DEPARTMENT OF ELECTRONICS AND
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INDIAN INSTITUTE OF INFORMATION
TECHNOLOGY, DESIGN AND
MANUFACTURING KANCHEEPURAM
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Synopsis Of

Acoustic Signal Enhancement in Hearing Aids and Mobile Audio Devices

A Thesis

To be submitted by

VANAMADI RAVI

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Of

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1.	Name	:	Vanamadi Ravi
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Signature of the Scholar
(Vanamadi Ravi)

Signature of the Guide
(Dr. Asutosh Kar)

1 Abstract

Acoustic feedback cancellation is a process to mitigate the acoustic coupling between the speaker and microphone of a hearing aid. Adaptive filters are used in this process and play a significant role by minimizing mean square error iteratively, which optimizes the overall feedback cancellation performance. The state-of-the-art research focuses on improving the convergence and steady-state of the adaptive filtering algorithms employed for feedback cancellation for time-varying feedback paths under various inputs and noise conditions. A convex combination of adaptive filters based on a combined proportionate least mean square approach has been proposed in this thesis. The proposed algorithms have been analysed for mean and mean-square convergence, and simulation results have been compared with the existing combined least mean square based approaches. The proposed feedback cancellation framework is found to achieve a perfect trade-off between the convergence and steady-state without degrading the overall feedback cancellation performance. The proposed algorithm outperforms the recent convex combination-based adaptive feedback cancellers as well.

In the past few decades, adaptive filtering algorithms have been used extensively for addressing monophonic and stereophonic acoustic echo that usually occurs in hands-free communication and audio conferencing. The biggest challenge in the present-day acoustic echo cancellers is the demand for higher-order adaptive filters, which degrade the overall convergence. State-of-the-art acoustic echo cancellation algorithms employ sub-filters based frameworks and variable tap-length adaptive algorithms to address the issue of slower convergence. The major issue with this framework is pseudo optimum tap-lengths and the absence of comprehensive performance analysis of sub-filters based variable tap-length frameworks in such scenarios. This thesis presents convergence analysis for variable tap-length, undermodelled multiple sub-filters based monophonic and stereophonic acoustic echo cancellers for various input signals under multiple noise conditions. This analysis helps to set the critical parameters of variable tap-length algorithms used for multiple sub-filters based acoustic echo cancellation to avoid the implications of pseudo optimum filter length.

2 Objectives

The main goals of the thesis are as follows:

- To improve the convergence performance of the transversal digital filter by proposing advanced adaptive filter-based intervention for feedback cancellation in digital hearing aids.

- To devise an improved adaptive feedback canceller with a better trade-off between the convergence rate and steady-state using a convex combination of proportionate adaptive algorithms.

- To analyse the effects of pseudo optimum tap-length for the state-of-the-art multiple sub-filters (MSF) based adaptive acoustic echo cancellers.

- To comprehensively evaluate the convergence performance of variable tap MSF based stereophonic acoustic echo cancellation.

3 Existing gaps which were bridged

- The least mean square (LMS) based linear adaptive algorithms are commonly used for acoustic feedback cancellation (AFC) in digital hearing aids, given the simplicity and robustness (Kar *et al.*, 2020). On the other hand, the LMS adaptive filter's primary limitation is slow convergence due to a compromised step-size selection (Kar *et al.*, 2019). In recent times, the convex combination (Zhao *et al.*, 2019) (Arenas-Garcia *et al.*, 2016) of LMS adaptive filters is found to depict an improved trade-off between the convergence rate and steady-state while employed for AFC. However, there still exists a scope for improving the convergence further without degrading the steady-state and overall feedback cancellation performance for a time-varying feedback path under different noise conditions and input signals.

- Adaptive MSF based algorithms have gained attention in recent times for their potential applications in the process of acoustic echo cancellation (AEC) (Kar and Chandra, 2015). The most advanced combined-error MSF algorithm maintains a trade-off between the different-error and common-error sub-filters based algorithms, and it is employed for echo cancellation applications that need faster convergence. Variable tap-length adaptive filtering algorithms (Schüldt *et al.*, 2009) find the optimum filter length and are used for optimizing the order of the sub-filters in MSF-based echo cancellers. In case of a long room impulse response, the MSF-based variable tap-length algorithm achieves a pseudo-optimum tap-length, rendering the overall design undermodelled (Kar, 2015). Therefore, it becomes apparent to analyze the effects of the undermodelled adaptive filters for the variable tap-length MSF (VT-MSF) based echo cancellers.

