A Review Paper on Speech Enhancement Techniques

D. Divya[1], Sandeep Kumar[2] and Deepika Ghai[3]
Student[1] and Professor[2], Sreyas Institute of Engineering and Technology, Hyderabad
Assistant Professor[3], Lovely Professional University, Punjab

ABSTRACT

Speech is an important way of communication between humans. various types of noise like background noise, atmospheric disturbances, reverberation, babble noise etc, will degrade the speech signals. The speech signal should be clean and clear for several applications such as hearing aids, Tele-communication system. The main aim of speech enhancement is to improve speech quality by using various enhancement methods like filtering techniques, spectral subtraction technique, model based methods and wavelet based methods etc. This paper provides a complete study analysis on different speech enhancement techniques, The paper contains the study of several enhancement models with their results. A complete discussion on the enhancement of speech and reduction of Noises are provided from the basis of the previous proposed works related to several speech enhancement algorithms which had been observed from the year 1998 to 2019. This work proposes the using of machine learning techniques for enhancement of speech signals and through comparing several enhancement algorithm for better performance.

INTRODUCTION

In today's sophisticated world speech is the most important way through which people can communicate. In communication medium due to the large distance between speaker and listener, the listener's ability to understand decreases due to the introduction of noise and the distortion in transmission media. The quality of communication is decreased and the speech will be less intelligible. Hence, speech enhancement is enhancing the quality of the degreded speech signal,intelligibility of noisy speechand enabling effective communication. Generally the noisy speech signal is combined with speech signal along with the background noise signal due to different environmental sources. It is difficult to track various noises which vary continuously with time. It is also not feasible to completely eliminate the noises in the speech signal. However the background noise level can be reduced if the variation in noise characteristics is slower than that of the speech signal. One of the important area where speech enhancement plays an important role is in mobile communication.

There are many speech enhancement algorithms available and each one of them is well suited for different conditions. The quality of speech signal is measured in terms of quality (subjective measure) and intelligibility (objective measure). The naturalness of the received speech is the quality of signal. Measure of correct identification percentage of a signal is termed as intelligibility. The signal which travel In a noisy environment, with degraded quality and poor intelligibility make voice communication difficult and fatiguing. Intelligibility loss occurs in speech sounds such as consonants, fricatives and stops which are often masked by noise. A scheme should be introduced to suppress the background noise which helps improving speech quality and intelligibility in the design of hearing aid. There has been many researches going on over the last few years in the area of speech enhancement algorithms development. Though there is a wider coverage for the term speech enhancement, in this the term is used to refer a background noise reduction. There are different ways of classifications for speech enhancement systems like single channel and multi channel which are based on number of input channels and the processing domain.single channel enhancement algorithms is used in applications like hearing aids and hands-free communication. The noise in background is eliminated by masking using a reference noise signal in multi-channel enhancement technique, Designing complexity is more in multichannel systems.Adaptive noise filtering method isOne of the powerful multichannel speech enhancement techniques.

THINK INDIA JOURNAL

ISSN:0971-1260

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SPEECH ENHANCEMENT

Enhancement mean improving the value or quality of something. When this is applied to speechsignal,it means improving the intelligibility or quality of the distorted speech signal by using various signal enhancement methods and algorithm.Enhancementof speech degraded by noise, or reduction of noise, is the most important feature of speech enhancement, and used in applications such as mobile phones,voip teleconferencing systems speech recognition systems and hearing aids. In past few years, many researches had been doing their work in the field of Speech enhancement. Clean speech isdegrsdedwith the noise in background or environment. So the main aim of Speech Enhancement is to improve the quality and intelligibility of noise which gets degraded when it is passed through a non stationary medium Also, the performance of the Speech Enhancement in real acoustic environment is not always predictable. Various Speech Enhancement techniques are developed in last few decades for improving Speech Signal-to-noise ratio (SNR). Thus an effective speech enhancement algorithm is required for speech enhancement by removing non stationary noises.

