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Combined feedback and adaptive feedforward active noise control in headsets

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ABSTRACT

In active control of acoustical noise, the combination of feedback and adaptive feedforward control enables broadband noise reduction. A simple combination strategy, which is based on an internal model feedback controller and an adaptive "filtered-x least mean square" feedforward filter, is commonly used for this purpose. Although this method enables saving of memory and calculating resources, it fails in terms of stability when the secondary path is subject to variations. Especially in headset applications varying leakage of the ear cup leads to instability in the internal model controller loop. To overcome this problem we propose an alternative combination method of a standard feedback controller and an adaptive feedforward filter in this paper. The feedback controller was developed to be stable under all leakage conditions. The combination strategy consists firstly in tuning each controller according to the aimed frequency range. Secondly the binding structure is realised to enable both controllers to act simultaneously in a constructive way. The proposed method was implemented for an active noise control headset using a DSP platform and proved stability and broadband noise reduction under different ear cup leakage conditions.

1 INTRODUCTION

Today commercially available headsets with Active Noise Control (ANC) are usually based on non-adaptive, analogue, and mostly feedback control techniques. With the ever-growing performance of digital signal processing (DSP) devices it became possible to implement adaptive controller design algorithms for ANC applications at a reasonable expense-benefit ratio and the digital signal processing has increasingly been used by researchers in this domain. Numerous publications such as [1][2][3] describe different active noise controller structures and optimisation algorithms by use of either feedback or feedforward strategies. Especially in headsets the simultaneous use of both control strategies can be of great benefit [1]. Some papers [4][5] report the outcome of combining feedforward

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and feedback control strategies. In this case, the Internal Model Control (IMC) is applied in the feedback control loop in order to make the two combined control algorithms work independently from each other, thus avoiding any interfering interaction. Under laboratory conditions this way of combining the two controllers provides rather good performance, but on closer examination some problems occur when tested under realistic conditions on a users head. In the following, the difficulties of the internal model control in a realistic application are discussed and a new combination method of a feedback and an adaptive feedforward controller is presented which overcomes this problem.

2 PROBLEM STATEMENT

2.1 Feedback versus Feedforward Noise Control

Active noise control systems attempt to neutralize undesired acoustic noise via some appropriate anti-noise signal generated by a secondary sound source (the speaker inside the ear cup). The noise cancellation takes effect in a specified area, where an error microphone is located.

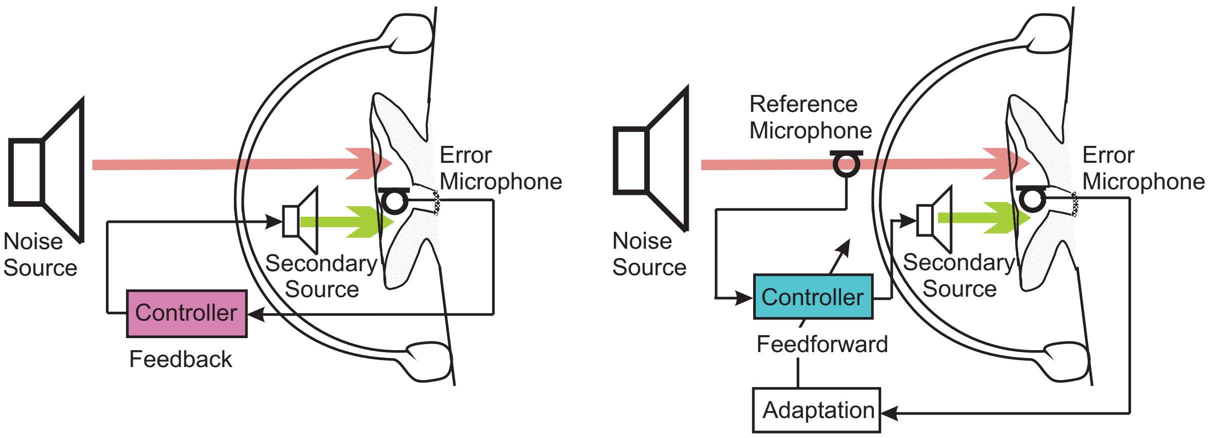

Figure 1: Feedback (left) and adaptive feedforward (right) active noise control in a headset.

