Comparative Analysis of EMD and VMD Algorithm in Speech Enhancement

Rashmirekha Ram, Department of Electronics and Communication Engineering, Siksha ‘O’ Anusandhan University, Bhubaneswar, India
Mihir Narayan Mohanty, Department of Electronics and Communication Engineering, Siksha ‘O’ Anusandhan University, Bhubaneswar, India

ABSTRACT

Signal enhancement is useful in many areas like social, medicine and engineering. It can be utilized in data mining approach for social and security aspects. Signal decomposition method is an alternative choice due to the elimination of noise and signal enhancement. In this paper, two different algorithms such as Empirical Mode Decomposition (EMD) and Variational Mode Decomposition (VMD) are used. The bands are updated concurrently and adaptively in each mode. That performs better than the traditional methods for non-recursive signals. Further it has been investigated that VMD outperforms EMD due to its self-optimization methods as well as adaptively using Wiener filter. It is shown in the result section. Different noise levels as 0dB, 5dB, 10dB and 15dB are considered for input signal.

KEYWORDS


INTRODUCTION

Removal of unwanted noise is a common issue in speech enhancement. Different types of noise such as airport noise, car noise, babble noise, street noise, fan noise etc. degrades the originality and quality of the speech signal which affects the relevance of various speech applications e.g. speech recognition, feature extraction, mobile communication, teleconferencing systems etc. Speech enhancement algorithms are very essential to enhance the quality of the degraded speech as well as to increase the signal-to-noise ratio (SNR). Due to rise in the usage of real time applications, researchers have focused more on speech enhancement. Different algorithms have been proposed from several years by researchers. Some of them are filtering methods, spectral subtractive algorithms, statistical model based methods, subspace algorithms, noise estimation algorithms, etc. But the selection of algorithms always creates difficulty between the speech distortion and the noise reduction (Loizou, 2007).

Due to the nonstationary nature of the speech signal it comes across many difficulties, hence spectral analysis of the signal is carried out before processing of the speech signal. To reduce the spectral effects and the background noise, S.F Boll proposed the Spectral Subtraction (SS) method in 1979. It is one of the first methods proposed for speech enhancement. The clean speech is prevailed when the estimated noise spectrum is subtracted from the noisy speech spectrum, but the residual noise remains in the clean speech spectrum, which is one of the major drawbacks of this method (Boll, 1979). Additionally, this method does not provide satisfactory results for nonstationary signals. So, adaptive algorithms can be used as an alternative to improve the SNR in noisy speech. Ram

DOI: 10.4018/IJNCR.2017010102

Copyright © 2017, IGI Global. Copying or distributing in print or electronic forms without written permission of IGI Global is prohibited.
and Mohnaty (2016) have applied Recursive Least Squares (RLS) method to enhance the degraded speech. The adaptive RLS algorithm results better enhanced signal than the SS method. The adaptive algorithms like least mean squares (LMS), RLS, Normalized Least mean squares (NLMS) are applied to the different SNR levels of the speech signal. RLS provides better convergence rate whereas the computational complexity is less in LMS and NLMS. So, the selections of adaptive algorithms are always difficult. The State Space RLS (SSRLS) algorithm is an extension of RLS algorithm and it provides better tracking performance. The results of SSRLS algorithms show that the mean square error (MSE) curves converge better as compared to other adaptive algorithms (R. Ram et.al, 2016). The cancellation of impulsive noise is also possible by using SSRLS algorithm for speech enhancement (R. Ram et.al, 2016).

Speech enhancement using signal subspace method is proposed by Yariv Ephraim. In this method, the noisy signal is decomposed into two different subspaces i.e. the noise subspace and the signal subspace. The clean speech spectrum is estimated from the signal subspace by removing the noise subspace. The signal decomposition is executed by using Karhunen-Loeve transform (KLT) to the noisy signal. The empirical estimation of the signal is involved in KLT whereas in the SS method Fourier transform is used to estimate the noise spectrum (Ephraim & Van Trees, 1995). Different noise estimation algorithms have been used for speech enhancement. But it is not easy to decrease the impact of transitory noise associated with the narrow band components. So, a standardized 4th order cumulant based adaptive method is projected to reduce the transient noise. The convergence of the curve is far better than the others. But a large step size is occurred. By using variable step size, different noise cancellation algorithms are also designed for speech enhancement which attenuates different types of noise present in the speech signal (Sasaoka et al., 2014; Hosseini et al., 2014). But when speech signal is affected by multiple noises, the adaptive self-switching noise removal method is well suited. The optimized cuckoo-search algorithm gives better result than the generalized LMS and RLS algorithms (Anoop, Rao & Nidhin, 2015).

Different statistical methods are also used for speech enhancement. The log spectral amplitude estimator is educed using the minimum mean square error (MMSE) for enhancement of speech. A very less amount of residual noise is remained as compared to spectral amplitude estimator which is short in time. The speech enhancement is also better in this noise estimation method. The model based systems are also designed for speech enhancement. Authors have designed Hidden Markov model by using MMSE principle. This method is very effective when the speech signals are affected by nonstationary noise. The noise adaptation method is highly efficient for any variations of noise affected speech (Ephraim & Malah, 1985; Sameti, 1998).

Decomposition methods are capable of extracting and removing the low and the high frequency components from the signal. Speech enhancement is employed by using the EMD method and the Teager- Kaiser Energy operator (TKEO). This operator helps in estimating the adaptive threshold of signal IMF's by separating the voiced frame and unvoiced signal frame. The Hilbert transform can be used in EMD for enhancing the speech. This method provides better result in nonstationary and the nonlinear signal (Huang, 1998; Khaldi, 2008). The EMD based filtering (EMDF) is designed for low frequency noise in (Khaldi, Boudraa, & Komaty, 2014). The filter works in the postprocessing stage in speech enhancement. The minima controlled recursive averaging algorithm is used for noise estimation. Different types of noise are tested to show the accuracy of EMDF method. L. Zao et al. have proposed Hurst based EMD method for speech enhancement. The EMD is employed to the noisy signal to find the set of IMF's. For the reconstruction of the speech signal the choice of IMF's are depend on the Hurst exponent. The most affected noise components are acquired by the Hurst exponent and the reconstruction of the signal is conceived by using the least affected IMF's (Chatlani & Soraghan, 2012). The same EMD method is used for noise reduction by decomposing the noisy signal adaptively into IMF's and the signal is reconstructed back using the shrinkage function (Zao, Coelho & Flandrin, 2014).
Inferring Gene Regulatory Networks from Genetical Genomics Data
www.igi-global.com/chapter/inferring-gene-regulatory-networks-genetical/38232?camid=4v1a

Customer Profiling in Complex Analytical Environments Using Swarm Intelligence Algorithms
www.igi-global.com/chapter/customer-profiling-in-complex-analytical-environments-using-swarm-intelligence-algorithms/161076?camid=4v1a