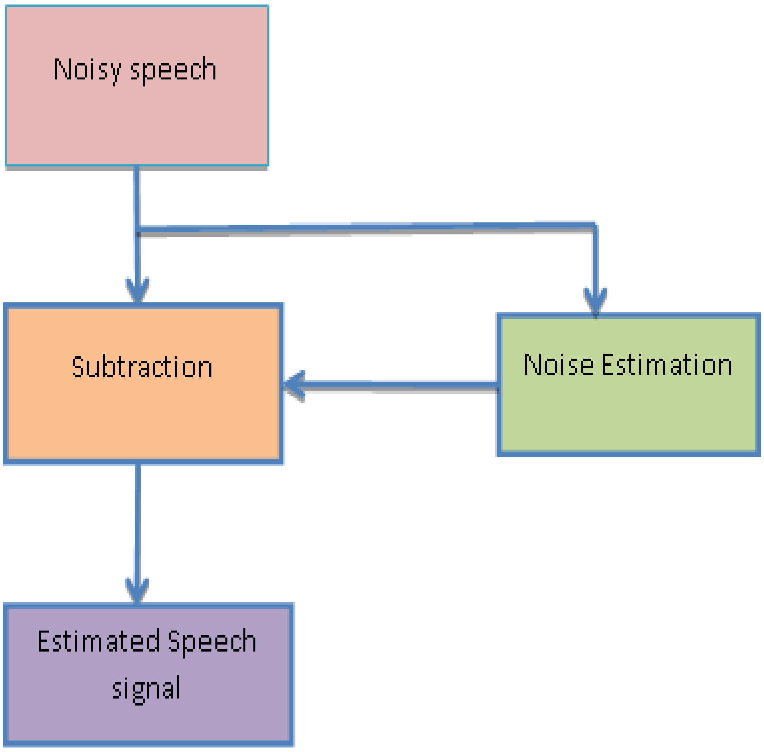

Fig: Block Diagram of Speech Enhancement

LITERATURE SURVEY

SPEECH ENHANCEMENT USING DEEP NEURAL NETWORKS

Qizheng Huang, et al. [ ], 2018 ,he proposed a speech enhancement method based on DNN by using MBE model which contains two stages, namely training stage and enhancing stage.In the training stage, two fully connected feed-forward DNNs are trained to predict two MBE parameters, including harmonic magnitude and band difference function. The harmonic magnitude and band difference function of clean speech,are used as taining targets respectively.Log-power spectra (LPS) of noisy speech is the input feature for two DNN'S used in training stage. The enhanced speech can be obtained by MBE speech synthesis using the output of DNNs and online estimated pitch period in enhancing stage. With the proposed method high quality enhanced speech is synthesized by estimating the parameters of MBE model accurately. At the same time, the noise between the harmonics is effectively eliminated .Estimated pitch of each frame along with the output of DNNs is used to enhance the noisy speech in enhancing stage. The experimental result shows the proposed method output from reference methods at different SNRs.

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Table-1: The average PESQ score result at different levels of SNR.

Method(dB)	-5dB	0dB	5dB
Ref x	1.8480	2.1740	2.5190
Ref y	1.8670	2.1860	2.4750
Proposed	1.9520	2.2460	2.5210

Table-2: The average SSNR improvement result.

Method(dB)	-5dB	0dB	5dB
Ref x	12.0790	10.3733	8.5411
Ref y	13.2680	10.5122	7.5122
Proposed	16.6540	13.0622	8.6822

