognition. *International Conference on Acoustics, Speech and Signal Processing, (ICASSP)*, 3917–3920.
- Chourasia, V., Samudravijaya, K., & Chandwani, M. (2005). Phonetically Rich Hindi Sentence Corpus for Creation of Speech Database. *Proc. O-Cocosda*, 132–137.
- Chunwijitra, V., Chotimongkol, A., & Wutiwiwatchai, C. (2016). A hybrid input-type recurrent neural network for LVCSR language modeling. *Eurasip Journal on Audio, Speech, and Music Processing*, (1), 1–12.
- Crystal, D. (2000). *Language Death*. Cambridge University Press.
- Davis, K. H., Biddulph, R., & Balashek, S. (1952). Automatic recognition of spoken digits. *Journal of the Acoustical Society of America*, 24(6), 637–642.
- Davis, S. B., & Mermelstein, P. (1980). Comparison of parametric representations for monosyllabic word recognition in continuously spoken sentences. *IEEE Transactions on Acoustics, Speech, and Signal Processing*, 28(4), 357–366.
- Day, P., & Nandi, A. K. (2007). Robust text-independent speaker verification using genetic programming. *IEEE Transactions on Audio, Speech, and Language Processing*, 15(1), 285–295.
- Dempster, A. P., Laird, N. M., & Rubin, D. B. (1997). Maximum likelihood from incomplete data via the EM algorithm. *Journal of the Royal Statistical Society. Series B (Methodological)*, 1–38.
- Deng, L., & O’Shaughnessy, D. (2003). *Speech processing: a dynamic and optimization-oriented approach*. CRC Press.
- Deng, L., Droppo, J., & Acero, A. (2005). Dynamic compensation of HMM variances using the feature enhancement uncertainty computed from a parametric model of speech distortion. *IEEE Transactions on Speech and Audio Processing*, 13(3), 412–421.
- Deng, L., Hinton, G., & Kingsbury, B. (2013). New types of deep neural network learning for speech recognition and related applications: An overview. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 8599–8603.
- Dey, A., Zhang, W., & Fung, P. (2014). Acoustic modeling for hindi speech recognition in low-resource settings. *International Conference on Audio, Language and Image Processing (ICALIP)*, 891–894.
- Dey, A., Lalhminghlu, W., Sarmah, P., Samudravijaya, K., Prasanna, S. M., Sinha, R.,

- & Nirmala, S. R. (2018). Mizo Phone Recognition System. Interspeech 2018.
- Doddington, G. R. (1985). Speaker recognition – identifying people by their voices. *Proceedings of the IEEE*, 73(11), 1651–1664.
- Dong, Z., Guo, Y., & Zeng, J. (2011). Recurrent hidden Markov models using particle swarm optimization. *International Journal of Modelling, Identification and Control*, 14(4), 258–264.
- Dua, M., Aggarwal, R. K., & Biswas, M. (2017). Discriminative Training using Heterogeneous Feature Vector for Hindi Automatic Speech Recognition System. *International Conference on Computer and Applications (ICCA)*, 158–162.
- Dua, M., Aggarwal, R. K., Kadyan, V., & Dua, S. (2012a). Punjabi automatic speech recognition using HTK. *International Journal of Computer Science Issues*, 9(4), 359–364.
- Dua, M., Aggarwal, R. K., Kadyan, V., & Dua, S. (2012b). Punjabi speech to text system for connected words. *Fourth International Conference on Advances in Recent Technologies in Communication and Computing, (ARTCom)*, 206–209.
- Dua, M., Aggarwal, R. K., & Biswas, M. (2018). GFCC based discriminatively trained noise robust continuous ASR system for Hindi language. *Journal of Ambient Intelligence and Humanized Computing*, 1-14.
- Dutta, K., & Sarma, K. K. (2012). Multiple feature extraction for RNN-based assamese speech recognition for speech to text conversion application. *International Conference on Communications, Devices and Intelligent Systems (CODIS)*, 600–603.
- Engelbrecht, A. P. (2007). *Computational intelligence: an introduction*. John Wiley & Sons. Inc., West Sussex, England.
- Farooq, O. & Datta, S. (2010). Robustness evaluation of wavelet based features for continuous speech recognition. *International Journal of Intelligent Systems Technologies and Applications*, 9(1), 1–14.
- Figielska, E., & Kasprzak, W. (2008). An evolutionary programming based algorithm for HMM training. *Computational Intelligence: Methods and Applications*, 166–175.
- Forney, G. D. (1972). The Viterbi Algorithm. *Proceedings of the IEEE*, 61(3), 268–278.
- Furui, S. (1981). Cepstral Analysis technique for automatic speaker verification. *IEEE Transactions on Acoustics, Speech, and Signal Processing*, 29(2), 254–272.
- Furui, S. (1986). Speaker independent isolated word recognition using dynamix features of speech spectrum. *IEEE Transactions on Acoustics, Speech, and Signal Processing*, 34(1), 52–59.
- Gaida, C., Lange, P., Petrick, R., Proba, P., Malatawy, A., & Suendermann-Oeft, D. (2014). Comparing open-source speech recognition toolkits. Tech. Rep., DHBW Stuttgart.
- Gales, M. J. F. (1995). *Model-based techniques for noise robust speech recognition*. University of Cambridge Dissertation.

