

Active Occlusion Cancellation with Hear-Through Equalization for Headphones

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Abstract—Occluding the ear canal alters the perception of the own voice. This is called the occlusion effect. Low frequency components of the own voice are amplified and high frequency components are attenuated. In this contribution, we extend our previously published Active Occlusion Cancellation (AOC) system for headphones and hearing aids. It uses the principle of Active Noise Cancellation (ANC) to enhance the perception of the own voice for an occluded ear canal. We combine robust feedback controller design with an adaptive mechanism to attenuate the low frequency components of the own voice. Furthermore, we include an equalizing hear-through filter to enhance high frequency components. The AOC system is evaluated in a hearing test with an in-ear headphone connected to a real-time processing system. Objective and subjective metrics show a clear improvement of the perception of the own voice. The occlusion effect is significantly reduced by the system.

I. INTRODUCTION

For hearing devices the focus often is on *listening* to outer sounds. However, an important factor for their acceptance is the own voice perception while *talking*. When the ears are occluded by the hearing devices, the own voice is often described to sound hollow or like talking in a barrel [1]. This *Occlusion Effect* (OE) is characterized by an amplification of low frequency components and an attenuation of high frequency components of the own voice. For hearing aids, the own-voice perception is typically improved by passive approaches, such as ventilation openings or deep insertion [2]. However, they create problems such as increased acoustic feedback or reduced comfort. As no solution free of drawbacks exists, the occlusion effect remains an open problem. Devices with integrated signal processing, including hearing aids, in-ear monitoring or communication headsets, allow for active enhancements relying on the principle of active noise cancellation (ANC). They use anti-phase compensation waves to cancel sound. In the recent literature, some attempts have been made using time-invariant feedback controllers e.g. [3], [4], [5], [6] or adaptive time-variant approaches e.g. [7], [8] for active occlusion cancellation (AOC). In a previous publication we proposed a time-invariant controller based on mixed sensitivity \mathcal{H}_∞ optimization that explicitly considers the OE characteristics, especially for low frequencies [9]. However, it considered time-varying acoustic conditions and the variability of speech only by a controller that is robust against changes and did not adjust to them. In this paper, we enhance the previously proposed solution by a combination of a time-invariant feedback controller with an adaptive approach as well as an equalizing hear-through filter. The significantly improved quality and acceptance of the proposed system is verified through hearing tests.

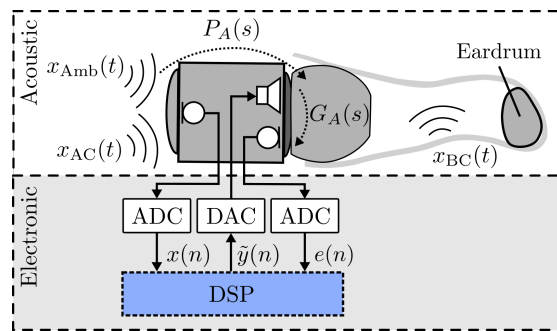


Fig. 1. Acoustic front-end with air-conducted and bone-conducted components of the own voice, $x_{AC}(t)$ and $x_{BC}(t)$ connected to the electronic front-end.

II. OCCLUSION EFFECT

The own voice is perceived through two different transmission paths originating from the voice excitation. The first is *air-conducted* (AC) sound $x_{AC}(t)$, also perceived by other listeners. The second component is *bone-conducted* (BC) sound $x_{BC}(t)$ that is only audible to the talker. The state of *talking*, involving $x_{AC}(t)$ and $x_{BC}(t)$, needs to be distinguished from *listening*, which includes only the ambient sound $x_{Amb}(t)$. These different acoustic signals are visualised as wavefronts in Fig. 1. While talking with occluded ears, the AC sound is attenuated by the occlusion and the BC sound in the ear canal, which is dominated by low frequencies, is amplified [10], [11]. The occlusion results in a strong amplification of low frequencies and an attenuation of high frequencies, due to the resonance characteristics of the occluded ears. The goal is to recreate a natural perception of the own voice.

