Investigation and Development of Digital Active Noise Control Headsets

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***Abstract*—Active Noise Control (ANC) applications for headsets are strongly influenced by practical constraints. Nevertheless, most contributions targeting the development of novel algorithms for ANC headsets are based on simplified simulations and neglect practical limitations. In other publications, real-time ANC prototypes are proposed which involve dedicated (and often expensive) audio hardware. In most cases, however, only very basic concepts of ANC can be realized in real-time due to a high involved programming effort and limited computational power.**
In this contribution, a novel approach for the investigation and development of algorithms for ANC headsets is presented. This new approach, on the one side, enables to deeply investigate different algorithms efficiently and without significant development (and cost) effort. On the other side, it involves real audio hardware, highly accurate measurement techniques and recorded audio signals. Therefore it is almost as valuable for the integration of algorithms in future ANC headset devices as a real-time prototype. In order to demonstrate the benefits of the new approach, exemplary simulation results will be presented which have been produced involving a prototype device built from a commercial in-ear headset and a miniature size microphone.

I. Introduction

Communication and the consumption of music have more and more become mobile applications. These applications are desired to function properly even in situations with adverse acoustic constraints such as a high level of ambient noise. The passive protection of the used headsets against unwanted ambient noise is often not sufficient. In this regard, in particular in combination with sophisticated signal enhancement techniques (e.g. proposed in [1]), ANC techniques can be a useful supplement to create moments of quiet and allows for undisturbed conversations in noisy environments.

The overall goal of ANC is to attenuate or cancel undesired noise signals. In order to achieve this goal, a compensation signal or *anti-noise* signal is produced by a loudspeaker such that at a specific point, the undesired ambient noise and the anti-noise superpose in a destructive way. In the literature, the zone around the specific point where the signals cancel out is called the *quiet zone*. In case of ANC headsets, the anti-noise is emitted by the loudspeaker of the headset and the quiet zone is supposed to be located around the ear drum. [1]

In the literature, research and development of novel algorithms for ANC systems is often based on very simple simulations and unrealistic assumptions. In particular, the so-called *unknown plant*, the acoustical path to be identified in feed forward ANC, is often modeled by a linear filter, e.g. [2][3][4]. Also, the impact of hardware components, in particular Sigma-Delta Analog-to-Digital Converters, (ADC) [5] is not or inadequately considered in most cases. These (low cost) converters are used in almost all commercial devices involving digital audio signal processing and cause signal delays which are critical for ANC.

Realistic ANC simulations in general involve digital signal processing in real-time, e.g., presented in [6]. Dedicated and expensive audio hardware (e.g., the DSpace 1103 Controller Board [7]) is necessary to enable an extremely low signal delay. In this context, algorithm development and research can not be realized efficiently since a high programming effort is required to operate the ANC algorithm in real-time and due to the limitation of computational power.

In this paper, a novel approach for the investigation and development of ANC headsets is proposed. This approach, on the one side, enables to develop new algorithms in the Matlab environment in a convenient (and cost-effective) way without any complexity restrictions. On the other side it enables the exploration of the influence of practical aspects and is almost as valuable as a real-time prototype since real audio hardware is involved. Coherence plots of real recordings involving a physical ANC in-ear headset prototype as well as ambient noise attenuation plots for feed forward ANC will be presented to demonstrate the benefit of the proposed system.

II. Active Noise Control in Headsets

A generic model for broadband ANC in headsets is shown in Figure 1. Note that for the sake of simplicity, all signals are assumed to be available in a time discrete representation in the model with time index k. The ambient noise signal in the figure reaches the ANC headset from the left side. It is recorded by the *reference microphone*, denoted as signal $x(k)$. The reference microphone is located outside the *ear cup* by which headsets are often shielded against ambient noise to a certain extend. The passive shielding does not attenuate the ambient noise perfectly, therefore, a specific part of the ambient noise enters the ear cup and reaches the position of the *error microphone*, denoted as signal $d(k)$. In ANC **headphones**, the error microphone is in most cases located in proximity to the entrance of the ear canal, in **in-ear headsets**, it may also be located more closely to the ear-drum inside the ear-canal. In order to attenuate the ambient noise reaching the ear, the *active noise control circuitry* calculates the compensation signal $y(k)$ which reaches the error microphone, denoted

[1]Note that in most cases, besides the creation of the quiet zone, a speech or audio signal should be emitted by the headset loudspeaker to reach the ear drum unaltered which, however, will not be of interest in the following.