Figure 1 presents the two main control strategies: Feedback and feedforward. In a feedback structure the superposition of the noise and the anti-noise at the error microphone is fed back to the controller. A feedforward system in contrast attempts to create an appropriate anti-noise signal by using an "upstream" signal from the reference microphone mounted on the outside of the ear cup, and uses this as input for the controller.

The feedback controller exhibits the advantage of reaching acceptable noise reduction performance at a minor cost, since automatic feedback control can effectively be realised non-adaptive with few analogue electrical components. Feedback control of noise however suffers from the plant latency mainly caused by the acoustical transfer path from the secondary source (the control loudspeaker) to the error microphone. The corresponding transfer function includes a delay, caused by the duration, the anti-noise needs to propagate from the loudspeaker to the error microphone with sound velocity. As known from the control theory this results in a substantial phase drop at high frequencies and hence in a decrease of the phase margin. Consequently, in feedback systems the ANC bandwidth is limited to low frequencies and in existing circum-aural ANC headsets no significant active noise reduction is achieved above 400 Hz.

Feedforward strategies overcome the latency problem by using the "upstream" reference signal of the outside microphone, which is intended to give a sufficiently advanced and coherent indication of the approaching noise. Thus, an adequately designed feedforward system can accomplish a higher frequency range for the ANC. This is a clear advantage of the feedforward strategy. Currently, non-adaptive feedforward control strategies are implemented in some commercial headsets such as the *HMEC 45* from *Sennheiser electronic*, which offers an active noise reduction up to 1.5 kHz.

So far, feedforward ANC in commercial products is limited to supra-aural "open" ear cups because of two acoustical properties: First, open ear cups offer only a minimum amount of passive attenuation on the primary path (acoustical transfer function between the outside and the eardrum). Therefore this transfer function shows only minor variations caused by different fits of the ear cup due to head anatomy. Second, the secondary path (acoustical transfer function from the inside speaker to the eardrum) also shows only minimum variations. In contrast to those open ear cups, primary and secondary paths in circum-aural closed ear cups vary significantly depending on the user's head anatomy and fitting pressure. For this reason, it is not possible to state a single filter transfer function for a feedforward controller, that grants considerable attenuation under the variable conditions of a closed headset. Nevertheless, the most effective high frequency noise attenuation can be achieved by the passive attenuation of closed circum-aural ear cups.

2.2 Potential for adaptive Solutions

To yield the profit of the combination of a feedforward noise control and the good passive attenuation of closed ear cups, the feedforward controller needs to be made adaptive to both acoustical path changes. Adaptive algorithms are therefore used to adapt the controller parameters, generally for the purpose of minimizing the power of the error signal. Through this principle, adaptive control is able to focus on the reduction of any dominant frequency band in the acoustical disturbing signal. This ability constitutes a second advantage of an adaptive feedforward controller.

As feedforward control systems provide only limited performance at low frequencies, an expedient arrangement is to assign a part of the cancelling task to a feedback controller. This can be accomplished by a combination of non-adaptive feedback and adaptive feedforward control, in which the adaptive feedforward component is intended to cancel high frequencies and to focus on specific dominant noise, while the feedback component is designed to cancel only low frequency noise. For this, the combination strategy must fulfil the condition that each of the two controllers works independently in its dedicated frequency band, and that any undesired interference between the two control algorithms is avoided.

3 STATE OF THE ART

3.1 Adaptive Feedforward Control

Figure 2 shows an active headset ear cup and the signal processing scheme of an adaptive feedforward controller. The reference signal $x(n)$ picked up by the outer reference microphone is passed to the adaptive Filter $W(z)$. This generates the actuating variable $y(n)$, which is the input for the plant $S(z)$. When the control loudspeaker is activated with $y(n)$, the plant reacts with the anti-noise signal $u(n)$. Inside the ear cup a superposition takes place of $u(n)$ with the disturbance $d(n)$, which arises from the primary noise source outside the headset propagated through the ear cup. The result of this superposition is the error signal $e(n)$, which is picked up by the inner error microphone. The adaptive feedforward controller $W(z)$ is designed as an FIR-filter (Finite Impulse Response) and is adapted by the well known "Filtered-x Least Mean Square" (FxLMS) algorithm [1].

