IMPROVED PROPORTIONATE AFFINE PROJECTION ALGORITHM FOR ADAPTIVE FEEDBACK CANCELLATION

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***Tóm tắt*—Acoustic feedback is a major problem in open-fit digital hearing aids, which significantly lowers the signal quality and limits the achievable maximum stable gain. Adaptive feedback cancellation (AFC) is a common and efficient approach, however, it introduces a biased estimate of the feedback path due to a high correlation between loudspeaker signal and the incoming signal, especially when the incoming signal is spectrally coloured, e.g., speech, music. The prediction error method (PEM) is well known for reducing this bias, resulting in significant performance improvement. To further improve the performance of the conventional PEM we propose to integrate the improved proportionate affine projection algorithm (IPAPA) into the PEM. The proposed method, namely PEMSC-IPAPA, leverages sparse characteristics of the feedback path and a fast adaptive filtering technique to enhance the convergence/tracking rates. A detailed derivation of the proposed AFC method and its stability analysis are also considered. We evaluate the performance of the proposed method with recorded speech and music as the incoming signals, and with an abrupt change of the feedback path. Simulation results show that the proposed method achieves higher convergence/tracking rates while retaining similar steady-state error and signal quality compared to the state-of-the-art baselines.**

***Từ khóa*—Adaptive feedback cancellation, prediction error method, IPAPA, maximum stable gain, convergence/tracking rates.**

I. INTRODUCTION

A major problem in open-fit hearing aids (HAs) and public address systems (PAs) is acoustic feedback produced by the loudspeaker signal coupling into the microphone(s). The HAs and PAs are considered closed-loop systems due to the presence of the forward path. The feedback signal is rendered through those closed-loop systems, resulting in signal quality degradation as well as achievable amplification limitation. Under certain situations, it leads HAs/PAs to unstable and/or howling. With a high demand for small and reliable open-fit hearing aids, acoustic feedback cancellation is still a challenge for hearing aid applications. In the last six decades, multiple acoustic feedback cancellation approaches have been proposed in the literature [1]–[3]. The adaptive feedback cancellation (AFC) emerges as a simple and efficient approach to reduce the negative effects of acoustic feedback. The main idea of this method is to adopt an FIR filter for estimating the acoustic feedback path. This feedback path estimate is used to compute the feedback signal which is then suppressed from the microphone signal (cf. Fig. 1). If the estimation is perfect, no acoustic feedback signal arrives in the loudspeaker. However, the feedback path estimate may produce a bias caused by a possibly high correlation between the incoming signal and the loudspeaker signal [1], [2], [4].

To reduce this bias, many decorrelation approaches have been introduced such as inserting a delay in the forward path [1], [5], adding a probe noise to the loudspeaker signal [6]–[9], frequency shifting [10], [11], phase modulation [12], and/or using pre-whitening filters [13], [14]. Among them the prediction error method based adaptive feedback cancellation (PEM-AFC) is one of the most popular approaches. The PEM-AFC is developed based on the idea that employs pre-filters to pre-whiten input signals of the adaptive feedback canceller, yielding lower correlation and so bias. It works effectively both in the time domain [14]–[19] and in the frequency domain [2], [20]–[24]. Other AFC methods leveraging subband techniques [25]–[28], multiple-microphones [19], [29]–[35], variable step-size (VSS) [11], [36]–[38], affine combination of filters [23], fast-converging adaptive filtering algorithms [14], [17], [18], [32], [39]–[41], decomposing a long adaptive filter into a Kronecker Product of two shorter filters [42], instability detection and control [43], [44] and/or combinations of those techniques have also been investigated. Although previous AFC approaches achieve performance improvement to some certain degree, there is still room for further enhancing the AFC performance.

Recently, AFC approaches that exploit sparse features of the feedback path have attracted audience attention, e.g., (improved) proportionate normalized least mean squares (I)PNLMS [18], [32], [45], de-correlated zero attracting least mean squares (DZALMS) and de-correlated re-weighted zero attracting least mean squares (DRZALMS) [41] algorithms. The idea is to assign variable weights for updating adaptive filter coefficients based on their

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strength. As a result, improvements in convergence rate, tracking rate and sound quality can be obtained.

To further improve the performance we propose a new AFC approach that integrates the improved proportionate affine projection algorithm (IPAPA) into the PEM-AFC with soft clipping, namely PEMSC-IPAPA. This approach leverages both sparse features of the feedback path and fast filtering techniques to improve the convergence and tracking rates of the conventional PEM-AFC. Note that the IPAPA has been successfully implemented for acoustic echo cancellation (AEC) [46], [47]. However, their applications for adaptive feedback cancellation are still challenging due to the possibly high correlation between the incoming signal and the loudspeaker signal. To overcome this challenge, the proposed approach firstly utilizes pre-filters to pre-whitening the adaptive filter inputs, which allows for correlation reduction. Moreover, a soft clipper (SC) [43] is applied to the error signal in order to limit the feedback contribution, resulting in a feedback cancellation improvement. Then the IPAPA is employed to recursively update the adaptive filter coefficients. In this paper the derivation of the proposed PEMSC-IPAPA is considered in detail. We also compare the computational complexity of the proposed approach with that of the state-of-the-art baselines. Simulation results using different incoming signals and different feedback paths show that the proposed approaches outperform state-of-the-art baselines in terms of convergence and tracking rates, while retaining a similar low steady-state error and good signal quality.

Throughout this paper, the lower and upper letters in bold are used to represent vectors and matrices, respectively. We use $E\{.\}$ for the expectation operation and the superscript T for transposition. We also use $\boldsymbol{R}_{\mathbf{x}}$, $\boldsymbol{R}_{\mathbf{xy}}$ and $\boldsymbol{r}_{\mathbf{x}\zeta}$ to denote the auto-correlation matrix of a vector $\mathbf{x}$, the cross-correlation matrix between two vectors $\mathbf{x}$ and $\mathbf{y}$, and the cross-correlation vector between a vector $\mathbf{x}$ and a scalar ζ, respectively, i.e., $\boldsymbol{R}_{\mathbf{x}} = E\left\{\mathbf{x}\mathbf{x}^T\right\}$, $\boldsymbol{R}_{\mathbf{xy}} = E\left\{\mathbf{x}\mathbf{y}^T\right\}$, and $\boldsymbol{r}_{\mathbf{x}\varsigma} = E\{\mathbf{x}\zeta\}$.

The paper is organized as follows. Section II and III review the standard AFC method and the conventional PEM-AFC, respectively. We theoretically analyzes the proposed PEMSC-IPAPA in Section IV. Section V provides a computational complexity analysis of the proposed method in comparison with baselines. Simulation results are described in Section VI. Section VII concludes the paper.