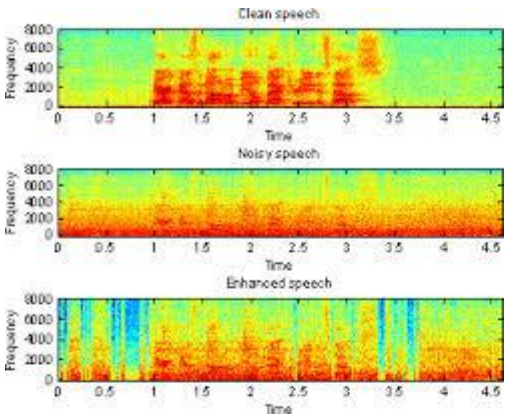

Figure: Spectrograms of (a) noisy speech (b) noisy speech (c) enhanced speech

HaichuanBai et al.[ ], proposed speech enhancement algorithm using DNN based on soft audible noise masking(SANM)which is used for residual noise reduction in single channel.To calculate the masking threshold from the estimated speech spectrum he used psychoacoustic model, and the SANM (soft audible noise masking) principle was brought into spectral weighting algorithm by using masking and estimated wind noise spectrum.He tested the proposed method with wind noise collected by a microphone without any wind screen on a windy day with wind speed of 15m/s.The speech samples were collected from a TIMIT dataset.

Table1 : The subjective preference tests result

SNR	**SANM-DNN**	**Direct DNN**	**No preference**
10dB	39%	25%	36%
5dB	43%	23%	34%
0dB	35%	19%	46%
-5dB	38%	24%	38%

Chen Jian-ming et al. [ ], proposed a speech enhancement algorithm using Ensemble Empirical Mode Decomposition (EEMD) and Deep Neural Network (DNN).Firstly EEMD is used to propose the original signal,and the better time-variation can be obtained by decomposinga series of time-frequency information of the IMF component, Secondly the weight of the IMF component by DNN is adusted and then synthesized to enhance the speech; Finally, the differences of speech enhancement performance between using different methods is compared.Like the performance of EEMD alone and EEMD along with fouriertransform.He concluded that the enhanced mothod which uses EEMD as a preprocessing will improve the scores of PESQ and STOI and effectively improve the speech quality and intelligibility.

ISSN:0971-1260
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Table: The values of PESQ and STIO in different scores

Evaluation indicators	different Scores			
	Noisy speech	No EEMD	Only EEMD	use EEMD
PESQ	1.381	2.0091	1.481	2.1265
STOI	0.652	0.7833	0.691	0.8211

Table: Comparision of various authors work on speech enhancement based on deep neural network

Author & Year	Methodology	Remarks	Dataset
QizhengHuang, et al. [2],2018	MBE model	SNR values at -5db=0.604 0db=0.685 +5db=0.761	TIMIT database
HaichuanBai et al. [9],2018	SANM masking	Increase in SNR %	TIMIT database
Chen Jian-ming et al. [31]2018	EEMD model	Improved PESQ and STOI	TIMIT database

SPEECH ENHANCEMENT USING A CODEBOOK BASED APPROACH

Yang Xiang, et al. [ ], ―proposed speech enhancementusing a novel codebook-driven method and performed experiment on test set taken from NOISEX database.Theresidual noise in harmonic bands is removed with this algorithm by combining speech harmonic structure and codebook. Firstly, the prior speech presence probability is estimated by speech harmonic structure. Then it is used to appraise the noise autoregressive (AR) spectral shapes for speech enhancement application. In addition, Wiener filter is modified by theprior speech presence probability. Finally,, the modified Wiener filter is build by combining the clean speech AR spectral shape codebook to acquire enhanced speech signal.HeCompared the performance of proposed method with traditional codebook-driven method (TCD) and concluded more residual noise in the harmonic bands of noisy speechcan be removed by using proposed method.

Mathew ShajiKavalekalam et al. [ ], proposed a method for enhancing dual channel speech signals based on codebook driven method. The proposed method involves the estimation of noise short term predictor (STP) parameters and speech signals. The estimated STP parameters are subsequently used for speech enhancement in a dual channel scenario only when we have access to binaural noisy signals. He also used kalman smoother for speech enhancement.

