COMBINED FEEDBACK AND ADAPTIVE FEEDFORWARD ACTIVE NOISE CONTROL

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ABSTRACT

This paper presents a strategy for combining a feedback and an adaptive feedforward active noise controller. Feedback control of noise based on a closed loop controller encounters the major constraint of the plant deadtime caused by the acoustical transfer path from the control-loudspeaker to the error-microphone. This results in a limit to the band width of the control. Therefore in Headset applications no sufficient noise reduction can be fulfilled above 400 Hz. Feedforward strategies overcome the dead time problem by using an earlier reference signal. Though control of high frequencies is hence made possible, the band width of noise reduction is still limited, since in practical applications the maximum number of controller parameters is fixed. In this paper we propose a combination strategy of both feedback and adaptive feedforward control in order to realise a broadband noise reduction. The strategy consists firstly in tuning each controller according to the desired frequency range. Secondly a binding structure is realised to enable both controllers act simultaneously in a constructive way.

1. INTRODUCTION.

Since ten years active headsets have been representing the unique widespread successful commercial application of active noise reduction. These commercial products have been based on non-adaptive, analogue, and mainly feedback control techniques. However, during the last two decades the digital signal processing has been increasingly used by researchers in the domain of active noise control. The trend of the ever-growing performance of processors simultaneously to the reduction of their size and costs has made possible the use of adaptive algorithms in practical applications. Particularly adaptive digital feedforward control techniques could now be implemented in active Headsets at a reasonable expense-benefit ratio. Numerous works such as [1][2][3] describe different active noise controller structures and optimisation algorithms by use of either feedback or feedforward strategies. Especially in active headsets the simultaneous use of both control strategies could be of great benefit [1].

Some works [4][5] report the outcomes of combining feedforward and feedback control strategies. In this case, the internal model control (IMC) is usually preferentially applied in the feedback control loop in order to guarantee that both combined algorithms work independently, thus avoiding any interfering interaction. As far as the headset application is concerned this combination offers satisfying performance under laboratory conditions. Usually very good noise reduction values could be measured on an artificial head but unfortunately the combination might malfunction under realistic conditions on a users head. In the following the problem of the internal model control (IMC) in a realistic application is explained and a new combination method of feedback and adaptive feedforward controllers is presented including the experimental realization and results.

2. PROBLEM STATEMENT

Active Noise Control (ANC) systems attempt to neutralize undesired acoustic noise via some appropriate antinoise signal generated by a secondary sound source. The Noise Cancellation occurs in a specified area, where an error microphone is placed.

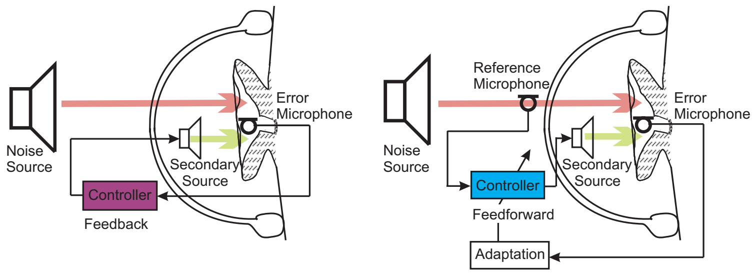

Fig. 1. Feedback (left) and adaptive Feedforward (right) active noise control in a Headset.

Fig. 1 presents the two main control strategies feedback and feedforward. For discrete-time signals, a feedback structure feeds only the superposition at the error microphone (also called error signal) to the controller while a feedforward system attempts to cancel the disturbing noise *d(n)* by taking also an "upstream" reference input *x(n)* into account.