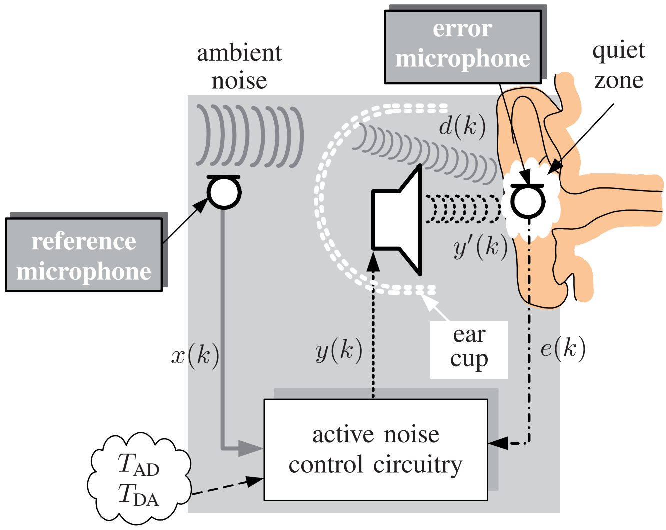

Fig. 1. Generic hardware setup for ANC headset.

as signal $y'(k)$. The signals $d(k)$ and $y'(k)$ are superposed to form the *residual ambient noise* signal

$$e(k) = y'(k) + d(k). \tag{1}$$

In order to adapt to different acoustic conditions, the residual ambient noise signal $e(k)$ is recorded by the error microphone and fed back to control the *ANC circuitry* such that the variance of signal $e(k)$ is minimized. Though in ANC headphones the quiet zone can be created in proximity to the error microphone and hence the entrance of the ear-canal only, significant noise cancellation (> 10 dB) is achieved for frequencies < 3000 Hz also at the eardrum [8].

A. Investigated Approaches for ANC in Headsets

Different approaches for broadband feed forward and feedback ANC are described, e.g., in [9] and have been integrated as part of the proposed system. The influence of practical constraints is more significant for broadband feed forward ANC than for feedback approaches since two signals (in general the reference signal $x(k)$ and the residual error signal $e(k)$) are involved in signal processing. Therefore, mainly this approach will be considered in the following.
In broadband feed forward ANC, the anti-noise signal $y(k)$ is computed from signal $x(k)$ as the output of a finite impulse response (FIR) filter of order N with the filter coefficients w_i,

$$y(k) = \sum_{i=0}^{N} x(k-i) \cdot w_i. \tag{2}$$

Assuming that signals $d(k)$ and $x(k)$ are stationary, the optimal filter coefficients can be computed based on a **Minimum Mean Square Error** (MMSE) approach (the Wiener-Hopf equations, e.g., in [9]). In practical ANC applications, the filter coefficients are in general time variant to contribute for a varying acoustic environment. Therefore, gradient descent algorithms such as the **Least Mean Square** (LMS) approach [10] are employed. However, the acoustic path from the loudspeaker to the error microphone, denoted as the *secondary path*, demands special consideration in a variant of the LMS, the *Filtered-x LMS* (FxLMS) approach [11].
The maximum theoretically achievable ANC performance following the broadband feed forward approach is [9]

$$\frac{\varphi_{e,e}(f)}{\varphi_{d,d}(f)} = \Big(1 - C_{d,x}(f)\Big). \tag{3}$$

In this equation, the terms $\varphi_{d,d}(f)$ and $\varphi_{e,e}(f)$ represent the (auto) power spectral densities corresponding to the stationary ambient noise signal $d(k)$ and residual ambient noise $e(k)$, respectively, given as a function of the frequency f. The term $C_{d,x}(f)$ is denoted as the *magnitude squared coherence* (MSC) [12] related to the signals $d(k)$ and $x(k)$.