II. STANDARD AFC MODEL

Fig. 1 illustrates the standard AFC scheme for a hearing aid with a single microphone and single loudspeaker. In this paper, we assume that the incoming signal is stationary and AFC systems are discrete and linear time-invariant. Since the loudspeaker signal injects into the microphone of a hearing aid, the microphone signal $m(k)$ consists of two components, called the incoming signal $x(k)$ and the feedback signal $v(k) = \mathbf{f}^T\mathbf{u}(k)$, i.e.,

$$m(k) = x(k) + \mathbf{f}^T\mathbf{u}(k), \tag{1}$$

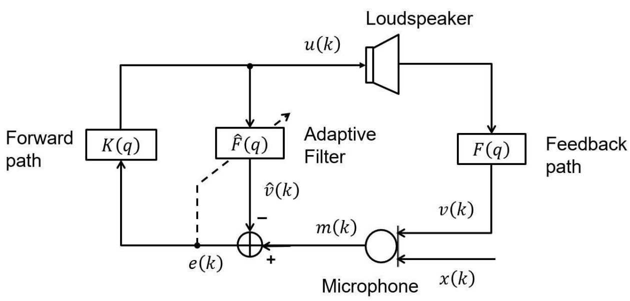
where k is the discrete-time index, and $\mathbf{u}(k) = [u(k), u(k-1), \ldots, u(k-L_f+1)]^T$ is a L_f-dimentional vector representing the loudspeaker signal. The vector $\mathbf{f} = \left[f_0, f_1, \ldots, f_{L_f-1}\right]^T$ denotes the true feedback path of length L_f, represented as a polynomial transfer function in q, i.e., $F(q) = \mathbf{f}^T\mathbf{q}$ with $\mathbf{q} = \left[\begin{array}{cccc} 1 & q^{-1} & \ldots & q^{-L_f+1} \end{array}\right]^T$. In the standard AFC system, the feedback path is firstly estimated by using an FIR adaptive filter. Then the estimated feedback signal $\hat{v}(k)$ computed based on the feedback path estimate is suppressed from the microphone signal $x(k)$, forming an error signal $e(k)$, i.e.,

$$\begin{aligned} e(k) &= m(k) - \hat{v}(k), & (2)\\ &= x(k) + \left[\mathbf{f}^T - \hat{\mathbf{f}}^T(k)\right]\mathbf{u}(k) & (3) \end{aligned}$$

where $\hat{v}(k) = \hat{\mathbf{f}}^T(k)\mathbf{u}(k)$ and $\hat{\mathbf{f}} = \left[\hat{f}_0(k), \hat{f}_1(k), \ldots, \hat{f}_{L_{\hat{f}}-1}(k)\right]^T$ denoting the $L_{\hat{f}}$-dimentional feedback path estimate. The loudspeaker signal is produced by passing the error signal through the forward path $K(q)$ of the hearing aid, i.e.,

$$u(k) = K(q)\,e(k), \tag{4}$$

where $K(q) = |K|\,q^{-d_k}$ with $|K|$ and d_k the gain and the delay in the forward path, respectively. We assume that there is at least one sample delay in the forward path ($d_k \geqslant 1$).



Hình 1. The diagram of a hearing aid using standard AFC.

An optimal solution for the feedback path estimate can be achieved by minimizing the cost function $J\left(\hat{\mathbf{f}}\right) = E\left\{e^2(k)\right\}$ with respect to (w.r.t.) $\hat{\mathbf{f}}$ as follows

$$\hat{\mathbf{f}}_0 = E\left\{\mathbf{u}(k)\,\mathbf{u}^T(k)\right\}^{-1} E\left\{\mathbf{u}(k)\,m(k)\right\}. \tag{5}$$

By substituting (1) into (5), we obtain [1]

$$\hat{\mathbf{f}}_0 = \mathbf{f} + \underbrace{\boldsymbol{R}_{\mathbf{u}}^{-1}(k)\,\boldsymbol{r}_{\mathbf{u}x}(k)}_{bias}. \tag{6}$$

From (6) we observe a bias in the feedback path estimate. This is due to the correlation between the incoming signal and the loudspeaker signal. The incoming signal now behaves as a disturbance to the feedback canceler [2]. By using normalized least mean square algorithm (NLMS), the optimal coefficients of the feedback path can be recursively approximated as

$$\hat{\mathbf{f}}(k) = \hat{\mathbf{f}}(k-1) + \frac{\mu}{\mathbf{u}^T(k)\,\mathbf{u}(k) + \delta}\mathbf{u}(k)\,e(k), \quad (7)$$

where μ denotes a step-size and δ denotes a small positive value added to avoid division by zero.

The transfer function $A(q)$ of the closed-loop system from the incoming signal to the loudspeaker signal is computed as follows

$$A(q) = \frac{K(q)}{1 - K(q)\left[F(q) - \hat{F}(q)\right]}, \quad (8)$$

where $\hat{F}(q)$ is an estimate of $F(q)$. According to the Nyquist stability criterion, a linear closed-loop system is unstable if both conditions for the loop gain and the loop phase are satisfied [3], i.e.,

$$\begin{cases} \left|K\left(e^{j\omega}\right)\left[F\left(e^{j\omega}\right) - \hat{F}\left(e^{j\omega}\right)\right]\right| \geq 1 \\ \angle K\left(e^{j\omega}\right)\left[F\left(e^{j\omega}\right) - \hat{F}\left(e^{j\omega}\right)\right] = 2\pi n, \quad n \in \mathbb{Z} \end{cases}, \quad (9)$$

where $\omega \in [0, 2\pi]$ denotes the angular frequency; $K\left(e^{j\omega}\right)$, $F\left(e^{j\omega}\right)$ and $\hat{F}\left(e^{j\omega}\right)$ denote frequency responses of the forward path, the feedback path and the feedback path estimate, respectively. Therefore, all AFC methods try to avoid at least one condition in (9) to be met.

III. CONVENTIONAL PEM-AFC

The conventional PEM-AFC [2], [14], [20] is popular for reducing the bias in the feedback path estimate. Fig. 2 depicts the PEM-AFC model for a hearing aid with a single microphone and single loudspeaker. This method employs pre-filters to pre-whiten the adaptive filter inputs. The incoming signal is assumed to be modelled by passing a white Gaussian noise sequence, $w(k)$, through a monic and inversely stable all-pole filter, $G^{-1}(q)$, i.e.,

$$x(k) = G^{-1}(q)\,w(k). \quad (10)$$

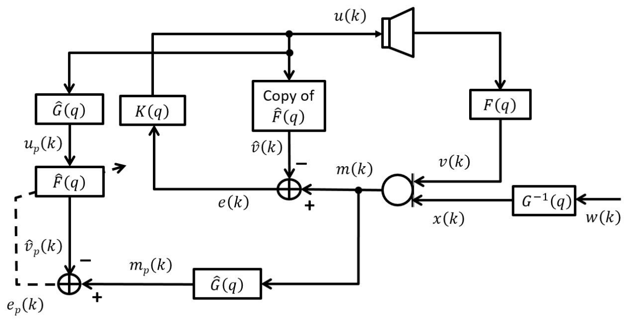
We define $\hat{G}(q)$ as the estimated version of $G(q)$ and use it to pre-whiten the loudspeaker and microphone signals as follows

$$u_p(k) = \hat{G}(q)\,u(k), \quad (11)$$

$$m_p(k) = \hat{G}(q)\,m(k), \quad (12)$$

where $u_p(k)$ and $m_p(k)$ are the pre-whitened loudspeaker and microphone signals, respectively. The incoming signal after pre-whitening, $x_p(k)$, is defined as

$$x_p(k) = \hat{G}(q)\,x(k). \quad (13)$$



Hình 2. PEM-AFC model.