- Gautama, T., & Van Hulle, M. M. (1999). Self-Organized Feature Extraction Achieved with a Parameterized Filterbank. *Neural Processing Letters*, 10(2), 131–137.
- Gelas, H., Abate, S. T., Besacier, L., & Pellegrino, F. (2011). Quality assessment of crowdsourcing transcriptions for African languages. *Twelfth Annual Conference of the International Speech Communication Association*, 3065–3068.
- Gelly, G., & Gauvain, J. L. (2018). Optimization of RNN-Based Speech Activity Detection. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, 26(3), 646–656.
- Gemello, R., Mana, F., Scanzio, S., Laface, P., & De Mori, R. (2006). Adaptation of hybrid ANN/HMM models using linear hidden transformations and conservative training. *IEEE International Conference on Acoustics, Speech and Signal Processing, (ICASSP)*, 1189–1192.
- Gen, M., & Cheng, R. (2002). *Genetic algorithms and engineering optimization*.
- Georgescu, A. L., Cucu, H., & Burileanu, C. (2017). Speed's DNN Approach to Romanian Speech Recognition. *International Conference on Speech Technology and Human-Computer Dialogue (SpeD)*, 1–8.
- Ghaffarzadegan, S., Bořil, H., & Hansen, J. H. (2017). Deep neural network training for whispered speech recognition using small databases and generative model sampling. *International Journal of Speech Technology*, 20(4), 1063–1075.
- Ghai, W., & Singh, N. (2013). Phone based acoustic modeling for automatic speech recognition for Punjabi language. *Journal of Speech Sciences*, 1(3), 69–83.
- Godambe, T., & Samudravijaya, K. (2011). Speech Data Acquisition for voice based Agricultural Information Retrieval. *Proceeding of 39th All India DLA Conference, Punjabi University, Patiala, India*, 1–8.
- Goffin, V., Allauzen, C., Bocchieri, E., Hakkani-Tur, D., Ljolje, A., Parthasarathy, S., & Saraclar, M. (2005). The AT&T Watson speech recognizer. *IEEE International Conference on Acoustics, Speech, and Signal Processing, (ICASSP)*, 1033–1036.
- Golik, P., Tüske, Z., Schlüter, R., & Ney, H. (2015). Convolutional neural networks for acoustic modeling of raw time signal in LVCSR. *Proceedings of the Annual Conference of the International Speech Communication Association, INTERSPEECH*, 26–30.
- Grézl, F., Karafiat, M., & Janda, M. (2011). Study of probabilistic and bottle-neck features in multilingual environment. *IEEE Workshop on Automatic Speech Recognition and Understanding (ASRU)*, 359–364.
- Guglani, J., & Mishra, A. N. (2018). Continuous Punjabi speech recognition model based on Kaldi ASR toolkit. *International Journal of Speech Technology*, 21(2), 1–6.
- Hansson-Sandsten, M., & Sandberg, J. (2009). Optimal cepstrum estimation using multiple windows. *IEEE International Conference on Acoustics, Speech and Signal Processing, (ICASSP)*, 3077–3080.
- Hansson-Sandsten, M. (2013). Mean square error optimal weighting for multitaper

- cepstrum estimation. *EURASIP Journal on Advances in Signal Processing*, (1), 158, 1-12.
- Hansson, M., & Salomonsson, G. (1997). A multiple window method for estimation of peaked spectra. *IEEE Transactions on Signal Processing*, 45(3), 778–781.
- Harris, F. J. (1978). On the use of windows for harmonic analysis with the discrete Fourier transform. *Proceedings of the IEEE*, 51–83.
- Hasnat, M., Mowla, J., & Khan, M. (2007). Isolated and continuous bangla speech recognition: implementation, performance and application perspective. *Proceedings of International Symposium on Natural Language Processing (SNLP)*, 1–6.
- Hegde, S., Achary, K. K., & Shetty, S. (2012). Isolated word recognition for Kannada language using support vector machine. *Wireless networks and computational intelligence*, 262–269.
- Hermansky, H., & Morgan, N. (1994). RASTA Processing of Speech. *IEEE Transactions on Speech and Audio Processing*, 2(4), 578–589.
- Hermansky, H., Morgan, N., & Hirsch, H. G. (1993). Recognition of speech in additive and convolutional noise based on RASTA spectral processing. In *IEEE International Conference on Acoustics, Speech, and Signal Processing, (ICASSP)*, 83–86.
- Hermansky, H. (1990). Perceptual linear predictive (PLP) analysis of speech. *The Journal of the Acoustical Society of America*, 87(4), 1738–1752.
- Hinton, G., Deng, L., Yu, D., Dahl, G. E., Mohamed, A. R., Jaitly, N., & Kingsbury, B. (2012). Deep neural networks for acoustic modeling in speech recognition: The shared views of four research groups. *Signal Processing Magazine, IEEE*, 29(6), 82–97.
- Holland, J. H. (1975). *Adaptation in natural and artificial systems. Adaptation in Natural and Artificial Systems*. Ann Arbor, MI: University of Michigan Press, 439-444.
- Hong, Q. Y., & Kwong, S. (2003). A training method for hidden Markov model with maximum model distance and genetic algorithm. *International Conference on Neural Networks and Signal Processing*, 465–468.
- Hsu, W. N., Zhang, Y., & Glass, J. (2016). A prioritized grid long short-term memory rnn for speech recognition. *Spoken Language Technology Workshop (SLT)*, 467–473.
- Hu, Y., & Loizou, P. C. (2004). Speech enhancement based on wavelet thresholding the multitaper spectrum. *IEEE Transactions on Speech and Audio Processing*, 12(1), 59–67.
- Huang, L. X., Evangelista, G., & Zhang, X. Y. (2011). Adaptive bands filter bank optimized by genetic algorithm for robust speech recognition system. *Journal of Central South University of Technology*, 18(5), 1595–1601.
- Huang, X., Acero, A., Hon, H. W., & Reddy, R. (2001). *Spoken Language Processing : A Guide to Theory , Algorithm and System Development*. Prentice Hall PTR

