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New feedback detection method for performance evaluation of hearing aids

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Abstract

New objective and accurate feedback detection method, transfer function variation criterion (TVC), has been developed for evaluating the performance of feedback cancellation techniques. The proposed method is able to classify stable, unstable, and sub-oscillatory stages of feedback in hearing aids. The sub-oscillatory stage is defined as a state where the hearing aid user may perceive distortion of sound quality without the occurrence of oscillation. This detection algorithm focuses on the transfer function variation of hearing aids and the relationship between system stability and feedback oscillation. The transfer functions are obtained using the FIR Wiener filtering algorithm off-line. An anechoic test box is used for the exact and reliable evaluation of different hearing aids. The results are listed and compared with the conventional power concentration ratio (PCR), which has been generally adopted as a feedback detection method for the performance evaluation of hearing aids. The possibility of real-time implementation is discussed in terms of a more convenient and exact performance evaluation of feedback cancellation techniques.

1. Introduction

Amplification of an acoustic signal is the basic goal of many sound systems that have both loudspeaker and microphone. In general, acoustic coupling between the loudspeaker and microphone occurs such that the entire system forms a closed loop via a feedback signal. In this case, feedback occurs at a particular frequency when the open-loop gain of the system is greater than the unity, and the open-loop phase response of the system is a 360° multiple of that frequency [1].

In hearing aids, feedback is hard to avoid due to the proximity of the receiver and microphone; feedback drives hearing aids into an unstable stage and makes them howl. Clinically, feedback is one of the most frequent complaints of hearing aid users, as it can reduce the sound quality and limit the gain. Sources of feedback paths in hearing aids include electrical feedback, vent holes, leakage from an imperfect seal between the earmold and tubing joint, as well as structural problems and acoustic transmission inherent within hearing aids, as illustrated in Fig. 1 [2]. Common approaches for preventing acoustic feedback include the attenuation

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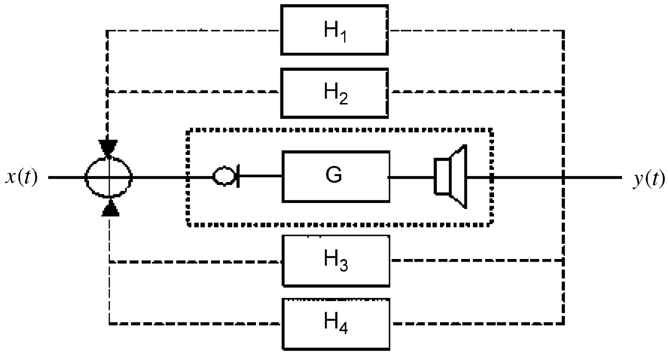

Fig. 1. Schematic block diagram of the signal paths of hearing aids. $x(t)$ is the sound source and G is the transfer function of hearing aids. H_1, H_2, H_3, H_4 are transfer functions of various feedback paths. $y(t)$ is the output signal which goes into the tympanic membrane (dashed lines for feedback signal path; dotted lines for hearing aid system).

of high frequency gain, or the insertion of a fixed phase shifter in the signal path [3]. However, these approaches are not adequate in the practical application of hearing aids because the sound quality of the output signal can be distorted. To overcome these current limitations, feedback cancellation techniques have recently been implemented in hearing aids [4–6]. In this technique, an adaptive filter generates an estimate of the feedback signal, and the generated signal is subsequently subtracted from the microphone input signal. This feedback cancellation technique is the most popular because it is able to track the variation of feedback path, and conserve the sound quality of the output signal. Nevertheless, though many feedback cancellation techniques have been implemented in majority of all digital hearing aids, a more accurate and objective evaluation methods and standards are not yet available [10].

Greenburg et al. [7] have proposed performance evaluation procedures. However, the accurate feedback detection method is the most important factor for the accurate performance evaluation of feedback cancellation technique. As such, several feedback detection methods have been proposed to evaluate the performance of hearing aids including the House Ear Institute (HEI)'s power concentration ratio (PCR) approach [8]. PCR was designed to obtain an objective estimate of early feedback.