The prediction error signal $e_p(k)$ can be calculated as

$$e_p(k) = m_p(k) - \hat{\mathbf{f}}^T(k)\,\mathbf{u}_p(k), \quad (14)$$

where $\mathbf{u}_p(k) = \left[u_p(k), \ldots, u_p\left(k - L_{\hat{f}} + 1\right)\right]^T$ denotes a $L_{\hat{f}}$-dimensional vector of the loudspeaker signal. For the PEM-AFC the optimal solution of the feedback path estimate can be obtained by minimizing the mean square prediction error, $E\left\{e_p^2(k)\right\}$, i.e.,

$$\hat{\mathbf{f}}_0 = E\left\{\mathbf{u}_p(k)\,\mathbf{u}_p^T(k)\right\}^{-1} E\left\{\mathbf{u}_p(k)\,m_p(k)\right\}. \quad (15)$$

The equation (15) can be rewritten as

$$\hat{\mathbf{f}}_0 = \mathbf{f} + \underbrace{\boldsymbol{R}_{\mathbf{u}_p}^{-1}(k)\,\boldsymbol{r}_{\mathbf{u}_p x_p}(k)}_{bias}. \quad (16)$$

Substituting (10) and (13) into (16), we obtain an unbiased solution for the feedback path estimate if the assumption (10) is fulfilled, $\hat{G}(q) = G(q)$ and at least one delay in the forward path is available. Note that both the feedback path $F(q)$ and the AR model $\hat{G}(q)$ can be identified in closed-loop without adding a probe signal or nonlinearities if the delay in the forward path is not smaller than the length of AR model $\hat{G}(q)$ [2], [14].

The optimal coefficients $\hat{\mathbf{f}}_0$ of the adaptive filter in the PEM-AFC can be recursively updated using the NLMS algorithm as follows

$$\hat{\mathbf{f}}(k) = \hat{\mathbf{f}}(k-1) + \frac{\mu}{\mathbf{u}_p^T(k)\,\mathbf{u}_p(k) + \delta}\mathbf{u}_p(k)\,e_p(k). \quad (17)$$

IV. PROPOSED APPROACH

A. PEMSC-IPAPA

In this subsection, we describe the proposed PEMSC-IPAPA in detail. The PEMSC-IPAPA model is similar to the PEM-AFC model, except a soft clipper (SC) block is added and the IPAPA algorithm is used for estimating coefficients of the block $\hat{F}(q)$ (cf. Fig. 3). We use similar formulae of the microphone, feedback and error signals as defined in (1) and (2), i.e.,

$$m(k) = x(k) + \mathbf{f}^T\mathbf{u}(k), \quad (18)$$

$$e(k) = m(k) - \hat{v}(k), \quad (19)$$
$$= x(k) + [v(k) - \hat{v}(k)]. \quad (20)$$

The pre-whitened loudspeaker, microphone and error signals are also defined similar to those in (11), (12), and (14), i.e.,

$$u_p(k) = \hat{G}(q)\, u(k), \quad (21)$$

$$m_p(k) = \hat{G}(q)\, m(k), \quad (22)$$

$$e_p(k) = m_p(k) - \hat{\mathbf{f}}^T \mathbf{u}_p(k). \quad (23)$$

To limit the feedback contribution to the error signal, a soft clipper is applied to the error signal, $e(k)$, i.e.,

$$e_{SC}(k) = \lambda \tanh\left(\frac{e(k)}{\lambda}\right), \quad (24)$$

where λ is a scaling parameter, $e_{SC}(k)$ is the soft-clipping error signal. The parameter λ is selected such that the most likely range of the incoming signal lies in the linear range of the tanh-function, $x(k) \approx \lambda \tanh\left(\frac{x(k)}{\lambda}\right)$. We adopt this $e_{SC}(k)$ to compute the pre-filter coefficients using Levison-Durbin algorithm. As a result, the performance of the acoustic feedback cancellation is improved [43]. The loudspeaker signal is generated by processing the soft-clipping error signal through the forward path as follows

$$u(k) = K(q)\, e_{SC}(k). \quad (25)$$

Let $\mathbf{U}(k) = [\mathbf{u}(k), \mathbf{u}(k-1), \ldots, \mathbf{u}(k-P+1)]$ be a matrix of P recent loudspeaker vectors and $\mathbf{m}(k)$ be a vector of P recent microphone signals,

$$\mathbf{m}(k) = [m(k), m(k-1) \ldots, m(k-P+1)]^T, \quad (26)$$

where P denotes projection order.

The vector of error signal is defined as

$$\mathbf{e}(k) = \mathbf{m}(k) - \mathbf{U}^T(k)\, \hat{\mathbf{f}}, \quad (27)$$

while the pre-whitened version of those signals can be expressed as follows

$$\mathbf{U}_p(k) = [\mathbf{u}_p(k), \mathbf{u}_p(k-1), \ldots, \mathbf{u}_p(k-P+1)], \quad (28)$$

$$\mathbf{m}_p(k) = [m_p(k), m_p(k-1), \ldots, m_p(k-P+1)]^T, \quad (29)$$

$$\mathbf{e}_p(k) = \mathbf{m}_p(k) - \mathbf{U}_p^T(k)\, \hat{\mathbf{f}}. \quad (30)$$

Instead of using the NLMS algorithm for estimating the feedback path as in the conventional PEM-AFC, we propose to adopt the IPAPA for this estimation. The IPAPA exploits the sparseness of the feedback path and the fast-converging adaptive filtering (APA). In particular, the IPAPA updates each coefficient of the adaptive filter by using an adaptive step-size in proportion to the estimated filter coefficient. Thus, faster convergence/tracking abilities can be achieved compared to the NLMS algorithm. In the proposed method, we minimize the cost function $J\left(\hat{\mathbf{f}}\right) = E\left\{\mathbf{e}_p^2(k)\right\}$ w.r.t. $\hat{\mathbf{f}}$, i.e.,

$$\min_{\hat{\mathbf{f}}} E\left\{\mathbf{e}_p^2(k)\right\}. \quad (31)$$

As a result, the optimal solution for the feedback path estimate in (31) can be approximately computed using the IPAPA as

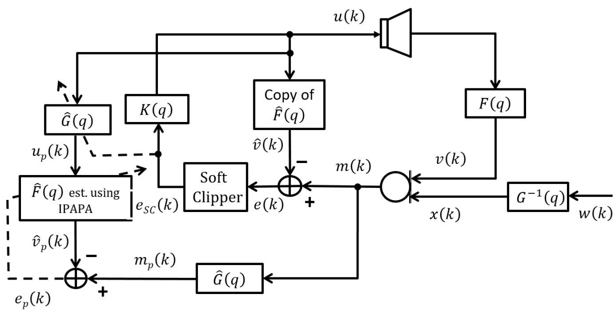
$$\hat{\mathbf{f}}(k) = \hat{\mathbf{f}}(k-1) + \mu \mathbf{B}(k-1)\, \mathbf{U}_p(k) \,. \left[\mathbf{U}_p^T(k)\, \mathbf{B}(k-1)\, \mathbf{U}_p(k) + \bar{\delta}\mathbf{I}_{P\mathrm{x}P}\right]^{-1} \mathbf{e}_p(k), \quad (32)$$

where $\mathbf{B}(k-1) = \mathrm{diag}\left\{b_0(k-1), \ldots, b_{L_{\hat{f}}-1}(k-1)\right\}$ is a diagonal matrix and $\bar{\delta} = \frac{(1-\beta)}{2L_{\hat{f}}}\delta$ is a regularization parameter.