B. Causality Aspects

In digital realizations of ANC headsets, the *ANC circuitry* block in Figure 1 involves AD and DA converters which introduce a signal delay. Given T_{AD} as the delay caused by the AD and T_{DA} as the delay of the DA converter, the overall delay ($T_{AD} + T_{DA}$) has a significant impact on the *system causality* in feed forward ANC: If the signal $y'(k)$ arrives at the error microphone later than the signal $d(k)$, the ANC performance is significantly degraded. In this case only periodic noise components can be canceled. The impact of these very practical aspects has been mostly neglected in the literature but will be investigated based on the proposed system.

III. Simulations of ANC Headsets

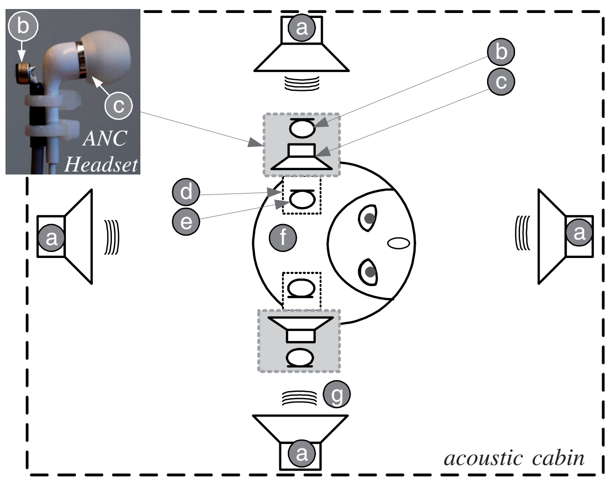

Fig. 2. Hardware setup for simulations of ANC headsets.

Basis for all simulations is the hardware setup shown in Figure 2: Four loudspeakers (a) are positioned around an artificial head (f) with a prototype ANC headset plugged into both ears. The prototype ANC headset is built of a commercial stereo in-ear headset including the loudspeakers (c) for the left and the right side. Reference microphones (b) are attached to the back of the left and the right side of the headset as shown by the photo in the left part of the figure. The headset is plugged into the ear-canals (d) of the artificial head, and the ear-drum microphone (e) of the artificial head acts as the error microphone. All audio components are located inside an audio cabin to protect all simulations against external acoustic influences and connected to a high quality soundcard (RME Multiface II) of a standard PC. With the described hardware setup, feed forward ANC could also be realized in real-time. Due to the high input-output audio signal delay, however, this real-time realization would not enable noise canceling of non-periodic ambient noise. Therefore, we propose the following alternative simulation procedure.

A. Simulation Procedure

During the simulation, ambient noise signals (g) are emitted by all four loudspeakers (a) in order to create a diffuse ambient noise field (similar to the standard [13]). The employed high quality soundcard enables *sample exact synchronous playback and recording* of audio signals. Therefore, the ambient noise can be considered as *reproducible*. Taking advantage of this, our approach enables to simulate very low or even zero signal delay systems in a *two-step procedure* as follows:

In the first step, the ambient noise field is produced by the four surrounding loudspeakers. Both, the signals $x(k)$ and $d(k)$ are recorded by the reference and the error microphone synchronously and stored to the harddrive.

In the second step, the exact same noise field is reproduced by emitting the same noise signals via the four loudspeakers. Due to the recording of the signals $x(k)$ and $e(k)$ in the first step, the signal $d(k)$ is known in advance. Therefore, a *pre-computed* anti-noise signal can be emitted simultaneously by the loudspeaker of the in-ear headset to arrive at the error microphone *in time*. The input-output system delay can be compensated by time-shifting the signal recorded in the first step before playback in the second step.

