

EXTENSION AND EVALUATION OF A SPECTRO-TEMPORAL MODULATION METHOD TO IMPROVE ACOUSTIC FEEDBACK PERFORMANCE IN HEARING AIDS

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ABSTRACT

Adaptive filters have been widely used for feedback cancellation in audio systems including hearing aids. However, the ability to cancel feedback in dynamic feedback situations is still a big challenge; the adaptive filters need to ensure a small enough steady-state error to facilitate sufficient amplification in hearing aids, hence their convergence rates are typically insufficient to handle fast feedback path changes. Recently, we proposed a novel method by using spectro-temporal modulation (STM) in the time-frequency regions where the adaptive filters have insufficient convergence rate. Applying STM prevents feedback to occur and replaces traditional loud/annoying feedback whistling sounds with soft/non-intrusive STM processed sounds. In this work, we introduce an extension to make the STM processed sound even less audible. Furthermore, we present novel evaluation results regarding feedback cancellation and sound quality from listening experiments, which confirm that without degrading sound quality in static feedback situations we significantly improve feedback cancellation performance upon fast feedback path changes.

Index Terms— Acoustic feedback cancellation, adaptive filters, spectro-temporal modulation, hearing aids.

1. INTRODUCTION

A hearing aid is a small medical device fitted in/on the ear, and it is designed to compensate for individual hearing loss. Main functionalities in modern hearing aids include dynamic amplification (known as compression), microphone array processing, and noise reduction to improve speech intelligibility and/or reduce listening effort [1–3]. However, due to the large amount of amplification needed in a hearing aid to compensate for hearing loss, and the fact that its microphone and loudspeaker (known as receiver in hearing aid terminology) are typically placed within a few centimeters to each other, the acoustic feedback problem is almost unavoidable.

Acoustic feedback problems occur when the output sound from an audio device, in this case a hearing aid, returns to its own microphone and thereby an acoustic loop is created. The audio system can be affected by the feedback signal that travels around this acoustic loop, and in the worst case the audio system becomes unstable and the output loudspeaker generates a loud/annoying whistling sound. Hence, the acoustic feedback problem can significantly lower the benefits of hearing aids.

A state-of-the-art solution, see examples in [4–22], for reducing the effects of feedback is an acoustic feedback cancellation (AFC) system using adaptive filters [23, 24] in a system identification setup [25, 26]. Fig. 1 illustrates a simple hearing aid system including

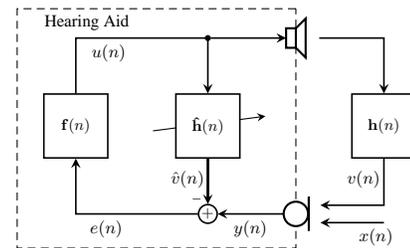


Fig. 1. A general hearing aid feedback cancellation system.

acoustic feedback and feedback cancellation by means of an adaptive filter. For convenience, we denote all signals as discrete-time signals with time index n .

The hearing aid system consists of a forward path denoted by the impulse response $\mathbf{f}(n)$, which represents the processing unit for generating the loudspeaker signal $u(n)$. The microphone signal $y(n)$ is a mixture of the desired incoming signal $x(n)$ and the undesired but unavoidable feedback signal $v(n)$; $v(n)$ is the result of the physical acoustic coupling from loudspeaker to microphone referred to as the acoustic feedback path denoted by the impulse response $\mathbf{h}(n)$. The adaptive filter $\hat{\mathbf{h}}(n)$ is used to create a feedback signal estimate $\hat{v}(n)$ to cancel $v(n)$, ideally $\hat{\mathbf{h}}(n) = \mathbf{h}(n)$ and thus $e(n) = x(n)$.

Typically, a compromise between steady-state error and convergence rate has to be made in adaptive filters. In the hearing aid application, the forward path $\mathbf{f}(n)$ needs to provide significant amplification (much more than 50 dB in the most extreme case [1]), hence, the adaptive filter has to provide a sufficiently small steady-state error to ensure system stability. Unfortunately, this typically induces too slow convergence rate of $\hat{\mathbf{h}}(n)$ upon fast feedback path changes in $\mathbf{h}(n)$, e.g., when a phone is moved towards the user’s ear.

