A Variable Step Size Improved Multiband-Structured Subband Adaptive Feedback Cancellation Scheme for Hearing Aids

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***Abstract*— Acoustic feedback degrades the sound quality and limits the maximum stable gain of modern open fitting behind the ear hearing aids. The standard adaptive feedback cancellation approach results in a biased estimation of the feedback path. Despite having several bias reduction methods, the convergence behaviour of the feedback canceller is deteriorated due to the highly correlated input signals such as speech and music. This paper proposes an improved delayless multiband-structured subband adaptive feedback cancellation scheme, which can be considered as a generalized form of the normalized least mean square algorithm, the affine projection algorithm and the multiband-structured subband algorithm. Both the prediction error method and frequency shifting are incorporated in the proposed scheme to reduce the bias. The improved proportionate technique along with a non-parametric variable step size technique is further developed to provide faster convergence and tracking rates. Simulation demonstrates improved transient and steady-state behaviour, improved tracking rate and enhanced speech quality.**

Index Terms— Digital hearing aid, feedback cancellation, frequency shifting, delayless subband.

I. INTRODUCTION

Acoustic feedback in hearing aids distorts the sound quality, limits the maximum gain that can be achieved and in the worst case causes howling [1-3]. One common solution is to cancel the acoustic feedback signal with an adaptive filter, which estimates the feedback signal by using a model of the acoustic feedback path, and subtracts the estimated signal from the microphone signal before it is amplified [4]. Adaptive filtering is necessary because the acoustic feedback path varies with the changes in the acoustic environment around the hearing aid such as head movement and use of mobile phones [5].

The coefficients of the adaptive canceller can be updated by using the normalized least mean square (NLMS) algorithm. However, the modelling of the feedback path might not be accurate for highly correlated input signal such as speech and music. Furthermore, the feedback canceller input signal is a delayed and processed version of the desired input signal, and the correlation between them leads to a bias estimation which results in distortion to the desired signal [6-7].

Many approaches have been proposed to reduce the effect of bias, which includes adding delay in the forward path [8], using frequency shifting (FS) and phase shifting in the forward path [9], injecting probe noise to the receiver [10], and using adaptive decorrelation filters [11]. The prediction error method (PEM) based feedback cancellation scheme adopts an adaptive prediction error filter to whiten the error signal and the input to the updating filter to reduce the correlation between the input to the loudspeaker and desired input to the microphone, which can be integrated with the frequency shifting operation to further reduce bias [12, 13].

The convergence of the system is a serious problem due to the highly correlated incoming signals. The affine projection algorithm (APA) is used to improve the convergence speed at the cost of computation load [14]. The multiband-structured subband adaptive filtering (MSAF) schemes have inherent decorrelating property, which can whiten the input signals before the adaptation process to increase convergence speed [15]. A feedback cancellation scheme using the PEM and delayless version of MSAF has been proposed by the authors to reduce bias and improve the convergence behaviour [16].

In this paper, we extend the work in [16] to further improve the performance of the adaptive feedback canceller in hearing aids. Motivated by the superior convergence performance of the improved multiband-structured subband subband adaptive filtering (IMSAF) scheme [17], we adopt the delayless closed-loop implementation of IMSAF scheme [18]. Unlike the MSAF scheme, the IMSAF scheme uses previous and present inputs in each subband to update the feedback canceller. The FS and PEM techniques are integrated with the IMSAF scheme to effectively handle the bias problem. In the proposed scheme, we also incorporate the improved proportionate technique by using the sparse nature of the acoustic feedback path [19], and develop a non-parametric variable step size approach to achieve faster convergence without sacrificing steady state performance.

The rest of the paper is organized as follows. In section II, the proposed feedback cancellation scheme which incorporates sparsity-aware learning rule and variable step size approach is presented in detail. Simulation results are reported in Section III to demonstrate the efficacy of the proposed schemes. Finally, the concluding remarks are drawn in Section IV.