The occlusion effect can be quantified by the ratio between the occluded and the open ear canal sound pressures, $e(n)|_{occl}$ and $e(n)|_{open}$, where $|_{occl}$ and $|_{open}$ indicates the current circumstances. However, for measurements a precise repeated voice reproduction of the same sound with and without occlusion is not possible. Thus, the effect is typically quantified by relating the sound pressures inside and outside the occluded ear canal, represented by the inner and outer microphone signals $e(n)|_{occl}$ and $x(n)|_{occl}$. We call the spectral domain transfer characteristic the measurable occlusion function $\widetilde{OE}(z)$.

$$\widetilde{OE}(z) = \frac{|E(z)|_{occl}}{|X(z)|_{occl}}. \quad (1)$$

¹ Note that the inner signal $e(n)|_{occl}$ contains the attenuated AC sound $x_{AC}(t)$ and the BC sound $x_{BC}(t)$ while speaking.

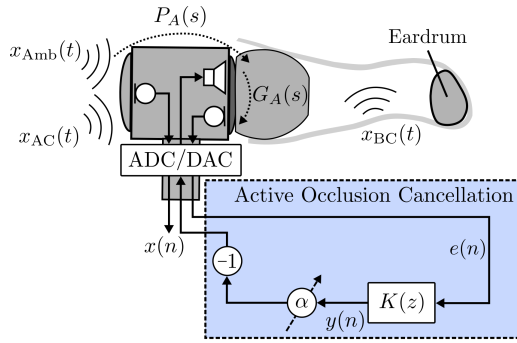


Fig. 2. Model and structure of the Active Occlusion Cancellation (AOC) System.

III. SYSTEM DESIGN

The structure of the Active Occlusion Cancellation (AOC) system is illustrated in Fig. 1. It consists of an acoustic front-end, in our case an in-ear headphone, and an electronic front-end including analog-digital (ADC), digital-analog conversion (DAC) and a digital signal processing unit. In the acoustic front-end we are typically dealing with continuous time t , or system representations in the continuous s -domain¹. In the digital domain, after the ADC, we deal with discrete time n and systems are described in the discrete z -domain.

The acoustic front-end is realized by an in-ear headphone, which contains not only a loudspeaker, but also two microphones. One microphone faces the outside to record ambient sound, the so-called reference signal $x(n)$, and one microphone faces the eardrum to capture the in-ear sound, the error signal $e(n)$. The transfer function between the outer and the inner microphone is called the *primary path*, which is digitally modeled by $P(z)$. It contains the acoustic primary path $P_A(s)$, the microphone characteristics, as well as the AD-conversions. The transfer function between the loudspeaker and the inner microphone is the *secondary path*, modeled by $G(z)$. Similarly, the model combines the acoustic secondary path $G_A(s)$, the microphone and loudspeaker characteristics, as well as the AD- and DA-conversion. The acoustic feedback path from the loudspeaker to the outer microphone is neglected, as measurements verified sufficient attenuation due to closed fitting of the hearing device.

The signal processing of the electronic front-end comprises two parts: the AOC to attenuate the amplified bone-conducted components, as introduced in Fig. 2, and a hear-through to restore the air-conducted components that have been blocked by the occlusion, added to the system in Fig. 3. The feedback loop is implemented as a time-invariant *controller* $K(z)$ scaled with an adaptive factor α . Its purpose is to calculate a cancellation signal $y(n)$ that cancels the unwanted signal portions within the ear canal.

As illustrated in Fig. 3, the hear-through preprocesses the reference signal $x(n)$ through an FIR filter $V(z)$ to create the hear-through signal. It can optionally be mixed with a desired

¹For comprehensibility we use the same names for discrete-time (n) and continuous-time (t) variables e.g. $e(t)$ and $e(n) = e(nT)$ or $G(s) = \mathcal{L}\{g(t)\}$ and $G(z) = \mathcal{Z}\{g(n)\}$, where $z = e^{sT}$.