In the past, linear time-varying (LTV) systems have been used for feedback control [27]. In [28], we further proposed a new LTV design by using spectro-temporal modulation (STM) in combination with traditional adaptive filters for feedback cancellation, we refer to it as AFC-STM. The main goal of the adaptive filter $\hat{\mathbf{h}}(n)$ was to provide small enough steady-state error for ensuring amplifications in relatively static feedback situations, while the STM processing is deployed in the time-frequency regions in which the adaptive filter is insufficient to cancel feedback upon fast feedback path changes. More specifically, *within the forward path $\mathbf{f}(n)$, a pre-defined STM processing is used upon a feedback detection, which prevents acoustic feedback to build up. In the meantime, the adaptive filter $\hat{\mathbf{h}}(n)$ converges to the new feedback path $\mathbf{h}(n)$, after that the STM processing is disabled.* A review of this method is given in Sec. 2.

In this work, we improve the original AFC-STM to decrease the audibility of the STM processing by introducing a new link be-

Example of determining $\alpha(m, k)$ based on $FD(m, k)$ and $\alpha_0(m, k)$

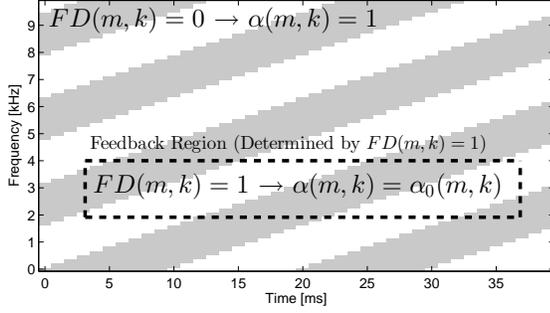


Fig. 2. An example of determining the applied STM pattern $\alpha(m, k)$ based on feedback detection $FD(m, k)$ and basis STM pattern $\alpha_0(m, k)$. The gray and white areas indicate basis values of $\alpha_0(m, k) = 0$ and $\alpha_0(m, k) = 1$, respectively. The STM processing $\alpha(m, k) = \alpha_0(m, k)$ is applied upon feedback detection $FD(m, k) = 1$, indicated by the spectro-temporal “Feedback Region” surrounded by the dashed box; otherwise $\alpha(m, k) = 1$, implying the STM processing is not applied.

tween the applied STM pattern and feedback risk, more details can be found in Sec. 3. Moreover, we conducted listening tests to compare a reference commercial hearing aid with traditional AFC system to a prototype hearing aid with improved AFC-STM system (it is otherwise identical to the reference hearing aid without STM); in a feedback performance test, participants rated the test hearing aids in different critical dynamic feedback situations in terms of annoyance ratings; in a sound quality test, participants assessed if there is a sound quality difference in static feedback situations between the two systems. More details are presented in Sec. 4.

2. REVIEW OF AFC-STM

The AFC-STM includes an STM processing in the forward path $f(n)$, and it can be facilitated in the short-time Fourier transform (STFT) domain. To obtain the desired STM processing in the STFT domain, a scaling factor $\alpha(m, k)$ is applied to each time-frequency unit, where m and k are frequency and time indices, respectively.

Fig. 2 illustrates an example of the STM processing in the STFT domain. The entire plot shows the basis STM pattern denoted by basis values $\alpha_0(m, k)$, where the gray and white areas indicate $\alpha_0(m, k) = 0$ and $\alpha_0(m, k) = 1$, respectively, and the basis pattern repeats over time. Furthermore, the STM processing is only applied in the forward path $f(n)$ upon feedback detection, i.e., when $FD(m, k) = 1$, as indicated by the marked “Feedback Region”. The applied values for the STM processing are denoted by $\alpha(m, k)$ and are derived as,

$$\alpha(m, k) = \begin{cases} \alpha_0(m, k) & \text{if } FD(m, k) = 1, \\ 1 & \text{otherwise.} \end{cases} \quad (1)$$

The STM pattern is specifically designed according to the acoustic loop delay in hearing aids, which is typically 4 – 8 ms [29]. For each frequency m over time in Fig. 2, there is a repeated pattern of $\alpha_0(m, k)$ as: 10 ms of $\alpha_0(m, k) = 0$ followed by 10 ms of $\alpha_0(m, k) = 1$. Moreover, the patterns at different frequencies are time shifted to ensure minimum audibility when being applied. With appropriate design of the STM pattern, it will prevent feedback to build up (to be noticeable) when applied [28].