The feedback controller exhibits the advantage of reaching acceptable noise reduction performance at a minor expense, since feedback automatic control can be realised with few non-adaptive analogue electrical components. Feedback control of noise however encounters the major constraint of the plant deadtime mainly caused by the acoustical transfer path from the control-loudspeaker to the error-microphone. As known from the control theory this results in a phase drop at high frequencies and hence in a decrease of the phase margin. Consequently the band width of control is still limited to low frequencies. For this reason, in headset applications with circumaural ear cups no sufficient noise reduction can be fulfilled above 400 Hz. By contrast, in a feedforward strategy the "upstream" reference *x(n)* is intended to give a sufficiently advanced and coherent indication of the approaching noise. If the time advance of the reference signal exceeds both the electrical and acoustical delay of the plant, than no deadtime limitation is to be observed. Thus, from this point of view an adequately outlayed feedforward system encounters no limitation in the frequency range of control, which represents the first advantage of the feedforward strategy. Currently non-adaptive feedforward control strategies are implemented in some commercial headsets such as the HMEC 45 from Sennheiser, which offers an active noise reduction up to 1,5 kHz. But the application of non-adaptive feedforward controllers is still limited to supraaural (open) ear cups. These ear cups exhibit almost the same plant and the same primary path ratio (Ratio of the two primary noise transfer paths from the noise source to the reference and the error microphone) independently from the user. By contrast, the acoustical plant and the primary path ratio in circumaural ear cups vary amply depending on the users head anatomy. The problem of a non-adaptive feedforward – so an open-loop - controller is that efficient noise reduction is guaranteed for only one plant and one primary path ratio. Thus, for the use in a circumaural ear cup a feedforward controller needs to be made adaptive to plant and primary transfer path changes. Therefore adaptive algorithms are used to adapt the controller parameters, generally for the purpose of minimizing the power of the error signal. The controller parameter adaptation based on the error signal provides a kind of automatic control, or in other words it enables to “close” the control loop. Through the purpose of minimizing the power of the error signal, adaptive algorithms are able by definition to focus on the reduction of the dominant frequency band of the disturbance. This ability constitutes the second advantage of using an adaptive feedforward controller. Adaptive algorithms however require the use of a powerful digital signal processor, which is linked with higher expenses. In a commercial application these expenses should be kept within a realistic limit. A meaningful issue is to confer a part of the cancelling task to a non-adaptive feedback controller, in order to save calculating and memory resources. This could be reached via a combination of 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 low frequency noise. For this, the combination strategy must fulfil the condition that each of the two controllers works independently for its intended frequency band in avoidance of any interfering interaction.

3. STATE OF THE ART

Lets consider the feedforward control strategy. Fig. 2 shows an active headset ear cup and the signal processing scheme of an adaptive Feedforward controller. The error signal *e(n)* is the superposition of the antinoise *u(n)* with the disturbance *d(n)*. The disturbance *d(n)* represents the sound captured by the inner error microphone when the control is switched off. This disturbance is arising from the primary noise source which propagated through the ear cup. The plant model $\hat{S}(z)$ reproduces the secondary path *S(z)*, which expresses the transfer behaviour from the output *y(n)* of the adaptive filter *W(z)* (the control variable) to the signal picked up by the error microphone *e(n)*. The electrical components for amplifications and AD/DA-conversions are not represented in the figure, but their effects are comprised in the modelled Plant. The adaptive feedforward controller is designed as a 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)* picked up by the outer reference microphone is filtered by the plant model $\hat{S}(z)$ to build the “filtered reference” *x’(n)*. This filtered reference 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

$$\vec{x}'(n) = [x'(n) \;\; x'(n-1) \; \ldots \; x'(n-L+1)]^T \quad \& \quad x'(n) = \hat{s}(n) * x(n) \tag{2}$$

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

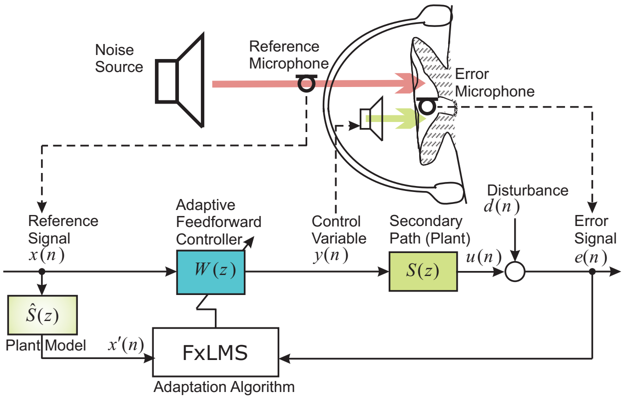

Fig. 2. Adaptive feedforward active noise control with FxLMS.