TVC method introduced in this paper is designed to discriminate stable, sub-oscillatory, and unstable stages of hearing aid amplification. This detection algorithm focuses on the variation of the transfer function of hearing aids, and the relationship between system stability and feedback oscillation, thereby providing new criteria for these three stages. The specific experimental setup including an anechoic test box is used in the evaluation of several different hearing aids. The experimental results of these hearing aids are subsequently presented and compared with PCR. This paper shows that the proposed TVC is a better and more reliable feedback detection method than conventional PCR for the performance evaluation of hearing aids.

2. Power concentration ratio method

HEI developed several methods to compare hearing aids with respect to the performance of feedback cancellation algorithms [8,9]. One of the more common or familiar measures is referred to as added stable gain (ASG), defined as the difference of the maximum stable gain (MSG) when the feedback cancellation algorithm is alternately switched on and off. MSG is the maximum gain in the stable stage:

$$\mathrm{ASG} = \mathrm{MSG}_{\mathrm{on}} - \mathrm{MSG}_{\mathrm{off}}, \tag{1}$$

where $\mathrm{MSG}_{\mathrm{on}}$ is the maximum stable gain with the feedback algorithm on, and $\mathrm{MSG}_{\mathrm{off}}$ is the maximum stable gain with the algorithm off.

Hearing aids first start to oscillate when they are in the unstable stage. Consequently, obtaining an appropriate MSG value requires measuring the gain increase until oscillation starts and the gain decrease until the oscillation stops. During this performance measuring process, we need a criterion to determine whether the oscillation is occurred at a specific moment.

One criterion, called the PCR, was introduced for this purpose. PCR measures the degree to which a large amount of power is concentrated at a small range of frequencies. The device output signal is recorded by a microphone while presenting a white noise signal to the hearing aid through a loudspeaker. Following this, the power spectrum of the output signal is divided into frequency bins with a bandwidth of 23 Hz. Among these frequency bins, the five bins having the highest power are selected; the ratio between the fraction of the total power contained in those five bins and the total power is defined as PCR given as

$$\mathrm{PCR} = \frac{P_5}{P_{\mathrm{total}}}, \tag{2}$$

where P_5 is the power of 5 bins with highest amplitude, and P_{total} is the total power of the output signal. Note that HEI defined as the primary feedback detection criterion as a moment when PCR is equal to 0.5, based on their informal listening tests.

In addition, PCR reduction (PCRR) defines how effectively the feedback canceller reduces sub-oscillatory peaks and is given as

$$\mathrm{PCRR} = \frac{\mathrm{PCR_{off}} - \mathrm{PCR_{on}}}{\mathrm{PCR_{off}}} \times 100, \tag{3}$$

where $\mathrm{PCR_{on}}$ is the PCR value with the feedback algorithm on, and $\mathrm{PCR_{off}}$ is the PCR value with the algorithm off.

3. Transfer function estimation

The main focus of this proposed feedback detection method is on transfer function variation. As a first step in feedback detection, transfer function estimation is required, which can be assumed as a system identification problem. The general concepts regarding system identification, and the finite impulse response (FIR) Wiener filter, which is used in the proposed method, will be overviewed.

3.1. System identification

System identification can be defined as a series of mathematical tools and algorithms that build models of any unknown systems from measured input and output data. The process may be categorized into several basic steps: the design of the experiment, choice of an appropriate model, and the estimation of the model parameters. During these steps, input signals should be broadband deterministic excitation signals that cover the range of the operating frequency band. Once the relevant frequency components are determined, it is required to concentrate on the energy and make a noise study within this band. While measuring the input and output signals, systematic measurement errors such as leakage, alias, and uncalibrated measurement devices should be avoided as much as possible. After the measurement, model parameters can be estimated using the selected algorithm.

3.2. FIR Wiener filter for system identification

The discrete form of the Wiener filtering problem [11–13], shown in Fig. 2, is to design a filter that will produce the minimum mean-square error estimate, $\hat{d}(n)$, of the desired signal, $d(n)$. The cost function is written in mathematical form as

$$\xi = E[e(n)^2], \tag{4}$$

where $e(n) = d(n) - \hat{d}(n)$.