II. The Proposed Scheme

Fig. 1 illustrates a schematic block diagram of the proposed acoustic feedback cancellation method in a behind the ear digital hearing aid using a finite impulse response (FIR) adaptive filter, where $s(n)$ represents the desired sound signal to the microphone, $u(n)$ is the input signal to the receiver, $f(n)$ is the acoustic feedback signal, $m(n)$ is the output signal of the microphone which is the mixture of $s(n)$ and $f(n)$. The transfer function of the forward path of the hearing aid is given by $G(z)=|G|z^{-\Delta}$, where Δ represents few sample delay. Assume the acoustic feedback path $F(z)$ can be represented with an L_f tap FIR filter, and it is to be modelled by an $L_{\hat{f}}$ tap adaptive FIR filter. An FIR filter $F_c(z)$ is used in the main cancellation path, which is a copy of the shadow filter $\hat{F}(z)$, updated regularly with $\hat{F}(z)$ during the process of adaptation. Signal $\hat{f}(n)$ is subtracted from $m(n)$ to obtain the resultant signal $e_f(n)$, which is processed by the forward path $G(z)$ to produce the signal $v(n)$. A frequency shifting (FS) operation is used in the forward path to further reduce bias [9]. $\overline{u}(n)$ and $\overline{e}(n)$ are obtained by filtering $u(n)$ and $e(n)=m(n)-\hat{f}_s(n)$ with a prediction error filter $A(z)$, which is updated by using the Levinson-Durbin recursion algorithm [20].

Inspired by the inherent advantage of data reuse nature of the APA, in this work we develop an improved multiband-structured subband adaptive feedback cancellation scheme which may be considered as a unifying framework for the PEM and FS based NLMS (PEM-NLMS), affine projection algorithm (PEM-APA) and delayless MSAF (PDMSAF) schemes. As shown in Fig. 1, the filtered error signal $\overline{e}(n)$ is passed through each analysis filter bank (paraunitary cosine-modulated filter bank) $B_i(z)$ followed by critical decimation to produce the resultant subband error signal

$$\overline{e}_{i,D}(k)=\sum_{l=0}^{L_b-1} b_i(l)\overline{e}(kN-l) \tag{1}$$

where k is the subband time index, n is the full band time index. The filter version of $u(n)$, i.e. $\overline{u}(n)$, is also passed through the identical analysis filter bank $B_i(z)$ to generate the subband input signal

$$\overline{u}_i(k)=\sum_{j=0}^{L_b-1} b_i(j)\overline{u}(n-j) \tag{2}$$

The i^{th} (i=1, 2 ,…,N) subband input signal vector is given by

$$\tilde{\mathbf{u}}_{\mathbf{i}}(k)=\left[\overline{u}_i(k),\overline{u}_i(k-1),\ldots,\overline{u}_i(k-L_{\hat{f}}+1)\right]^T \tag{3}$$

The input signal matrix used in the weight update can be written as

$$\overline{\mathbf{U}}(k)=\left[\overline{\mathbf{U}}_0(k),\overline{\mathbf{U}}_1(k),\ldots,\overline{\mathbf{U}}_{N-1}(k)\right] \tag{4}$$

which is a matrix of size ($L_{\hat{f}}\times NP$)

where

$$\overline{\mathbf{U}}_i(k)=\left[\tilde{\mathbf{u}}_i(k),\tilde{\mathbf{u}}_i(k-1),\ldots,\tilde{\mathbf{u}}_i(k-P+1)\right] \tag{5}$$

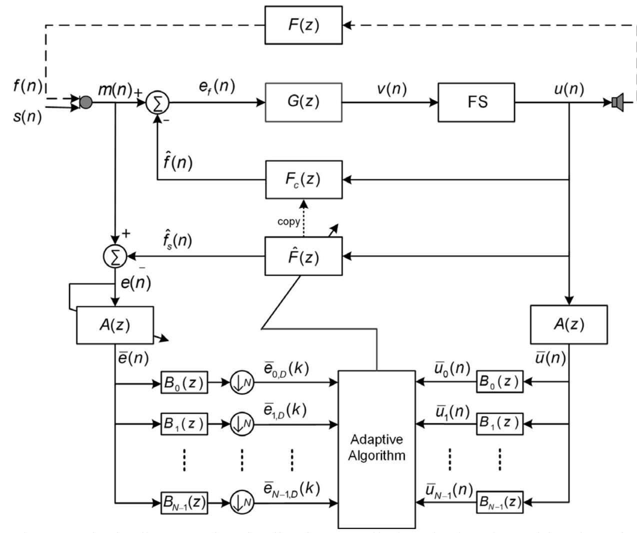

Fig. 1. Block diagram for feedback cancellation in hearing aid using the proposed approach.