Within this algorithm, the reference signal $x(n)$ is simultaneously filtered by the mathematical plant model $\hat{S}(z)$, producing the "filtered reference" signal $x'(n)$. $\hat{S}(z)$ is a representation of the secondary path $S(z)$, which describes the transfer behaviour from the controller output $y(n)$ to the sensor signal $e(n)$, comprising, besides the acoustical, all the electrical analogue as well as digital effects.

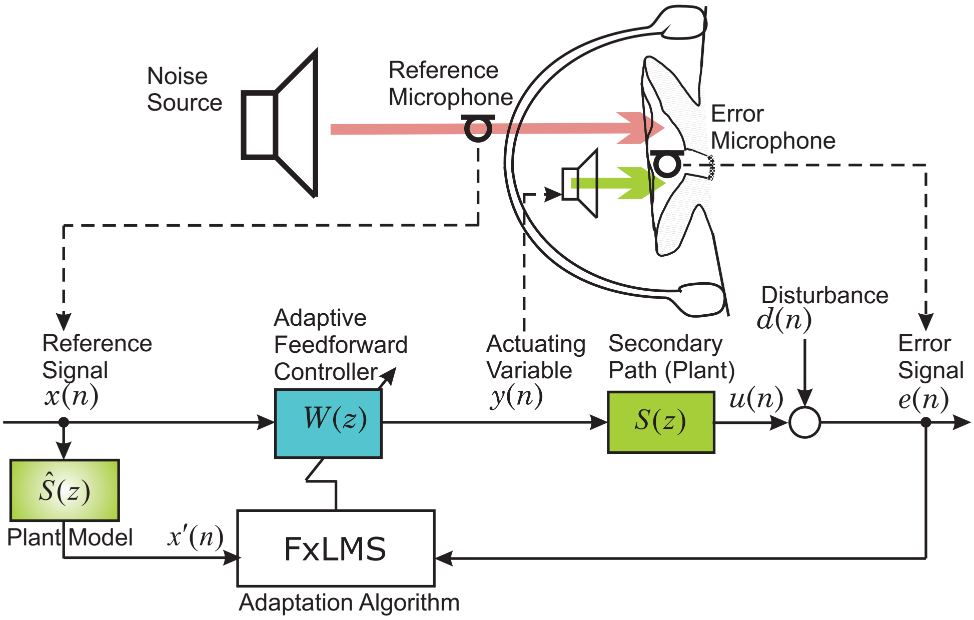

Figure 2: Adaptive feedforward active noise control with FxLMS.

The filtered reference $x'(n)$ is than used to adapt the parameters of the FIR-filter $W(z)$ according to the update equation

$$\vec{w}(n+1)=\vec{w}(n)+\mu\cdot e(n)\cdot\vec{x}'(n) \tag{1}$$

with

$$\vec{x}'(n)=\left[x'(n) \quad x'(n-1)\ldots \quad x'(n-L+1)\right]^T \tag{2}$$

and

$$x'(n)=\hat{s}(n)*x(n) \tag{3}$$

where $\hat{s}(n)$ is the impulse response of the plant model $\hat{S}(z)$ in time-domain and * denotes linear convolution. The constants μ and L represent the update step and the filter length, respectively.

3.2 Feedforward/Feedback Combination with Internal Model Control

When integrating any feedback controller within the presented feedforward scheme, the feedforward output will pass through the feedback loop and undergo some modification. It is important to study this modification since, from the feedforward point of view, it leads to a changing secondary path.

In Figure 3 we consider the integration of an Internal Model Control (IMC) feedback controller within the feedforward loop. The IMC strategy is favourably applied in the feedback control loop when it is connected to an adaptive feedforward loop. The figure shows an active headset ear cup and the signal processing scheme of the IMC-feedback / adaptive feedforward combination.