If we assume the unit sample response of the Wiener filter is $w(n)$, the $W(z)$ can be rewritten as

$$W(z) = \sum_{n=0}^{p-1} w(n) z^{-n}, \tag{5}$$

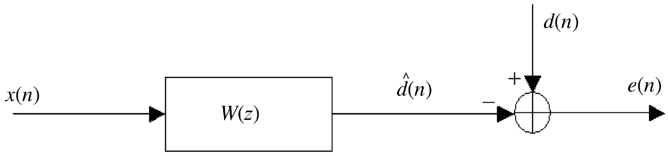

Fig. 2. Illustration of the FIR Wiener filtering problem. $x(n)$ is the input signal and $d(n)$ is the desired signal, $\hat{d}(n)$ is the estimated signal by FIR Wiener filter, $W(z)$ and $e(n)$ is the error signal.

where $(p-1)$ represents the filter order. Additionally, the filtered output denoted by $\hat{d}(n)$, can be defined as the convolution of $w(n)$ with $x(n)$, as

$$\hat{d}(n) = \sum_{l=0}^{p-1} w(l)x(n-l). \tag{6}$$

The Wiener filter design problem for system identification is used to find the filter coefficients $w(n)$ that minimize the mean-square error, ξ as

$$\xi = E[e(n)^2] = E[(d(n) - \hat{d}(n))^2]. \tag{7}$$

In order to find the optimum value that minimizes the mean-square error ξ, the equivalent form of ξ in Eq. (7) has to be differentiated by $w(l)$:

$$\frac{\partial \xi}{\partial w(l)} = -2r_{\mathrm{dx}}(l) + 2\sum_{m=0}^{p-1} w(m)\, r_x(m-l), \tag{8}$$

where $r_x(k)$ is the auto-correlation function of $x(n)$, $r_{dx}(k)$ is the cross-correlation function between $d(n)$ and $x(n)$ with time delay k. If Eq. (8) is equal to zero, then the filter coefficient $w(m)$ is the optimum value that minimizes the mean-square error $e(n)$:

$$\sum_{m=0}^{p-1} w(m)\, r_x(m-l) = r_{\mathrm{dx}}(l), \quad l = 0, \ldots, p-1. \tag{9}$$

When the input $x(n)$ is the white noise or impulse, the optimum filter coefficients, $w(m)$, can be easily obtained because the autocorrelation of those inputs are the impulse function.

4. Proposed feedback detection method

4.1. Concept of proposed feedback detection method

The proposed feedback detection method is based on the transfer function variation of a system. It is known that feedback drives the system into unstable stage; when a system becomes unstable, its transfer function can significantly change within a specific frequency range from that in the stable stage, as illustrated in Fig. 3. If the gain difference is compensated for with respect to the stable transfer function, information about the degree of variation occurring in that unknown stage can be obtained. Standards for the stability classification are required to set up a feedback detection criterion based on the observation in a fixed experimental environment. In this context, the FIR Wiener algorithm in Section 3 is used for estimating transfer functions.

4.2. Experimental setup

The experimental setup illustrated in Fig. 4 is designed to apply the proposed method for evaluation of the feedback cancellation algorithm. Every measurement is carried out in a fixed experimental setup in order to maintain the same acoustic feedback pathway. A hearing aid is connected with tubing, an ear mold with venting, a $2\,\mathrm{cm}^3$ coupler, and a cable socket to interface with the control software. These components are all

