Switched Combination of Simplified Kalman Filter and Affine Projection Algorithm for Acoustic Feedback Cancellation

S.Siva Prasad[1*], CB Rama Rao[2]

[1]Research Scholar, Department of ECE, NationalInstitute of Technology, Warangal - 506004, India
*Email: sivaphd.nitw@gmail.com
[2]Professor, Department of ECE, National Institute of Technology, Warangal - 506004, India

Abstract
Acoustic coupling between the microphone and the loudspeaker is a major issue in open-fit digital hearing aids. When compared to a close-fit hearing aid, an open-fit dramatically reduces signal quality and limits the potential maximum stable gain. Adaptive feedback cancellation (AFC) is a practical method for reducing the influence of acoustic coupling. However, because to the high correlation between the loudspeaker signal and the incoming signal, it might induce bias in calculating the feedback path if not carefully considered, especially when the incoming signal is spectrally coloured, as in speech and music. For decreasing this bias, the prediction error method (PEM) is well recognised. In this paper we proposed a switched PEM based hybrid combination of simpliedkalman filter and affine projection algorithms (H-SKF-APA), with soft-clipping that allows for further increase in convergence/tracking rates, resulting in a better ability to recover from an unstable or howling state. Simulation results showed that the proposed algorithm performed better.

Keywords: Acoustic feedback cancellation, adaptive signal processing, Kalman filters, digital hearing aids, prediction-error method

1. Introduction
Acoustic feedback is a serious problem in digital hearing aids[1][2]. It is the primary source of howling, whistling, or screeching in hearing aids. Acoustic feedback is mostly caused by the acoustic coupling of the desired input signal with the loudspeaker signal at the microphone's input[3]. An acute degradation of the desired signal and troublesome howling increase as the gain increases in acoustic coupling.

There are numerous strategies for acoustic feedback cancellation (AFC); among them, adaptive filtering is effective, as seen in Fig. 1, which comprises the continuous estimation of the feedback signal and its subtraction from the microphone input signal[4]. In fig. 1, W represents the actual acoustic path between loudspeaker and microphone, w represents the estimated acoustic path, which is continuously estimated by an adaptive filter, and G represents the gain of the hearing aid.The primary shortcoming of this continuous adaptation strategy is that the desired signal is a combination of both feedback and input signals [5]. As a result of the correlation between the desired source signal and the loudspeaker signal, the feedback signal is in correlation with the source signal. Hence, the adaptive filter may suffer from a bias estimation problem [6].

