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Low Complex IIR Adaptive Hear-Through Ambient Filtering for Overcoming Practical Constraints in Earbuds

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Abstract

The materials used in the headphones and earbuds typically attenuate sound passively above frequency of 500-700 Hz. Hear-Through (HT) techniques can compensate for the passive attenuation characteristics of listening devices to achieve a listening experience similar to the open ear for enhancing situational awareness of the users. Most studies have proposed usage of fixed-filter or adaptive finite impulse response (FIR) HT techniques, which can lead to mismatch between the modelled and desired open ear response for different users and fittings, or lead to higher computations and larger processing delays. This paper proposes a virtual sensing model with an infinite impulse response (IIR) based adaptive equalization method to model the open ear response at user's eardrum. The proposed method reduces the computational complexity by 34%, while maintaining stability and providing superior HT performance upto 10 dB compared to existing methods in dynamic acoustic scenarios.

Index Terms: Hear-Through, ambient filter, adaptive IIR filter, adaptive FIR filter, ear canal model

1. Introduction

With advancements in Digital signal processors and algorithms for audio technology, several smart headphones and earbuds have been developed which provide higher fidelity audio, active noise control (ANC) and hear-through (HT) modes (also referred as ambient/transparency mode) for enhancing immersion, comfort and awareness of the user [1], [3-6] [14]. In all earbuds and headphones, the external ambient sound is attenuated by passive filtering due to physical structure of the devices [7, 8]. The goal of HT mode is to make the earbuds acoustically transparent thereby allowing the user to have a listening experience over headphones or earbuds which is perceptually similar to the open ear listening. The HT mode ensures that the user has environmental awareness in day-to-day listening such as conversations, music, pet sounds and paying attention to emergency situations such as alarm sounds, sirens or horns [9], [10]. To achieve this goal, external ambient signal captured by the microphone facing the ambient environment (referred to as reference microphone) should be processed by the HT Equalization (EQ) filter to generate the pseudo-ambient sound signal. The pseudo ambient signal is played back by the loudspeaker which combines with leaked signal present due to passive isolation to generate the ambient signal heard by users in earbuds devices [11].

Previous studies have proposed HT techniques primarily using fixed and adaptive FIR filtering and fixed IIR filtering methods. Ramo and Valimiki proposed and evaluated HT filtering method based on the all-pass filtering [12, 13]. Ranjan and Gan described the adaptive equalization methods for fusion of the virtual and real sounds using the prototype open-ear headphones with up to 10dB of isolation for higher frequencies (above 5 kHz) [9]. Patel et al. proposed a super directive beamformer combined with adaptive FIR filtering techniques using directional microphone arrays for HT [15]. Zhaung et al. proposed a frequency domain-based implementation for HT EQ similar to the constrained Active Noise Cancellation (ANC) design based on solving cone programming problem [16]. Approaches combining deep learning and dimensionality reduction techniques with fixed or adaptive filtering techniques to improve HT performance have also been explored in past literature. In our previous paper, we proposed a parametric HT equalization method using NN models to compute direction of arrival (DoA) of incident sound source and selecting the appropriate HT filter pre-computed using adaptive approaches from the database [17]. Jin et al. proposed the derivation of individualized HT equalization filters by leveraging secondary path estimates to predict sound pressure at the eardrum using the Principle Component Analysis (PCA) method [18]. Huang et.al. has proposed a FIR filter Least mean square (LMS) based adaptive HT for in-ear headphones [24].

However, previous HT studies mentioned above have mostly focused on utilizing fixed FIR/IIR or adaptive FIR filters for HT filtering. For HT techniques, minimizing computational complexity is important to reduce processing delay, since large processing delays can lead to unnatural HT experience and comb-filtering effects. While fixed filters analyzed in previous studies have low processing delay in real-time operations, these filters can often lead to mismatch between modelled and desired open ear response for different users and different fittings for each user. Adaptive FIR filters overcome this issue and provide a closer match to the target open ear response while maintaining stability, however, they often lead to higher computational complexity, larger memory requirements due to higher filter orders and higher processing delays for earbud devices which often have strict memory and computational resource constraints. Although adaptive IIR filters can help to reduce the computational complexity, memory usage and processing delays, based on our review, most previous HT studies have not presented a systematic analysis for ensuring IIR filter stability and achieving superior performance for different real-world HT scenarios.