FengBao, et al. [ ], proposed a speech enhancement method based on a few shapes of speech spectrum. Instead of training the noise codebooks used in conventional method he estimated noise by utilizing Minima Controlled Recursive Averaging (MCRA) algorithm. Then, by minimizing the spectral distortion between the noisy speech and the combination of noise and speechthe spectral shapes and the spectral gains of speech and noise are optimized. Next,spectral gains of speech and noise are modified byusing normalized cross-correlation coefficients between the spectra of noisy speech and noise. Finally, the noisy speech is passed through the reconstructed Wiener filter to obtain the enhanced speech.he performed it using utterances from different speaker voices in NTT database. The length of each utterance is 8s. white, babble, office and street are different noises chosen from NOISE 92 database.and he concluded that any noise classification is not required in proposed process and is better suitable for real time applications.

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Table-1: The test results between Ref X and the proposed method

dB	Prefer the ref X(%)	Prefer the proposed method(%)	No preference(%)
10	27	58	15
5	26	56	18
0	34	57	09

Qi He1, et al. [ ], proposedcodebook-driven Wiener filteringin the speech enhancement method for estimating autoregressive (AR) parameters of speech and noise by using novel technique. He trained the codebook shape of speech spectrum offline, and the noise spectrum shape is estimated online for solving the problem of noise classification. Unlike conventional codebook-driven methods, the proposed method uses a multiplicative update rule for more accurate estimation of speech and noise AR gains, The Bayesian parameter-estimator without the noise codebook is also developedby utilizing a very simple method of combining the codebook-driven Wiener filter along with speech-presence probability(SPP) for removing the residual noise between the harmonics of noisy speech.

Table-1: Test results of PESQ at different levels (perceptual evaluation of speech quality)

Enhancement methods	0dB	5dB	10dB
Ref.X	2.05	2.45	2.71
Ref.Y	2.33	2.64	2.90
Noisy speech	1.87	2.20	2.55
proposed	2.40	2.76	3.06

Table-2: test results of SSNR (spectral signal to noise ratio)

Enhancement methods	0dB	5dB	10dB
Ref.X	9.74	8.71	7.59
Ref.Y	13.23	11.97	10.69
proposed	16.42	15.46	14.16
Noisy speech	----	----	----

Table: Comparision of various authors work on speech enhancement based on code book driven method

Author & Year	Methodology	Remarks	Dataset
Yang Xiang, et al. [3] 2017	codebook-driven method	More residual noise is removed	NOISEX database
Mathew ShajiKavalekalametal. [5],2016	codebook-driven method	binaural noisy signals are enhanced	Babble noise CHIME database
FengBao, et al. [22],2014	(MCRA) algorithm	Proposed results 0dB=57% 5dB=56% 10dB=58%	NTT database
Qi He1, et al. [32], 2016	codebook-driven Wienr filtering method	Proposed LSD at 0db=7.41 5db=6.11 10db=5.15	NTT database.

Speech enhancement using support vector machine

VinayshankarSomalaraNataraj, et al. [ ], 2017 used Speech enhancement using adaptive filtering methods for recovering good speech signal from the noisy speech signal, he usedLeast Mean Square (LMS) and

ISSN:0971-1260

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Recursive Least Squares (RLS) algorithms. He gave correlating noise as a referencesignal for de-noising in this algorthms,. A novel method is used to capture noise signals which identifies the best correlating part of the noise signal with respect to noise in noisy speech and this can be used as the reference for speech enhancement in adaptive algorithms. Therefore in all the adaptive algorithms, two microphones are used, one for capturing noisy speech and the other for capturing noise signal alone Variable Step Size LMS (VSSLMS) and RLS algorithms are used for speech enhancement after finding best correlating noise. Prior to this, to identify the type of noise present in the speechNoise classification is done by using Bark features and Support Vector Machine (SVM).By using svm classification he identified the best correlating part of noise and as reference signal in adaptive algorithms for single channel speech enhancement.He concluded that a very good performance can be achived by using the above proposed system even at very low SNR of speech mixed with non-stationary noise.