- In a long room impulse response, stereophonic acoustic echo cancellation (SAEC) is carried out with the help of more than one adaptive filter, each having hundreds to thousands of filter coefficients (Kar and Swamy, 2017). The large filter order in SAEC degrades the convergence and increases the structural filter design complexity. Multiple sub-filters (MSF) and variable tap-length algorithms are independently proposed for SAEC scenarios to address these issues (Messini and Djendi, 2019). The MSF-based framework improves the convergence; on the other hand, the variable tap-length algorithm optimizes the weight requirement for adaptive filters. The effects of pseudo optimum filter order for VT-MSF based SAEC needs to be analysed as it helps to set the critical parameters in the process of dual-channel AEC.

4 Most important contributions

4.1 Convex combination of adaptive filters for feedback cancellation

A new approach with the convex combination of two adaptive filters for feedback cancellation in digital hearing aids has been presented in this Section. The mixing parameter controls the adaptation of the convex combination by selecting the appropriate step size for the adaptive filter (Arenas-Garcia *et al.*, 2016). The optimization of the mixing parameter has been evaluated, and it improves the overall feedback cancellation and thee by convergence performance in comparison to the algorithms presented in (Das *et al.*, 2017)(Anand *et al.*, 2017).

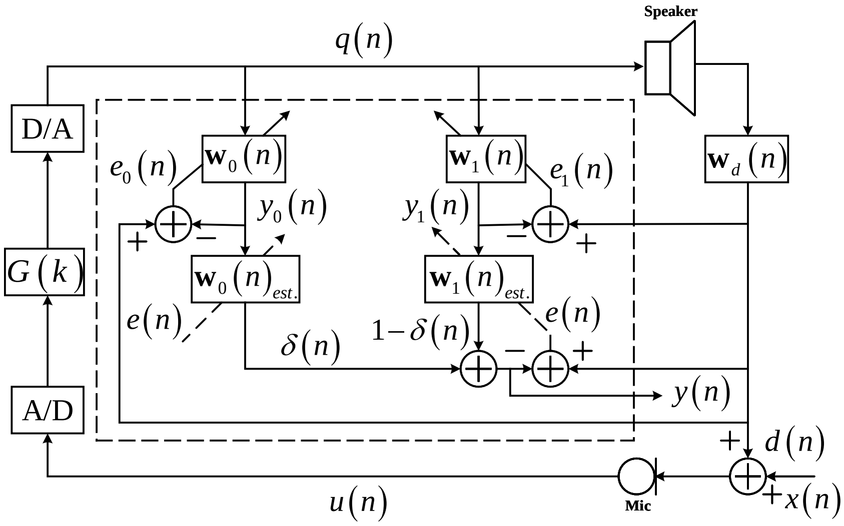


Figure 1: The proposed convex combination of two adaptive filters for AFC.

The proportionate adaptive filtering algorithms have been employed for the weight update of the transversal filters used in the convex combination. The proposed AFC is shown in Figure 1, where $\mathbf{w}_0(n)$ and $\mathbf{w}_1(n)$ are the two adaptive filters with equal step-sizes (Song and Zhao, 2019) (Anand *et al.*, 2018). The mean square deviation (MSD)