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[*] Equal contribution
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Moreover, most of the previous studies have examined HT performance close to the internal microphone present near to the loudspeaker of the headphones or earbuds. However, the HT performance typically has some difference at the internal microphone and at the ideal location of the error microphone, which is at the eardrum of the user. Evaluating performance near the eardrum is difficult because it is difficult to place an error microphone safely close to the eardrum. It is also difficult to model the unique characteristics of each user's individual ear canal [18]. The equivalent circuit models which have been typically used for estimating sound pressure at eardrum, are Norton/Thevenin models [19], Transmission line equivalence model [20], modified horn equation model [21], reflectance phase and pressure minima-based methods [25]. Although most of the previous studies have been focused on the lumped parameter models, practical usage of these models for earbuds or headphones is challenging due to their limited accuracy across the wide frequency range of interest (typically 100 Hz to 8 kHz) and their lack of generalizability across different users and fittings.

This study aims to address the issues using the proposed virtual sensing based in-ear modelling for HT system. Here, virtual sensing is proposed to estimate the sound pressure at the eardrum based on the sound pressure at the internal microphone. The approach of virtual sensing is inspired by its usage in ANC to control the noise at the desired zone of quiet region [22, 23]. In this paper, virtual sensing based adaptive IIR HT EQ filter is developed for reducing computational complexity and processing delay for HT system. The stability of IIR filter is ensured using an equation error model. The virtual path is been selected from the database which has been created to store different virtual paths based on the dominant frequency of the leakage signal between the reference microphone and internal microphone. The proposed method provides better ear canal estimation and improved HT performance compared to other techniques with lower computational complexity.

2. Proposed Virtual Sensing based HT System

The $x(n)$ is signal received by the reference microphone and the pseudo-ambient signal captured at the internal microphone is given by

$$d'(n) = p_e(n) + y'(n) \tag{1}$$

Where the residual ambient sound signal due to passive attenuation by the device can be denoted as $p_e(n)$ and the compensating signal generated by the HT system is $y'(n)$. To achieve our goal for HT system, ideally the pseudo ambient signal and the open ear signal must be equal i.e., $d'(n) = d(n)$. The open ear response $\hat{T}(z)$ is obtained using the IKS Open-source dataset [24]. Since HT responses at internal microphone and eardrum are not identical, we utilize the virtual sensing technique.

2.1. Remote microphone based virtual sensing technique using online secondary path of HT filter

In order to obtain the HT EQ filter, we use remote microphone based virtual sensing technique to estimate the pseudo ambient signal at the ear drum of the user as shown in Fig. 1. Secondary path $(S(z))$ is the path between the secondary speaker and internal microphone, whereas virtual secondary path $(S_v(z))$ is the path between the secondary speaker and error mic at ear drum. Primary path $(P(z))$ is the path between ambient sound source and the internal microphone respectively. The mean square error signal minimized using the gradient descent method is given by

$$E[e^2(n)] \approx 0 \tag{2}$$

The error signal at the eardrum can be given by

$$e(n) = d(n) - (p_d(n) + y''(n)) \tag{3}$$

Where, $d(n)$ is the open ear response, $p_d(n)$ is the residual ambient sound due to passive isolation at the error mic at ear drum of the user, and $y''(n)$ is the compensation signal which can be given by $y''(n) = y'(n) * h(n)$. The $h(n)$ is the virtual path or ear canal path, $*$ is the convolution operator and $y'(n)$ is the estimated compensation signal at the internal microphone which can be given by

$$y'(n) = e'(n) - (y(n) * \hat{s}(n)) \tag{4}$$

Here, $\hat{s}(n)$ denotes the estimate of the secondary path between the secondary speaker and internal microphone. The input signal to the loudspeaker is given by

$$y(n) = \boldsymbol{w}^T(n)\boldsymbol{x}(n) \tag{5}$$

where, $\boldsymbol{w}(n) = [w_0(n),\ w_1(n), \dots, w_{L-1}(n)]^T$ is the finite impulse response (FIR) EQ filter of length L. The input ambient signal from the feedforward microphone can be given by $\boldsymbol{x}(n) = [x(n), x(n-1), \dots,\ x(n-L+1)]^T$. Based on the generic formulation for filtered input-based adaptive algorithms, update equation can be formulated as

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \mu\, f[e(n)]\, \boldsymbol{x}'(n) \tag{6}$$