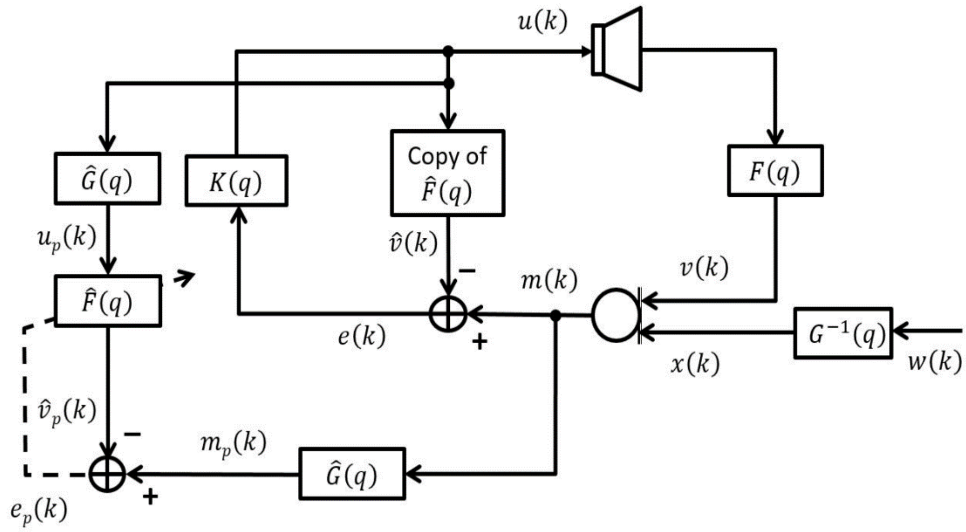

Figure.1 PEM-AFC System

The PEM-based technique is an effective one in obtaining an unbiased model of the feedback path by decorrelating the signals in the feedback loop, i.e., by pre-whitening the loudspeaker and the microphone signals with the inverse of the near-end source signal model and then these prefiltered signals are used to estimate the feedback path using the adaptive algorithm[6][7][8]. The design of an effective adaptive feedback controller demands a trade-off between improving steady-state performance with an unbiased estimate of the feedback path and the adaptive algorithm's convergence rate. The least mean square (LMS) algorithm has a low convergence rate for coloured signals and a high computational complexity when modelling a acoustic path with a large number of filter coefficients[8][9]. The step-size used to control the adoption of the adaptive filter coefficients is an important parameter in AFC, since it provides better trade-off between convergence speed and steady state misalignment. Different algorithms such as variable step-size(VSS) schemes[10][11], subband NLMS with VSS ref, affine projection algorithm with VSS [12] have been developed for better trade-off. The Kalman filter in time-domain and frequency-domain has been used in [13][14][15] to implement PEM-AFC system. Due to limited power handling capability of hearing aids, simplified versions of time-domain Kalman filter has been developed for PEM-AFC [13]. In [16], multiband structure Kalman filter is proposed for system identification problem, the same can be used for PEM-AFC system as it is less complex and having low processing delay due to parallel processing with subbands. In [14], [17], hybrid combination of SKF and NLMS algorithms has been proposed to overcome reconvergence inability when the howling occurs due to sudden change in feedback path.

In this article contents are organised as follows: section 2 discusses the signal model of PEM-AFC system; section 3 discuss about simplified Kalman filter for PEM-AFC and APA algorithm for AFC; section 4 discuss about the switched combination of SKF and APA algorithms; section 5 contains the simulation results and the final section provides the conclusion.

2. Signal Model of PEM-AFC System

The microphone signal is provided by eq(1), where $s(k)$, $x(k)$, $u(k)$ represent microphone, incoming signals and loud-speaker signals, respectively, and $F(q)$ is the real feedback path's polynomial transfer function in q. The loud-speaker signal is given by eq (2), G represents forward gain of the hearing aid,

$$m(k) = x(k) + F(q)u(k), \tag{1}$$

$$u(k) = G(q)m(k). \tag{2}$$

The Prediction-error method (PEM) is widely used to overcome biasing issue while evaluating the filter weights. The prefiltered signals of the microphone $s_p(k)$ and loud-speaker $u_p(k)$ are given by,

$$m_p(k) = \hat{H}(q)m(k), \tag{3}$$

$$u_p(k) = \hat{H}(q)u(k), \tag{4}$$

The Prewhitened error signal of PEM-AFC is,

$$e_p(k) = m_p(k) - \hat{\mathbf{f}}^T \mathbf{u}_p(k), \tag{5}$$

$$x(k) = H^{-1}(q)w(k), \tag{6}$$

3. Existing Algorithms

3.1 Simplified Kalman filter

The standard or generalized Kalman filtering (GKF) algorithm requires the O(M3) multiplications per sample, where M is length of the filter, which is too expensive in practice. Thus a simplified structure of Kalman filter is used in ref to reduce computational complexity. The simplified Kalman filter (SKF) was proposed in[13]. It was demonstrated that when the appropriate settings are used, it functions as a variable step-size filter. Its quick convergence and low misadjustmenthave been demonstrated for use in PEM-AFC. By using the notations in figure 1, if $r_v(n)$ represents the correlation matrix of priori state estimation error signal, $\hat{e}(n)$ is the apriori error signal vector between microphone signal and estimate of Prewhitened feedback signal, the simplified kalman filter algorithm equations [13] for PEM-AFC are given below:

$$r_v(k) = \{1 - \frac{tr\{[S_U(k-1) + \delta(k-1)I_P]^{-1} S_U(k-1)\}}{MP}\} r_v(k-1) + \sigma_x^2(k), \tag{7}$$

$$e_p(k) = M_p(k) - \boldsymbol{U}_P^T(k)\hat{f}(n-1), \tag{8}$$

$$\delta(k) = \frac{\sigma_x^2(k)}{r_v(k)} \tag{10}$$

$$S_U(k) = \boldsymbol{U}_P^T(k)\boldsymbol{U}_P(k), \tag{11}$$

$$\hat{f}(k) = \hat{f}(n-1) + \boldsymbol{U}_P(k)\{S_U(k) + \delta(k)I_P\}^{-1} e_p(k) \tag{12}$$

When the SKF technique begins to converge or there is a feedback change the possibility of howling is high. As a result, permitting certain NLMS repetitions during these observed times may be advantageous. It is proposed in [14] to use a switching combination of the NLMS and SKF algorithms, which is controlled by a stability detector. When instability is detected, the NLMS iterations are taken for weight estimation, otherwise, the SKF algorithm is used.