The anti-noise signal is computed in between both simulation steps and involves the ANC algorithm to be explored (e.g., the FxLMS approach). For the computation of the anti-noise, the signals $d(k)$ and $x(k)$ recorded in the first simulation step as well as models of specific acoustic paths are required. Therefore, a complete ANC simulation is composed of the following phases:

- **Phase I:** Measurement of involved hardware components and acoustic paths (calibration).
- **Phase II:** Production of an acoustic ambient noise field.
- **Phase III:** Pre-computation of the anti-noise signal.
- **Phase IV:** Actual ANC Simulation.
- **Phase V:** Post Processing.

Each simulation produces audio samples of the perceived ambient noise $d(k)$ from simulation Phase II (headset with only passive noise attenuation) and the perceived residual ambient noise $e(k)$ from simulation Phase IV (headset with passive and active noise attenuation). Feedback ANC approaches can be simulated in the post processing phase in offline manner involving signal $d(k)$ recorded in Phase II and the acoustic paths measured in Phase I. In order to enable reproducible synchronous playback and recording of audio signals (this feature is not supported by Matlab), the `RTProc` real-time prototyping system [14] is operated in the background of Matlab.

B. Evaluation of the Simulation

During and after each simulation, a sequence of signal processing tasks is conducted for the evaluation of the measured data:

- Determination of the acoustic properties of all involved hardware components and in particular the system delay.
- Modeling of all involved acoustic paths based on long linear (FIR) filters. Most prominent and important example for an acoustic path to be modeled is the secondary path (headset loudspeaker to error microphone) required for the FxLMS approach.
- Evaluation of the measurements based on the MSC (3) to determine the maximum theoretical performance.
- Simulation of a specific algorithm (e.g, the FxLMS) based on the recorded signals $d(k)$ and $x(k)$ to compute $y(k)$.

All acoustic measurements involve a novel excitation signal type proposed in [15] in order to minimize the influence of undesired measurement noise.

IV. Simulation Results

Simulations have been conducted based on different types of ambient noise signals with a sampling rate of $f_s = 48$ kHz. The ambient noise signals have a duration of 10 seconds. Exemplary results for the prototype headset from Figure 2 will be presented in the following.

A. Measured MSC

The maximum theoretically attainable ANC noise attenuation is given in (3) and is a function of the MSC. Given the recorded signals $x(k)$ and $d(k)$, the MSC was evaluated and is shown in Figure 3 as a function of the frequency $0 \leq f < 16$ kHz. The MSC measure reflects physical constraints given

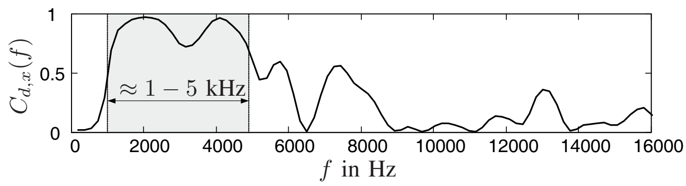

Fig. 3. Measured MSC based of the signals $x(k)$ and $d(k)$. A high ANC performance can be expected in the frequency areas between approximately 1 and 5 kHz.

by the simulation setup. The diagram shown in Figure 3 indicates that with the investigated hardware setup high ANC performance can be expected only in the range between 1 and 5 kHz (area with high MSC values). Surprisingly, no attenuation seems to be possible below 1 kHz. It could be shown, however, that the low MSC values below 1 kHz are due to the acoustic closure of the ear canal by the in-ear headset.

B. Possible Noise Attenuation

Noise attenuation curves based on the recording of the signals $d(k)$ and $e(k)$ are shown in three subplots in Figure 4. The three subplots have been produced for different simulation setups and hence system delays $(T_{\mathrm{AD}} + T_{\mathrm{DA}})$ defined as follows:

- Setup A: $(T_{\mathrm{AD}} + T_{\mathrm{DA}}) = 0.104$ ms (or 5 samples)
- Setup B: $(T_{\mathrm{AD}} + T_{\mathrm{DA}}) = 0$ ms (or 0 samples)

- Setup C: $(T_{\mathrm{AD}} + T_{\mathrm{DA}}) = -0.125$ ms (or -6 samples)

Note that also a negative delay (in this case reference signal $x(k)$ is preponed) has been simulated which can not be realized in practice but should be considered an edgework experiment. Independent low-pass filtered Gaussian noise signals were used for the production of the ambient noise.