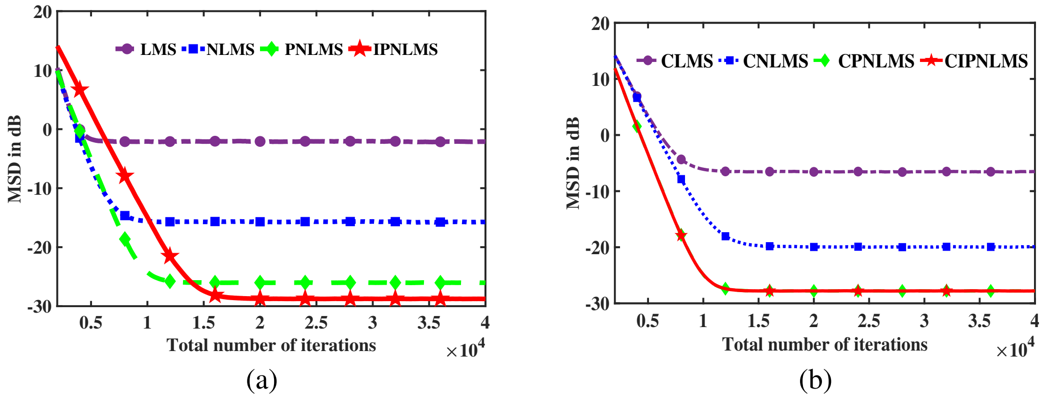
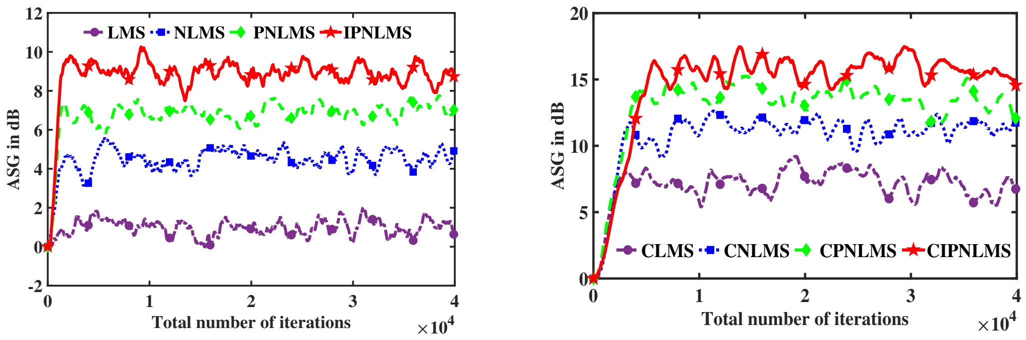


Figure 2: MSD performance comparison for the proportionate adaptive algorithms for the feedback cancellation with the white noise input signal (a) without convex combination (b) with a convex combination.

is used as a figure-of merit for performance comparison of the proposed proportionate normalised least mean square (PNLMS) and Improved PNLMS (IPNLMS) algorithms for AFC. The comparison without combination have been presented in Figure 2(a). It can be observed that the PNLMS algorithms outperform the LMS and normalised least mean square (NLMS) algorithms, and the IPNLMS algorithm outperforms the PNLMS algorithm. This convex combination of PNLMS (CPNLMS) and convex combination of IPNLMS (CIPNLMS) for hearing aid systems has been evaluated with a white noise input signal. The MSD performance for convex combination of LMS (CLMS) (Anand *et al.*, 2017), CNLMS, CPNLMS, and CIPNLMS algorithms have been presented in Figure 2(b). However, the proposed convex combination of CPNLMS and CIPNLMS feedback canceled algorithms outperform both above algorithms by achieving the highest MSD. Thus, it can be concluded that the MSD performance of the CPNLMS and CIPNLMS feedback canceller is higher than the existing state-of-the-art algorithms (Anand *et al.*, 2017), at the cost of a slower convergence.



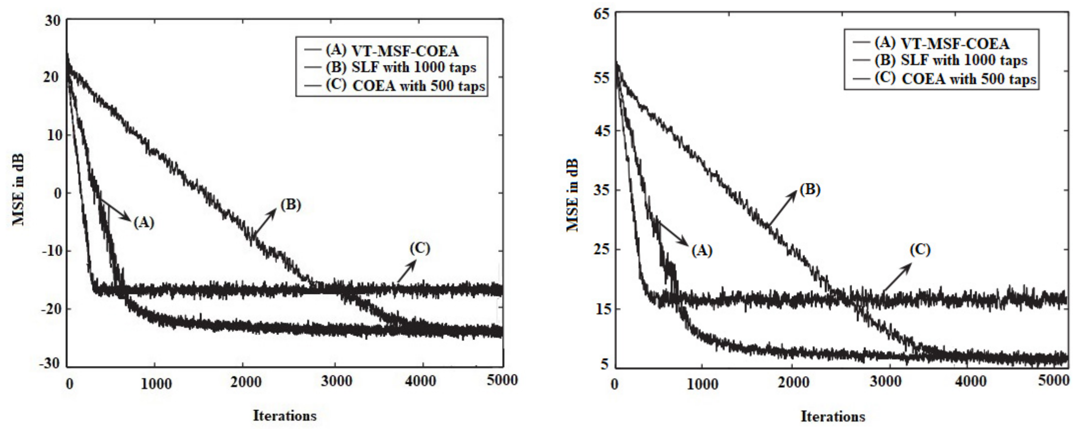
(a) (b)

Figure 3: ASG performance comparison for the proportionate adaptive algorithms for the feedback cancellation with the white noise input signal (a) without convex combination (b) with a convex combination.