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 it leads to a changing secondary path from the feedforward point of view. Lets now consider the integration of the Internal Model Control (IMC) strategy within the feedforward loop. The IMC strategy is favourably applied in the feedback control loop when it is connected to an adaptive feedforward loop. Fig. 3 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 to calculate an approximation $\hat{u}(n)$ of the antinoise $u(n)$. The difference between the superposition $e(n)$ and the approximated antinoise $\hat{u}(n)$ produces an approximation $\hat{d}(n)$ of the primary disturbance noise $d(n)$, which is then fed to the IMC feedback Controller $C(z)$. With an accurate approximation of the disturbance noise, no output from the feedforward path is contained in the feedback controller input $\hat{d}(n)$ and so the feedback loop does not react to the feedforward part. Also from the feedforward point of view, no changes occurred to its output through the IMC feedback loop. Thus the plant model remains $\hat{S}(z)$ and no modification is needed in the feedforward loop. Hence, this combination-strategy fulfils the condition set above without any need for adjustment to the loops.

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 approximation error which is subsequently amplified by the controller $C(z)$, and could destabilize the closed loop behaviour. Designing an IMC controller which avoids instability in any leakage-case signifies a poor overall noise reduction performance. In [6] the authors reported this problem and suggested to control the plant *S(z)* with a subordinate standard feedback controller which was specifically designed to make the secondary path less dependent from the leakage condition. Thus, the superordinate 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 $\hat{S}(z)$ via some generated incoherent test noise. Even though this approach is consistent, it is practically not reasonable, since it damages the comfort of the listener by adding the test noise to the loudspeaker input. Moreover, the approach significantly burdens the computing resources since it requires a full online identification algorithm.

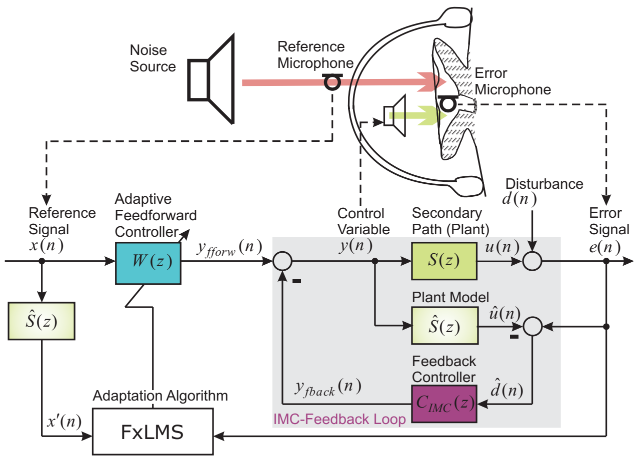

Fig. 3. Internal Model Control Feedback – adaptive Feedforward combination.

4. CONTROLLER DESIGN

A standard closed loop feedback controller is more robust to plant changes due to leakage variations, since 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 Fig. 4 the subordinate standard feedback controller directly processes the error signal *e(n)* picked up by the error microphone. With this feedback approach an emphasis is put on the absence of any plant model in the signal transmission path of the feedback controller. The absence of this uncertainty factor reduced the stability analysis complexity, and the controller design turned out to be more manageable. For the controller design, firstly several plants, each corresponding to a different ear cup leakage situation, were measured. Secondly 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 of each plant-case.

When a feedback loop is linked to a superordinate adaptive feedforward loop as we proposed in Fig. 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

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

constitutes the new secondary path, since it is defined as the transfer behaviour from the output of the adaptive filter to the signal picked up by the error microphone. Therefore 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) analogue to the case of the separate feedforward loop stated above. Summarised, within the proposed approach the subordinate feedback control loop, independently from the feedforward loop, presents always 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 accurately to take this effect into account. This makes the proposed algorithm fulfil the conditions set in the problem statement.