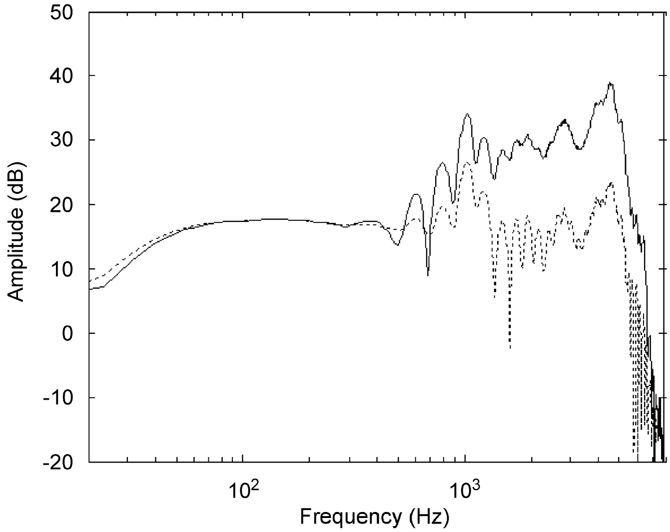

Fig. 3. Transfer functions of a hearing aid in two different stages (dashed lines for transfer function in stable stage; solid lines for transfer function in unstable stage).

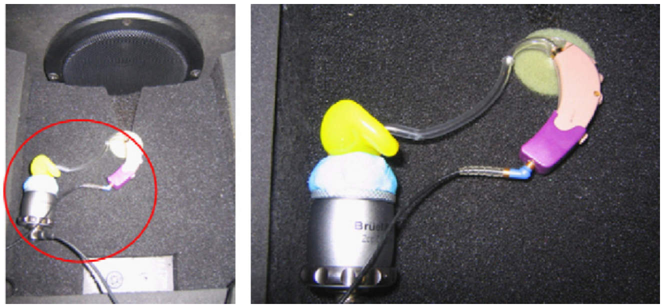

Fig. 4. Experimental setup inside an anechoic test chamber (Type 4232, B&K) and conections between hearing aid equipments. The magnified region inside red circle is represented on the right.

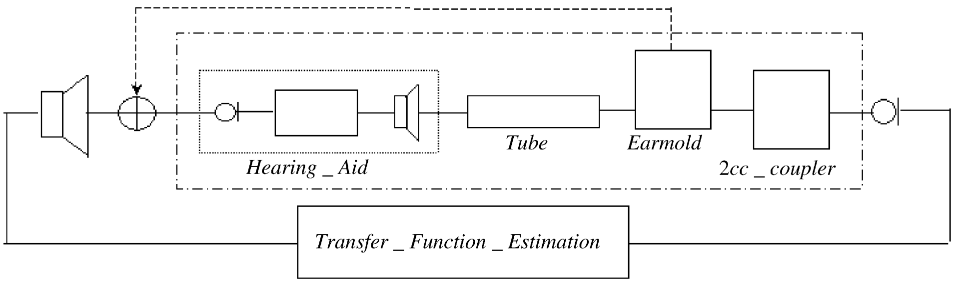

Fig. 5. Block diagram of experimental setup in Fig. 3. Dashed lines represent the acoustic feedback and the components in dash-dotted square are assumed as a system–human ear wearing a hearing aid.

located in an anechoic test chamber (Type 4232, B&K). A loudspeaker and a microphone are used to present the input signal to the hearing aid, and collect the resulting output signal.

Fig. 5 shows the block diagram of the experimental setup in Fig. 4. Components inside the dash-dotted square—the hearing aid, tubing, ear-mold and 2 cm^3 coupler—are considered the system for transfer function estimation, as these components simulate a human ear wearing the hearing aid. Note that the dashed lines represent acoustic feedback caused by the vent in the ear mold.

The loudspeaker presents the input signal, and the microphone connected to the 2 cm^3 coupler measures and records the output signal from the hearing aid. The input signal should be white noise, a broadband deterministic excitation signal, for the best performance of the transfer function estimation. In these experiments, the sampling frequency was set at 16 kHz due to the fact that the focus of hearing aids is on the informative parts of speech, with a maximum frequency of approximately 7 kHz. The transfer function was obtained from the input and output signals using the FIR Wiener filter algorithm. In this case, the transfer function in the stable stage was obtained with an initial gain set with respect to the audiogram, as illustrated in Fig. 6. While increasing the gain in 1 dB step, transfer functions in various stages of the stable, sub-oscillatory, and unstable gain, could be obtained. Note that during the measurement of these transfer functions, all other algorithms in the hearing aid were turned off. The compression, directional microphone, noise reduction and speech enhancement algorithms are turned off for evaluating the performance of the feedback cancellation technique [14]. Consequently, the maximum gain difference between the feedback cancellation algorithm on and off states would explain the feedback cancellation performance.