Upper Saddle River, NJ, USA.

- Huemmer, C., Schwarz, A., Maas, R., Barfuss, H., Astudilló, R. F., & Kellermann, W. (2016). A new uncertainty decoding scheme for DNN-HMM hybrid systems with multichannel speech enhancement. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 5760–5764.
- Hung, J. W. (2004). Optimization of Filter-Bank to Improve The Extraction of MFCC Features in Speech Recognition. *International Symposium on Intelligent Multimedia, Video and Speech Processing*, 675–678.
- Hwang, J. N., & Li, H. (1992). Interactive query learning for isolated speech recognition. *Proceedings of the IEEE-SP Workshop*, 93–102.
- Imseng, D., Boulard, H., & Garner, P. N. (2012). Using KL-divergence and multilingual information to improve ASR for under-resourced languages. *IEEE International conference on acoustics, speech and signal processing*, 4869–4872.
- Imseng, D., Boulard, H., Dines, J., & Garner, P. N. (2011). Improving non-native ASR through stochastic multilingual phoneme space transformations. *Idiap*.
- Jayanna, H. S., & Prasanna, S. M. (2010). Limited data speaker identification. *Sadhana*, 35(5), 525–546.
- Jayanna, H. S. (2009). Limited data speaker recognition. PhD Thesis, Indian Institute Of Technology Guwahati.
- Jelinek, F. (1976). Continuous speech recognition by statistical methods. *Proceedings of the IEEE*, 64(4), 532–556.
- Jiang, W., Chen, Y., Jin, H., Zheng, R., & Chi, Y. (2016). A novel GPU-based efficient approach for convolutional neural networks with small filters. *Journal of Signal Processing Systems*, 86(2–3), 313–325.
- Jiang, H. (2010). Discriminative training of HMMs for automatic speech recognition: A survey. *Computer Speech and Language*, 24(4), 589–608.
- Johansen, F. T. (2007). A comparison of hybrid HMM architecture using global discriminating training. *Fourth International Conference on Spoken Language, (ICSLP)*, 498–501.
- Karanasou, P., Wu, C., Gales, M., & Woodland, P. C. (2017). I-vectors and structured neural networks for rapid adaptation of acoustic models. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, 25(4), 818–828.
- Kay, S. M. (1988). *Modern spectral estimation*. Pearson Education India.
- Khelifa, M. O., Elhadj, Y. M., Abdellah, Y., & Belkasmi, M. (2017). Constructing accurate and robust HMM/GMM models for an Arabic speech recognition system. *International Journal of Speech Technology*, 20(4), 937–949.
- Kim, C., & Stern, R. M. (2016). Power-normalized cepstral coefficients (PNCC) for robust speech recognition. *IEEE Trans on Audio, Speech, and Language Processing*, 24(7), 1315–1329.
- Kingsbury, B. (2009). Lattice-based optimization of sequence classification criteria for neural-network acoustic modeling. *IEEE International Conference on Acoustics,*