From (6), μ is the step size of the algorithm and $\boldsymbol{x}'(n) = [x'(n), x'(n-1), \dots,\ x'(n-L+1)]^T$ is the tap delayed filtered input reference signal with $x'(n) = x(n) * \hat{s}_v(n)$. The $s_v(n)$ is the virtual secondary path between the secondary speaker and the error mic at ear drum, and $f[e(n)]$ is the error signal function that varies for different adaptive approaches. In many scenarios, secondary path varies due to different fittings of the user. So, we propose using online secondary path modelling-based approach. The excitation signal $v(n)$ is played through the device loudspeaker, which is independent of the ambient sound. The error signal can be modified as

$$e(n) = d(n) - (p_d(n) + y''(n)) + v_1'(n). \tag{7}$$

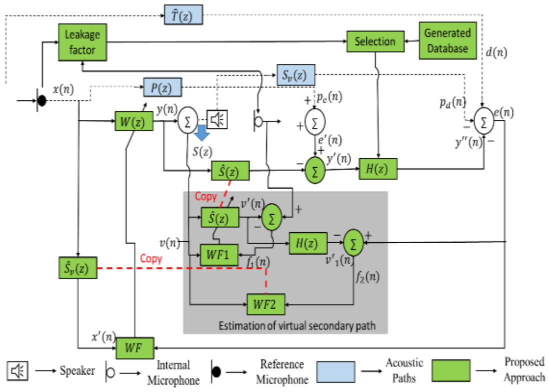

Figure 1: *Block diagram of the proposed HT EQ filter*

The estimated secondary path signal $v'(n) = v(n) * s(n)$ and the corresponding secondary path is estimated using the LMS adaptive algorithm whose update equation can be given by

$$\hat{\boldsymbol{s}}(n+1) = \hat{\boldsymbol{s}}(n) + \mu_s f_1(n)\boldsymbol{v}(n) \tag{8}$$

with $f_1(n) = e'(n) - v'(n)$ and $\boldsymbol{v}(n)$ is the tap delayed excitation signal of the length equal to length of the estimated secondary path. The corresponding HT EQ filter can be given by

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \mu\,(f[e(n)]\,\boldsymbol{x}'(n) + v_1{}'(n)\boldsymbol{x}'(n)) \tag{9}$$

The virtual secondary path estimate can be similarly obtained as

$$\hat{\boldsymbol{s}}_v(n+1) = \hat{\boldsymbol{s}}_v(n) + \mu_{s1} f_2(n)\boldsymbol{v}(n) \tag{10}$$

Where, $f_2(n) = e(n) - v_1{}'(n)$, virtual secondary path output $v_1{}'(n) = v_1(n) * s_v(n)$, $v_1(n) = v(n) * h(n)$ where $h(n)$ being the virtual path.

In practical implementation, in order to obtain the HT EQ, the order of finite impulse response (FIR) of the EQ filter must be very large, which is not practical for low-latency applications in earbud devices. Hence, we propose infinite impulse response (IIR) based adaptive approach which can be approximated with smaller filter orders. The optimization function for the derivation of adaptive IIR filter coefficients is the mean square error signal obtained at the eardrum of the user represented as

$$E[e^2(n)] \approx e^2(n) \approx 0 \tag{11}$$

The E[.] is the expectation operator and it is minimized using the gradient descent algorithm to provide the optimal coefficients of the adaptive filter given by

$$\boldsymbol{b}(n+1) = \boldsymbol{b}(n) - \mu\frac{\partial e^2(n)}{\partial b} \tag{12}$$

$$\boldsymbol{a}(n+1) = \boldsymbol{a}(n) - \mu\frac{\partial e^2(n)}{\partial a} \tag{13}$$

Where,

$$\frac{\partial e^2(n)}{\partial a} = 2e(n)\frac{\partial e(n)}{\partial a},\quad \frac{\partial e^2(n)}{\partial b} = 2e(n)\frac{\partial e(n)}{\partial b}. \tag{14}$$

The error signal e(n) is given by

$$e(n) = d(n) - (p_d(n) + y''(n)) + v_1{}'(n), \tag{15}$$

and

$$y''(n) = (e'(n) - \sum_{i=0}^{P} b_i(n)\boldsymbol{x}'(n-i) - \sum_{j=1}^{Q} a_j(n)\boldsymbol{y}'(n-i) * \hat{s}(n) * h(n). \tag{16}$$