The diagonal elements of the diagonal matrix $\mathbf{B}(k-1)$ are calculated as follows

$$b_j(k) = \frac{1-\beta}{2L_{\hat{f}}} + (1+\beta)\frac{\left|\hat{f}_j(k)\right|}{2\left\|\hat{\mathbf{f}}(k)\right\|_1 + \xi}, \quad (33)$$

where ξ is a small positive constant added to avoid division by zero. Noting that the IPAPA behaves like the APA for $\beta = -1$, whereas it is similar to the PAPA for $\beta \approx 1$. The suggested selections for the parameter β are -0.5 or 0 [48].



Hình 3. Block diagram of the proposed PEMSC-IPAPA approach.

B. Stability analysis of the PEMSC-IPAPA

In this subsection, we analyze the convergence of the proposed PEMSC-IPAPA in detail. We subtract both sides of (32) from $\hat{\mathbf{f}}_0$ and let $\mathbf{c}(k) = \hat{\mathbf{f}}_0 - \hat{\mathbf{f}}(k)$, $\mathbf{\Gamma}(k) = \mathbf{U}_p^T(k)\, \mathbf{B}(k-1)\, \mathbf{U}_p(k) + \bar{\delta}\mathbf{I}_{P\mathrm{x}P}$, $\boldsymbol{\gamma}(k) = \mathbf{U}_p^T(k)\, \mathbf{B}(k-1)\, \mathbf{U}_p(k)$, resulting in

$$\mathbf{c}(k) = \mathbf{c}(k-1) - \mu \mathbf{B}(k-1)\, \mathbf{U}_p(k)\, \mathbf{\Gamma}^{-1}(k)\, \mathbf{e}_p(k). \quad (34)$$

Hence,

$$\mathbf{c}(k) - \mathbf{c}(k-1) = -\mu \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\mathbf{\Gamma}^{-1}(k)\,\mathbf{e}_p(k). \tag{35}$$

A priori pre-whitening error vector, $\boldsymbol{\varepsilon}_a(k)$, and *a posteriori* pre-whitening error vector, $\boldsymbol{\varepsilon}_{post}(k)$, are defined as

$$\boldsymbol{\varepsilon}_a(k) = \mathbf{U}_p^T(k)\left[\hat{\mathbf{f}}_0 - \hat{\mathbf{f}}(k-1)\right], \tag{36}$$

$$\boldsymbol{\varepsilon}_{post}(k) = \mathbf{U}_p^T(k)\left[\hat{\mathbf{f}}_0 - \hat{\mathbf{f}}(k)\right]. \tag{37}$$

Subtracting (36) from (37) and taking (35) into account we obtain

$$\begin{aligned}\boldsymbol{\varepsilon}_{post}(k) - \boldsymbol{\varepsilon}_a(k) &= -\mathbf{U}_p^T(k)\left[\hat{\mathbf{f}}(k) - \hat{\mathbf{f}}(k-1)\right] \\ &= -\mu\boldsymbol{\gamma}(k)\,\mathbf{\Gamma}^{-1}(k)\,\mathbf{e}_p(k) \quad (38)\\ &= \mathbf{U}_p^T(k)\left[\mathbf{c}(k) - \mathbf{c}(k-1)\right]. \quad (39)\end{aligned}$$

Hence,

$$\begin{aligned}\mathbf{c}(k) = \mathbf{c}(k-1) + \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,(k)\,\boldsymbol{\gamma}^{-1}(k)\,.\\ \left[\boldsymbol{\varepsilon}_{post}(k) - \boldsymbol{\varepsilon}_a(k)\right],\end{aligned} \tag{40}$$

i.e.,

$$\begin{aligned}\mathbf{c}(k) + \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k)\,\boldsymbol{\varepsilon}_a(k) = \mathbf{c}(k-1) + \\ \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k)\,\boldsymbol{\varepsilon}_{post}(k).\end{aligned} \tag{41}$$

From (38) *a posteriori* pre-whitening error vector can be rewritten as

$$\boldsymbol{\varepsilon}_{post}(k) = \boldsymbol{\varepsilon}_a(k) - \mu\boldsymbol{\gamma}(k)\,\mathbf{\Gamma}^{-1}(k)\,\mathbf{e}_p(k). \tag{42}$$

To evaluate the energy we apply the square $l_2 - \mathrm{norm}$ to both sides of (41) yielding

$$\begin{aligned}\left\|\mathbf{c}(k) + \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k)\,\boldsymbol{\varepsilon}_a(k)\right\|_2^2 = \\ \left\|\mathbf{c}(k-1) + \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k)\,\boldsymbol{\varepsilon}_{post}(k)\right\|_2^2,\end{aligned} \tag{43}$$

$$\begin{aligned}\|\mathbf{c}(k)\|_2^2 + \boldsymbol{\varepsilon}_a^T(k)\,\boldsymbol{\gamma}^{-1,T}(k)\,\mathbf{U}_p^T(k)\,\mathbf{B}^T(k-1).\\ \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k)\,\boldsymbol{\varepsilon}_a(k) = \\ \|\mathbf{c}(k-1)\|_2^2 + \boldsymbol{\varepsilon}_{post}^T(k)\,\boldsymbol{\gamma}^{-1,T}(k)\,\mathbf{U}_p^T(k)\,\mathbf{B}^T(k-1).\\ \mathbf{B}(k-1)\,\mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k)\,\boldsymbol{\varepsilon}_{post}(k)\end{aligned} \tag{44}$$

Let

$$\begin{aligned}\mathbf{a}(k) = \boldsymbol{\gamma}^{-1,T}(k)\,\mathbf{U}_p^T(k)\,\mathbf{B}^T(k-1)\,\mathbf{B}(k-1)\,.\\ \mathbf{U}_p(k)\,\boldsymbol{\gamma}^{-1}(k),\end{aligned} \tag{45}$$

the equation (44) can be rewritten as

$$\begin{aligned}\|\mathbf{c}(k)\|_2^2 + \boldsymbol{\varepsilon}_a^T(k)\,\mathbf{a}(k)\,\boldsymbol{\varepsilon}_a(k) = \|\mathbf{c}(k-1)\|_2^2 + \\ \boldsymbol{\varepsilon}_{post}^T(k)\,\mathbf{a}(k)\,\boldsymbol{\varepsilon}_{post}(k).\end{aligned} \tag{46}$$

Applying expectation to both sides of (46) we obtain

$$\begin{aligned}E\left\{\|\mathbf{c}(k)\|_2^2\right\} + E\left\{\boldsymbol{\varepsilon}_a^T(k)\,\mathbf{a}(k)\,\boldsymbol{\varepsilon}_a(k)\right\} = \\ E\left\{\|\mathbf{c}(k-1)\|_2^2\right\} + E\left\{\boldsymbol{\varepsilon}_{post}^T(k)\,\mathbf{a}(k)\,\boldsymbol{\varepsilon}_{post}(k)\right\}.\end{aligned} \tag{47}$$

Let $\mathbf{d}(k) = E\left\{\|\mathbf{c}(k)\|_2^2\right\}$, we reformulate (47) as

$$\begin{aligned}\mathbf{d}(k) + E\left\{\boldsymbol{\varepsilon}_a^T(k)\,\mathbf{a}(k)\,\boldsymbol{\varepsilon}_a(k)\right\} = \mathbf{d}(k-1) + \\ E\left\{\boldsymbol{\varepsilon}_{post}^T(k)\,\mathbf{a}(k)\,\boldsymbol{\varepsilon}_{post}(k)\right\}.\end{aligned} \tag{48}$$

