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# Design and analysis of a BLPC vocoder-based adaptive feedback cancellation with probe noise

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ABSTRACT

The band-limited linear predictive coding (BLPC) vocoder-based adaptive feedback cancellation (AFC) removes the high-frequency bias, while the low frequency bias persists between the desired input signal and the loudspeaker signal in the estimate of the feedback path. In this paper, we present a BLPC vocoder-based adaptive feedback canceller with probe noise with an objective of reducing the low-frequency bias in digital hearing-aids. A step-wise mathematical analysis of the proposed feedback canceller is presented employing the recursive least square and normalized least mean square adaptive algorithms. It is observed that the optimal solution of the feedback path is unbiased for an unshaped probe noise, but is biased for a shaped probe signal; the bias term does not consist of correlation between the desired input and the loudspeaker output. The identifiability conditions are analysed and it is shown that a delay, greater than or equal to the length of the adaptive filter, must be introduced in the forward path to achieve an unbiased feedback path estimate. Algorithm analysis and computer simulations presented in this paper justify the reason for selecting the proposed design over the existing BLPC vocoder-based feedback cancellation algorithm.



## 1. Introduction

Hearing-aids are electro-acoustic assistive listening devices designed to amplify incoming acoustic signals to compensate for the hearing deficiency in the users. An ill-fitted or a vented hearing aid facilitates acoustic leakage [1]. Consequently, system microphone picks up a part of the output acoustic signal, resulting in the formation of a feedback path from the loudspeaker to the microphone. Acoustic feedback degrades the performance of the audio-reinforcement system and is the prime cause of feedback oscillations in hearing-aids. Inefficient attenuation of the feedback compared to the amplification provided by the assistive listening device leads to system instability, resulting in howling and feedback whistling while limiting the maximum achievable gain [2–4]. To achieve speech intelligibility and target insertion gain for the user, feedback suppression is of paramount importance in hearing-aids. A widely employed method of feedback suppression is adaptive feedback cancellation (AFC) [5]. Adaptive filters are used due to their ability to identify and track changes in the acoustic environment which has a major influence upon the feedback path transfer function [3,6]. Owing to the presence of the closed loop, existence of correlation between the original acoustic signal and the loudspeaker signal leads to biased estimate of the feedback path [7–9]. Achieving a perfect match between the feedback path and its estimate is essential for the system to remain stable at any amount of signal amplification. However, a biased estimate is one of the prime reasons for the occurrence of a modelling error in practice [10].

Several feed-forward suppression and feedback cancellation-based decorrelation techniques have been proposed to deal with the bias problem [5,7,11]. Decorrelation by delay insertion can be performed either in the forward signal processing path or in the filter estimation part [8]. It aids in partly bypassing strong signal correlations at lower time lags. But it is not widely practiced as relatively short time delays are allowed in real-time applications [10–12]. Phase modulation [13] and frequency shifting [14] are feed-forward suppression based phase modification techniques [15,16] which move the feedback sound to a different frequency, thereby breaking the acoustic loop to aid decorrelation. They also tend to smooth the open-loop magnitude function leading to a smaller maximum value, and lack the ability to preserve harmonic structures in voiced speech and other audio signals thereby

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degrading the loudspeaker signal [12,17]. Pre-whitening approaches are based on prediction error method, and these techniques generate white signals for adaptive filter estimation, assuming that the incoming sound signal can be modelled as a white noise signal processed by an all-pole filter [18]. However, apart from speech signals that are unvoiced, not all acoustic signals can be assumed to be auto-regressive [18]. In probe noise approach, an unbiased feedback path estimate can be obtained on the basis of a probe signal, which is normally assumed to be uncorrelated with the desired signal. The estimate comprises of a biased and an unbiased part if a mixture of loudspeaker signal and probe signal is used [19]. Probe noise approach can be used in closed loop identification or to derive an adaptive estimate directly based on it [20–24]. It is required that the probe signal be powerful in comparison to the loudspeaker signal in AFC, resulting in an audible probe noise signal [22–24]. Since the probe signal must be inaudible in the presence of the loudspeaker signal, a shaped probe signal can be utilized for feedback cancellation applications [24]. Decorrelation can be reduced and the loudspeaker sound quality can be preserved by BLPC vocoder-based adaptive feedback cancellation proposed in [10]. It involves replacing the high frequency part of the loudspeaker signal with a synthetic signal that sounds perceptually similar to, but is uncorrelated with, the desired signal. BLPC-vocoder based approach effectively reduces the high-frequency bias, but a small bias still persists in the feedback estimate due to correlation between the incoming acoustic signal and the low-frequency component of the reinforced signal. This biased estimate can result in an inefficient performance in environments where low frequency noise is high. Use of probe noise, which is assumed to be uncorrelated with the incoming signal, in the feedback estimation path can further reduce the bias.