TABLE I. Accuracy ofNoise classification for different types of noise and SNR levels

Noise	-5dB	0dB	5dB	10dB
White	100%	100%	100%	100%
Factory	100%	100%	100%	100%
Babble	100%	100%	100%	100%

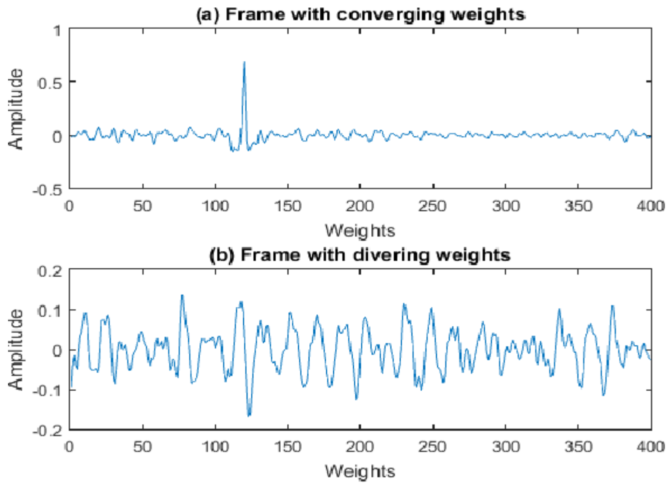

Fig : weight convergence

Joon-Hyuk Chang, et al. [ ], 2008 propose a speech enhancement technique usinga support vector machine (SVM), based on global soft decision.In which the probabilistic outputs of the SVM are employed rather than the conventional Bayes' rule.Heresigmoid function is used to determine global speech absence probability (GSAP) based on key parameters estimated by the model-trust minimization algorithm of the SVM output.He concluded that when the proposed SVMisadopted inthe global soft decisionforspeech enhancement improved results of speech qualitymeasures for various types of noise at different signal-to-noise ratio (SNR) levels are achived.

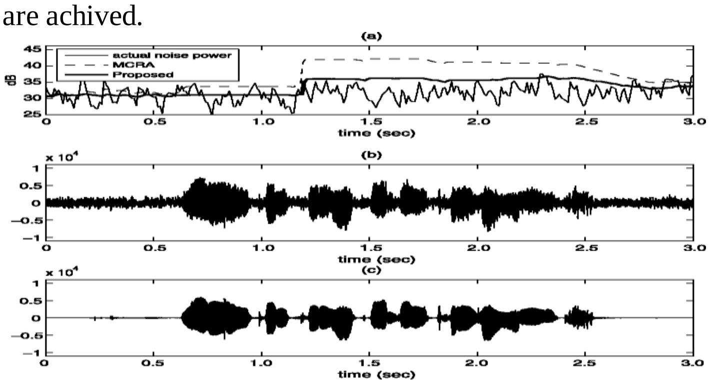

Fig. 1.GSAP Comparison of F16 noise. (a) GSAP. (b) Clean speech signal. (c) Noisy speech signal.

THINK INDIA JOURNAL

ISSN:0971-1260

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Mahesh et al. [ ], 2018 proposed a DWT algorithm for speech enhancement and speaker identification. In this he used a Five level discrete wavelet decomposition and binary mask thresholding function to enhance the speech patterns by giving all the five level DWT outputs as inputs. The speaker identification process is based on support vector machine is used for speaker identification. MFCC feature is extracted from the enhanced speech and is used as the speaker features. Different wavelet transforms are used to calculate and compare Output Signal to Noise Ratio and cepstrum distance of the enhanced speech signal,by using NOIZEUS databasespeech signals of different noises such as train noise,carnoise,station noise in several SNRs, he tested and obtained results based on processing and compared the performance of the different wavelet transform and its corresponding SNR ratio. He also compared the Performance of speaker identification in terms of its accuracy for different kernel functions.