In [28], feedback detection $FD(m, k)$ is based on the open loop transfer function $\Theta(m, k) = F(m, k)(H(m, k) - \hat{H}(m, k))$, where

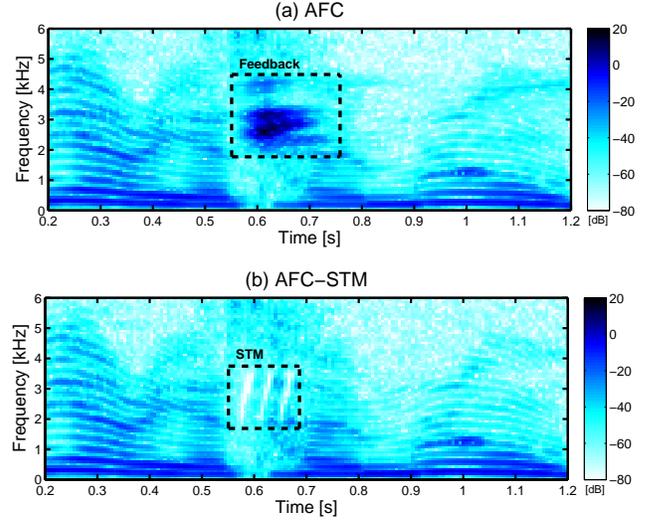


Fig. 3. Spectrograms of hearing aid output speech signals with an abrupt feedback path change after 0.55 s. (a) Traditional AFC system becomes unstable and loud feedback sound appears. (b) AFC-STM system remains stable and the STM processed sound (vertical strips) replaces the feedback sound and feedback cannot build up.

$F(m, k)$, $H(m, k)$, and $\hat{H}(m, k)$ are the frequency responses of $f(n)$, $h(n)$, and $\hat{h}(n)$. More specifically, we determined $FD(m, k)$ using the open loop transfer function estimate $\hat{\Theta}(m, k)$, and the threshold value θ_m on the open loop magnitude $|\hat{\Theta}(m, k)|$, as

$$FD(m, k) = \begin{cases} 1 & \text{if } |\hat{\Theta}(m, k)| \geq \theta_m, \\ 0 & \text{otherwise.} \end{cases} \quad (2)$$

The value $\theta_m \approx 1$, thus the detection in (2) is based on the magnitude condition of the Nyquist stability criterion [30, 31].

Roughly speaking, the STM processing is only active in time and frequency regions, where the AFC system is not able to cancel feedback, e.g., during and shortly after a rapid change of the feedback path. We refer to [28] for more details.

Fig. 3 is re-produced from Fig. 8 in [28], showing simulation results with focus on the abrupt feedback path change after 0.55 s. It shows that the AFC system without STM becomes unstable and feedback whistling sound appears upon a rapid feedback path change, whereas for the AFC-STM system the STM is active upon feedback detection right after the feedback path change, and prevents the feedback to build up over time. Moreover, the STM is only active in time-frequency regions where the AFC system cannot efficiently cancel feedback.

3. EXTENSION OF AFC-STM

In this work, we improve the AFC-STM by applying *continuous* values in the STM pattern, instead of using binary values as proposed in [28]. The motivation for this is to reduce the audibility of the STM processing. The pattern with these continuous values is referred to as the *soft STM pattern* and denoted by $\alpha_s(m, k)$, where $0 \leq \alpha_s(m, k) \leq 1$.

Applying the binary values of $\alpha(m, k)$, as done in [28], can be considered as the most efficient way to prevent/remove feedback. However, this also implies that we introduce maximum possible modulation over time and frequency to the hearing aid output signal,

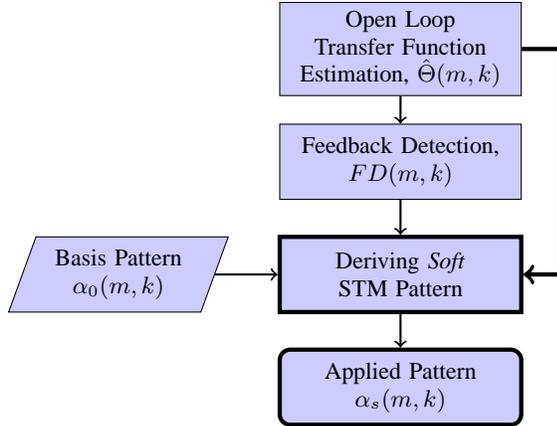


Fig. 4. Deriving *soft* STM pattern with $0 \leq \alpha_s(m, k) \leq 1$. The arrow and the boxes with thick lines indicate the extension to the original method in [28].

which can potentially degrade sound quality; it should thereby only be applied if the feedback risk is very high.