- Speech and Signal Processing, (ICASSP), 3761–3764.
- Kinnunen, T., Saeidi, R., Sandberg, J., & Hansson-Sandsten, M. (2010). What else is new than the Hamming window? Robust MFCCs for speaker recognition via multitapering. *Interspeech*, 2734–2737.
- Kipyatkova, I., & Karpov, A. (2015). Recurrent neural network-based language modeling for an automatic Russian speech recognition system. *Artificial Intelligence and Natural Language and Information Extraction, Social Media and Web Search FRUCT Conference (AINL-ISMW FRUCT)*, 33–38.
- Kipyatkova, I., & Karpov, A. (2016). DNN-Based Acoustic Modeling for Russian Speech Recognition Using Kaldi. *International Conference on Speech and Computer, (SPECOM)*, (9811), 246–253.
- Kiruthiga, S., & Krishnamoorthy, K. (2012). Design Issues in Developing Speech Corpus for Indian Languages – A survey. *International Conference on Computer Communication and Informatics*, 1–4.
- Krishnamoorthy, P. (2009). Combined temporal and spectral processing methods for speech enhancement. *Indian Institute of Technology Guwahati, Guwahati-781039, Assam, India*.
- Kuamr, A., Dua, M., & Choudhary, A. (2014). Implementation and performance evaluation of continuous Hindi speech recognition. *International Conference on In Electronics and Communication Systems (ICECS)*, 1–5.
- Kuamr, A., Dua, M., & Choudhary, T. (2014). Continuous Hindi speech recognition using Gaussian mixture HMM. *IEEE Students' Conference on Electrical, Electronics and Computer Science (SCEECS)*, 1–5.
- Kumar, K., & Aggarwal, R. K. (2011). Hindi Speech Recognition System using HTK. *International Journal of Computing and Business Research*, 2(2), 1–12.
- Kumar, K., Aggarwal, R. K., & Jain, A. (2012). A Hindi speech recognition system for connected words using HTK. *International Journal of Computational Systems Engineering*, 1(1), 25–32.
- Kumar, M., Rajput, N., & Verma, A. (2004). A large-vocabulary continuous speech recognition system for Hindi. *IBM Journal of Research and Development*, 48(5.6), 703–715.
- Kumar, R., & Singh, M. (2011). Spoken isolated word recognition of Punjabi language using dynamic time warp technique. In: Singh C., Singh Lehal G., Sengupta J., Sharma D.V., Goyal V. (eds) *Information systems for Indian languages*, 139, 301.
- Kumar, R., Kishore, S., Gopalakrishna, A., Chitturi, R., Joshi, S., Singh, S., & Sitaram, R. (2005). Development of Indian language speech databases for large vocabulary speech recognition systems. *Proceedings of SPECOM*, 1–6.
- Kumar, Y., & Singh, N. (2017). An automatic speech recognition system for spontaneous Punjabi speech corpus. *International Journal of Speech Technology*, 20(2), 297–303.
- Kumar, R. (2010). Comparison of HMM and DTW for isolated word recognition system

- for punjabi language. *International Journal of Soft Computing*, 5(3), 88–92.
- Kundu, S., Mantena, G., Qian, Y., Tan, T., Delcroix, M., & Sim, K. C. (2016). Joint acoustic factor learning for robust deep neural network based automatic speech recognition. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 5025–5029.
- Kurian, C. (2015). A Review on Speech Corpus Development for Automatic Speech Recognition in Indian Languages. *Int. J, Advanced Networking and Applications*, 6(6), 2556–2558.
- Kwong, S., Chau, C. W., Man, K. F., & Tang, K. S. (2001). Optimisation of HMM topology and its model parameters by genetic algorithms. *Pattern Recognition*, 34(2), 509–522.
- Kwong, S., He, Q. H., Ku, K. W., Chan, T. M., Man, K. F., & Tang, K. S. (2002). A genetic classification error method for speech recognition. *Signal Processing*, 82(5), 737–748.
- Kwong, S. & Chau, C. W. (1997). Analysis of parallel genetic algorithms on hmm based speech recognition system. *IEEE Transactions on Consumer Electronics*, 43(4), 1229–1233.
- Lasserre, J. A., Bishop, C. M., & Minka, T. P. (2006). Principled Hybrids of Generative and Discriminative Models. *IEEE Computer Society Conference on Computer Vision and Pattern Recognition*, 87–94.
- Lata, S., & Arora, S. (2013). Laryngeal tonal characteristics of Punjabi—An experimental study. *IEEE International Conference on Human Computer Interactions (ICHCI)*, 1–6.
- Lee, A., Kawahara, T., & Shikano, K. (2001). Julius - an Open Source Real-Time Large Vocabulary Recognition Engine. *EUROSPEECH 7th European Conference on Speech Communication and Technology*, 1691–1694.
- Lee, C. H., Gauvain, J. L., Pieraccini, R., & Rabiner, L. R. (1993). Large vocabulary speech recognition using subword units. *Speech Communication*, 13(3–4), 263–279.
- Lee, S. J., Kang, B. O., Chung, H., & Park, J. G. (2015). A useful feature-engineering approach for a LVCSR system based on CD-DNN-HMM algorithm. *23rd European Signal Processing Conference (EUSIPCO)*, 1421–1425.
- Lee, S. M., Fang, S. H., Hung, J. W., & Lee, L. S. (2001). Improved MFCC feature extraction by PCA-optimized filter-bank for speech recognition. *IEEE Workshop on Automatic Speech Recognition and Understanding, (ASRU)*, 49–52.
- Li, B., & Sim, K. C. (2010). Comparison of discriminative input and output transformations for speaker adaptation in the hybrid NN/HMM systems. *Eleventh Annual Conference of the International Speech Communication Association*, 526–529.
- Li, Q., & Huang, Y. (2010). Robust speaker identification using an auditory-based feature. *IEEE International Conference on Acoustics Speech and Signal*