3.2 APA

We propose a novel rule to update the feedback route estimate to increase the convergence/tracking rates and steady-state error while maintaining high output sound quality. Standard adaptive algorithms like LMS, NLMS, and APA exhibit a trade-off between quick convergence/tracking rates and low steady-state error. When the step-size is big, such algorithms provide quick convergence/tracking rates but significant steady-state error, and vice versa. Figure 5 depicts the trade-off for AFC utilizing PEMSC-NLMS with various step-size values.
The update rule for APA algorithm is written as,

$$\hat{f}(k)=\hat{f}(k-1)+\mu_2\boldsymbol{U}_P(k)\left[\boldsymbol{U}_P^T(k)\boldsymbol{U}_P(k)+\delta_{APA}\mathbf{I}_P\right]^{-1}\mathbf{e}_p(k) \tag{13}$$

Furthermore, the affine projection algorithm (APA) is subject to this trade-off in relation to its projection order (P), i.e., the APA delivers quick convergence/tracking but significant steady-state error when P is big and vice versa[17]. When the projection order grows to a specific level, for example, from P=2 to P=6 in the experiment, the PEMSC-APA converges quicker but produces a bigger steady-state error. P=8 results in nearly no improvement in PEMSC-APA convergence, but the worst steady-state error when compared to the identical experiment with P=6.

4. Proposed algorithm: Switched combination of PEM-SKF and APA algorithms(SW-SKF-APA)

In this paper, we have proposed a hybrid algorithm with the combination of PEM-SKF and APA algorithms. As it is discussed, the PEM-SKF algorithm fails to track the sudden feedback changes and which will result in howling effect. To overcome this we are proposing the switched combination of PEM-SKF and APA. In the normal scenario i.e., when there is no change in feedback path, PEM-SKF runs and provides the good tracking. When the howling is observed with the help of stability detector, the APA algorithm is used in evaluating the feedback path[17].

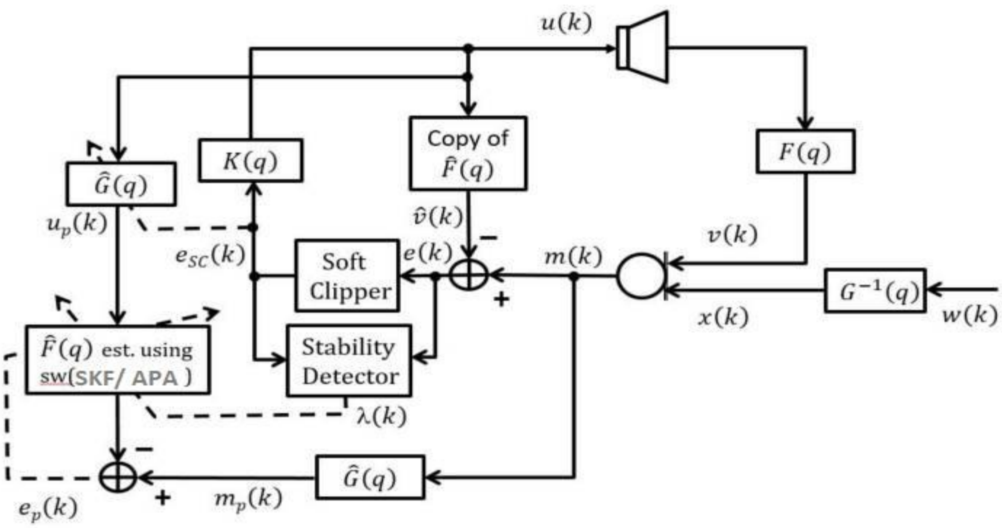

Figure.2HSKF-APA algorithm for PEM-AFC System

The update equation for the proposed hybrid algorithm is given as,

$$\hat{f}(k) = \hat{f}(k-1) + \lambda(k)\boldsymbol{U}_P(k)\{S_U(k) + \delta(k)I_P\}^{-1}e_p(k)$$
$$+\mu_2\left[1-\lambda(k)\right]\boldsymbol{U}_P(k)\left[\boldsymbol{U}_P^T(k)\boldsymbol{U}_P(k)+\delta_{APA}\mathbf{I}_P\right]^{-1}\mathbf{e}_p(k) \quad (14)$$