Also, less modulation is in many cases sufficient to prevent feedback, which has the potential to make the STM processing less audible. Compared to the original method shown in Fig. 2, this improvement would apply to the marked feedback region; in the gray areas, we now use $0 \leq \alpha_s(m, k) \leq 1$ instead of $\alpha_0(m, k) = 0$; in the white areas, $\alpha_s(m, k) = 1$ which is identical to [28].

The flow chart in Fig. 4 illustrates how to achieve the values of $\alpha_s(m, k)$. The parts involving open loop transfer function estimate $\hat{\Theta}(m, k)$ and feedback detection $FD(m, k)$ are unchanged compared to [28]. In the following, we explain the improved derivation of $\alpha_s(m, k)$ in more details.

Upon feedback detection, i.e., $FD(m, k) = 1$, the values $0 \leq \alpha_s(m, k) \leq 1$ are derived based on the open loop transfer function $\hat{\Theta}(m, k)$ and the basis STM pattern $\alpha_0(m, k)$ with binary values, as

$$\alpha_s(m, k) = \max\left(f(\hat{\Theta}(m, k)), \alpha_0(m, k)\right), \quad (3)$$

where $f(\hat{\Theta}(m, k))$ is a mapping function, and $0 \leq f(\hat{\Theta}(m, k)) \leq 1$; it is used to associate the applied STM pattern $\alpha_s(m, k)$ with the feedback risk reflected by the open loop magnitude $|\hat{\Theta}(m, k)|$. An example mapping function could be

$$f(\hat{\Theta}(m, k)) = \begin{cases} 0 & \text{if } |\hat{\Theta}(m, k)| \geq 1, \\ 1 & \text{if } |\hat{\Theta}(m, k)| \leq 0.5, \\ 2 - 2 \cdot |\hat{\Theta}(m, k)| & \text{otherwise.} \end{cases} \quad (4)$$

Inserting (4) to (3) and applying an example threshold value $\theta_m = 0.5$ in (2) implies, that for very high open loop magnitude estimates and thereby high feedback risks, e.g., $|\hat{\Theta}(m, k)| \geq 1$, we apply the maximum modulation in $\alpha_s(m, k)$ as $\alpha_s(m, k) = \alpha_0(m, k)$ upon feedback detection. Compared to the original method shown in Fig. 2, this implies no change, i.e., $\alpha_s(m, k) = 0$ for the gray areas and $\alpha_s(m, k) = 1$ for the white areas within the “Feedback Region”.

On the other hand, for smaller open loop magnitude estimates and thereby lower feedback risks, e.g., $|\hat{\Theta}(m, k)| \leq 0.5$, $\alpha_s(m, k) = 1$, i.e., the STM processing in principle would not be active even if there is a feedback detection. Compared to the original method shown in Fig. 2, this implies $\alpha_s(m, k) = 1$ for both the gray and white areas within the “Feedback Region”.

Example of soft STM pattern $\alpha_s(m, k)$, where $0 \leq \alpha_s(m, k) \leq 1$

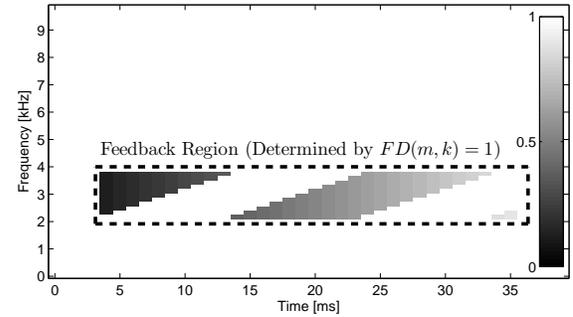


Fig. 5. An example of applied *soft* STM pattern $\alpha_s(m, k)$. Within the feedback region $0 \leq \alpha_s(m, k) \leq 1$, otherwise $\alpha_s(m, k) = 1$.