For robustness considerations, one may criticise that the algorithm still depends on the uncertainty factor of plant modelling. But we emphasise here that the plant modelling is switched to the FxLMS update path, which is much more insensitive 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].

2.1. 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)

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

where *E[.]* denotes the expected value. By adequate filtering of the error signal *e(n)* one can define any weighting function desired for the noise reduction across the frequency range. Thus, as shown in Fig. 4, we applied a high-pass filter on the error signal *e(n)* propagating to the feedforward adaptation algorithm in order to attenuate low frequency parts from the cost functional. Since this filtering process also belongs to the secondary path as it was defined above, the same high-pass filter must be used to build 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 at 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 instable, 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.

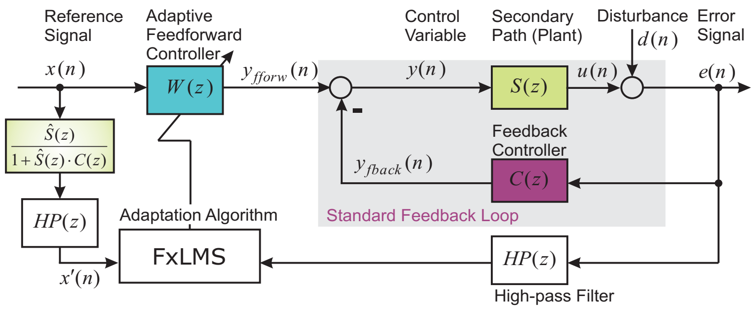

Fig. 4. Standard Feedback Control – adaptive Feedforward combination.

5. IMPLEMENTATION AND RESULTS

For the achievement 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 series product headset *Sennheiser HMEC350* as 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 under a sampling frequency of 20 kHz. Analogue-digital data transfer was accomplished by 16 bit AD and DA converters. For the evaluation of noise reduction a self constructed artificial head with ear simulator was used. The results presented in Fig. 5 were reached under reproduction of an average ear cup / head leakage.

Fig. 5. presents a comparison between the passive noise reduction of the ear cups, the overall passive and active noise reduction of the series product used as acoustical platform, and the overall noise reduction of the new prototype. The series product 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 we stated above, to guarantee for robustness, the controller design implied that the ANC effect ceased 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 reaching up to 2 kHz.

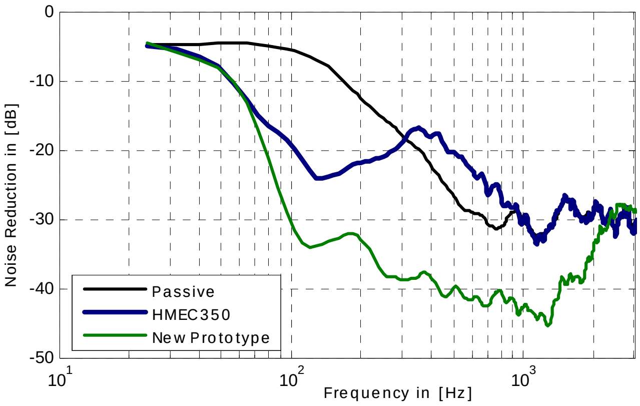

Fig. 5. Comparison between passive and active noise reduction (Series product HMEC350 and new prototype.

6. CONCLUSIONS

In this paper the development of an active noise control headset prototype was reported. The control strategy consisted in combining a non-adaptive standard feedback, and an adaptive feedforward loop based on the FxLMS algorithm. For the feedback loop a standard controller was used, motivated by its robustness qualities while maintaining good noise reduction at low frequencies. For the superordinate feedforward adaptive loop a FxLMS algorithm was applied and modified to take account of the effects of the subordinate feedback loop. Finally a high-pass filter was added to asses the effective range of the feedforward part to higher frequencies. The constructed headset prototype permitted to extend the frequency range of active noise cancellation to 2 kHz.

7. AKNOWLEDGMENT

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