with P representing the projection order. The vector containing the NP recent errors can be represented as

$$\tilde{\mathbf{e}}_D(k)=\left[\tilde{\mathbf{e}}_{0,D}^T(k),\tilde{\mathbf{e}}_{1,D}^T(k),\ldots,\tilde{\mathbf{e}}_{N-1,D}^T(k)\right]^T, \tag{6}$$

where

$$\tilde{\mathbf{e}}_{i,D}^T(k)=\left[\overline{e}_{i,D}^T(k),\overline{e}_{i,D}^T(k-1),\ldots,\overline{e}_{i,D}^T(k-P+1)\right] \tag{7}$$

The weigh update equation can be given by,

$$\hat{\mathbf{f}}(k+1)=\hat{\mathbf{f}}(k)+\mu\overline{\mathbf{U}}(k)\mathbf{\Omega}^{-1}(k)\tilde{\mathbf{e}}_D(k) \tag{8}$$

where

$$\mathbf{\Omega}(k)=\overline{\mathbf{U}}^T(k)\overline{\mathbf{U}}(k)+\delta\mathbf{I}_{NP}, \tag{9}$$

is a normalization matrix with $\mathbf{I}_{NP}$ representing an identity matrix of size $NP\times NP$. This PEM and frequency shifting method based the Delayless Improved Multiband-structured Subband Adaptive Filtering algorithm is hence forth referred to as the PDIMSAF scheme.

The impulse response of the acoustic feedback path usually contains few active coefficients (significant coefficients) and rest coefficients are zeros or close to zeros. In such a scenario, the convergence behaviour is affected if identical step size is assigned to all coefficients of the adaptive filter irrespective of their magnitude. Several sparsity-aware algorithms have been reported to address this issue [21]. Inspired by the convergence behaviour of the improved proportionate technique in [22], we incorporate this technique into the above proposed PDIMSAF algorithm by assigning magnitude dependent step size to the coefficients of the adaptive filter. This is achieved by including magnitude dependent adaptation gain factor in the update rule, so the new update equation is

$$\hat{\mathbf{f}}(k+1)=\hat{\mathbf{f}}(k)+\mu\mathbf{\Theta}(k)\overline{\mathbf{U}}(k)\mathbf{\Omega}^{-1}(k)\tilde{\mathbf{e}}_D(k), \tag{10}$$

where

$$\mathbf{\Theta}(k)=\mathrm{diag}[\theta_0(k),\theta_1(k),\ldots,\theta_l(k),\ldots,\theta_{L_{\hat{f}}-1}(k)], \tag{11}$$

with

$$\theta_t(k) = \frac{1-\beta}{2L_{\hat{f}}} + (1+\beta)\frac{|\hat{f}_t(k)|}{2\|\hat{\mathbf{f}}(k)\| + \varepsilon} \quad (12)$$

and

$$\mathbf{\Omega}(k) = \bar{\mathbf{U}}^T(k)\mathbf{\Theta}(k)\bar{\mathbf{U}}(k) + \delta \mathbf{I}_{NP}. \quad (13)$$

where $-1 \leq \beta < 1$ is a control parameter for scaling the adaptive coefficients and ε is a small constant to avoid divide by zero conditions. This Improved Proportionate PDIMSAF scheme is hereafter referred to as the IP-PDIMSAF scheme.

In order to further improve the convergence of the proposed IP-PDIMSAF algorithm, a variable step size is incorporated in the update mechanism by replacing the fixed step size with a variable step size. Because the inverse of a block diagonal matrix is composed of the inverse of the constituent blocks, Eq. (10) may be rewritten as

$$\hat{\mathbf{f}}(k+1) = \hat{\mathbf{f}}(k) + \mu_i(k)\sum_{i=0}^{N-1}\mathbf{\Theta}(k)\bar{\mathbf{U}}_i(k) \left[\bar{\mathbf{U}}_i^T(k)\mathbf{\Theta}(k)\bar{\mathbf{U}}_i(k) + \delta\mathbf{I}\right]^{-1}\tilde{\mathbf{e}}_{i,D}(k) \quad (14)$$

where the step size is updated with a non-parametric variable step size approach [13] by using

$$\mu_i(k) = \left(1 - \frac{\sigma_{s_i}(k)}{\sigma_{e_{f_i}}(k) + \zeta}\right) \quad (15)$$