The represented IMC feedback loop uses a model of the plant $\hat{S}(z)$ to calculate an estimate $\hat{u}(n)$ of the anti-noise $u(n)$. The difference between the superposition $e(n)$ and the estimated anti-noise $\hat{u}(n)$ produces an estimate $\hat{d}(n)$ of the primary disturbance noise $d(n)$, which is then fed to the IMC feedback controller $C_{IMC}(z)$. The actuating variable $y(n)$ now is a combination of the feedforward controller output $y_{fforw}(n)$ and the feedback controller output $y_{fback}(n)$. With an accurate approximation of the disturbance noise, the input for the feedback controller $\hat{d}$(n) contains no output from the feedforward path and thus the feedback loop does not react to the feedforward controller output. Accordingly, from the feedforward point of view, no changes occur to the plant transfer function caused by the presence of the IMC feedback loop. Thus, the plant model $\hat{S}(z)$ stays valid for the feedforward path and no modification is needed. Hence, this combination-strategy fulfils the condition stated above without any need for adjustment to the loops.

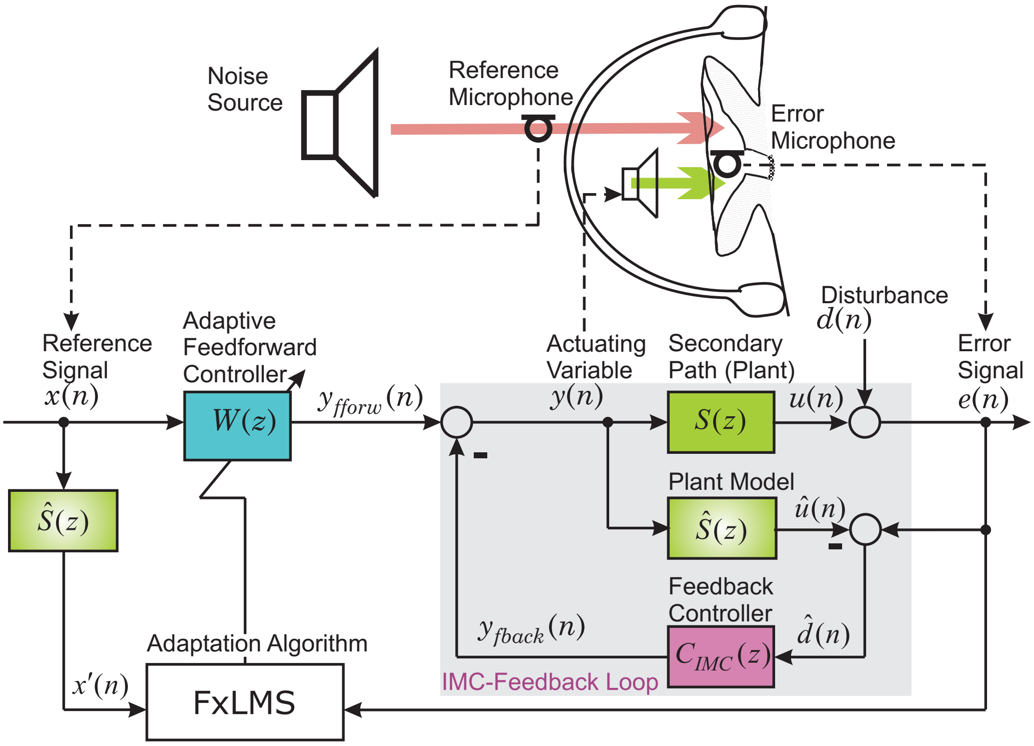

Figure 3: Internal model control feedback – adaptive feedforward combination.

Unfortunately, measurements on real subjects have demonstrated that the secondary path $S(z)$ is subject to significant variations, especially caused by the different leakage conditions between a headset and the subject's heads. This produces an estimation error which is subsequently amplified by the controller $C_{IMC}(z)$, and could destabilise the closed loop behaviour. Designing an IMC controller which avoids instability in any leakage-case, results in poor overall noise reduction performance.

In [6] the authors reported this problem and suggested to control the plant $S(z)$ with an additional auxiliary standard feedback controller which was specifically designed to make the secondary path less dependent on the leakage condition. Thus, the IMC feedback loop would be less sensitive to the subject specificity. In [7] the authors proposed another approach which consists in an online identification of the secondary path $S(z)$ via some generated incoherent test noise. Even though this approach is consistent, it is practically not reasonable, since it derogates the comfort of the listener by adding the test noise to the loudspeaker input. Moreover, the approach significantly burdens the computing resources by requiring a full online identification procedure.