Table : Different kernel functions with accuracy

Kernel functions	Accuracy(%)
Polynomial of order 2	82.54
Polynomial of order 3	84.34
Radial basis function	85.8

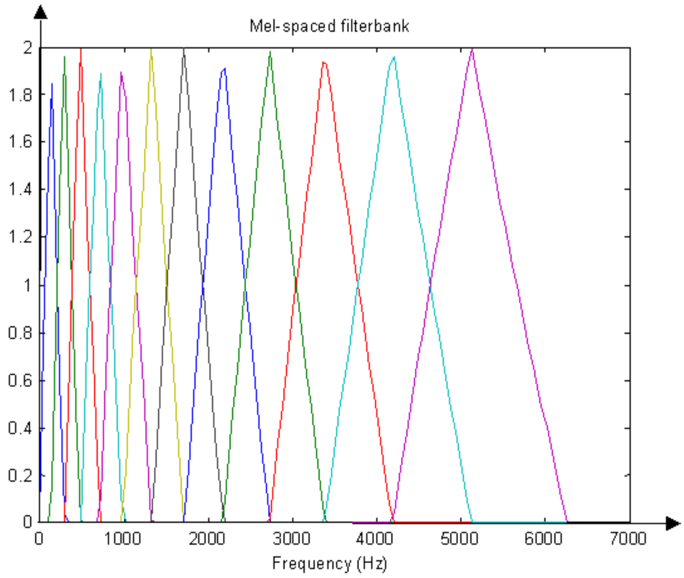

Fig: Mel Filter Bank Frequency Response

Table :comparision of various authors work on speech enhancement based on Support vector machine

Author & Year	Methodology	Remarks	Dataset
VinayshankarSomalaraNataraj, et al. [33], 2017	SVM with adaptive filtering	It is very fast and known to give very good speech enhancement even under very low SNR conditions.	Noisex92
Joon-Hyuk Chang, et al. [34], 2008	SVM with global soft decision		NTT database.
Mahesh et al. [12], 2018	DWT and SVM	Accuracy is more than 80%	NOIZEUS database

ISSN:0971-1260
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DISCUSSION

Speech Enhancement is difficult for two reasons. one is the characteristics and nature of the noise signals can change with time,Application to application. Second is different performance measurement is defined for various applications. In past 15 years, many researches are carring their research in the field of Speech enhancement,We have reviewed different speech enhancement technique like deep neural networks, codebook driven techniques ,support vector machine and matrix factorization. The speech signal is degraded because of different types of noises in background or when ever it travel trough non stationary medium i.e the statistical properties of a medium which change over time. DNN based models along with soft audible noise masking and ensemble methods are widely used speech enhancement technique may not completely remove the residual noises and it has both training and testing stages many of the authors do not provide their information of testing and training data, which is the main aspect during evaluation. The code book driven technique along with wiener filter or speech spectrum does not require any noise classification and is best suitable for real time applications. The speech enhancement based on SVM classification along with DWT will give accuracy up to 86% in real time applications. However estimation of different types of noise and their time variations is difficult and hence complete noise cancellation is not possible. The selected speech enhancement technique for particular application may suppress the noise to certain acceptable level but not eliminated completely. Spectral subtraction method is used to eliminateWide-band noise which occurs due to breathing, wind and used in applications like speech or speaker recognition. Stationary filters or adaptive filters are used to eliminate Periodic noise which occurs in industrial environment due to machinery ,engines electrical interference .