The recommended method applies soft-clipping (SC) to the error signal, resulting in the soft-clipping error signal,

$$e_{SC}(k) = \alpha \tanh\left(\frac{e(k)}{\alpha}\right), \quad (15)$$

where α is a scaling parameter. We choose α such that the most likely range of the incoming signal is within the linear range of the tanh-function, i.e., $x(k) \approx \alpha \tanh\left(\frac{x(k)}{\alpha}\right)$ may be used to identify AFC instability. This SC enables the error signal to have regulated nonlinearity. The nonlinearity is therefore known, and the AFC may be maintained linear. As a consequence, the performance of feedback cancellation has improved. We use a SCSD to generate a control signal, $\lambda(k)$,as seen below

$$\lambda(k) = \Gamma\left\{\left|e_{SC}(k) - e(k)\right| \gamma\right\}, \quad (16)$$

where γ is a decision threshold that determines the detector's sensitivity and is a binary function that returns 1 if the inequality holds and 0 if it does not.

5. Simulation Results

The performance of the H-NLMS, H-SKF, and SW-SKF-APA algorithms was investigated using similar feedback path parameters measured in both normal and nearest feedback pathway circumstances. The arriving signal was a concatenation of male and female NOIZEUS database voice. The amplitude responses of the observed auditory feedback channels are presented in Fig. 3, where the first (H1(f)) and second (H2(f)) were measured in free-field and with a telephone receiver situated close to the ear, respectively .

Two metrics are used to compare the algorithms: estimation error and achievable stable amplification. The first one, i.e., the estimation erroris measured in terms of misalignment (Mis), which is defined as the normalized difference between the true (acoustic impulse response) and estimated feedback paths, which is often expressed in decibels (dB).

$$\mathrm{Mis}(l) = 20\log_{10}\frac{\|\mathbf{f}_r(l)\|}{\|\mathbf{f}_t(l)\|} \quad (17)$$

where $\mathbf{f}_r(\kappa) = \mathbf{f}_t(\kappa) - \hat{\mathbf{f}}(\kappa)$. Added stable gain (MSG) is the second parameter and it is defined as the maximum stable gain that may be achieved at a given time if the forward path is spectrally constant.

$$\mathrm{ASG}(l) = -20\log_{10}\left[\max_{l\in \mathrm{P}(l)}\left|F_r(l,k)\right|\right] \quad (18)$$

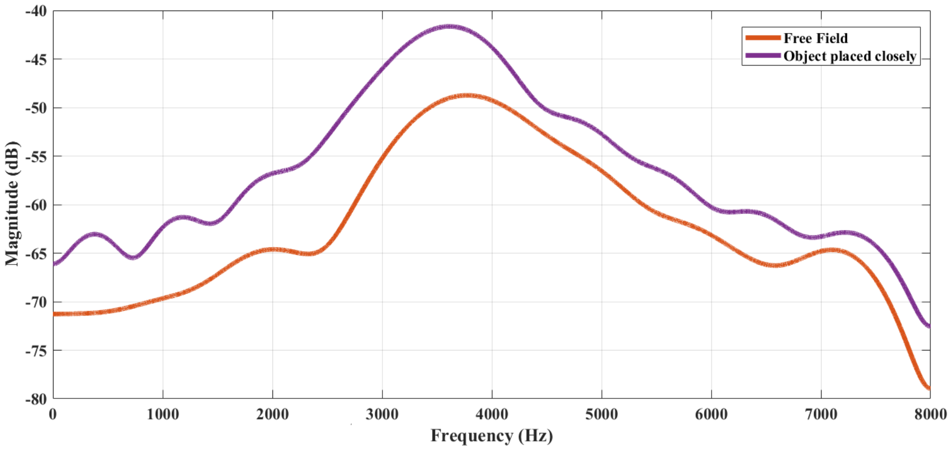

Figure 3. Frequency characteristics of the acoustic feedback paths.