4 CONTROLLER DESIGN

A standard closed loop feedback controller is more robust to plant changes due to leakage variations, as it does not contain any plant modelling in its signal processing path. From our experience we can report that a standard feedback controller is more likely to ally robustness with good noise reduction than the IMC-feedback. For this reason we used a standard feedback closed-loop control for the combination with the adaptive feedforward part.

As shown in Figure 4, the subordinate standard feedback controller $C(z)$ directly processes the error signal $e(n)$. With this approach an emphasis is put on the absence of any plant model inside the signal transmission path of the feedback controller. The absence of this uncertainty factor reduces the stability analysis complexity, and the controller design turns out to be more manageable. First, for the feedback controller design, several plant transfer functions were measured, each corresponding to a different ear cup leakage situation. Second, the measured plants were identified and finally a controller was designed to offer good noise reduction, while always remaining within the stability margins of the closed loop control for each plant-case.

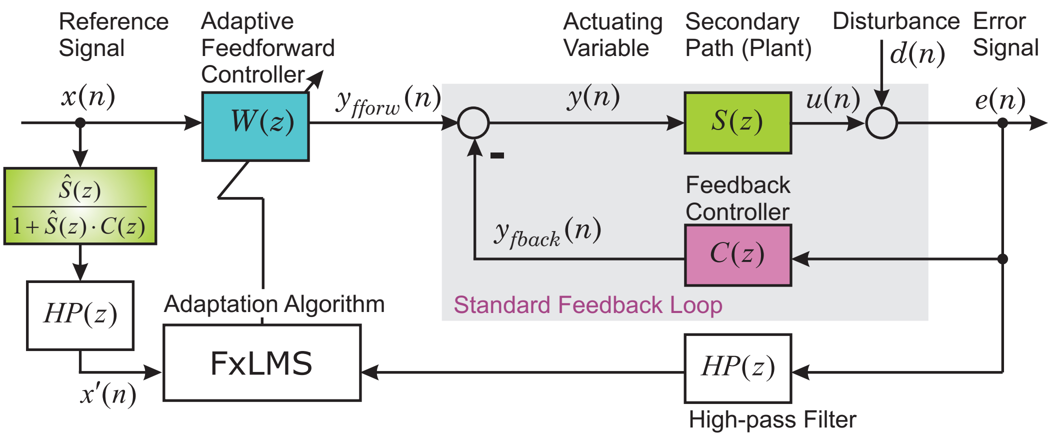

Figure 4: Standard feedback control – adaptive feedforward combination.

When a feedback loop is linked to a superordinate adaptive feedforward controller as we proposed in Figure 4, the output of the feedforward filter $y_{fforw}(n)$ fully passes through the feedback loop before flowing into the error signal $e(n)$. Thus, from the feedforward algorithm point of view, the closed loop transfer function $S(z)$ changes to

$$S^*(z) = \frac{S(z)}{1+S(z)\cdot C(z)} \tag{4}$$

$S^*(z)$ constitutes the new secondary path to the feedforward algorithm, since it is defined as the transfer behaviour from the output $y_{fforw}(n)$ of the adaptive feedforward filter $W(z)$ to the signal $e(n)$ picked up by the error microphone. Therefore, in Figure 4 the closed loop transfer behaviour is modelled by a transfer function $\hat{S}^*(z)$, which is used in the FxLMS update path to filter the reference signal $x(n)$ equivalent to the case of the separate feedforward loop stated in section 3.1. $\hat{S}^*(z)$ is the estimate for $S^*(z)$ as denoted in equation (4), but using the plant model $\hat{S}(z)$ instead of $S(z)$. Summarised, within the proposed approach, the subordinate feedback control loop, independently of the feedforward controller, always presents the same transfer function from the disturbance $d(n)$ to the error signal $e(n)$, which describes the noise reduction behaviour of the feedback controller. The output of the feedforward filter is affected by the presence of the feedback loop, but the algorithm was modified specifically to take this effect into account. This allows the proposed algorithm to fulfil the conditions set in the problem statement.