Here, * is the convolution operator. Substituting equation (14) in (12) and (13), and solving using the equation error-based approach [28], weight update equations of the IIR adaptive filter can be derived as

$$\boldsymbol{b}(n+1) = \boldsymbol{b}(n) + \mu(e(n)\,\boldsymbol{x}'(n) + v_1{}'(n)\boldsymbol{x}'(n)) \tag{17}$$

$$\boldsymbol{a}(n+1) = \boldsymbol{a}(n) + \mu\big(e(n)\boldsymbol{y}'(n) + v_1{}'(n)\boldsymbol{x}'(n)\big) \tag{18}$$

The limitations for the step size parameter (μ) can be derived by the similar procedure as described in [31] for further analysis.

2.2. Generation of database of the virtual paths

Due to the different anthropometric features of ears, the transfer function between the internal microphone (approximation of error microphone) and the eardrum microphone (ideal error microphone location) has to be modelled accurately. Typically, the length of the ear canal of the person varies from 23mm to 29mm, and the dominant frequency of the ear canal changes correspondingly. The dominant frequency is inversely proportional to twice the length of ear canal for open ear response. However, for the occluded ear response the dominant frequency is inversely proportional to four times the length of ear canal. In this paper, we used leakage signal, which is computed using difference between the reference and internal microphone. The leakage signal's dominant frequency is calculated for different fittings. The corresponding virtual path is determined using the adaptive system identification [29] approach for corresponding fitting based on the measured responses. The computed frequency, estimated ear canal length and estimated virtual path are stored during the pertaining stage. During the implementation phase, the dominant frequency of leakage signal is obtained to calculate ear canal length, and corresponding virtual path needs to be loaded in device memory as shown in Fig. 1. Considering the computational memory, and construction of earbuds, the virtual path length of 128 and total 10 different characteristics of ear canal is taken. To store each frequency and length component of each virtual path, we require 2 bytes, and each virtual path of length 128 needs 256 bytes of memory, so we need 2.5 kB of space to store the database in INT16 format on earbuds, which is reasonable storage for most earbuds based on our review.

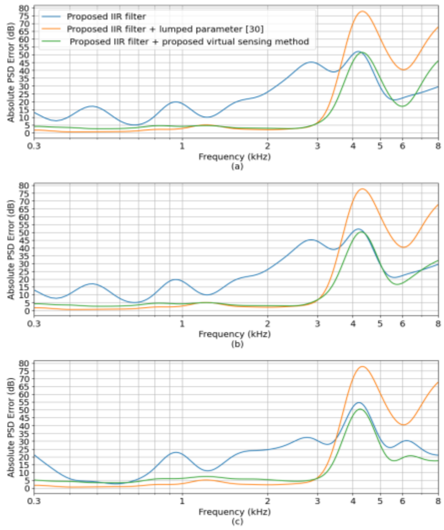

Figure 2: *PSD performance at the eardrum compared to open ear response for different fittings*

3. Results and Discussion

To evaluate the performance of the proposed approach, we carry out the simulations at the sampling frequency of 16 kHz studying different effects. The state of art adaptive FIR used for comparison are sample based least mean square (LMS) algorithm, block based LMS and block based total least square (TLS) algorithms.

3.1. Comparison of the HT EQ filter performance at the ear drum

To compare the effect of virtual sensing on the HT playback of the earbuds, a white noise with zero mean and unit variance is used as an excitation signal. The absolute value of power spectral density (PSD) error is computed at eardrum with respect to open ear response is shown in Fig. 2. From Fig. 2, we can observe that proposed virtual sensing based technique has similar performance as the state-of-the-art lumped parameter model [30] below 3 kHz frequency range. It has significantly less PSD error compared to that of lumped parameter-based ear canal model above 3 kHz range by more than 20 dB for certain frequencies between 4 to 8 kHz. With our proposed ear canal modelling technique combined with IIR filtering, the performance improves slightly above 3 kHz without ear modelling, with maximum benefit of our proposed technique around 5.5 kHz with about 3 to 6 dB lesser error compared to existing techniques.

3.2. Proposed HT-EQ filter evaluation for white noise and simulated acoustic environments

3.2.1. Experiment-I: White noise signal

The HT performance is evaluated using the mean square error (MSE). Using white noise signal of length 10s, it can be observed from Fig. 4(a) that proposed IIR filter was able to achieve low MSE compared to existing algorithms, however, convergence was slightly slower in some cases due to addition of filter stability criterion compared to state-of-the-art adaptive FIR filters.