In Fig. 4 and 5, the proposed and existing algorithms' MIS and ASG performance are compared when the incoming signal is speech at 0 dB loudness level. It is clear that the suggested algorithm achieves superior MIS and ASG performance, as well as faster convergence and tracking abilities. Furthermore, the HSKF-APA had a PESQ value of 3.81, whereas the H-SKF and H-NLMS had PESQ scores of 3.74 and 3.42, respectively.

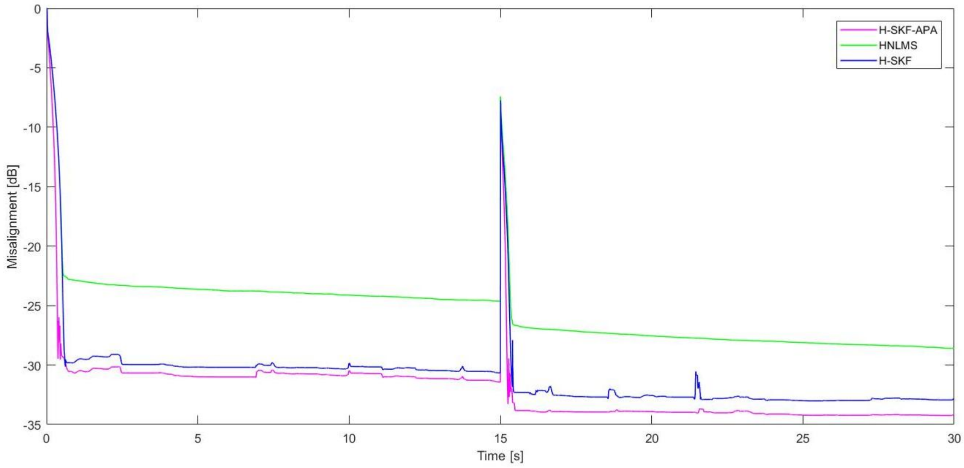

Figure 4: Misalignment comparison of HSKF-APA, HSKF and HNLMS for the speech input signal

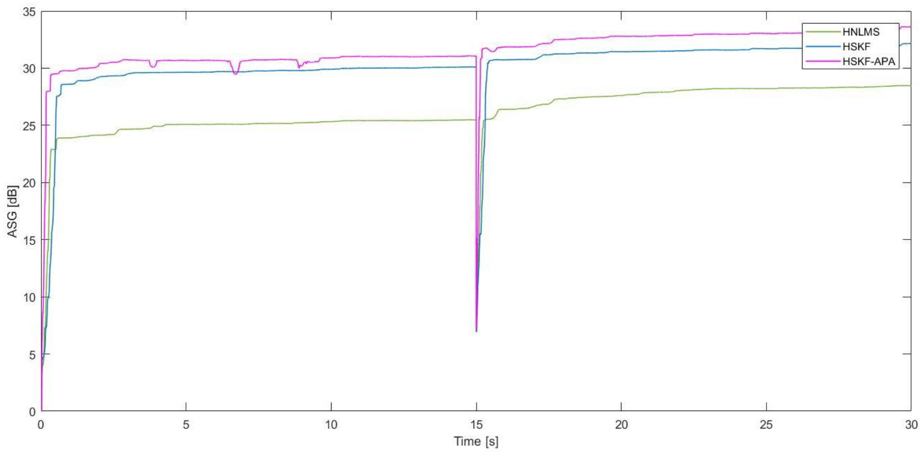

Figure 5: ASG comparison of HSKF-APA, HSKF and HNLMS for the speech input signal

When the incoming music input is set to 0 dB, the MIS and ASG performance of the analysed algorithms is shown in Fig. 6 and 7. The performance of proposed algorithm shown better performance compared to the existing algorithms.