Substituting (42) into (48) and assuming that $\bar{\delta}\mathbf{I}_{P\times P}$ is negligible ($\boldsymbol{\gamma}(k) \approx \mathbf{\Gamma}^{-1}(k)$) yields

$$\begin{aligned}\mathbf{d}(k) - \mathbf{d}(k-1) \approx \mu^2 E\left\{\mathbf{e}_p^T(k)\,\mathbf{a}(k)\,\mathbf{e}_p(k)\right\}\\ -2\mu E\left\{\mathbf{e}_a^T(k)\,\mathbf{a}(k)\,\mathbf{e}_a(k)\right\}.\end{aligned} \tag{49}$$

The stability of the algorithm is guaranteed if $\mathbf{d}(k) < \mathbf{d}(k-1)$ for $\forall k$, i.e.,

$$\begin{aligned}\mu^2 E\left\{\mathbf{e}_p^T(k)\,\mathbf{a}(k)\,\mathbf{e}_p(k)\right\} - \\ 2\mu E\left\{\mathbf{e}_a^T(k)\,\mathbf{a}(k)\,\mathbf{e}_a(k)\right\} < 0.\end{aligned} \tag{50}$$

Hence,

$$0 < \mu < \frac{2E\left\{\mathbf{e}_a^T(k)\,\mathbf{a}(k)\,\mathbf{e}_a(k)\right\}}{E\left\{\mathbf{e}_p^T(k)\,\mathbf{a}(k)\,\mathbf{e}_p(k)\right\}}. \tag{51}$$

V. COMPUTATIONAL COMPLEXITY

In this section, we compare computational complexity between the proposed PEMSC-IPAPA and other baselines such as the PEMSC-NLMS, PEMSC-IPNLMS, and PEMSC-APA. Table I summarizes the number of real multiplications per output sample [49] for all mentioned AFC methods. We assume that a real multiplication and a real division have equal complexity. The computational complexity for estimating the linear predictor coefficients (LPC) using the autocorrelation matrix and the Levinson-Durbin algorithm is $\frac{5N^2+2LN+N}{2L}$ multiplications, where N is the AR-model order and L is the frame length. Additionally, each pre-whitened signal is computed using N multiplications and the soft-clipping needs 2 multiplications. Thus the PEMSC requires $M = \frac{5N^2+2LN+N}{2L} + 2N + 2$ multiplications per output sample. For the NLMS algorithm the complexity for estimating the adaptive filter coefficients is $3L_{\hat{f}} + 2$ multiplications, where $L_{\hat{f}}$ is the adaptive filter order [50]. In total, the complexity for computing PEMSC-NLMS is $M + 3L_{\hat{f}} + 2$ multiplications. The IPNLMS, APA, and IPAPA need $8L_{\hat{f}} + 2$, $\left(P^2 + 2P\right)L_{\hat{f}} + 2P$, $\left(P^2 + 3P + 4\right)L_{\hat{f}} + 2P$ multiplications, respectively.

In general, the PEMSC-NLMS has the lowest computational complexity. The complexity of PEMSC-APA is higher than that of PEMSC-IPNLMS, but they are marginal if the projection order is small (e.g., for the case of $P = 2$). Although the proposed method yields a higher computational complexity than baselines, its performance outperforms other baselines.

Bảng I
COMPUTATIONAL COMPLEXITY PER OUTPUT SAMPLE.

AFC methods	Computational complexity	#
PEMSC-NLMS	$M+3L_{\hat{f}}+2$	263
PEMSC-IPNLMS	$M+8L_{\hat{f}}+2$	583
PEMSC-APA	$M+\left(P^2+2P\right)L_{\hat{f}}+2P$	585
PEMSC-IPAPA	$M+\left(P^2+3P+4\right)L_{\hat{f}}+2P$	969

A numerical value is given for $N=20$, $L=160$, $L_{\hat{f}}=64$, and $P=2$.

VI. SIMULATION RESULTS

In this section, we evaluate the performance of the proposed method in comparison with considered baselines. We adopt feedback paths measured in two acoustic environments, namely free-field and telephone-near. The free-field (F_1) feedback path is measured without obstacle between loudspeaker and microphone, while the telephone-near (F_2) feedback path is measured with a telephone placed very close to the ear [51]. Fig. 4 depicts the amplitude and phase responses of the measured feedback paths.

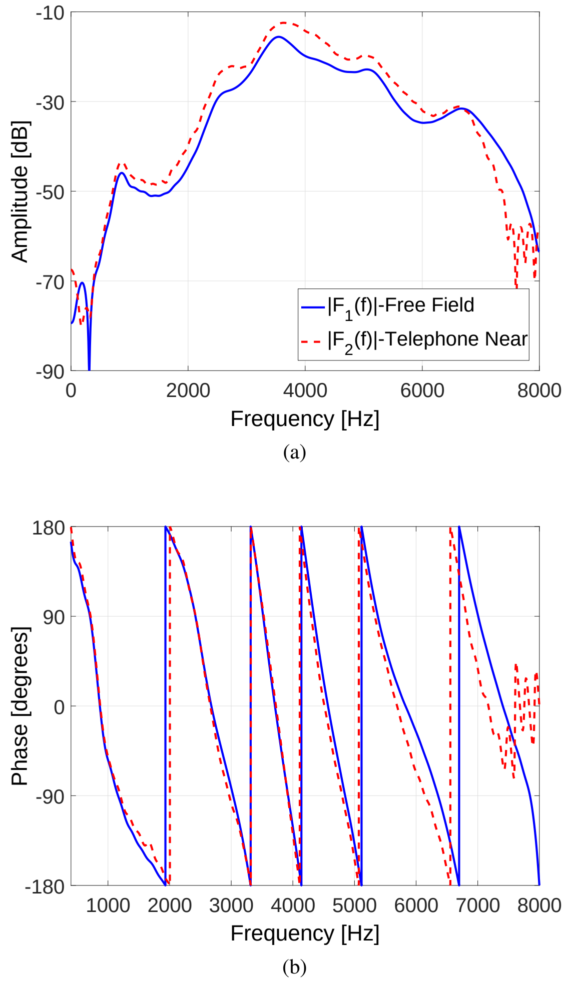
We utilize three common metrics to evaluate the performance of the AFC methods such as the normalized misalignment (MIS), added stable gain (ASG) and perceptual evaluation of speech quality (PESQ) [52]. A good AFC method will have a low value of MIS, high values of ASG and PESQ. The PESQ scores are defined in a range from -0.5 to 4.5, where the values of -0.5 and 4.5 denote bad and excellent speech quality, respectively. For the PESQ measures, the incoming signal $x(k)$ and the error signal $e_{SC}(k)$ are chosen as the reference and the test signals, respectively. The normalized misalignment is defined as [20]

$$\mathrm{MIS}=10\log_{10}\left(\frac{\int_0^\pi\left|F\left(e^{j\omega}\right)-e^{-j\omega d_{fb}}\hat{F}\left(e^{j\omega}\right)\right|^2\mathrm{d}\omega}{\int_0^\pi\left|F\left(e^{j\omega}\right)\right|^2\mathrm{d}\omega}\right), \tag{52}$$

while the added stable gain is defined as [20], [53]

$$\begin{aligned}\mathrm{ASG}=&10\log_{10}\left(\min_\omega\frac{1}{\left|F\left(e^{j\omega}\right)-e^{-j\omega d_{fb}}\hat{F}\left(e^{j\omega}\right)\right|^2}\right)\\&-10\log_{10}\left(\min_\omega\frac{1}{\left|F\left(e^{j\omega}\right)\right|^2}\right),\end{aligned} \tag{53}$$

where d_{fb} is a delay in the feedback canceler's path; $F\left(e^{j\omega}\right)$ and $\hat{F}\left(e^{j\omega}\right)$ are frequency responses of the true and the estimated feedback paths at the normalized angular frequency ω, respectively. We select the following



Hình 4. Measured feedback paths: a) Amplitude responses, b) Phase.