It might be criticised, that still a model $\hat{S}(z)$ of the plant is used inside the algorithm, although the modelling error was identified as a major uncertainty factor. But the benefit of this structure resides in the fact that the plant modelling is located exclusively inside the FxLMS update path, which is much less sensitive to model uncertainty than the feedback signal processing path. From the literature it is well known that the FxLMS algorithm still converges even with modelling phase errors up to 90° [1].

4.1 Frequency selective Adaptation

As we stated in the objectives of the combination, the controllers are intended to operate on different frequency ranges. While the effective frequency range of the non-adaptive feedback controller can be directly assessed by the design, the feedforward adaptive algorithm needs a modification in the signal processing path. In fact, the adaptive feedforward algorithm aims at minimizing a cost functional which is generally determined by the Mean Square Error (MSE)

$$\xi(n) = E\left[e^2(n)\right] \tag{5}$$

where $E[.]$ denotes the expected value. By adequate filtering of the error signal $e(n)$ one can define any desired weighting function for the noise reduction across the frequency range. Thus, as shown in Figure 4, a high-pass filter $HP(z)$ is applied on the error signal $e(n)$ propagating to the feedforward adaptation algorithm in order to attenuate low frequency components of the cost functional. Since this filtering process also belongs to the secondary path as it was defined above, the same high-pass filter $HP(z)$ must be additionally used to bring up the filtered reference $x'(n)$. This "frequency selective adaptation" was an important step in tuning the feedback-feedforward combination to reach a good overall noise reduction.

Moreover, the high-pass filtering of the error signal propagating to the feedforward adaptation algorithm prevents the feedforward part from malfunctioning in a well known problematic situation occurring in active headsets: That is, when movements of the ear cup with respect to the user’s head cause extensive low frequency (<15 Hz) pressure fluctuations inside the ear cup. The common feedforward adaptive algorithm tries to react to these pressure variations, since they are measured by the error microphone, but cannot eliminate them as there is no corresponding reference measured outside the ear cup. Although in this case the algorithm does not become unstable, the adaptation is disturbed, causing some unwanted momentary loss of noise reduction. The high-pass filtering of the error signal removes these low frequency pressure fluctuations from the cost functional of the adaptive feedforward algorithm and prevents it from a disturbing reaction.

4.2 Clipping Protection

The extensive low frequency pressure fluctuations inside the ear cup, caused by fast movements of a user's head can bring up a further problem: When the feedback controller reacts to the high amplitude at its input $e(n)$, also the corresponding output $y_{fback}(n)$ may get to excessive level. If the actuating variable $y(n)$ exceeds the operating range of the control loudspeaker and the associated amplifying circuit, the speaker starts clipping and instead of the desired anti-noise signal $u(n)$, a high level distortion noise is generated at the users ear. The same problem occurs, when the headset is used in extremely noisy environment, so that the anti-noise required for active compensation exceeds the obtainable level. In this case, the control loop must be modified, to prevent the speaker from clipping.

For the feedback loop, this can simply be done by reducing the feedback controller gain, whenever $y_{fback}(n)$ approaches an accordant limit. In Figure 5, the factor *Reduce* is integrated into the feedback loop for this purpose. In normal operation, *Reduce* remains at value 1.0, which means that it doesn't effect the controller loop. In extreme level situations *Reduce* is automatically diminished to keep $y_{fback}(n)$ in the desired range. Of course, the noise reduction effect ceases, but instead of producing distortions the best available compensation is performed. To return to normal mode, *Reduce* continuously trends towards the value 1.0 as soon as the extreme level situation is over.