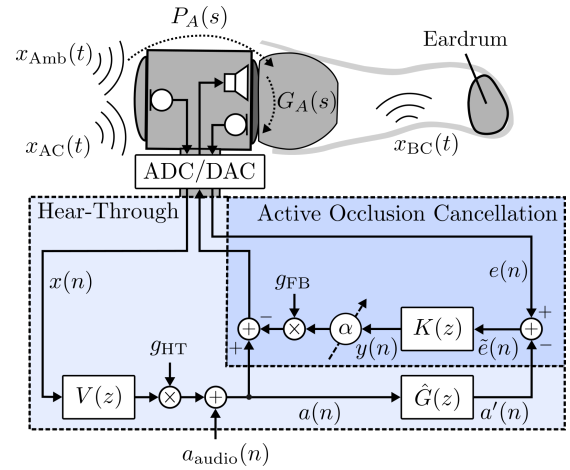


Fig. 3. Model and structure of the Active Occlusion Cancellation (AOC) System with equalized hear-through and optimal audio signal $a_{audio}(n)$.

audio signal $a_{audio}(n)$, e.g. in a communication headset, to create the additional signal $a(n)$. In order to keep $a(n)$ mostly free from the influence of the feedback loop, it is filtered by an estimate of the secondary path $\hat{G}(z)$ and subtracted from the error signal $e(n)$. This yields the corrected error signal $\tilde{e}(n)$ which is then fed into the controller $K(z)$. If $\hat{G}(z) = G(z)$, the hear-through is not altered by the feedback loop.

For the hearing test, the system offers two gains g_{FB} and g_{HT} which can be manually adjusted by the participants according to their sound preferences. g_{FB} tunes the feedback loop and g_{HT} scales the loudness of the hear-through signal $a(n)$.

A. Controller Design

The controller $K(z)$ is designed using mixed-sensitivity \mathcal{H}_∞ optimization as presented in [9]. It requires a continuous model of the nominal secondary path $G(s)$ as well as specifications of the performance and stability requirements in the s -domain. The requirements are modeled as frequency-dependent weighting functions. The performance requirements define the design goal for the closed loop transfer function of the feedback loop without the adaptive factor ($\alpha = 1$). Following Fig. 2 this closed loop transfer function, also known as sensitivity $S(z)$, yields:

$$S(z) = \frac{E(z)|_{\text{AOC on}}}{E(z)|_{\text{AOC off}}} = \frac{1}{1 + \alpha \cdot G(z)K(z)}. \quad (2)$$

The performance requirements are chosen based on the measured occlusion function $\widetilde{OE}(z)$ as presented in [9].

The stability requirements take the uncertainty of the secondary path $G(s)$ into account. We used the variations of $G(s)$ for 11 different persons wearing the in-ear headphone. Furthermore, it contains the free field case, which is critical w.r.t. stability while handling the headphone.

The sensitivity of the designed controller $S(z)$ is visualized in Fig. 4 as the case where $\alpha = 1$ (—). For the nominal secondary path it has a gain margin of 11.4 dB and a phase margin of 83.7°.

B. Adaptive Factor

Robust controller design considers all possible scenarios that may occur and designs a controller that is stable for all of these

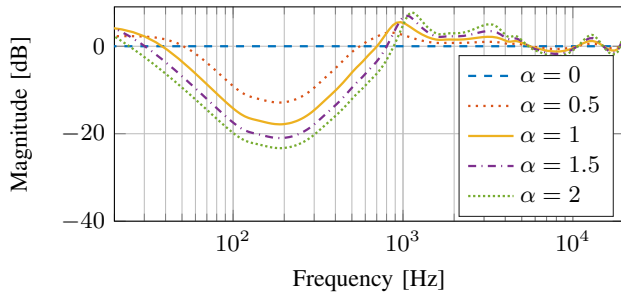


Fig. 4. Influence of the adaptive factor α on the sensitivity $S(z)$.

cases. To enhance the controller performance, we propose to focus on the *common* use cases in the controller design and handle the extreme cases with the adaptive factor α .