Figure 3 shows the added stable gain (ASG) performance comparison results for a with and without combination of adaptive filters excited with the white noise input signal. The ASG characteristics of proportionate algorithms present improvements throughout all iterations, as illustrated in Figure 3(a). The convex combination of the proposed AFC model achieves better ASG performance over the existing algorithms, as shown in Figure 3(b). It can see that the proposed AFC algorithm outperforms the existing CLMS (Anand *et al.*, 2017) and CNLMS algorithms.

4.2 Effects of undermodelling in MSF based acoustic echo cancellation

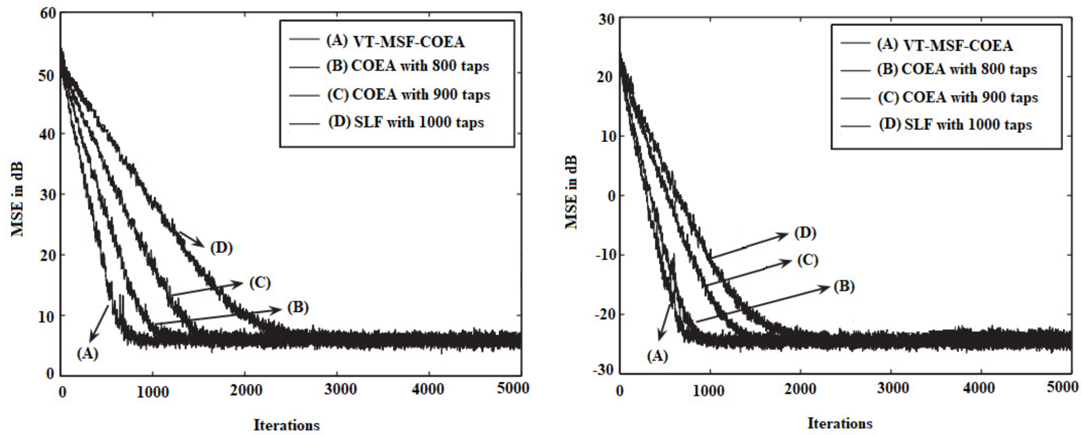
The effects of deficient-tap-length (Messini and Djendi, 2019)adaptive sub-filters in the VT-MSF-based combined-error (VT-MSF-COEA) monophonic echo canceller have been investigated in this Section. The performance of the undermodelled design is studied by evaluating the different-error and the common-error signals and analyzing them for the mean and mean-square convergence. For an improved understanding of the convergence rate under the constraints of pseudo-optimum filter length, the mean-square stability analysis of undermodelled VT-MSF-COEA is also presented. Moreover, the steady-state performance evaluation of the undermodelled VT-MSF-COEA presents a thorough understanding of the statistical behavior of the undermodelled MSF in the AEC scenario.



(a) (b)

Figure 4: Comparison of MSE at different noise conditions for VT-MSF-COEA, SLF and undermodelled MSF design (a) at SNR=25 dB (Low Noise). (b) at SNR=5 dB (High Noise)

Figure 4 depicts the variation of mean square error (MSE) concerning the total number of iterations for single long filter (SLF) and the most advanced MSF-based echo cancellers and its undermodelled variant due to the pseudo optimum tap-length occurring at 500. It can be seen that the undermodelled algorithm converges faster, at about 200 iterations, due to the lower number of coefficients. However, in low, moderate as well as high-noise conditions, the steady-state is reached at a compromised value of SNR, as evident in Figure 4(a) and Figure 4(b). Moreover, it can be clearly observed that VT-MSF-COEA maintains a perfect trade-off between the convergence rate and the steady-state MSE.