- Processing, (ICASSP), 4514–4517.
- Li, Q., & Huang, Y. (2011). An Auditory-Based Feature Extraction Algorithm for Robust Speaker Identification Under Mismatched Conditions. *IEEE Transactions on Audio, Speech, and Language Processing*, 19(6), 1791–1801.
- Li, Q., Soong, F. K., & Siohan, O. (2000). A high-performance auditory feature for robust speech recognition. *Sixth International Conference on Spoken Language Processing*, 51–54.
- Li, Y., Feng, J., & Hu, J. (2016). Covariance and crossover matrix guided differential evolution for global numerical optimization. *SpringerPlus*, 5(1), 1–22.
- Li, Y. X., Kwong, S., He, Q. H., He, J., & Yang, J. C. (2010). Genetic algorithm based simultaneous optimization of feature subsets and hidden Markov model parameters for discrimination between speech and non-speech events. *International Journal of Speech Technology*, 13(2), 61–73.
- Li, Z., & Gao, Y. (2016). Acoustic feature extraction method for robust speaker identification. *Multimedia Tools and Applications*, 75(12), 7391–7406.
- Liporace, L. (1982). Maximum likelihood estimation for multivariate observations of Markov sources. *IEEE Transactions on Information Theory*, 28(5), 729–734.
- Liu, S., & Sim, K. C. (2014). On combining DNN and GMM with unsupervised speaker adaptation for robust automatic speech recognition. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 195–199.
- Maas, A. L., Qi, P., Xie, Z., Hannun, A. Y., Lengerich, C. T., Jurafsky, D., & Ng, A. Y. (2017). Building DNN acoustic models for large vocabulary speech recognition. *Computer Speech and Language*, 41, 195–213.
- Macaš, M., Novák, D., & Lhotská, L. (2006). Constraints in particle swarm optimization of hidden Markov models. *International Conference on Intelligent Data Engineering and Automated Learning*, 1399–1406.
- Maganti, H. K., & Matassoni, M. (2010). An Auditory Based Modulation Spectral Feature for Reverberant Speech Recognition. *Eleventh Annual Conference of the International Speech Communication Association*, 570–573.
- Makhoul, J. (1975). Linear Prediction: A Tutorial Review. *Proceedings of the IEEE*, 63(4), 561–580.
- Maldonado, Y. P., Morales, S. O. C., & Ortega, R. O. C. (2012). GA approaches to HMM optimization for automatic speech recognition. *Mexican Conference on Pattern Recognition*, 313–322).
- Mandal, P., Jain, S., Ojha, G., & Shukla, A. (2015). Development of Hindi speech recognition system of agricultural commodities using deep neural network. *Sixteenth Annual Conference of the International Speech Communication Association*, 1241–1245.
- Mantena, G. V., Rajendran, S., Rambabu, B., Gangashetty, S. V., Yegnanarayana, B., & Prahallad, K. (2011). A Speech-Based Conversation System for Accessing Agriculture Commodity Prices in Indian Languages. *Joint Workshop on Hands-*

- Free Speech Communication and Microphone Arrays, 153–154.
- Mao, R., Zhu, H., Zhang, L., & Chen, A. (2006). A new method to assist small data set neural network learning. *Sixth International Conference on Intelligent Systems Design and Applications, (ISDA)*, 17–22.
- Mason, J. S., & Zhang, X. (1991). Velocity and acceleration features in speaker recognition. *International Conference on Acoustics, Speech, and Signal, (ICASSP)*, 3673–3676.
- Mehla, R. & Aggarwal, R. K. (2013). Improving HMM Training Process Using Hybrid Baum-Welch Genetic Algorithm. *International Conference on Emerging Research in Computing, Information, Communication and Applications (ERCICA)*, 1–7.
- Meriem, F., Farid, H., Messaoud, B., & Abderrahmene, A. (2017). New front end based on multitaper and gammatone filters for robust speaker verification. In M. Chadli, S. Bououden, & I. Zelinka (Eds.), *Recent Advances in Electrical Engineering and Control Applications*, 411, 344–354.
- Milner, B., & Darch, J. (2011). Robust acoustic speech feature prediction from noisy mel-frequency cepstral coefficients. *IEEE Transactions on Audio, Speech, and Language Processing*, 19(2), 338–347.
- Minh, V. D., & Lee, S. (2004). PCA-based human auditory filter bank for speech recognition. *International Conference on Signal Processing and Communications, (SPCOM)*, 393–397.
- Mitra, V., Sivaraman, G., Nam, H., Espy-Wilson, C., Saltzman, E., & Tiede, M. (2017). Hybrid convolutional neural networks for articulatory and acoustic information based speech recognition. *Speech Communication*, 89, 103–112.
- Mitra, V., Wang, W., Franco, H., Lei, Y., Bartels, C., & Graciarena, M. (2014). Evaluating robust features on Deep Neural Networks for speech recognition in noisy and channel mismatched conditions. *Annual Conference of the International Speech Communication Association, INTERSPEECH*, 895–899.
- Mittal, S., & Sharma, R. K. (2014). Development of Phonetic Engine for Punjabi Language. M.Tech. Thesis, Thapar University, Punjab, India.
- Mittal, T., & Sharma, R. K. (2016). An improved SVM using predator pre optimization and hook's jeeve method. *Journal of Engineering Research*, 4(1), 1–20.
- Mittal, T. & Sharma, R. K. (2016). Integrated search technique for parameter determination of SVM for speech recognition. *Journal of Central South University*, 23(6), 1390–1398.
- Mizuta, S., & Nakajima, K. (1992). A discriminative training method for continuous mixture density HMMs and its implementation to recognize noisy speech. *Journal of the Acoustical Society of Japan (E)*, 13(6), 389–393.
- Mohamed, A., & Nair, K. R. (2012). HMM/ANN hybrid model for continuous Malayalam speech recognition. *Procedia Engineering*, 30, 616–622.
- Moon, S. & Hwang, J. N. (1997). Robust speech recognition based on joint model and feature space optimization of hidden Markov models. *IEEE Transactions on Neural*