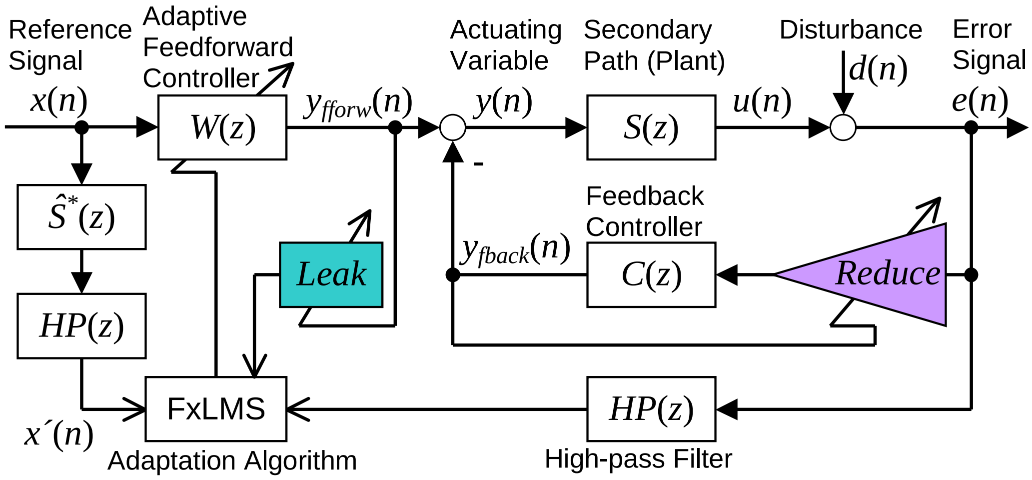

Figure 5: Implementation with clipping protection.

For the adaptive feedforward controller, the problem is a bit more complicated: A simple reduction factor for $y_{fforw}(n)$ would affect the adaptation algorithm in a way, that it tries to counteract by rising the parameters of the FIR filter $W(z)$ in order to keep the noise reduction effect unchanged, countervailing the reduction factor. Thus, for the adaptive case, the reduction has to be done directly inside the adaptation algorithm. In [8] a "leak factor" is introduced, that can be used for this purpose. The adaptation equation (1) is then extended by the new factor *Leak* to

$$\vec{w}(n+1) = Leak \cdot \vec{w}(n) + \mu \cdot e(n) \cdot \vec{x}'(n) \tag{6}$$

In each sampling step the former values $\vec{w}(n)$ of all FIR parameters in $W(z)$ are multiplied by the *Leak* factor before using them for calculation of the new parameters $\vec{w}(n + 1)$. In [8] the leak factor has a fixed value slightly below 1.0. For our application, we propose to make *Leak* variable. Similar to the *Reduce* factor, in normal operation *Leak* remains 1.0 and does not affect the controller. As shown in Figure 5 *Leak* can be diminished, whenever $y_{fforw}(n)$ is about to overrun its limit. This results in a reduction of all FIR parameters, also reducing the output $y_{fforw}(n)$, which is the desired effect to prevent the speaker from clipping. Like the *Reduce* factor *Leak* also continuously trends towards 1.0 as soon as the environmental noise allows the return to normal operation.

5 IMPLEMENTATION AND RESULTS

For the creation of the prototype we designed a robust standard feedback controller with 11 poles and 11 zeros, offering noise reduction for a frequency range from 20 up to 200 Hz. The filtering of the error signal considered by the cost functional of the adaptive feedforward algorithm was designed to shape the effective range of the adaptive algorithm to begin at 100 Hz. The experimental realisation was fulfilled using the product series headset *Sennheiser HMEC350* as an acoustical platform. The headset was complemented with a reference microphone outside the ear cup. Both feedback and adaptive feedforward control were processed by a DSP at a sampling frequency of 20 kHz. Analogue-digital data transfer was accomplished by 16-bit A/D and D/A converters.

For the evaluation of noise reduction, a self constructed artificial head with an ear simulator was used. The results presented in Figure 6 were achieved under conditions reproducing average ear cup / head leakage.