(a) (b)

Figure 5: Comparison of MSE at different noise conditions for VT-MSF-COEA, SLF and MSF designs with more than optimum tap-length (a) at SNR=25 dB (Low Noise). (b) at SNR=5dB (High Noise)

In Figure 5(a) and Figure 5(b), the MSE comparison of VT-MSF-COEA and SLF is shown at different SNR values for the tap-length set at a value higher than the optimum filter length, i.e., 750. It can be observed from the both figures that the variable tap-length algorithm, at the optimum filter length, has better convergence speed and maintains a perfect MSE for low, moderate, and high SNR values. On the other hand, SLF performs equally well for the steady-state MSE but at the cost of a slower convergence speed.

4.3 Convergence analysis of adaptive filter for undermodelled stereophonic acoustic echo cancellation

The VT-MSF-SAEC mean and mean-square convergence behavior for independent white Gaussian input data under deficient-length adaptive filter conditions have been presented in this Section (Kar and Swamy, 2017) (Messini and Djendi, 2019). The

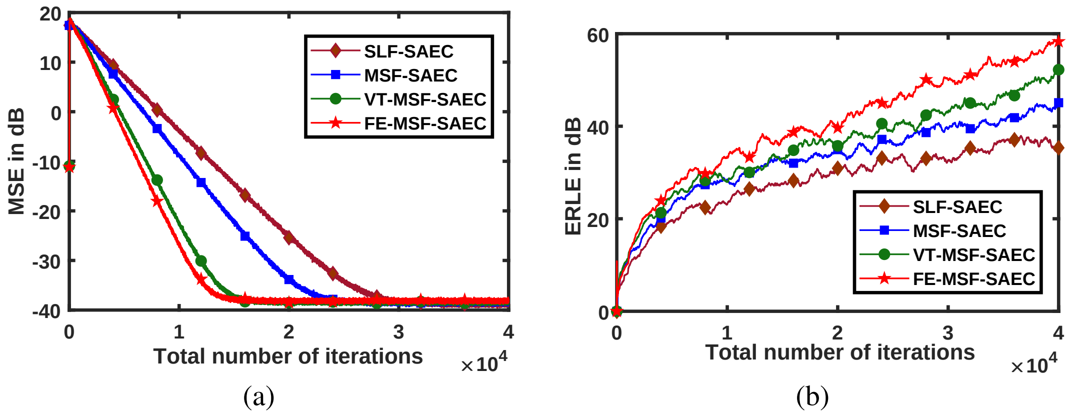


Figure 6: Performance characteristics comparision for proposed VT- MSF- based SAEC model for strongly correlated white Gaussian input signal (a) MSE. (b) ERLE.

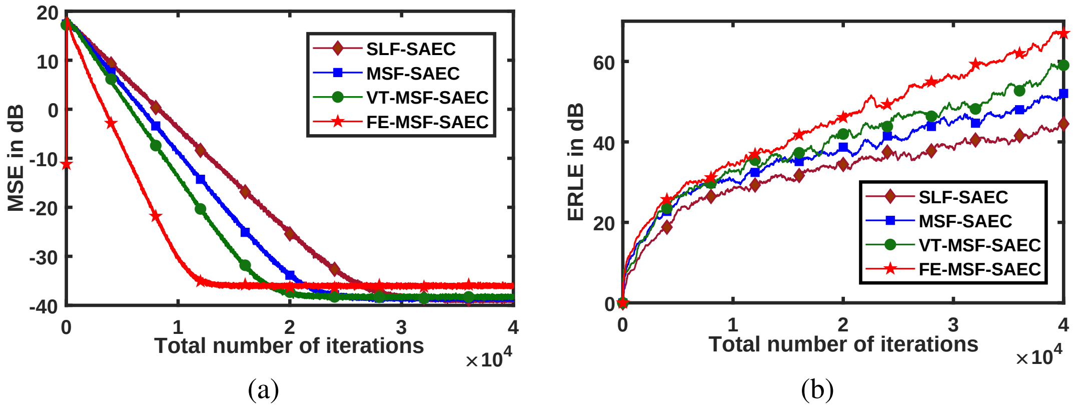
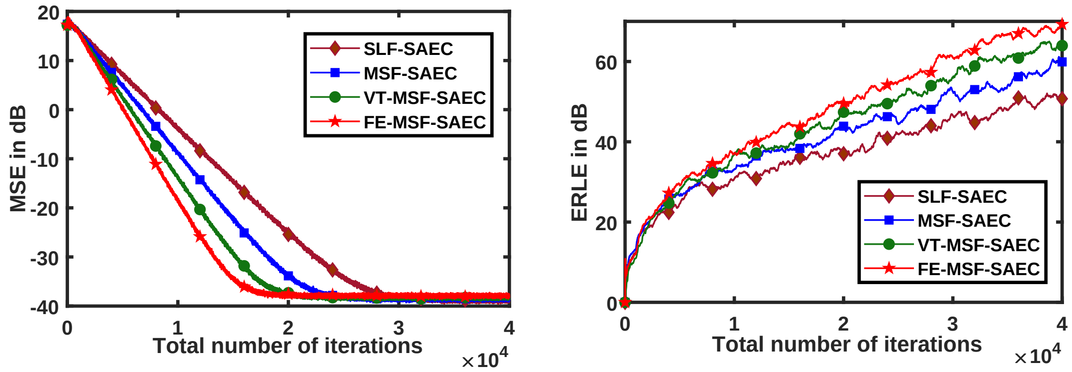