More interestingly, in between these two extreme cases, i.e., $0.5 < |\hat{\Theta}(m, k)| < 1$, a soft STM pattern will be applied and the modulation depends on the open loop magnitude estimate $|\hat{\Theta}(m, k)|$. Compared to the original method shown in Fig. 2, this implies $0 \leq \alpha_s(m, k) \leq 1$ for the gray areas and $\alpha_s(m, k) = 1$ for the white areas within the “Feedback Region”.

Fig. 5 illustrates an example of using continuous values of $\alpha_s(m, k)$ based on (3). Within the “Feedback Region”, the values of $\alpha_s(m, k)$ alternate between $0 \leq \alpha_s(m, k) \leq 1$ (indicated by grayscale colors) and $\alpha_s(m, k) = 1$ (indicated by white color). Moreover, in the areas with $0 \leq \alpha_s(m, k) \leq 1$, the values fade from $\alpha_s(m, k) = 0$ towards $\alpha_s(m, k) = 1$, indicating that in this example the feedback risk is highest right after feedback detection, and it is decreasing over time.

4. LAB EVALUATIONS

In this section, we present the lab evaluations regarding feedback performance and sound quality. Fourteen participants (ten males and four females), with normal hearing, were recruited for this test. Average age was 36.8 years old (min = 20 and max = 51 years old). The results confirm that we achieved a significant improvement in feedback performance while maintaining sound quality with the AFC-STM using the soft STM pattern, where $0 \leq \alpha_s(m, k) \leq 1$.

4.1. Feedback Performance Test

This test includes in total six test conditions. Three conditions were based on a commercial hearing aid with traditional AFC system using NLMS update of $\hat{h}(n)$ fitted 0 dB (AFC-0), 6 dB (AFC-6), and 10 dB (AFC-10) into feedback¹. Furthermore, three conditions based on a prototype hearing aid with improved AFC-STM system fitted 0 dB (AFC-STM-0), 6 dB (AFC-STM-6), and 10 dB (AFC-STM-10) into feedback were tested. The AFC-0 was considered as the reference.

We have identified five situations of hearing aid manipulations from daily life, where the feedback system is critically challenged: hearing aid insertion; covering the ear with a hand; phone calls; wearing a hat; removing the hearing aid. In an exploratory blind test, each participant was asked to do the manipulations on a KEMAR manikin. At the same time, the participant listened to the hearing aid

¹Fitting “ x ” dB into feedback implies that adjusting $F(\omega)$ to reach $\max_{\omega} (20 \log_{10} |F(\omega)H(\omega)|) = x$ dB, where $F(\omega)$ denotes the programmed gain at frequency ω in the hearing aids, and $H(\omega)$ denotes the static feedback path magnitude response in a particular fitting situation.

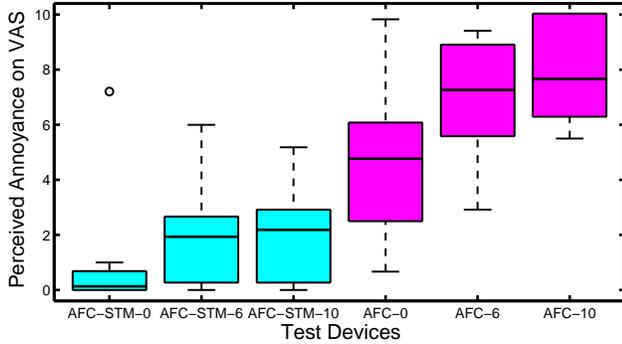


Fig. 6. Statistics of feedback performance test. At VAS=0 the feedback is not annoying, whereas at VAS=10 the feedback is extremely annoying.

output sounds through KEMAR couplers/ears and headphones with preadjusted sound level. This was necessary to ensure that participants were not exposed to very loud feedback sounds.

The experiment took place in a quiet room to ensure the same test condition for each participant. After a training session, which allows each participant to get familiar with the manipulations and different types/levels of artifacts (feedback and/or STM processed sounds), each participant had to rate the hearing aid output sound for each test device and each manipulation. They reported the annoyance on a visual analogue scale (VAS) from 0 (not annoying) to 10 (extremely annoying).

Fig. 6 shows the box plot² across all manipulations for all participants. A robust ANOVA on repeated measures shows a significant effect of the test condition ($p < 0.001$). Post hoc comparisons reveal that all comparisons are statistically significant except the following: AFC-STM-0 and AFC-STM-6, AFC-STM-0 and AFC-STM-10, AFC-STM-6 and AFC-STM-10, AFC-6 and AFC-10.

