Abstract

The Conventional acoustic echo canceller encounters problems like slow convergence rate (especially for speech signal) and high computational complexity as the identification of the echo path requires filter with more than a thousand taps, non-stationary speech input, slowly time-varying systems to be identified. The demand for fast convergence and less MSE level cannot be met by conventional adaptive filtering algorithms. There is a need to be computationally efficient and rapidly converging algorithm. The LMS algorithm is easy to implement and computationally inexpensive. This feature makes the LMS algorithm attractive for echo cancellation applications. The results show that the steady state value of the output estimation error increases with increasing the step size parameter and the optimality of the LMS algorithm is no longer hold. The results also reveal that choosing the smallest value of the step size parameter guarantees the smallest mis-adjustment but might not meet the convergence criteria.

References

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Analysis the results of Acoustic Echo Cancellation for speech processing using LMS Adaptive Filtering Algorithm.


**Index Terms**

Computer Science  
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**Keywords**

LMS Algorithm  
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