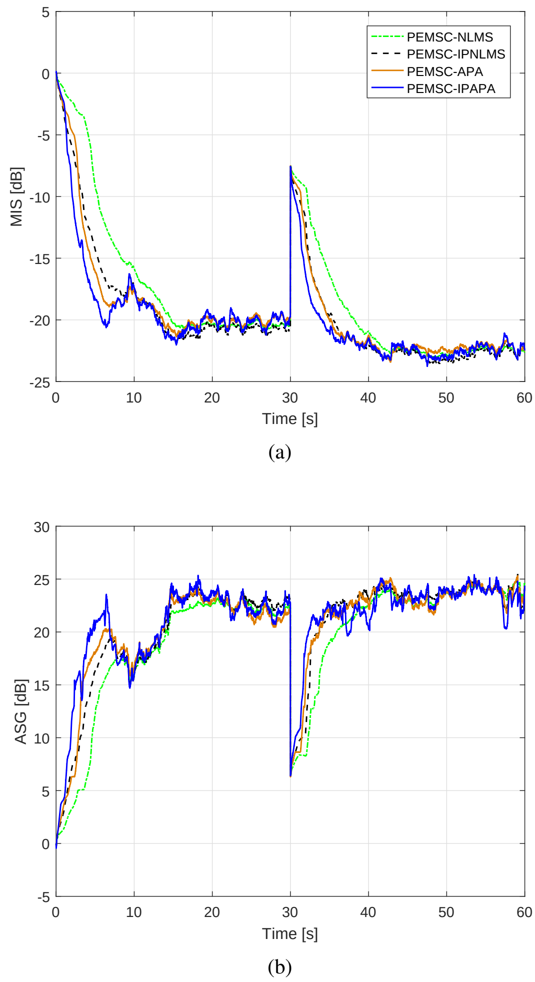
parameters for all simulations: the sampling frequency $f_s=16$ kHz, the delay in the forward path $d_k=96$ samples, the gain in the forward path $|K|=30$ dB, the delay in the feedback canceler's path $d_{fb}=1$ sample, and the regularization parameter $\delta=10^{-6}$. The lengths of the true and estimated feedback paths are $L_f=100$ and $L_{\hat{f}}=64$, respectively. To avoid a highly increase in the computational complexity we select a small projection order, e.g., $P=2$, for all AFC methods using affine projection algorithms like PEMSC-APA and PEMSC-IPAPA. The incoming signals are recorded as described in [30]. We select the step-sizes for all AFC methods such that they provide a similar initial convergence rate. For example, the step size $\mu=0.001$ is chosen for the PEMSC-NLMS and PEMSC-IPNLMS, whereas $\mu=0.0008$ is chosen for both the PEMSC-APA and the PEMSC-IPAPA. In all AFC methods using the PEM, a 20-order AR model of the incoming signal is computed for every frame of 160 samples by using the Levinson-Durbin algorithm [54].

Scenario 1: In this scenario, recorded speech is used as the incoming signal. The speech source is constructed by using 30 IEEE sentences spoken by 3 male and 3 female speakers from NOIZEUS database [52]. In particular, the speech input is produced by concatenating all 30 IEEE sentences together. Furthermore, the feedback path suddenly changes from F_1 to F_2 after half of the simulation time.

Fig. 5 depicts the normalized misalignment and added stable gain of the proposed PEMSC-IPAPA in comparison with those of PEMSC-NLMS, PEMSC-IPNLMS, PEMSC-APA. It can be seen that the PEMSC-IPNLMS and PEM-APA converge and track the change of the feedback path quicker than the PEMSC-NLMS. The PEMSC-APA converges faster while yielding a similar tracking rate compared to the PEMSC-IPNLMS. The proposed method, PEMSC-IPAPA, outperforms all mentioned baselines.

Table II compares the performance of AFC methods in terms of average misalignment ($\overline{MIS_i}$), average added stable gain ($\overline{ASG_i}$) corresponding to the F_i feedback path ($i = 1, 2$), misalignment measured after τ seconds in the F_i feedback path ($MIS_{i,\tau}$) and the necessary time to reach κ_i dB of the misalignment corresponding to the F_i feedback path (τ_{κ_i}). Among them, the first two terms are used to evaluate the steady-state error and the added stable gain, while the last two terms are used to evaluate the convergence/tracking abilities. The best values are indicated in boldface. We select $\tau = 4$ s and $\kappa_i = -15$ dB for this scenario. We observe that the proposed method outperforms all other baselines for both feedback paths. Particularly, it achieves the highest values of $\overline{MIS}_1$, $\overline{MIS}_2$ and $\overline{ASG}_1$. It also provides higher $\overline{ASG}_2$ than that of the PEMSC-NLMS and PEMSC-APA, but a slightly smaller $\overline{ASG}_2$ than that of the PEMSC-IPNLMS. After 4 s the proposed method can reach approximately -16.4 dB and -20 dB of misalignment corresponding to the first and the second feedback paths, while those values for the PEMSC-NLMS, PEMSC-IPNLMS and PEMSC-APA are -4.677 and -14.633 dB, -12.201 and -18.436 dB, -13.986 and -18.391 dB, respectively. Moreover, the needed time for the proposed method reaches to -15 dB of misalignment is much shorter than other baselines for both feedback paths. These results are consistent with the results shown in Fig. 5. Note that the average misalignment ($\overline{MIS_i}$) and average added stable gain ($\overline{ASG_i}$) are computed over 30 s (i.e., 480000 samples) of each realization.

Fig. 7 illustrates the output signal received at the loudspeaker of the HA for all considered AFC methods. It can be seen that all AFC methods suffer from howling at initialization and the sudden change of the feedback path. The PEMSC-NLMS has the longest howling periods. The howling periods in the PEMSC-IPNLMS are shorter than those in the PEMSC-NLMS but longer than those in the PEMSC-APA. Among them, the proposed PEMSC-IPAPA provides the shortest howling periods. These observations match well with the results shown in Fig. 5 and Table II, which demonstrate that the proposed method achieves quicker convergence and tracking rates compared to those of baselines.



Hình 5. Performance of the proposed methods, speech input, feedback path changes from free-field (F_1) to telephone-near (F_2) after 30 s, a) MIS, b) ASG.

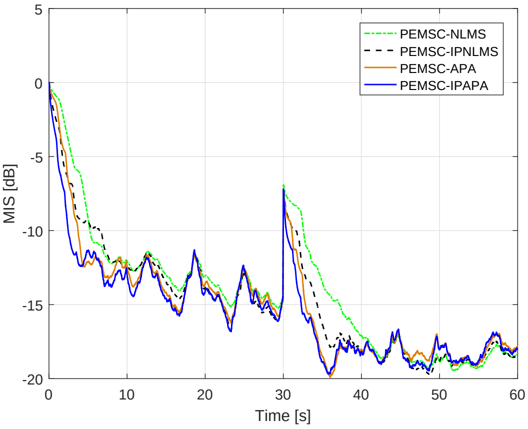
Table III shows the PESQ measures during the last 15 seconds of the incoming signal corresponding to the ith feedback path, namely PESQ_1 for the period 15 s - 30 s and PESQ_2 for the period 45 s - 60 s. It is shown that all considered AFC methods achieve very good speech quality (PESQ scores > 4 for both feedback paths) when the system has converged. Although the PESQ scores of the proposed methods are comparable with those of baselines, it outperforms all baselines in terms of convergence/tracking rates.