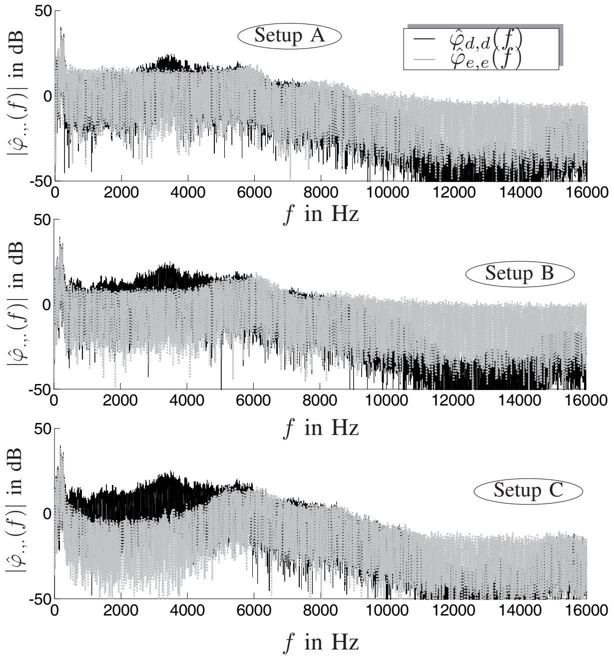

Fig. 4. Measured MSC based on measurements of signals $x(k)$ and $d(k)$.

For the evaluation of the recorded signals, approximations $\hat{\varphi}_{e,e}(f)$ and $\hat{\varphi}_{d,d}(f)$ of the power spectral density were computed based on the magnitude spectra for $e(k)$ and $d(k)$, respectively. In the figure, the estimation of the power spectra $\hat{\varphi}_{d,d}(f)$ and $\hat{\varphi}_{e,e}(f)$ are shown in black color in the background and light gray color in the foreground, respectively. Frequency regions in which the ambient noise is attenuated due to ANC are regions in which the black curve in the background is visible.

In all simulations, one fixed set of optimal filter w_i were computed in simulation Phase III following the MMSE approach (the ambient noise was assumed to be stationary). Very similar results have been produced for the simulation of ANC headsets involving the FxLMS approach.

As a conclusion of the three curves, it is obvious that feed forward ANC does not work well for the prototype in-ear ANC device in case of a system delay of 5 samples (Setup A) since only a small portion of the ambient noise is canceled. The performance is only slightly higher in case of the zero-delay system (Setup B). Really significant noise canceling (> 20 dB) could only be observed for Setup C. In compliance with the measured MSC the impact of the noise canceling is limited to the frequency area between 1 and 5 kHz. Setup C, however, can not be realized in practice since it would require that the ambient noise would be recorded 6 samples before arrival at the reference microphone. Clearly, with the investigated ANC prototype headset, the anti-noise arrives too late for efficient noise canceling, even if the ANC device would be realized with zero system delay.

V. Conclusion

In this contribution, a novel approach for the investigation of algorithms for ANC headsets was proposed. This approach, on the one side, enables to develop new algorithms in a convenient and cost-effective way in Matlab. On the other hand, it enables the exploration of the influence of very practical aspects yielding results which are of highly practical relevance. Key element of the new approach is a simulation procedure in which a test setup involving a prototype ANC headset is exposed to a reproducible ambient noise field. Due to the reproducibility of the noise, systems with arbitrary signal delays can be simulated in a realistic way.

In order to demonstrate the benefit of the new approach, example results were presented which show that only very limited performance can be expected by feed forward ANC for the investigated prototype in-ear headset device due to the late arrival of the anti-noise.

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