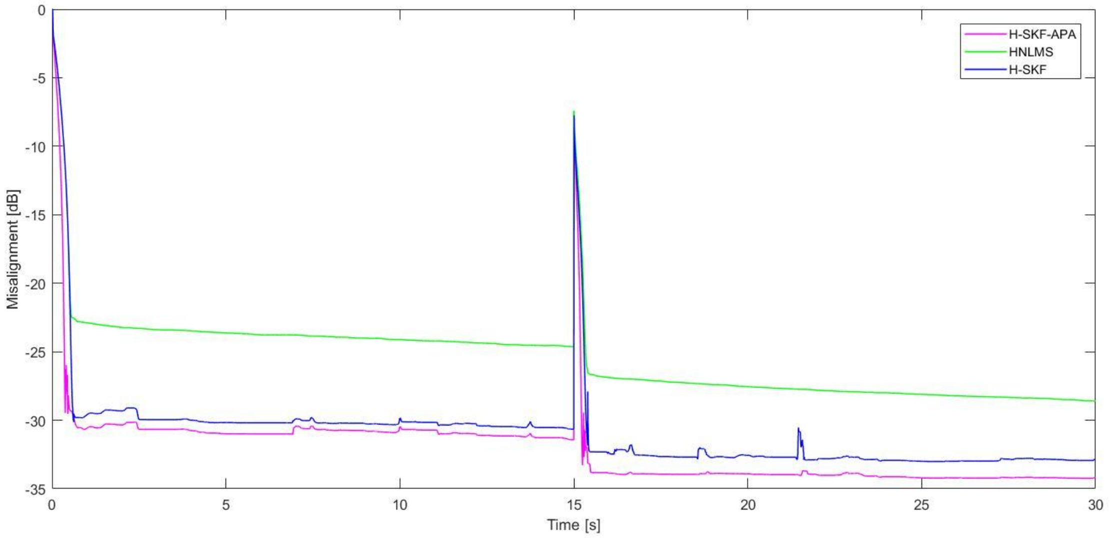

Figure 6: Misalignment comparison of HSKF-APA, HSKF and HNLMS for the music input signal

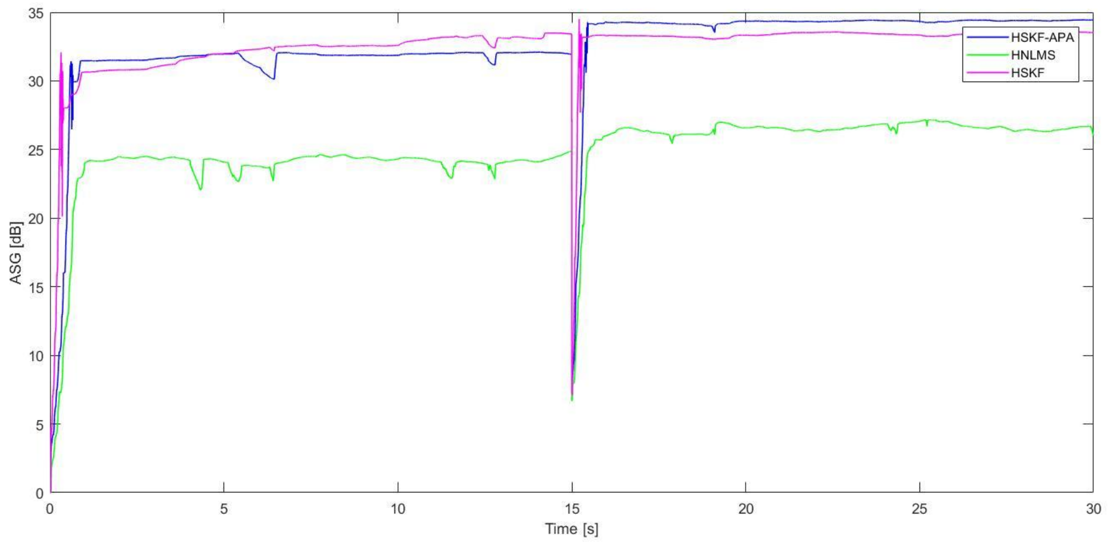

Figure 7: ASG comparison of HSKF-APA, HSKF and HNLMS for the music input signal

6. Conclusion:

In this paper, we proposed a switched combination of simplifiedKalman algorithm and affine projection algorithm i.e., HSKF-APA to implement PEM-AFC system. This algorithm helps to overcome the reconvergence inability when the feedback path changes or howling occurs. Computer simulation indicated that the proposed algorithms performance is better in terms of both faster convergence and low steady state error compared to HSKF and HNLMS algorithms. Also, the PESQ score of the proposed algorithm has shown slight improvement.