Scenario 2: In this scenario, recorded music is used as the incoming signal. Particularly, the song "Imagine" by John Lennon is selected as the music incoming signal. Moreover, the feedback path suddenly changes from F_1 to F_2 after half of the simulation time. We choose $\tau = 4$ s, $\kappa_1 = -8$ dB and $\kappa_2 = -15$ dB for this scenario.

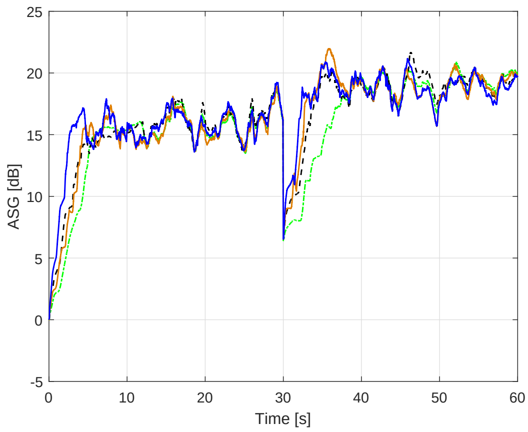
Fig. 6 depicts the performance of all considered AFC methods in the second scenario. It can be observed that both PEMSC-IPNLMS and PEMSC-APA yield quicker

Bảng II
EVALUATE PERFORMANCE OF PEMSC-NLMS, PEMSC-APA, PEMSC-IPNLMS, PEMSC-IPAPA FOR SPEECH AND MUSIC AS THE INCOMING SIGNALS, FEEDBACK PATH CHANGES FROM FREE-FIELD (F_1) TO TELEPHONE-NEAR (F_2) AFTER 30 S.

AFC methods	Incoming signals	$\overline{MIS}_1$[dB]	$\overline{ASG}_1$[dB]	$MIS_{1,\tau}$[dB]	$\tau_{1,\kappa}$[s]	$\overline{MIS}_2$[dB]	$\overline{ASG}_2$[dB]	$MIS_{2,\tau}$[dB]	$\tau_{2,\kappa}$[s]
PEMSC-NLMS	Concat. speech	-16.023	17.656	-4.677	8.312	-20.115	21.194	-14.633	4.190
PEMSC-IPNLMS		-17.739	19.008	-12.201	5.357	-21.154	**22.221**	-18.436	2.470
PEMSC-APA		-17.616	19.141	-13.986	4.401	-20.851	22.084	-18.391	2.302
PEMSC-IPAPA		**-18.266**	**19.829**	**-16.403**	**3.505**	**-21.335**	22.112	**-19.967**	**1.669**
PEMSC-NLMS	Music	-11.493	14.066	-6.184	4.629	-16.568	17.354	-12.324	7.172
PEMSC-IPNLMS		-12.143	14.638	-9.270	3.251	-17.290	18.149	-14.421	4.340
PEMSC-APA		-12.461	14.359	-11.147	3.049	-17.428	18.302	-16.522	2.842
PEMSC-IPAPA		**-13.031**	**15.116**	**-12.391**	**2.076**	**-17.656**	**18.318**	**-16.988**	**2.295**



(a)



(b)

Hình 6. Performance of the proposed methods with music input, feedback path changes from free-field (F_1) to telephone-near (F_2) after 30 s, a) MIS, b) ASG.

convergence and tracking rates than the PEMSC-NLMS. The PEMSC-APA obtains a similar convergence rate to that of the PEMSC-NLMS, but a faster tracking rate. The PEMSC-IPAPA provides further improvement in convergence and tracking rates compared to the PEMSC-APA while retaining a similar steady-state error. These observations are consistent with the results shown in Table II in which the proposed PEMSC-IPAPA achieves the best scores of all evaluated terms for the music incoming signal.

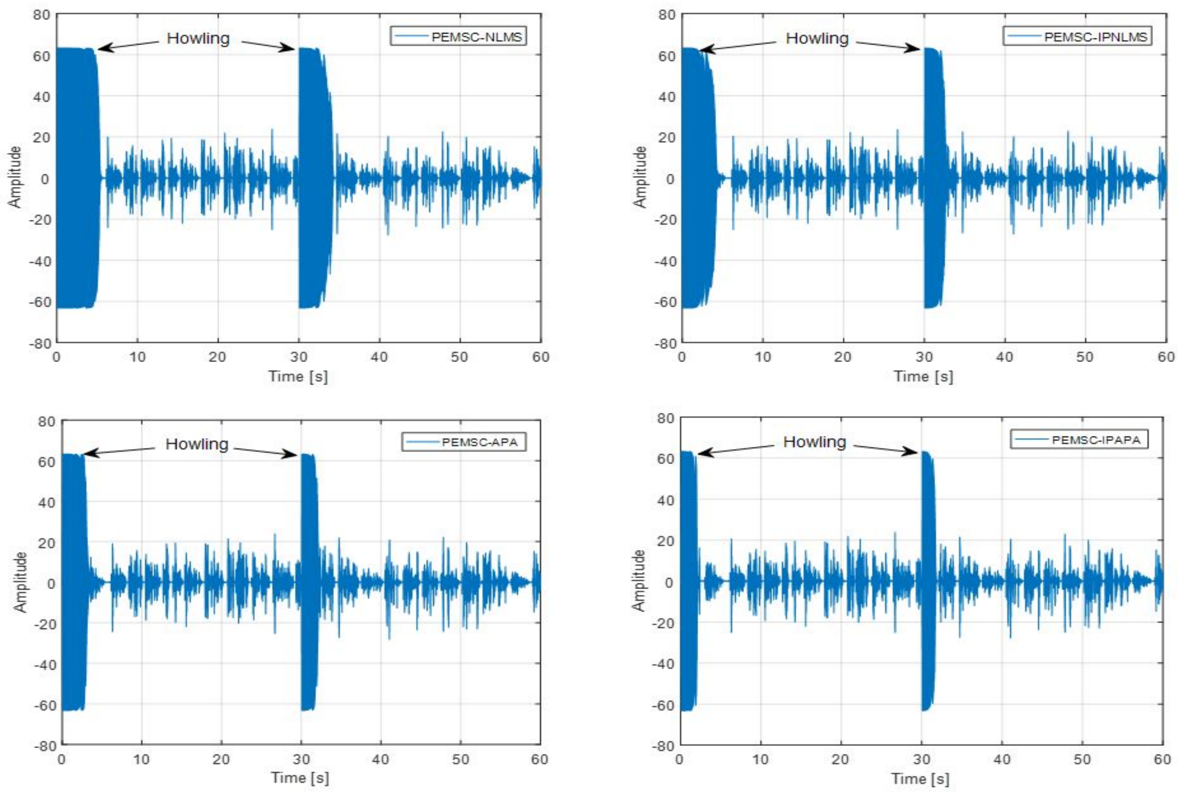
Bảng III
PESQ MEASURES OF THE PEMSC-NLMS, PEMSC-APA, PEMSC-IPNLMS, PEMSC-IPAPA, WITH A SUDDEN CHANGE OF FEEDBACK PATHS FROM F_1 TO F_2 AFTER 30 S, CONCATENATED SPEECH AS INCOMING SIGNAL.