In this paper, we focus on the application of probe noise signal in the feedback estimation path of the BLPC vocoder-based adaptive feedback cancellation for bias reduction. We derive expressions providing a system description and a detailed mathematical analysis of the proposed feedback-cancellation and bias reduction algorithm. It has been accepted that an estimation process driven by probe noise, which is assumed to be uncorrelated with the incoming acoustic signal, results in an unbiased optimal solution of the feedback path [19]. In this work, we show that the insertion of probe noise signal to drive the feedback canceller does not guarantee a perfectly unbiased solution, but achieves better performance compared to that of the existing BLPC vocoder-based design [10]. Here, we derive an optimal solution for both the unshaped and shaped probe signal-based adaptive feedback canceller. In addition, we also find the conditions of identifiability of the feedback path which facilitate an unbiased solution of the feedback canceller for the probe noise being correlated and uncorrelated with the white noise signal to the estimated desired signal model. The paper is divided into five sections including the introduction. We present a brief background of the basic AFC process and the BLPC vocoder-based feedback cancellation in Section 2. In Section 3, we propose a BLPC vocoder-based feedback canceller design with probe noise in the filter estimation path and also, a mathematical analysis of the proposed algorithm for both unshaped and shaped probe signal is presented using recursive least squares (RLS) algorithm for achieving faster convergence as compared to normalized least mean square (NLMS) algorithm. In Section 4, we define the identifiability conditions that facilitate optimal set of coefficients for the feedback canceller to obtain an unbiased solution, with the probe signal which is correlated and uncorrelated with the white noise input to estimated desired signal model. In Section 5, computer simulations are presented in support of the analysis.

The following notations have been adopted throughout the paper; $[\cdot]^T$ for the transpose of a matrix, $[\cdot]^{-1}$ for the inverse of a matrix, $E[\cdot]$ for the expectation operation, $n$ for discrete-time index, $k$ for discrete-time delay operator such that $k^{-1}m(n) = m(n-1)$, bold-faced capital letters for matrices, bold-faced small letters for vectors and $\mathbb{Z}$ for integers. A discrete-time filter of length $L$ is represented as a polynomial $F(k)$ in terms of $k^{-1}$ as $F(k) = f_0 + f_1 k^{-1} + \cdots + f_{L-1} k^{-L+1}$ or by its coefficient vector $\mathbf{f} = [f_0, f_1, \ldots, f_{L-1}]^T$. The signal $m(n)$ is filtered by $F(k)$ as $F(k)m(n) = \mathbf{f}^T(n)\mathbf{m}(n)$, with $\mathbf{m}(n) = [m(n), m(n-1), \ldots, m(n-L+1)]^T$. $F(\Omega, n)$ denotes the time-varying short-term frequency spectrum of $F(k)$.

## 2. Preliminaries

In this section, we present a brief background of adaptive feedback cancellation and the BLPC vocoder-based AFC. The analysis is presented considering that all time-domain signals are real valued and the filter order of the adaptive filter is sufficiently large to prevent undermodelling of the feedback path.

### 2.1. Adaptive feedback cancellation

A general block diagram of the adaptive feedback cancellation system is shown in Fig. 1. The incoming acoustic signal is assumed to be a wide sense stationary process and is denoted as $x(n)$. During the hearing-aid operation, the loudspeaker output signal $u(n)$ is processed by the acoustic environment and reaches the microphone as a feedback signal $f_B(n)$. Thus, it forms an acoustic feedback path as shown in the figure. For the analysis, original feedback path $F(k)$ is considered as an FIR filter of length $L_f$ and coefficient vector $\mathbf{f}(n) = [f_0, f_1, \ldots, f_{L_f-1}]^T$. The adaptive filter $\widehat{F}(k)$ is also an FIR filter of length $L_{\hat{f}}$ and the coefficient vector $\hat{\mathbf{f}}(n) = \left[\hat{f}_0, \hat{f}_1, \ldots, \hat{f}_{L_{\hat{f}}-1}\right]^T$. The microphone output $y(n)$ is fed into the system as shown in Fig. 1 and it is expressed as

$$y(n) = x(n) + f_B(n), \tag{1}$$

where

$$f_B(n) = F(k)u(n). \tag{2}$$

The time-varying nature of $F(k)$ is tracked by $\widehat{F}(k)$ such that an estimate of the feedback signal $f_B(n)$, i.e., $v(n)$ can be obtained as

$$v(n) = \widehat{F}(k)u(n). \tag{3}$$

The feedback estimate $v(n)$ is subtracted from the microphone output $y(n)$ to produce an estimate of the original speech signal without the feedback signal, i.e. $e(n)$, which is represented as

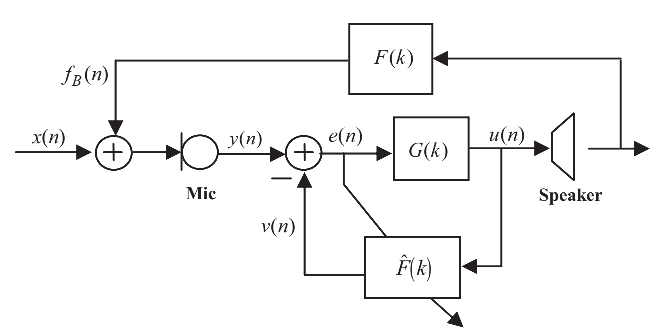


**Fig. 1.** General block diagram of AFC.