7. References:

[1] J. M. Kates, "Digital Hearing Aids," *Plural Publishing*, vol. Thieme, 2008.

[2] A. Spriet, G. Rombouts, M. Moonen and J. Wouters, "Feedback Control in Hearing Aids.," in *In: Benesty J., Sondhi M.M., Huang Y.A. (eds) Springer Handbook of Speech Processing.*, Berlin, Heidelberg, Springer., 2008, pp. 979-1000.

[3] M. G. Siqueira and A. Alwan, "Steady-State Analysis of Continuous Adaptationin Acoustic Feedback Reduction Systems for Hearing-Aids," *IEEE Transactions on Speech and Audio Processing*, vol. 8, July 2000.

[4] Vasundhara, G. Panda and N. B. Puhan, "An improved block adaptive system for effective feedback cancellation in hearing aids," *Digital Signal Processing*, vol. 48, pp. 216-225, Jan 2016.

[5] C. S. B. J. Paleologu C., "An overview on optimized NLMS algorithms for acoustic echo cancellation," *EURASIP J. Adv. Signal Process.*, vol. 97, 2015.

[6] T. v. W. J. W. a. M. M. G. Bernardi, "Adaptive Feedback Cancellation Using a Partitioned-Block Frequency-Domain Kalman Filter Approach With PEM-Based Signal Prewhitening," *in IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 25, pp. 1784-1798, Sept. 2017.

[7] R. C. Prasad S.S., "Acoustic Feedback Cancellation Using Optimal Step-Size Control of the Partition Block Frequency-Domain Adaptive Filter," in *Advances in Communications, Signal Processing, and VLSI. Lecture Notes in Electrical Engineering*, vol. 722, 2021, pp. 139-151.

[8] K. W. Schneider.M, "The generalized frequency-domain adaptive filtering algorithm as an approximation of the block recursive least-squares algorithm," *EURASIP Journal on Advances in Signal Processing*, 2016.

[9] F. T. L. T. &. N. S. Albu, " A combined variable step size strategy for two microphones acoustic feedback cancellation using proportionate algorithms," in *Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA ASC), IEEE*, 2017.

[10] M. A. F. &. C. H. Rotaru, "A variable step size modified decorrelated NLMS algorithm for adaptive feedback cancellation in hearing aids," in *In 2012 10th International Symposium on Electronics and Telecommunications (pp. 263-266), IEEE.*, 2012.

[11] L. T. S. H. D. S. D. H. H. &. N. S. Tran, "Improved practical variable step-size algorithm for adaptive feedback control in hearing aids.," in *In 2016 10th International Conference on Signal Processing and Communication Systems (ICSPCS) (pp. 1-8). IEEE.*, 2016.

[12] F. &. P. H. Strasser, "Sub-band feedback cancellation with variable step sizes for music signals in hearing aids," in *In 2014 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP) (pp. 8207-8211). IEEE.*, 2014.

[13] C. Y. F. &. Y. J. Lu, "An Adaptive Time- Domain Kalman Filtering Approach to Acoustic Feedback Cancellation for Hearing Aids.," *Chinese Journal of Electronics*, vol. 29(1), pp. 139-146., 2020.

[14] F. T. L. T. &. N. S. Albu, "The hybrid simplified Kalman filter for adaptive feedback cancellation," in *In 2018 International conference on communications (COMM) (pp. 45-50). IEEE.*, 2018.

[15] F. Yang and J. Yang, "Optimal Step-Size Control of the Partitioned Block Frequency-Domain Adaptive Filter," *IEEE Transactions on Circuits and Systems II: Express Briefs*, pp. 814-818, Dec, 2017.

[16] F. &. Y. J. Yang, "Multiband- structured Kalman filter. IET Signal Processing," *IET Signal Processing*, vol. 12(6), pp. 722-728, 2018.

[17] Tran LTT, Nordholm SE. A Switched Algorithm for Adaptive Feedback Cancellation Using Pre-Filters in Hearing Aids. *Audiology Research*. 2021; 11(3):389-409.