The radius and shape of vent, length of tubing and location of microphone inside the hearing aid are important parameters that affect the results. The vent used in this paper was a straight type with radius 0.8 mm, and the length of tubing was 45 mm. The location of the microphone inside the hearing aid was at the center of the test box and facing the loudspeaker.

4.3. Transfer function variation criterion

There are several procedures in the proposed hearing aid feedback detection method. First, the transfer function in its initial stable stage is estimated by using the FIR Wiener filter algorithm. The initial gain is set with respect to the audiogram in Fig. 6. Second, it is required to incrementally increase the hearing aid gain in 1 dB steps in the overall frequency range by using control software, and the hearing aid transfer function in each gain stage is estimated. Third, the gain difference between the initial stable gain and the increased gain is calculated as

$$\mathrm{Gain}_{\mathrm{diff}}(\mathrm{dB}) = \mathrm{Gain}_{\mathrm{inc}}(\mathrm{dB}) - \mathrm{Gain}_{\mathrm{init}}(\mathrm{dB}), \tag{10}$$

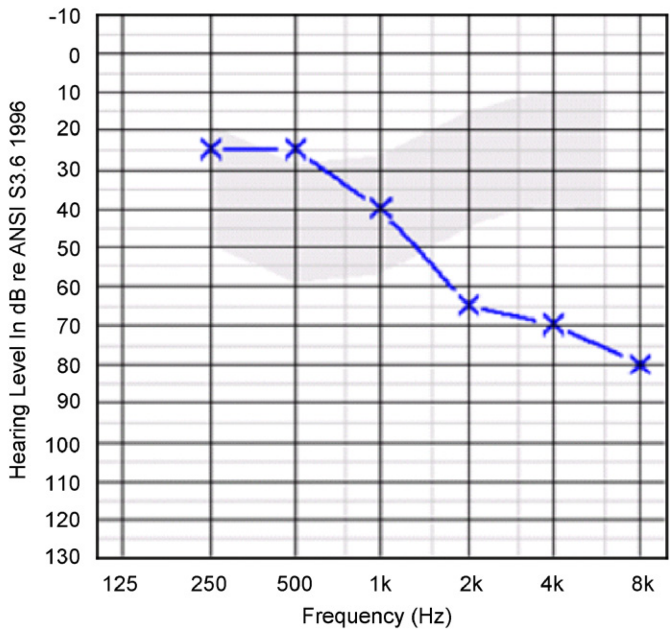

Fig. 6. Audiogram for general hearing loss caused by aging.

where $\text{Gain}_{\text{diff}}$ is gain difference, Gain_{inc} is the increased gain in certain stages, and $\text{Gain}_{\text{init}}$ is the initial gain in a stable stage. Eq. (10) shows that the gain difference is frequency independent, whereas the initial gain is frequency dependant. Fourth, this frequency independent gain difference is added to the initial stable transfer function for direct comparison with those in the increased gain stage. Fifth, transfer functions in certain increased gain stages are subtracted by the initial stable transfer function to which those gain differences are added. Sixth, these subtracted transfer functions are limited to the specific frequency boundary from 1 to 5 kHz since the frequency range of feedback is restricted to high frequencies in that range. These resultant graphs represent the difference between the transfer function in specific stages and the gain-adjusted stable stage, as shown in Fig. 7. These resultant functions are referred to as the transfer function variation function (TVF). These TVFs are obtained using off-line procedures by recording input, output signals and estimating transfer functions in all stages.