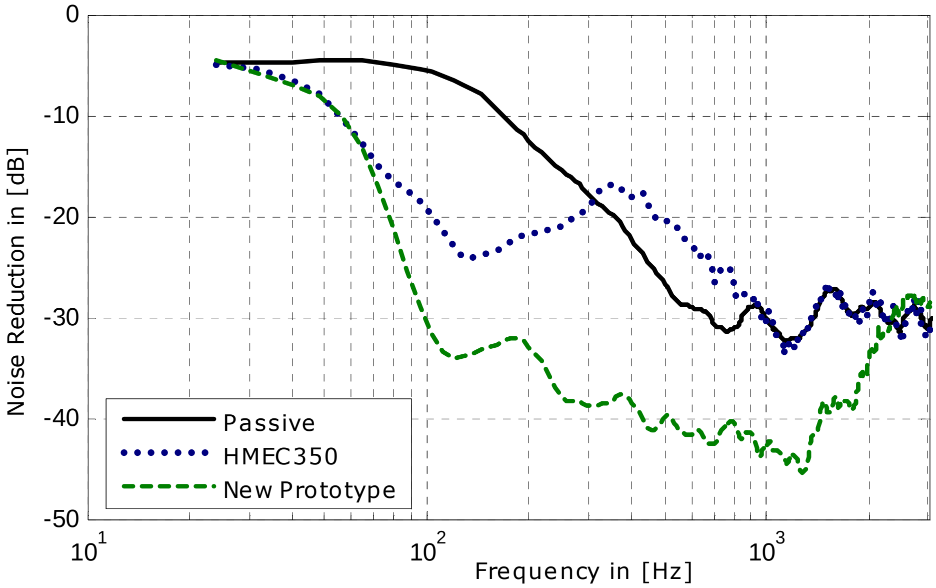

Figure 6: Comparison between passive and active noise reduction (Series HMEC350 and new prototype).

Figure 6 presents a comparison between the passive noise reduction of the ear cups, the overall passive and active noise reduction of the product used as acoustical platform, and the overall noise reduction of the new prototype. The existing product, the *HMEC350*, is based on an analogue non-adaptive standard feedback controller and was designed to actively compensate for the gap of passive noise reduction at low frequencies. As stated above, to guarantee for robustness, the controller design implies that the ANC effect ceases at 300 Hz. Through the use of an added feedforward controller with the proposed combination strategy, the new prototype enables a higher and broader band active noise reduction.

6 CONCLUSIONS

This paper details the development of an active noise control headset prototype. The control strategy consists of combining a non-adaptive standard feedback with an adaptive feedforward controller based on the FxLMS algorithm. By use of adaptive control, it became possible, to combine the very good passive attenuation of a closed circum-aural headset with the benefits of the feedforward strategy active noise control, additionally to the feedback control. For the feedback loop, a standard controller was used, motivated by its robustness while maintaining good noise reduction at low frequencies. The feedforward adaptive controller applies the FxLMS algorithm and was modified to take into account the effects of the feedback loop. Finally, a high-pass filter was added to focus the effective range of the feedforward part on higher frequencies. The demonstrator built during this work allows for circum-aural ear cups to extend the effective upper frequency limit of the ANC from 300 Hz to 2 kHz.

7 AKNOWLEDGMENT

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8 REFERENCES

[1] S. M. Kuo and D. R. Morgan, *Active Noise Control Systems Algorithms and DSP Implementations* (Wiley-Interscience, New York, 1996).

[2] S. J. Elliott, *Signal Processing for Active Control* (Academic Press, San Diego, 2001).

[3] C. H. Hansen, "Active Noise Control – From Laboratory to Industrial Implementation," *Proceedings of NOISE-CON 97,* Vol. 1, pp. 3-38, 1997.

[4] W. K. Tseng, B. Rafaely and S. J. Elliott, "Combined Feedback-Feedforward Active Control of Sound in a Room," Journal of the Acoustical Society of America, **104**(6), 3417-3435 (1998).

[5] E. Esmailzadeh, A. Alasty and A. R. Ohadi, "Hybrid Active Noise Control of a One-Dimensional Acoustic Duct," Transactions of the ASME, **12**, 10-18 (2002).

[6] Y. Song, Y. Gong and S. M. Kuo, "A robust Hybrid Feedback Active Noise Cancellation Headset," IEEE Transactions on Speech and Audio Processing, **13**(4), 607-617 (2005).

[7] W. S. Gan and S. M. Kuo, "An Integrated Audio and Active Noise Control Headsets," IEEE Transactions on Consumer Electronics, **48**(2), 242-247 (2002).

[8] S. J. Elliott, I. M. Stothers, and P. A. Nelson, "A multiple error LMS algorithm and its application to the active control of sound and vibration," IEEE Trans. Acoust., Speech, signal Processing, ASSP, **35**, 1423-1434 (1987).