REFERENCES

1. Bolner, Federico, Tobias Goehring, Jessica Monaghan, Bas Van Dijk, Jan Wouters, and Stefan Bleeck. "Speech enhancement based on neural networks applied to cochlear implant coding strategies." In IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 6520-6524, 2016.
2. Huang, Qizheng, Changchun Bao, Xianyun Wang, and Yang Xiang. "DNN-Based Speech Enhancement Using MBE Model." In 16th International Workshop on Acoustic Signal Enhancement (IWAENC), pp. 196-200, 2018.
3. Xiang, Yang, and Changchun Bao. "A codebook-driven speech enhancement method by exploiting speech harmonicity." In IEEE International Conference on Signal Processing, Communications and Computing (ICSPCC), 2017.
4. Tantibundhit, Charturong, Franz Pernkopf, and GernotKubin. "Speech enhancement based on joint time-frequency segmentation." In IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 4673-4676. 2009.
5. Kavalekalam, Mathew Shaji, MadsGræsbøll Christensen, and Jesper B. Boldt. "Binaural speech enhancement using a codebook based approach." In IEEE International Workshop on Acoustic Signal Enhancement (IWAENC), pp. 1-5,2016.
6. Bao, Feng, Hui-jing Dou, Mao-shenJia, and Chang-chunBao. "Speech enhancement based on a few shapes of speech spectrum." In IEEE China Summit & International Conference on Signal and Information Processing (ChinaSIP), pp. 90-94, 2014
7. Lu, Yang, and Philipos C. Loizou. "Speech enhancement by combining statistical estimators of speech and noise." In IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 4754-4757, 2010.
8. Leitner, Christina, and Franz Pernkopf. "Speech enhancement using pre-image iterations." In 2012 International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 4665-4668. IEEE.
9. Bai, Haichuan, FengpeiGe, and Yonghong Yan. "DNN-based speech enhancement using soft audible noise masking for wind noise reduction." China Communications 15, no. 9 (2018): 235-243.

ISSN:0971-1260

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10. Patton, Brian, Jan Skoglund, Jeremy Thorpe, John Hershey, Kevin Wilson, Michael Chinen, Richard F. Lyon, and Rif A. Saurous. "EXPLORING TRADEOFFS IN MODELS FOR LOW-LATENCY SPEECH ENHANCEMENT." (2018).
11. Tu, Yan-Hui, Ivan Tashev, ShuaybZarar, and Chin-Hui Lee. "A hybrid approach to combining conventional and deep learning techniques for single-channel speech enhancement and recognition." In IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 2531-2535. 2018.
12. Mahesh, R. "Speech Enhancement for Speaker Identification." In 2018 9th International Conference on Computing, Communication and Networking Technologies (ICCCNT), pp. 1-4. IEEE, 2018..
13. Stahl, Johannes, and PejmanMowlaee. "A pitch-synchronous simultaneous detection-estimation framework for speech enhancement." IEEE/ACM Transactions on Audio, Speech and Language Processing (TASLP) 26, no. 2 (2018): 436-450
14. Bosco, Julien, and Eric Plourde. "Speech enhancement using both spectral and spectral modulation domains." In IEEE 30th Canadian Conference on Electrical and Computer Engineering (CCECE), pp. 1-4, 2017.
15. Rehr, Robert, and TimoGerkmann. "On the importance of super-Gaussian speech priors for machine-learning based speech enhancement." IEEE/ACM Transactions on Audio, Speech and Language Processing (TASLP) 26, no. 2 (2018): 357-366
16. Liu, Yun, Hui Zhang, Xueliang Zhang, and Linju Yang. "Supervised Speech Enhancement with Real Spectrum Approximation." In ICASSP IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 5746-5750, 2019.
17. Kavalekalam, Mathew Shaji, MadsGræsbøll Christensen, and Jesper B. Boldt. "Binaural speech enhancement using a codebook based approach."In IEEE International Workshop on Acoustic Signal Enhancement (IWAENC), pp. 1-5, 2016.
18. Lu, Yang, and Philipos C. Loizou. "Speech enhancement by combining statistical estimators of speech and noise." In IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 4754-4757, 2010.
19. Ren, Yao, and Michael T. Johnson. "Auditory coding based speech enhancement." In IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 4685-4688, 2009.
20. He, Qi, Chang-chunBao, and FengBao. "Multiplicative update of AR gains in codebook-driven speech enhancement." In 2016 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 5230-5234. IEEE, 2016.
21. You, Chang Huai, SusantoRahardja, and Haizhou Li. "Speech enhancement for telephony name speech recognition." In 2008 IEEE International Conference on Multimedia and Expo, pp. 973-976. IEEE, 2008.
22. Zhang, Dong-ming, Chang-chunBao, and Feng Deng. "Integrating codebook and Wiener filtering for speech enhancement." In 2015 IEEE International Conference on Signal Processing, Communications and Computing (ICSPCC), 2015.
23. Rao, Yu, YiyaHao, Issa MS Panahi, and Nasser Kehtarnavaz. "Smartphone-based real-time speech enhancement for improving hearing aids speech perception." In 38th Annual International Conference of the IEEE Engineering in Medicine and Biology Society (EMBC), pp. 5885-5888, 2016.
24. Fu, Zhong-Hua, and Jhing-Fa Wang. "Speech presence probability estimation based on integrated time-frequency minimum tracking for speech enhancement in adverse environments." In IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 4258-4261, 2010.