3.2.2. Experiment-II: Simulated Acoustic environments

To evaluate the performance of proposed algorithm in real-world scenarios, complex indoor and outdoor acoustic environments are simulated. In all cases, the elevation for directional sources is taken to be at 0°. The outdoor environment had the user listening to siren on a vehicle moving at constant speed of 15 km/hr from azimuthal direction of 0^0 to 150^0, together with a diffuse wind signal, and a speech signal from a speaker situated at 75^0 azimuth. For simulating an indoor environment, the user is listening to music from a loudspeaker at 60^0 together with diffuse construction noise, and distant sound of car passing by on the road from 0° to 150° azimuth at 15 km/hr, to mimic car sound passing by a window. Fig. 3 shows the pole plot for the simulated acoustic environments and the white noise signal. The proposed IIR filter is stable for simulated indoor and outdoor acoustic environments, with all poles lying inside unit circle as shown in Fig. 3. The Fig. 4 (b) and (c) shows the MSE plot for outdoor and indoor cases respectively, where the blue lines represent the DoA switching for siren and car sounds. It can be observed that proposed technique is able to track the changes in the sound with lower MSE compared to other state-of-the art adaptive FIR algorithms, and fixed IIR filter in both the indoor and outdoor cases.

Table I. *Computational Complexity*

Algorithm	MAC Computation	MACs
FxLMS	$4L_v + 6L_s + 4L_w$+4	2308
FxBLMS	$4L_v + 6L_s + 4L_w$+4	2308
FxBTLS	$4L_v + 6L_s + 5L_w$+6	2565
Fixed IIR filter	$2L_v + 2L_s + 2L_a + 2L_b$+5	637
Proposed IIR adaptive filter	$\mathbf{4L_v + 6L_s + 4L_a + 4L_b+5}$	**1525**

3.2.3. Computational complexity analysis

The computational complexity for all the algorithms is shown in Table I. Here, L_v is the length of the virtual path taken as 128, L_s is the length of secondary path which is also taken as 128,

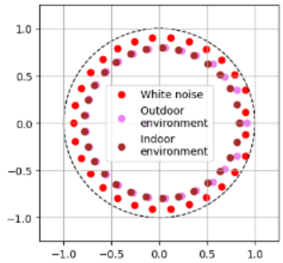

Figure 3: *Pole plot of the proposed IIR filters, which is stable with all poles inside the unit circle for white noise, simulated indoor, and outdoor environments signals*

L_w is the length of FIR adaptive filter taken as 256, L_a, and L_b are the number of poles and zeros of the proposed IIR filter,

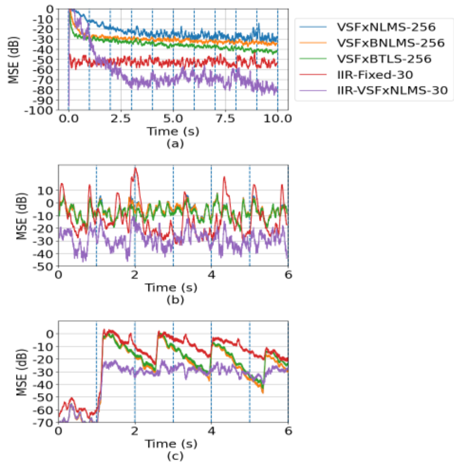

Figure 4: *MSE performance comparison for proposed adaptive IIR filter with state-of-the-art adaptive FIR and fixed IIR filter for (a) white noise, (b) simulated outdoor and (c) simulated indoor environments*

which are taken as 30 each. It can be depicted that proposed approach has lower MAC operations compared to other state-of-art adaptive FIR filters by more than 34%.

4. Conclusion

In this paper, an IIR adaptive HT EQ filter using the remote microphone based virtual sensing technique has been described. The proposed virtual sensing technique combined with adaptive IIR technique reduces the computational complexity in terms of MAC operations by about 34%, while improving the HT performance by about 10-20 dB for white noise signals and simulated real world environments over existing state-of-the-art adaptive FIR techniques and fixed IIR filters. Our proposed method also improves the ear canal modelling performance over existing lumped parameter methods above 3 kHz range. Future works can consider a utilizing a bigger database of eardrum responses across multiple users and fittings to derive the estimated response, and further evaluation of adaptive IIR filter techniques in different complex real-world acoustic environments.

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