- Networks, 8(2), 194–204.
- Morgan, N., & Hermansky, H. (1992). RASTA extensions: Robustness to additive and convolutional noise. *Speech Processing in Adverse Conditions*, 115–118.
- Nahid, M. M. H., Purkaystha, B., & Islam, M. S. (2017). Bengali speech recognition: A double layered LSTM-RNN approach. *20th International Conference of Computer and Information Technology (ICCIT)*, 1–6.
- Najkar, N., Razzazi, F., & Sameti, H. (2010). A novel approach to HMM-based speech recognition systems using particle swarm optimization. *Mathematical and Computer Modelling*, 52(11–12), 1910–1920.
- Neal, R. M., & Hinton, G. E. (1998). A view of the EM algorithm that justifies incremental, sparse, and other variants. In: Jordan, M.I. (Ed.), *Learning in Graphical Models*. Kluwer Academic Publishers.
- Ney, H. (1984). The use of a one-stage dynamic programming algorithm for connected word recognition. *IEEE Transactions on Acoustics, Speech, and Signal Processing*, (ICASSP), 188–196.
- Ni, H., Yi, J., Wen, Z., Liu, B., & Tao, J. (2016). Improving accented Mandarin speech recognition by using recurrent neural network based language model adaptation. *10th International Symposium on Chinese Spoken Language Processing (ISCSLP)*, 1–5.
- P1: Punjabi Speech Corpus [http://tdil-dc.in/index.php?option=com_download&task=viewtoolsbyParam&categoryid=84&lang=en] available on 15 Oct 2015 10:30 p:m.
- Patil, H. A., & Basu, T. K. (2008). Development of speech corpora for speaker recognition research and evaluation in Indian languages. *International Journal of Speech Technology*, 11(1), 17-32.
- Palaz, D., Magimai.-Doss, M., & Collobert, R. (2015). Analysis of CNN-based speech recognition system using raw speech as input. *Proceedings of the Annual Conference of the International Speech Communication Association, INTERSPEECH*, 11–15.
- Pan, S. T., Chen, C. F., Chang, W. D., & Tsai, Y. H. (2011). Performances comparison between improved DHMM and Gaussian mixture HMM for speech recognition. *4th International Congress on Image and Signal Processing (CISP)*, 2426–2430.
- Pandey, A., Srivastava, B. M. L., & Gangashetty, S. V. (2017). Adapting monolingual resources for code-mixed hindi-english speech recognition. *International Conference on Asian Language Processing (IALP)*, 218–221.
- Paramonov, P., & Sutula, N. (2016). Simplified scoring methods for HMM-based speech recognition. *Soft Computing*, 20(9), 3455–3460.
- Parthasarathi, S. H. K., Hoffmeister, B., Matsoukas, S., Mandal, A., Strom, N., & Garimella, S. (2015). FMLLR based feature-space speaker adaptation of DNN acoustic models. *Sixteenth Annual Conference of the International Speech Communication Association*, 3630–3634.

- Povey, D., Burget, L., Agarwal, M., Akyazi, P., Kai, F., Ghoshal, A., & Rose, R. C. (2011). The subspace Gaussian mixture model—A structured model for speech recognition. *Computer Speech & Language*, 25(2), 404–439.
- Povey, D., Ghoshal, A., Boulianne, G., Burget, L., Glembek, O., Goel, N., & Silovsky, J. (2011). The Kaldi speech recognition toolkit. In *IEEE 2011 workshop on automatic speech recognition and understanding* (No. EPFL-CONF-192584). IEEE Signal Processing Society.
- Prahalad, K., Black, A. W., Kumar, R., & Sangal, R. (2003). Experiments with unit selection speech databases for Indian languages. *National Seminar on Language Technology Tools: Implementation of Telugu*, 1–6.
- Prasanna, S. M. (2004). *Event based analysis of speech*. Indian Institute of Technology Madras, India.
- Price, K. V. (1996). Differential evolution: a fast and simple numerical optimizer. *Biennial Conference of the North American Fuzzy Information Processing Society, (NAFIPS)*, 524–527.
- Pujol, P., Pol, S., Nadeu, C., Hagen, A., & Boulard, H. (2005). Comparison and combination of features in a hybrid HMM/MLP and a HMM/GMM speech recognition system. *IEEE Transactions on Speech and Audio Processing*, 13(1), 14–22.
- Qi, J., Wang, D., Jiang, Y., & Liu, R. (2013). Auditory features based on gammatone filters for robust speech recognition. *IEEE International Symposium on Circuits and Systems (ISCAS)*, 305–308.
- Quatieri, T. F. (2006). *Discrete-time speech signal processing: principles and practice*. Pearson Education.
- Rabiner, L. R. (1989). A tutorial on hidden Markov models and selected applications in speech recognition. *Proceedings of the IEEE*, 77(2), 257–286.
- Rabiner, L. R., & Juang, B. H. (1993). *Fundamentals of Speech Recognition*. PTR Prentice Hall.
- Radha, V. (2012). Speaker independent isolated speech recognition system for Tamil language using HMM. *Procedia Engineering*, 1097–1102.
- Rajagede, R. A., & Dewa, C. K. (2017). Recognizing Arabic letter utterance using convolutional neural network. *18th IEEE/ACIS International Conference on Software Engineering, Artificial Intelligence, Networking and Parallel/Distributed Computing (SNPD)*, 181–186.
- Ranjan, R., Singh, S. K., Shukla, A., & Tiwari, R. (2010). Text-dependent multilingual speaker identification for indian languages using artificial neural network. *3rd International Conference on Emerging Trends in Engineering and Technology (ICETET)*, 632–635.
- Rao, K. S. (2011). Application Prosody model for Developing speech system. *International Journal of Speech Technology*, 14(1), 19–33.
- Ravanelli, M., Brakel, P., Omologo, M., & Bengio, Y. (2016). Batch-normalized joint