TVFs of a BTE-type hearing aid are illustrated in Fig. 7. As can be seen in the figure, TVFs in various stages give information about the degree of change in transfer functions as a function of the gain setting. These TVFs can also be used as a measure of stability. Peaks of the TVFs can be observed in order to determine criterion for categorizing stable, sub-oscillatory and unstable stages of gain. When feedback oscillation occurs, at least one peak over ± 10 dB is observed in the TVF; there is no peak exceeding ± 10 dB when no feedback oscillation occurs. The stable stage allows for the exact relationship between the maximum TVF peak and hearing aid gain to be observed. Furthermore, it can be seen that sudden and abnormal changes of the transfer function just before the oscillation (unstable) stage drives hearing aids into the sub-oscillatory stage; however, this stage shows no relationship between gain increase and maximum peak value of TVF. Despite a 1 dB gain increase, the maximum peak value unexpectedly changes. These abrupt changes occur when peaks exist between ± 5 and ± 10 dB. At this moment, the hearing aid does not yet enter the unstable stage. This phenomenon provides some explanation for the sub-oscillatory stage in which hearing aid users report or perceive a difference of sound quality, just before the unstable oscillating stage.

Three measurement results for two different BTE hearing aids, produced by two different companies, are obtained by using the above experimental setup. As explained in Section 4.2, the compression, directional microphone, noise reduction and speech enhancement algorithms are turned off for both hearing aids. The maximum stable gains of those two hearing aids are characterized in Table 1 when they are fitted for the audiogram presented in Fig. 6 without feedback cancellation. Three measurements demonstrate consistent results, due in part to the fixed experimental setup. Table 2 shows the transfer function variation using the number of peaks in TVFs over the specific criteria, ± 5 and ± 10 dB, as was previously determined. During the experiments, both hearing aids started to generate oscillating sounds when peaks are present over the ± 10 dB criterions. The gain in this oscillating moment is referred to as the reference of gain difference. It was determined that sudden changes occur when peaks are present between ± 5 and ± 10 dB in the TVF, and that hearing aids are stable when no peak exceeds ± 5 dB.

As a result of this effort, a new feedback detection criterion has been developed from the observation of these TVFs. The new feedback detection method is called the transfer function variation criterion (TVC). Using TVF, TVC is clearly defined as

$$\text{TVC} = \max(|\text{TVF}|). \tag{11}$$

TVC is not only a feedback detection method, but a classification scheme for the three feedback stages in hearing aids (stable, sub-oscillatory and unstable stages). TVC thresholds for each stage are 5 and 10 dB:

$$\begin{cases} \text{TVC} < 5\,\text{dB}, & \text{stable stage,} \\ 5\,\text{dB} \leqslant \text{TVC} < 10\,\text{dB}, & \text{sub-oscillatory stage,} \\ \text{TVC} \geqslant 10\,\text{dB}, & \text{unstable stage.} \end{cases} \tag{12}$$

As mentioned in first paragraph of this section, this paper obtains TVF and implements TVC off-line. However, the proposed performance evaluating algorithm, TVC, can be implemented in real-time as it has the potential to implement the real-time tracking of the transfer function using a digital signal processor. Once the transfer function in the initial stable stage is obtained and memorized in physical memory, TVF can be given by estimating the transfer function in a certain gain stage. Furthermore, TVC can be implemented by setting