Hence, it is interesting to note that the perceived annoyance of AFC-STM is statistically significantly lower than AFC for all programmed gains. In other words, *perceived annoyance with AFC-STM fitted 10 dB into feedback is even lower than the reference AFC fitted 0 dB into feedback.*

4.2. Sound Quality Test

AFC-STM might recognize some specific input signals as feedback, even without any change to the feedback situation; these false recognitions/detections, even though with very short durations, lead to STM activations, which would then undesirably change the hearing aid output signal. Hence, the question we want to answer is: “Are these undesired STM activations audible?”

An example of false detection and hence changes in hearing aid output signals can be found in a “Song” signal which leads to an STM attenuation of 12 dB around 2 – 4.5 kHz for 12 ms. Similarly, false detections are found in “Bird” and “Flute” signals around 2 – 5 kHz for some milliseconds. These examples represented the most extreme measurable changes, and we assessed if these changes are audible when compared to signals from a reference system with

²A “box-and-whisker plot” or simply “box plot” describes the distribution of a continuous variable. The horizontal line inside the box indicates the median. The bottom and top of the box are the first and third quartiles, the difference between these two quartiles is defined as the interquartile range. Each whisker extends to the most extreme data point, which is no more than the 1.5 times the interquartile range. Values outside this range are considered as outliers and shown as circles.

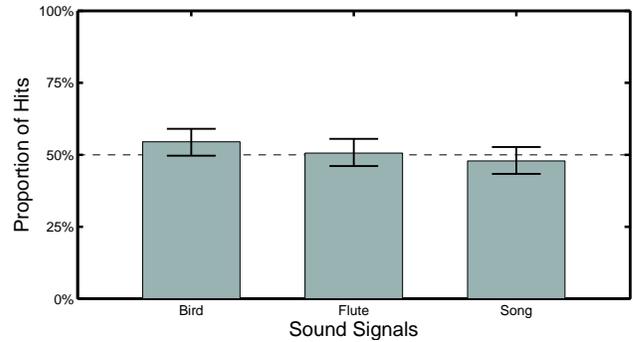


Fig. 7. Statistics of sound quality test. The proportion of hits close to 50% indicates that discrimination between the two test stimuli under comparison is not possible beyond the chance level.

optimal/fixed feedback cancellation and without STM activations.

An AXB discrimination test is used for this verification; in the AXB test, the listener compares which stimulus, the A or the B, is identical or most similar to the X stimulus, which is randomly selected from either A or B. The X stimulus is always presented in the middle of the series of three. The three stimuli appear in four possible orders: AAB, ABB, BAA, and BBA. This experiment is similar to verifying if a coin is well balanced; under the null hypothesis, 50% heads and 50% tails are expected.

In our experiment, we expect to have 50% A’s and 50% B’s if discrimination between both sounds is not possible. We model this experiment as a repetition of a Bernoulli experiment (two outcomes: hit or miss for correctly or wrongly identifying the X) following a binomial distribution.

For each test sound (Bird, Flute, Song), we compared the output signals of AFC-0/AFC-10 to the output signals of AFC-STM-0/AFC-STM-10 with the above mentioned STM activations. Fig. 7 shows the proportion of hit and its standard deviation for each test sound, where the results for each test sound at two different gain levels were pooled together for better parameter estimation.

The results suggest that the measurable differences between both systems cannot be perceptually detected above the chance level for any test sound, i.e., *sound quality is not affected in static feedback situations although there might be measurable differences in hearing aid output signals due to undesired STM activations.*

4.3. Summary

The results from the feedback performance test and the sound quality test suggest that there is a consistent and systematic improvement of feedback experience with the AFC-STM system compared to the AFC system. Furthermore, the measurable differences between AFC-STM and AFC processed output signals do not lead to audible differences in static feedback situations.

5. CONCLUSION

We presented an extended system using spectro-temporal modulation (STM) to improve feedback cancellation. The extension makes the STM processing even less audible compared to the recently introduced STM method. The evaluation results, from listening experiments, show that compared to traditional feedback cancellation systems the improved STM method minimizes feedback annoyance significantly—even with 10 dB additional gain in hearing aids—while maintaining sound quality in static feedback situations.

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