Figure 7: Performance characteristics comparision for proposed VT- MSF- based SAEC model for white Gaussian input signal (a) MSE. (b) ERLE.

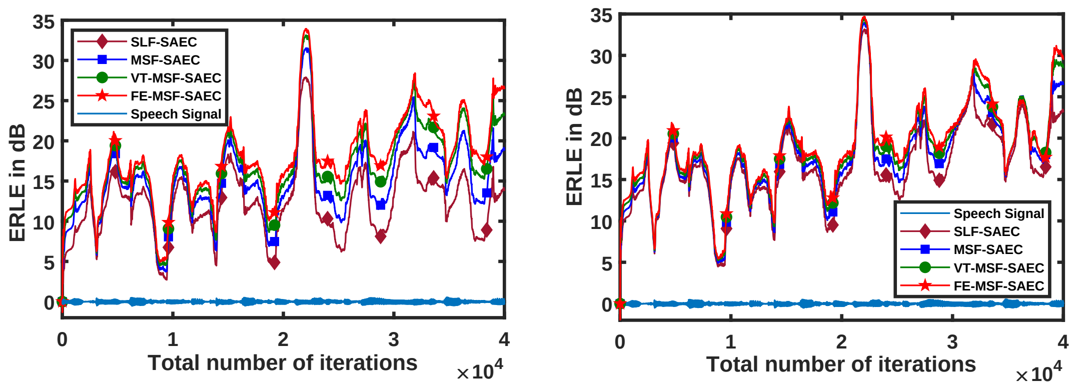
convergence and the MSE of the VT-MSF-SAEC algorithm and final error-based VT-MSF-SAEC (FE-MSF-SAEC) algorithm on the stereophonic channels have been performed (Djendi and Bounif, 2012). These analyses are done based on the assumption that the current input signal and the present coefficients of the adaptive filter are statistically independent. The undermodelled VT-MSF-SAEC algorithm mean and mean-square convergence update equations are analyzed mathematically.

The MSE performance comparison for the proposed undermodelled VT-MSF-SAEC adaptive algorithm for strongly correlated Gaussian input (Djendi and Bounif, 2012) is shown in Figure 6(a). The proposed adaptive algorithms converge faster than existing algorithms. The echo return loss enhancement (ERLE) characteristics of the proposed undermodelled VT-MSF-SAEC adaptive algorithm are illustrated in Figure 6(b). The proposed algorithm's ERLE performance is improved than the other counter parts. The



(a) (b)

Figure 8: Performance characteristics comparision for proposed VT- MSF- based SAEC model for weakly correlated white Gaussian input signal (a) MSE. (b) ERLE.



(a) (b)

Figure 9: ERLE characteristics comparison for proposed VT- MSF- based SAEC model for speech input signal at (a) SNR=30dB. (b) SNR=40dB.