The idea of the adaptive factor α is to monitor the correlation between the error signal $e(n)$ and the auxiliary signal $y'(n) = \hat{g}(n) * y(n)$, calculated in the background, as the error signal $e(n)$ is dominated by the controller output $y(n)$ for an unstable system. We use a first order IIR filter with the smoothing parameter β to estimate the cross-correlation with lag zero:

$$\hat{\varphi}_{ey'}(n) = \beta \cdot \hat{\varphi}_{ey'}(n-1) + (1-\beta) \cdot e(n) \cdot y'(n). \quad (3)$$

Furthermore, we normalize the cross-correlation $\hat{\varphi}_{ey'}(n)$ by the similarly estimated autocorrelations $\hat{\varphi}_{ee}(n)$ and $\hat{\varphi}_{y'y'}(n)$ and calculate the adaptive factor

$$\alpha = 1 + \frac{\hat{\varphi}_{ey'}(n)}{\sqrt{\hat{\varphi}_{ee}(n)\hat{\varphi}_{y'y'}(n)}} = 1 + \hat{\gamma}_{ey'}(n). \quad (4)$$

The influence on the feedback sensitivity $S(z)$ are shown in Fig. 4. Investigations have shown improved performance vs stability trade-off, however, a detailed examination is beyond the scope of this paper.

C. Hear-Through Equalization

For realizing a natural perception of the own voice, we also need to amplify the air-conducted component which is passively attenuated by the earpiece, described by the primary path $P(z)$, and actively attenuated by the feedback loop, with the sensitivity $S(z)$. Inside the ear, we would perceive $x_{AC}(t)$ filtered by $P(z)S(z)$. Thus, for a transparent perception of the air-conducted sound we need to add the missing part of $x_{AC}(t)$, which is $x_{AC}(t)$ filtered by $1 - P(z)S(z)$. In order to create the desired signal within the ear canal, the characteristic of the loudspeaker needs to be compensated for. We approximate $G_{spk}(z) \approx \hat{G}(z)$ for the hear-through filter design. By doing this, we assume a flat microphone characteristic and define the following design goal for the hear-through filter $V(z)$:

$$V(z) = \frac{1 - P(z)S(z)}{\hat{G}(z)}. \quad (5)$$

For this design target we assume the adaptive factor to be $\alpha = 1$. We need to consider that the secondary path $G(z)$ is an acoustic path and thus non-minimum phase. Therefore it may not easily be invertible. To nevertheless achieve a flat overall magnitude response, we allow a constant delay of the system. We are thus replacing $1 - P(z)S(z)$ by $z^{-10} - P(z)S(z)$ with an experimentally determined delay of 10 samples at

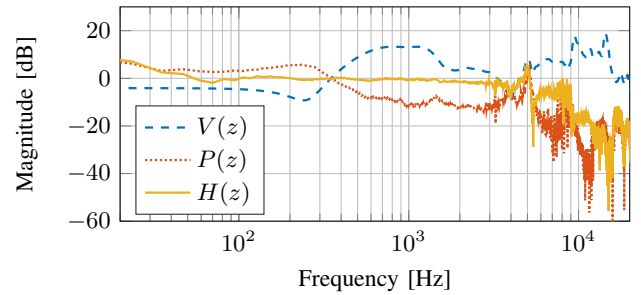


Fig. 5. Characteristics and influence of the hear-through filter $V(z)$.

a samplerate of $f_s = 48$ kHz. Furthermore, we implement $V(z)$ as an FIR filter as a causal approximation of the ideal filter. In order to determine the optimal filter in the minimum mean-square error sense, we are using the FIR-solution of the Wiener-Hopf-Equation [12]:

$$v = \Psi_{\hat{g},\hat{g}}^{-1} \cdot \varphi_{\xi,\hat{g}}, \quad (6)$$

with $\Psi_{\hat{g},\hat{g}}$ being the auto-correlation matrix for a filter $\hat{g}(n)$ and $\varphi_{\xi,\hat{g}}$ being the cross-correlation vector between $\xi(n) = \delta(n-10) - p(n) * s(n)$ and $\hat{g}(n)$. v is the impulse response of the hear-through filter $V(z)$ in vector form. In a last step, we apply a lowpass filter with a cutoff frequency of $f_g = 5$ kHz to $V(z)$ to prevent too large amplification of noise at high frequencies. In Fig. 5, we can see the final design of the hear-through filter $V(z)$, the primary path $P(z)$ as well as the overall transfer function of the whole system, including feedback loop and hear-through filter, which is