where $\sigma_{s_i}(k)$ is the standard deviation of $s(n)$, $\sigma_{e_{f_i}}(k)$ is the standard deviation of $e_f(n)$ and ζ is a small constant to avoid divide by zero condition. In applications, $s(n)$ is not accessible independently, so an attempt has been made in [13] to estimate $\sigma^2_{s_i}(k)$ as

$$\begin{aligned}\sigma^2_{s_i}(k) &= \sigma^2_{e_f}(k) \\ &+ \mathbf{r}^H_{eu}(k)\left(\mathbf{R}^{-1}_{uu}(k)\right)^H \mathbf{r}_{eu}(k) \\ &- 2\mathrm{Re}\left\{\mathbf{r}^H_{eu}(k)\left(\mathbf{R}^{-1}_{uu}(k)\right)^H \mathbf{r}_{eu}(k)\right\}\end{aligned} \quad (16)$$

In the scenarios where the power of the desired input signal $s(n)$ increases very fast, the estimation of $\sigma^2_{s_i}(k)$ becomes inaccurate. In an endeavour to handle this situation, a modification is incorporated by replacing $\mathbf{R}_{uu}$ with the combination of autocorrelation matrices of $u(n)$ and $e_f(n)$ [13]

$$\bar{\mathbf{R}}_{uu}(k) = 0.5\left(\mathbf{R}_{uu}(k) + \mathbf{R}_{e_f e_f}(k)\frac{\sigma^2_u(k)}{\sigma^2_{e_f}(k)}\right) \quad (17)$$

The terms $\mathbf{R}_{uu}(k)$, $\mathbf{R}_{e_f e_f}(k)$, $\sigma^2_u(k)$, $\sigma^2_{e_f}(k)$ and $\mathbf{r}_{eu}$(k) are estimated recursively as

$$\bar{\mathbf{R}}_{uu}(k+1) = \lambda_1\bar{\mathbf{R}}_{uu}(k) + (1-\lambda_1)\mathbf{u}(n)\mathbf{u}^H(n) \quad (18)$$

$$\bar{\mathbf{R}}_{e_f e_f}(k+1) = \lambda_2\bar{\mathbf{R}}_{e_f e_f}(k) + (1-\lambda_2)\mathbf{e}_f(n)\mathbf{e}_f^H(n) \quad (19)$$

$$\bar{\sigma}^2_u(k+1) = \lambda_1\bar{\sigma}^2_u(k) + (1-\lambda_1)\left|u(n)\right|^2 \quad (20)$$

$$\bar{\sigma}^2_{e_f}(k+1) = \lambda_1\bar{\sigma}^2_{e_f}(k) + (1-\lambda_1)\left|e_f(n)\right|^2 \quad (21)$$

$$\bar{\mathbf{r}}_{eu}(k+1) = \lambda_1\bar{\mathbf{r}}_{eu}(k) + (1-\lambda_1)e_f(n)\mathbf{u}^*(n) \quad (22)$$

Substituting Eqs. (17)-(22) in Eq. (16), the signal power $\sigma^2_{s_i}(k)$ can be estimated and using the obtained value in Eq. (15), the variable step size is calculated. With an objective to achieve a further smooth variation of step size, we limit the upper and lower bound of step size as

$$\mu_{v_i}(k+1) = \begin{cases}\mu_{\max,}\ if\ \mu_c(k+1) > \mu_{\max} \\ \mu_{\min,}\ if\ \mu_c(k+1) < \mu_{\min} \\ \mu_c(k), \text{otherwise}\end{cases} \quad (23)$$

where $\mu_c(k) = \mu_i(k)\mu_{max}$ with $\mu_{\max}$ and $\mu_{\min}$ denoting the selected upper and lower bound of the step size. The above discussed feedback cancellation algorithm is hereafter referred to as the VSS-IP-PDIMSAF scheme.

III. SIMULATION RESULTS

The performance of the proposed VSS-IP-PDIMSAF scheme is compared with other schemes such as the NLMS, PEM-NLMS, PEM-APA, PDMSAF, PDIMSAF, and the IP-PDIMSAF. In the simulations, the sampling frequency is 8 kHz, the adaptive prediction error filter uses a 21 order filter with window length of 160 samples, the frequency shifting used is 3 Hz to the left of the spectrum [23]. The misalignment (MIS) and the added stable gain (ASG) are used as the performance matrices, which are defined as

$$\mathrm{MIS} = 10\log_{10}\left(\frac{\sum_{i=0}^{N_f-1}|F(e^{j\omega_i}) - \hat{F}(e^{j\omega_i})|^2}{\sum_{i=0}^{N_f-1}|F(e^{j\omega_i})|^2}\right) \quad (24)$$

$$\mathrm{ASG} = \mathrm{MSG} - 20\log_{10}\left(\min_\omega \frac{1}{|F(e^{j\omega_i})|}\right), \quad (25)$$

$$\mathrm{MSG} = 20\log_{10}\left(\min_\omega \frac{1}{|F(e^{j\omega_i}) - \hat{F}(e^{j\omega_i})|}\right). \quad (26)$$