MSE and ERLE characteristics of the undermodelled VT-MSF-SAEC and FE-MSF-SAEC algorithms have been presented in Figure 6(a) and Figure 6(b), respectively. From Figure 6, It can be observed that the proposed adaptive algorithm improves the steady-state MSE performance for fewer iterations as compared with the other algorithms. The MSE and ERLE characteristics for white noise and weakly correlated input signals (Djendi and Bounif, 2012) have been presented in Figure 7 and Figure 8 respectively. The FE-MSF-SAEC algorithm performs better compared with the other existing algorithms. The ERLE performance characteristics for speech signal input have been presented in Figure 9 under different SNR conditions. It clearly shows that the FE-MSF-SAEC algorithm depicts better ERLE performance over the other existing algorithms.

5 Conclusions

In this thesis, advanced adaptive filter based interventions have been proposed for fast converging acoustic feedback cancellation. The proposed convex combination of proportionate adaptive filters not only brings a perfect trade off between convergence and steady state performance but also depicts improved overall performance in comparison to the state-of-the-art feedback cancellation algorithms. A detailed mathematical analysis along with the simulation results have been presented in supporting of proposed proportionate adaptive algorithms.

An undermodelled adaptive filter degrades the overall acoustic echo cancellation performance in a system identification framework. In this thesis, the effect of undermodelled adaptive filter to pseudo optimum filter length in a variable tap length algorithm have been thoroughly analysed. The analysis have been extended to the most recent VT-MSF based acoustic echo cancellers. The effect of pseudo fractional tap-lengths for both monophonic and stereophonic sub-filters based different error, common error,combined error and final error have been presented and analysed respectively. The mean and mean square convergence analysis, stability and tracking analysis of the undemodelled AEC and SAEC helps to fix the key parameters which in return improves the overall echo cancellation.

6 Organization of the thesis

- Chapter 1: Introduction.
- Chapter 2: Literature Survey.
- Chapter 3: Feedback cancellation in digital hearing aids using proportionate adaptive algorithms.
- Chapter 4: Performance evaluation of an undermodelled monophonic acoustic echo canceller.
- Chapter 5: Effects of pseudo optimum tap-length on a stereophonic acoustic echo canceller.
- Chapter 6: Conclusion and future Scope.
- References.

7 List of publications

I. REFEREED JOURNALS BASED ON THE THESIS

1. **Ravi Vanamadi** and Asutosh Kar, "Convergence analysis for an undermodelled variable tap-length MSF-based stereophonic acoustic echo canceller", *Circuit Systems, and Signal Processing, Springer*, *(In press/ Accepted on 23 March 2021)* *Doi:10.1007/s00034-022-02033-3*.
2. **Ravi Vanamadi** and Asutosh Kar, "Feedback cancellation in digital hearing aids using convex combination of proportionate adaptive algorithms", *Applied Acoustics, Elsevier*, Vol. 182, pp.1-12, (2021) https://www.sciencedirect.com/science/article/pii/S0003682X21002693.
3. **Ravi Vanamadi**, Asutosh Kar, Ankita Anand, Banshidhar Majhi, and MNS Swamy, "Analyzing the effects of pseudo-optimum tap-length for an MSF-based acoustic echo canceller", *Applied Acoustics, Elsevier*, Vol. 150, pp.198-206, (2019) https://www.sciencedirect.com/science/article/abs/pii/S0003682X18311034.

II. PRESENTATIONS/PUBLICATIONS IN CONFERENCES BASED ON THE THESIS

1. **Ravi Vanamadi** and Asutosh Kar, "Performance Analysis of Transversal Filter Combination for Feedback Suppression in Hearing Aids", *2021 IEEE 18th India Council International Conference (INDICON)*, pp. 1-5, (2021) https://ieeexplore.ieee.org/abstract/document/8955136.
2. **Ravi Vanamadi**, Asutosh Kar, Srikanth Burra and Ankita Anand "Convergence Performance Evaluation of MSF-Based LMS Adaptive Algorithm", *2019 16th International Conference on Electrical Engineering/Electronics, Computer, Telecommunications and Information Technology (ECTI-CON)*, pp. 597-600, (2019) https://ieeexplore.ieee.org/document/9691582.

III. PRESENTATIONS/PUBLICATIONS IN CONFERENCES (Others)

1. **Ravi Vanamadi** and Asutosh Kar, "Power Transfer Function based Feedback Cancellation using Convex Combination of Adaptive Filters", *2022 IEEE 19th International Conference on Electrical Engineering/Electronics, Computer, Telecommunications and Information Technology (ECTI-CON)*, *(Accepted on 4 April 2022)*.

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