$$H(z) = P(z)S(z) + V(z)G(z), \quad (7)$$

for $\hat{G}(z) = G(z)$. The flat characteristic of the overall transfer function $H(z)$ extends up to roughly $f = 4$ kHz. This means that sounds from the outside are mostly unaltered in this frequency range.

IV. EVALUATION

We used a *Bose QC 20* in-ear headphone without the ANC electronics [13] as our acoustic front-end. It was connected to a *dSPACE DS1005* real-time system with the *DS2004* and *DS2102* extension boards. Excluding the acoustics, the DSP system has a round trip delay of 1 sample at a sampling rate of $f_s = 48$ kHz. We chose $\beta = 0.999$ for the smoothing factor in Eq. (3).

To evaluate the subjective performance of the occlusion cancellation system, we conducted a full factorial paired comparison scaling test in an acoustic booth (*STUDIOBOX Premium*) without ambient noise ($x_{Amb}(t) \approx 0$). The test covers the situation of *talking*, not listening. It's design and evaluation principles were previously described in [3].

		Hear-Through	
		off	on
AOC	off	A	B
	on	C	D

TABLE I
CONDITIONS EVALUATED IN PART (A1) OF THE HEARING TEST.

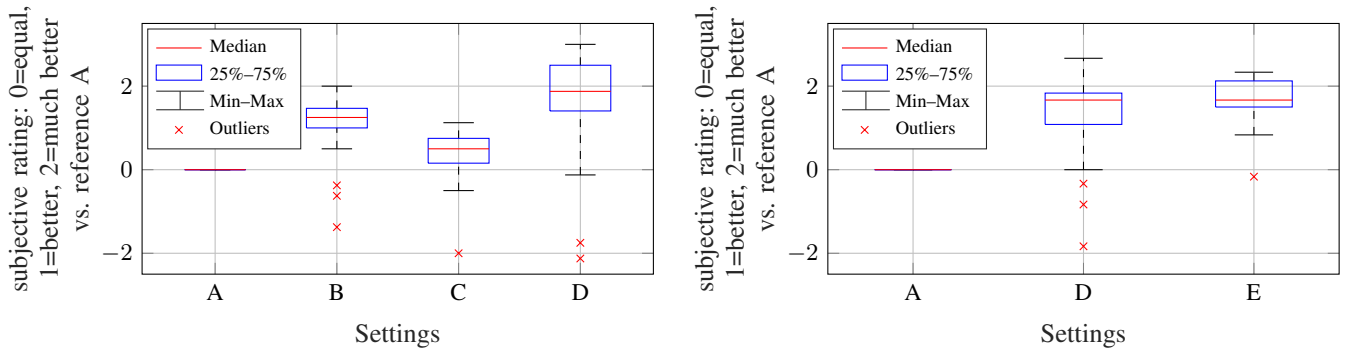


Fig. 6. Scores from hearing test part 1 with default settings A, B, C and D (left), and subsequently conducted part 2 including individual tuning E (right).

Test part 1: Comparing four predefined settings (see Table I). The 23 participants (normal-hearing, 20 male, 3 female) were presented with all possible pairs of settings in randomized order and had to compare them while reading three test sentences (sa1, sx32, and sx198 from the TIMIT acoustic-phonetic speech corpus [14]) aloud. For every comparison, we asked them to rate the difference in own-voice naturalness between the two settings on a 5-point Likert scale going from -2 to $+2$. This yielded an over-determined set of 12 linear equations (i.e. 12 partially redundant answers) for each participant. We chose setting A, i.e. AOC and HT off, as our reference with a score of zero. Then, we determined scores for B, C, and D by solving the set of linear equations by means of least squares [3].