- training for DNN-based distant speech recognition. *Spoken Language Technology Workshop (SLT)*, 28–34.
- Renjith, S., & Manju, K. G. (2017). Speech based emotion recognition in Tamil and Telugu using LPCC and hurst parameters—A comparative study using KNN and ANN classifiers. *International Conference on In Circuit, Power and Computing Technologies (ICCPCT)*, 1–6.
- Renshaw, D., & Hall, K. B. (2015). Long short-term memory language models with additive morphological features for automatic speech recognition. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 5246–5250.
- Reynolds, D. A. (2002). An overview of automatic speaker recognition technology. *IEEE International Conference on Acoustics, Speech, and Signal Processing*, 4, 4072–4075.
- Reza, M., Rashid, W., & Mostakim, M. (2017). Prodorshok I: A Bengali Isolated Speech Dataset for Voice-Based Assistive Technologies - A comparative analysis of the effects of data augmentation on HMM-GMM and DNN classifiers. *5th IEEE R10 HTC*, 1–4.
- Riedel, K. S., & Sidorenko, A. (1995). Minimum bias multiple taper spectral estimation. *IEEE Transactions on Signal Processing*, 43(1), 188–195.
- Rodríguez, L. J., & Torres, I. (2003). Comparative Study of the Baum-Welch and Viterbi Training Algorithms Applied to Read and Spontaneous Speech Recognition. *Iberian Conference on Pattern Recognition and Image Analysis*, 847–857.
- Sahidullah, M., & Saha, G. (2012). Design, analysis and experimental evaluation of block based transformation in MFCC computation for speaker recognition. *Speech Communication*, 54(4), 543–565.
- Samudravijaya, K., Rao, P. V. S., & Agrawal, S. S. (2000). Hindi Speech Database. *Sixth International Conference on Spoken Language Processing (ICSLP)*, 456–459.
- Sandberg, J., Hansson-Sandsten, M., Kinnunen, T., Saeidi, R., Flandrin, P., & Borgnat, P. (2010). Multitaper estimation of frequency-warped cepstra with application to speaker verification. *IEEE Signal Processing Letters*, 17(4), 343–346.
- Schluter, R., Bezrukov, I., Wagner, H., & Ney, H. (2007). Gammatone features and feature combination for large vocabulary speech recognition. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 4, 649–652.
- Schmidhuber, J. (2015). Deep learning in neural networks: an overview. *Neural Networks*, 61, 85–117.
- Schwartz, R., Austin, S., Kubala, F., Makhoul, J., Nguyen, L., Placeway, P., & Zavaliagkos, G. (1992). New uses for the N-best sentence hypotheses within the BYBLOS speech recognition system. *IEEE International Conference on Acoustics, Speech, and Signal Processing, (ICASSP)*, 1–4.
- Seide, F., Li, G., Chen, X., & Yu, D. (2011). Feature engineering in context-dependent

- deep neural networks for conversational speech transcription. *IEEE Workshop on Automatic Speech Recognition and Understanding (ASRU)*, 24–29.
- Sarma, B. D., Sarmah, P., Lalhminglui, W., & Prasanna, S. M. (2015). Detection of Mizo Tones. *Sixteenth Annual Conference of the International Speech Communication Association (Interspeech)*, 934-937.
- Siddhant, A., Jyothi, P., & Ganapathy, S. (2017). Leveraging Native Language Speech for Accent Identification using Deep Siamese Networks. *CoRR*, 1–8.
- Singh, A., Pandey, D., & Agrawal, S. S. (2015). Analysis of Punjabi tonemes. *2nd International Conference on Computing for Sustainable Global Development (INDIACom)*, 1694–1697.
- Song, X., Zou, Y., Huang, S., Chen, S., & Liu, Y. (2017). Investigating multi-task learning for automatic speech recognition with code-switching between mandarin and english. *International Conference on Asian Language Processing (IALP)*, 27–30.
- Stuttle, M. N., & Gales, M. J. F. (2002). Combining a Gaussian mixture model front end with MFCC parameters. *7th International Conference on Spoken Language Processing*, 1565–1568.
- Takamichi, S. (2017). Modulation spectrum-based speech parameter trajectory smoothing for DNN-based speech synthesis using FFT spectra. *Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA ASC)*, 1308–1311.
- Takara, T., Iha, Y., & Nagayama, I. (1998). Selection of the optimal structure of the continuous hmm using the genetic algorithm. *Fifth International Conference on Spoken Language Processing*, 1–4.
- Thalengala, A., & Shama, K. (2016). Study of sub-word acoustical models for Kannada isolated word recognition system. *International Journal of Speech Technology*, 19(4), 817–826.
- Thatphithakkul, N., & Kanokphara, S. (2004). HMM parameter optimization using tabu search. *IEEE International Symposium on Communications and Information Technology, (ISCIT)*, 2, 904–908.
- Thomson, D. J. (1982). Spectrum estimation and harmonic analysis. *Proceedings of the IEEE*, 70(9), 1055–1096.
- Tomashenko, N., & Khokhlov, Y. (2015). GMM-derived features for effective unsupervised adaptation of deep neural network acoustic models. *Sixteenth Annual Conference of the International Speech Communication Association*, 2882–2886.
- Tomashenko, N., Khokhlov, Y., & Estève, Y. (2016). A New Perspective on Combining GMM and DNN Frameworks for Speaker Adaptation. *International Conference on Statistical Language and Speech Processing*, 120–132.
- Variani, E., Sainath, T. N., Shafran, I., & Bacchiani, M. (2016). Complex Linear Projection (CLP): A Discriminative Approach to Joint Feature Extraction and Acoustic Modeling. *Interspeech*, 808–812.