Fig. 2 shows the impulse response and magnitude of the frequency response of the acoustic feedback paths considered in the simulations, where one is for the hearing aid being used without obstruction and the other is that with obstruction [16]. The acoustic feedback path considered for the first 20 sec of the input speech segment (Part A) is the one without obstruction. The feedback path suddenly changes after the 20th sec, and a modified version of the feedback path is used for the rest of 20 seconds (Part B) which is the one with obstruction. The forward path considered contains a delay of $\Delta = 60$ samples followed by a gain of $G = 12$.

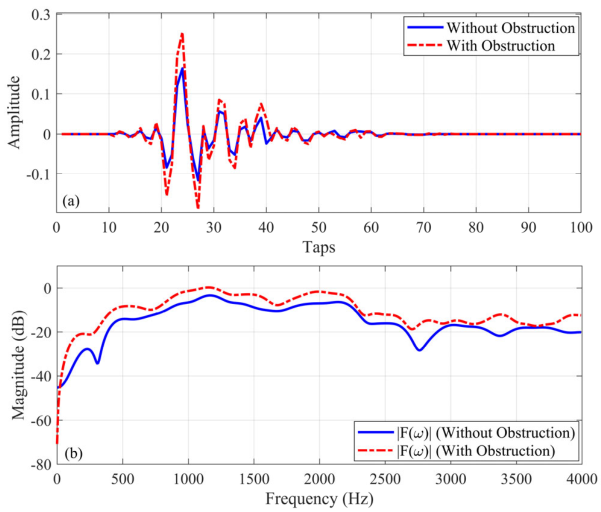

Fig. 2. Acoustic feedback paths. (a) Impulse response (b) Magnitude of frequency response.

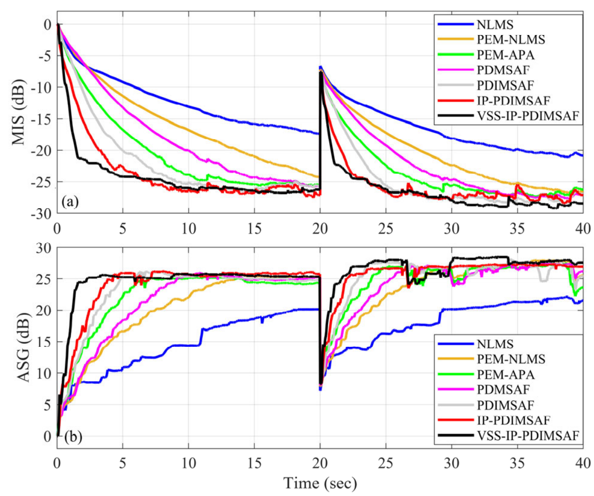

Fig. 3. (a) Variation of Misalignment (b) Variation of ASG with respect to time for the algorithms under study.

A speech segment of 40 seconds from the NOIZEUS database is used as the input signal [24]. Fig. 3(a) shows the variation of MIS with respect to time for the two feedback paths considered, where the incorporation of PEM and FS is shown to be able to reduce the correlation to a satisfactory margin. Furthermore, the faster convergence characteristics of the proposed scheme for Part A and Part B reflect the superiority over all other schemes. It is important to notice that the faster convergence is achieved without sacrificing the lower steady state misalignment. The improved cancellation performance is also confirmed by the variation of ASG shown in Fig. 3(b).