AFC methods	$PESQ_1$	$PESQ_2$
PEM-NLMS	4.40	4.30
PEM-IPNLMS	4.42	4.30
PEM-APA	4.39	4.27
PEM-IPAPA	4.41	4.26

Fig. 8 shows the output signal received at the loudspeaker of the HA for all considered AFC methods with music as the incoming signal. Similar to scenario 1 with the speech incoming signal, the proposed method also yields the shortest howling periods, while the PEMSC-NLMS yields the longest howling periods. The PEMSC-APA has shorter howling periods than both PEMSC-NLMS and PEMSC-IPNLMS. These observations also match well with the results shown in Fig. 6 and Table II.

VII. CONCLUSIONS

In this paper, we propose a new AFC approach for HAs. In the proposed approach, PEMSC-IPAPA, we employ a soft clipper on the error signal as well as integrates the IPAPA for adaptive feedback canceller. The soft clipper limits feedback contribution. The IPAPA takes advantage of a fast-convergence filtering technique (APA) and the feedback path sparseness, which allows for a tap-dependent step-size for updating the adaptive filter coefficients. Stability analysis of the proposed approach is also provided. Simulation results show that the proposed PEMSC-IPAPA achieves a significant improvement in convergence/tracking rates, average MIS and average ASG



Hình 7. The output signal of the PEMSC-IPAPA and baselines, with a sudden change of feedback paths from F_1 to F_2 after 30 s, concatenated speech as incoming signal.

compared to other state-of-the-art AFC approaches in most scenarios while retaining good signal quality. Furthermore, the PEMSC-IPAPA yields the shortest howling periods for both recorded speech and music as the incoming signals and a sudden change of the feedback paths. However, the improvements from using the proposed PEMSC-IPAPA come at an increased cost in computational complexity.

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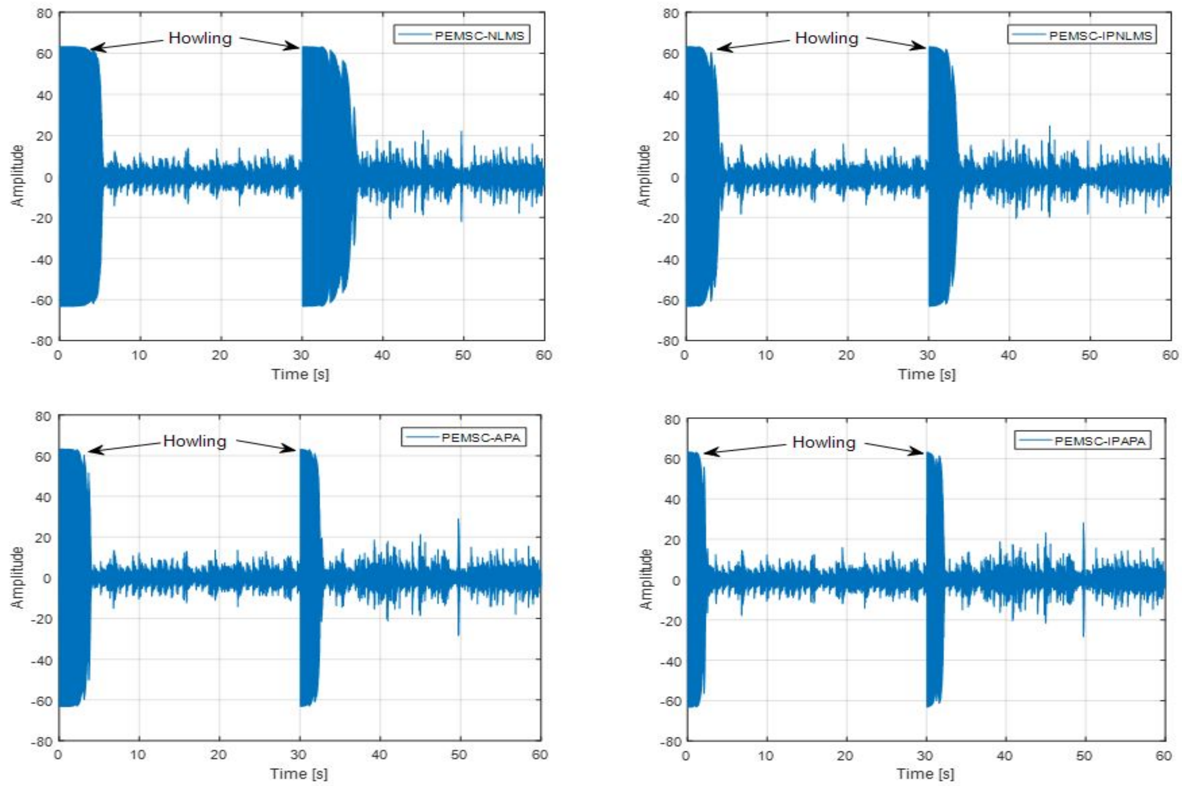

Hình 8. The output signal of the PEMSC-IPAPA and baselines, with a sudden change of feedback paths from F_1 to F_2 after 30 s, music as the incoming signal.

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THUẬT TOÁN AFFINE PROJECTION TỶ LỆ CẢI TIẾN CHO LOẠI BỎ PHẢN HỒI ÂM THANH TRONG MÁY TRỢ THÍNH

***Tóm tắt*—Phản hồi âm thanh là một vấn đề chính trong máy trợ thính dạng mở. Nó làm giảm đáng kể chất lượng tín hiệu và hạn chế độ lợi ổn định cực đại có thể nhận được. Loại bỏ phản hồi thích nghi (AFC) là một phương pháp phổ biến và hiệu quả, tuy nhiên, phương pháp này có sai số trong việc ước lượng kênh phản hồi do sự tương quan cao giữa tín hiệu tại loa và tín hiệu đầu vào mic, đặc biệt là khi tín hiệu đầu vào mic là tín hiệu phổ màu, ví dụ như tiếng nói, âm nhạc. Phương pháp dự đoán lỗi (PEM) khá phổ biến để loại trừ sai số này, kết quả là hiệu suất được nâng lên đáng kể. Để cải thiện hơn nữa hiệu suất của phương pháp PEM thông thường chúng tôi đề xuất tích hợp thuật toán IPAPA vào PEM. Phương pháp được đề xuất, tên là PEMSC-IPAPA, sử dụng các đặc điểm thưa thớt của kênh phản hồi âm thanh và kỹ thuật lọc thích nghi nhanh để nâng cao tốc độ hội tụ/theo dấu. Bài báo cũng phân tích chi tiết phương pháp AFC đề xuất và phân tích độ ổn định của phương pháp. Chúng tôi đánh giá hiệu suất của phương pháp đề xuất sử dụng tiếng nói, âm thanh làm tín hiệu đầu vào mic và kênh phản hồi thay đổi đột ngột. Các kết quả mô phỏng cho thấy phương pháp đề xuất có tốc độ hội tụ/theo dấu cao trong khi vẫn duy trì chất lượng tín hiệu và lỗi ở trạng thái ổn định tương đương so với các phương pháp hiện thời.**

***Từ khóa*—Loại bỏ phản hồi âm thanh, phương pháp dự đoán lỗi, IPAPA, độ lợi ổn định cực đại, tốc độ hội tụ/theo dấu.**

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