Test part 2: Individual tuning of g_{HT} and g_{FB} .

The participants had to adjust their own-voice sound using sliders for the hear-through gain g_{HT} and feedback loop gain g_{FB} until they found their preferred configuration. The resulting new setting E was then compared to settings A and D using the same hearing test procedure as before.

Test part 3: Measurements of the occlusion function $\widetilde{OE}(z)$. To obtain objective OE measurements, we asked the participants to read the test sentences aloud once under each of the five settings. We calculated $\widetilde{OE}(z)$ as defined in Eq. (1) using the headphone's built-in microphones, and obtained an averaged version of the occlusion function.

A. Subjective Ratings

Box plots of the scores obtained in the two subsequently conducted parts of the hearing test are shown in Fig. 6. As expected, the hear-through setting B was rated to significantly

(p-value $p < 0.000005$ [15]) improve naturalness of the own voice (mean rating increase $\mu_{\Delta r} = 1.02$). Additionally turning on active cancellation in setting D resulted in another significant improvement over the hear-through-only state ($p < 0.008$, $\mu_{\Delta r} = 0.58$). Notable is that the hear-through alone (setting B) scored higher than the AOC system alone (setting C). Finally in part 2, letting the subjects come up with their own setting E led to a third significant, albeit smaller, rating increase ($p < 0.027$, $\mu_{\Delta r} = 0.38$), (cmp. Fig. 6 right).

We identified two reasons for the outliers seen in both plots. First, some participants reportedly disliked the hear-through because of audible microphone and quantization noise. The latter comes from the limited bit depth of the ADCs in the *dSPACE* system (16 bits), rendered more noticeable by the high-frequency boost of the hear-through filter. Second, the control loop's gain margin at the default settings of $g_{FB} = 1$ and $0 \leq \alpha \leq 2$ turned out to be too large for two of the subjects. When they spoke, they heard signs of beginning instability, i.e. high-frequency ringing.

B. Objective Measurement Results

As an example, Fig. 7 shows $\widetilde{OE}(z)$ for the left ear of subject 5 with the settings A (system completely off) and E (system individually tuned by subject). Under setting A (---) the amplification of low and the attenuation of high frequency components is clearly visible. Looking at setting E (—), we can see that the transfer function has a flatter character. The low-frequency amplification and the high-frequency attenuation have been compensated for. A flatter magnitude spectrum of $\widetilde{OE}(z)$ seems to correlate with a high rating in the subjective hearing test, which needs further investigation in future work.

V. CONCLUSION

We presented an enhanced approach for the active cancellation of the occlusion effect. It is based on the combination of a time-invariant feedback controller designed using mixed-sensitivity \mathcal{H}_∞ optimization with a novel adaptive scaling factor and an equalized hear-through filter. The adaptive factor operates based on the correlation between the filtered controller output $y'(n)$ and the error signal $e(n)$. A hearing test with 23 participants using a real-time system with a *Bose QC 20* as the acoustic front-end, shows a major improvement of the perception of the own voice and significant reduction of the occlusion effect in the situation of *talking*. Overall the participants reported a clear, transparent own voice.

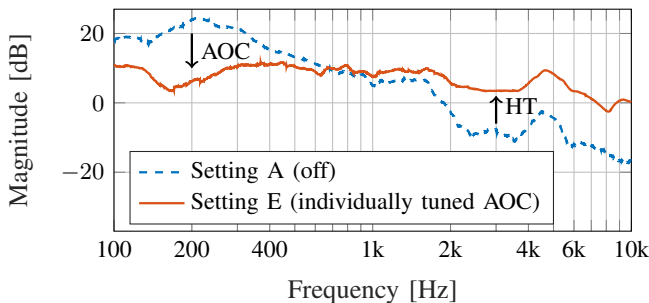


Fig. 7. Occlusion functions $\widetilde{OE}(z)$ from part 3 for the left ear of subject 5 (data for setting E: rating: 1.83, $g_{FB} = 0.67$, $g_{HT} = 2.01$) with $\frac{1}{3}$ -octave ⁴ band smoothing.

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