The PESQ scores [25] for Part A and Part B are: NLMS(3.51, 3.57), PEM-NLMS(3.69, 3.77), PEM-APA(4.11, 4.14), PDMSAF(4.10, 4.01), PDIMSAF(4.13,4.17), IP-PDIMSAF(4.22,4.24), VSS-IP-PDIMSAF(4.31, 4.35). The

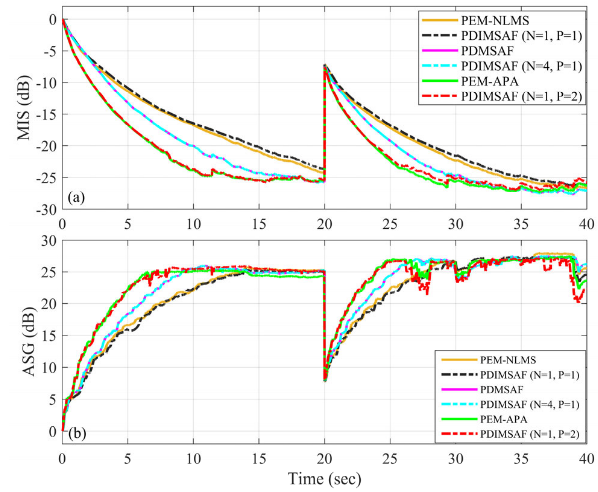

Fig. 4. The proposed PDIMSAF scheme as a generalized form of the PDMSAF, PEM-APA and PEMNLMS schemes (a) Variation of Misalignment (b) Variation of ASG with respect to time.

improved speech quality is evident from the PESQ score obtained. The other parameters considered are: $\mu = 0.003$, $\delta = 1\times10^{-5}$, $P = 2$, $N = 4$, $\beta = -0.5$, $\varepsilon = 1\times10^{-5}$, $\zeta = 1\times10^{-5}$, $\lambda_1 = 0.99$, $\lambda_2 = 0.95$, $\mu_{\max} = 0.008$, $\mu_{\min} = 0.00001$.

One can notice that the NLMS based feedback cancellation scheme is not able to cancel the feedback effectively due to the associated bias problem. The PEM and FS based PEM-NLMS scheme can reduce the biased estimation of feedback path. The PEM-APA and the PDMSAF scheme accelerate the convergence behaviour of the adaptive filter, while the PDIMSAF scheme exploits the advantages of both APA and MSAF scheme for performance improvement. The IP-PDIMSAF scheme further improves the performance by taking into account the sparse nature of the feedback path. The proposed VSS-IP-PDIMSAF scheme outperforms all the other schemes compared. This improvement in feedback cancellation performance is attributed to the combination of the PEM and FS to reduce the bias, the decorrelating property of the delayless improved multiband structure, the improved proportionate learning scheme to improve convergence and the VSS technique to further improve convergence without deteriorating steady state performance.

The proposed PDIMSAF scheme can be treated as a unifying framework for the PEM-NLMS, PEM-APA and PDMSAF schemes. In a special case where $N = 4$ and $P = 1$, the proposed PDIMSAF scheme reduces to the PDMSAF scheme. In a special case where N=1 and P=2, the proposed PDIMSAF scheme reduces to the PEM-APA scheme. Similarly, the PDIMSAF scheme reduces to the PEM-NLMS scheme when N=1 and P=1.

To verify this, the input signal, feedback paths and simulation parameters remain the same as the previous case. The variation of MIS and ASG with respect to time is shown in Figs. 4 (a) and (b) respectively, where it is clear that the PEM-NLMS and PDIMSAF (N=1, P=1) schemes behave similarly, the PEM-APA and PDIMSAF (N=1, P=2) schemes behave similarly, and the PDMSAF and PDIMSAF (N=4,

P=1) scheme behave similarly. The PEM-APA scheme improves the convergence rate by data-reuse, the PDMSAF scheme improves convergence with subband decomposition and the proposed PDIMSAF accelerates the convergence by both subband decomposition and data-reuse.

IV. Conclusion

In this paper, a delayless improved multiband-structured subband adaptive feedback canceller has been developed, which can be considered as a generalized form of the normalized least mean square algorithm, the affine projection algorithm and the multiband-structured subband algorithm. Both the prediction error method and frequency shifting are incorporated in the proposed scheme to reduce bias, and the improved proportionate technique is integrated to provide faster convergence and tracking rates. A non-parametric variable step size technique is further developed in the proposed scheme to improve the convergence behaviour and tracking rate without sacrificing the steady-state performance. Simulations have been carried out to test the efficacy of the proposed scheme, which demonstrates improved transient and steady-state behaviour, improved tracking rate and enhanced speech quality. Future work include applying the proposed scheme to acoustic echo cancellation and public address system and develop low complexity implementation of proposed scheme.

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