- Veisi, H., & Sameti, H. (2013). Speech enhancement using hidden Markov models in Mel- frequency domain. *Speech Communication*, 52(2), 205–220.
- Vesterstrom, J., & Thomsen, R. (2004). A comparative study of differential evolution, particle swarm optimization, and evolutionary algorithms on numerical benchmark problems. *Evolutionary Computation*, (CEC), 1980–1987.
- Viterbi, A. (1967). Error bounds for convolutional codes and an asymptotically optimum decoding algorithm. *IEEE Transactions on Information Theory*, 13(2), 260–269.
- Walker, W., Lamere, P., Kwok, P., Raj, B., Singh, R., Gouvea, E., & Woelfel, J. (2004). *Sphinx-4: A flexible open source framework for speech recognition*. Sun Microsystems, Inc., Mountain View, CA, USA.
- Wang, L., Minami, K., Yamamoto, K., & Nakagawa, S. (2010). Speaker identification by combining MFCC and phase information in noisy environments. *IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP)*, 4502–4505.
- Xue, S., Abdel-Hamid, O., Jiang, H., Dai, L., & Liu, Q. (2014). Fast adaptation of deep neural network based on discriminant codes for speech recognition. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, 22(12), 1713–1725.
- Yan, Z., Liu, P., Du, J., Soong, F., & Wang, R. (2006). Training Discriminative HMM by Optimal Allocation of Gaussian Kernels. *International Symposium on Chinese Spoken Language Processing*, 1–10.
- Yang, F., Zhang, C., & Bai, G. (2008). A novel genetic algorithm based on tabu search for HMM optimization. *Fourth International Conference on Natural Computation*, (ICNC), 57–61.
- Yang, F., Zhang, C., & Sun, T. (2008). Comparison of particle swarm optimization and genetic algorithm for HMM training. *19th International Conference on Pattern Recognition*, (ICPR), 1–4.
- Yao, K., Yu, D., Seide, F., Su, H., Deng, L., & Gong, Y. (2012a). Adaptation of context-dependent deep neural networks for automatic speech recognition. *IEEE on Spoken Language Technology Workshop (SLT)*, 366–369.
- Yi, J., Ni, H., Wen, Z., Liu, B., & Tao, J. (2016). CTC regularized model adaptation for improving LSTM RNN based multi-accent Mandarin speech recognition. *10th International Symposium on Chinese Spoken Language Processing (ISCSLP)*, 1–5.
- Young, S., Evermann, G., Gales, M., Hain, T., Kershaw, D., Liu, X., & Valtchev, V. (2002). *The HTK book*. Cambridge University Engineering Department, vol . 3, 1-336.
- Yu, D., & Deng, L. (2015). *Automatic Speech Recognition - A Deep Learning Approach*. Springer-Verlag London.
- Zhang, X., Wang, Y., & Zhao, Z. (2008). A hybrid speech recognition training method for HMM based on genetic algorithm and Baum Welch algorithm. *Second International Conference on Innovative Computing, Information and Control*, (ICICIC), 25–28.

- Zhao, T., Zhao, Y., & Chen, X. (2016). Ensemble Acoustic Modeling for CD-DNN-HMM Using Random Forests of Phonetic Decision Trees. *Journal of Signal Processing Systems*, 82(2), 187–196.
- Zhao, X., & Wang, D. (2013). Analyzing noise robustness of MFCC and GFCC features in speaker identification. *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 7204–7208.
- Zolnay, A., Kocharov, D., Schlüter, R., & Ney, H. (2007). Using multiple acoustic feature sets for speech recognition. *Speech Communication*, 49(6), 514–525.
- Zolnay, A., Schluter, R., & Ney, H. (2005). Acoustic feature combination for robust speech recognition. *IEEE International Conference on Acoustics, Speech, and Signal Processing, (ICASSP)*, 457–460.
- Zouhir, Y., & Ouni, K. (2014). A bio-inspired feature extraction for robust speech recognition. *SpringerPlus*, 1–8.