ISSN:0971-1260

Vol-22-Issue-41-December-2019

25. Takada, Shintaro, Tetsuji Ogawa, KenzoAkagiri, and Tetsunori Kobayashi. "Speech enhancement using square microphone array for mobile devices." In 2008 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 313-316. IEEE, 2008.
26. Tantibundhit, Charturong, Franz Pernkopf, and GernotKubin. "Joint time–frequency segmentation algorithm for transient speech decomposition and speech enhancement." IEEE Transactions on Audio, Speech, and Language Processing 18, no. 6 (2009): 1417-1428.
27. Deisher, Michael E., and Andreas S. Spanias. "HMM-based speech enhancement using harmonic modeling." In 1997 IEEE International Conference on Acoustics, Speech, and Signal Processing, vol. 2, pp. 1175-1178. IEEE, 1997.
28. Jeong, So-Young, Jae-HoonJeong, and Kwang-Cheol Oh. "Dominant speech enhancement based on SNR-adaptive soft mask filtering." In 2009 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 1317-1320. IEEE, 2009.
29. Premananda, B. S., and B. V. Uma. "Speech enhancement using temporal masking in presence of near-end noise." In International Conference on Circuits, Communication, Control and Computing, pp. 263-266. IEEE, 2014.
30. So, Stephen, Kamil K. Wójcicki, James G. Lyons, Anthony P. Stark, and Kuldip K. Paliwal. "Kalmanfitler with phase spectrum compensation algorithm for speech enhancement." In 2009 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 4405-4408. IEEE, 2009.
31. Chen, Jianming, and Zhicheng Liang. "The Application of Deep Neural Network in Speech Enhancement Processing." In 2018 5th International Conference on Information Science and Control Engineering (ICISCE), pp. 1263-1266. IEEE, 2018.
32. He, Qi, FengBao, Changchun Bao, Qi He, FengBao, and Changchun Bao. "Multiplicative update of auto-regressive gains for codebook-based speech enhancement." IEEE/ACM Transactions on Audio, Speech and Language Processing (TASLP) 25, no. 3 (2017): 457-468.
33. Nataraj, VinayshankarSomalara, M. S. Athulya, and SathideviPuthumangalathuSavithri. "Single channel speech enhancement using adaptive filtering and best correlating noise identification." In 2017 IEEE 30th Canadian Conference on Electrical and Computer Engineering (CCECE), pp. 1-4. IEEE, 2017.
34. Chang, Joon-Hyuk, Q-Haing Jo, Dong Kook Kim, and Nam Soo Kim. "Global soft decision employing support vector machine for speech enhancement." IEEE Signal Processing Letters 16, no. 1 (2008): 57-60.
35. Mohammadiha, Nasser, Paris Smaragdis, and Arne Leijon. "Supervised and unsupervised speech enhancement using nonnegative matrix factorization." IEEE Transactions on Audio, Speech, and Language Processing 21, no. 10 (2013): 2140-2151.
36. Andy.C and Sandeep Kumar, “An Appraisal on Speech and Emotional Recognition Technologies based on Machine Learning” in International Journal of Recent Technology and Engineering (IJRTE), Vol. 8, No. 5, July 2019 with ISSN: 2277-3878.
37. Rehr, Robert, and TimoGerkmann. "On the importance of super-Gaussian speech priors for machine-learning based speech enhancement." IEEE/ACM Transactions on Audio, Speech and Language Processing (TASLP) 26, no. 2 (2018): 357-366.