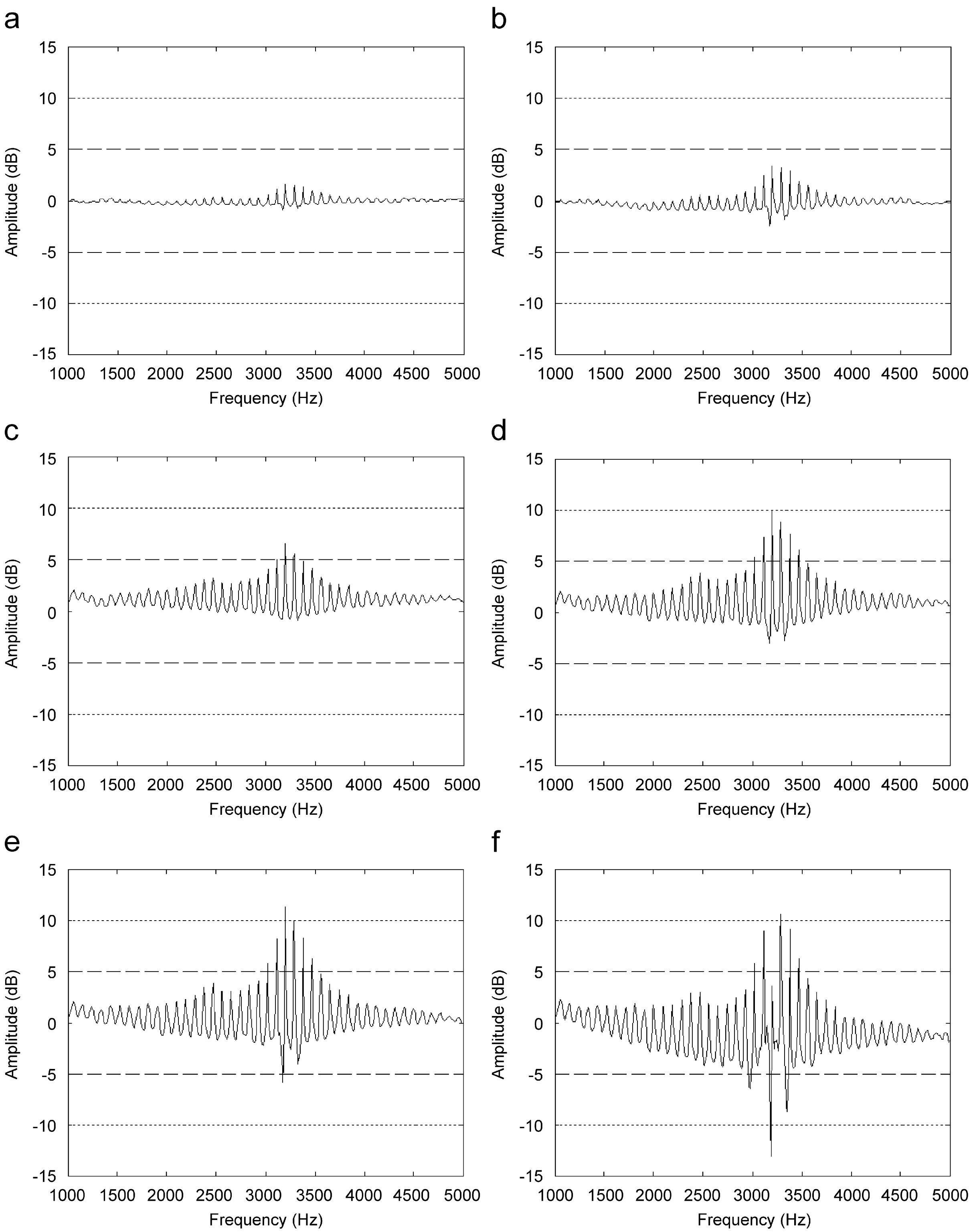

Fig. 7. Transfer function variation function (TVF). Differences between transfer functions in specific stages and gain adjusted stable stage with criterion for each stage (dashed lines—± 5 dB criterion for sub-oscillatory stage; dotted lines—± 10 dB criterion for unstable stage): (a, b) results from stable stage, (c, d) results from sub-oscillatory stage, (e, f) results from unstable stage.

Table 1
Maximum stable gain (MSG) without feedback cancellation technique

		Center frequency (Hz)				
		500	1000	2000	4000	8000
MSG (dB)	Hearing aid I (Company A)	10	16	26	27	26
	Hearing aid II (Company B)	9	17	25	29	26

Table 2
Results of transfer function variation expressed by the number of peaks in TVFs over the specific criteria: ± 5 and ± 10 dB (gap from the feedback oscillation is the gain difference)

Gain difference (dB)	Hearing aid I (Company A)					Hearing aid II (Company B)				
	−10	−5	5	10		−10	−5	5	10	
−5	0	0	—	0	0	0	0	—	0	0
−4	0	0	—	0	0	0	0	—	0	0
−3	0	0	—	2	0	0	0	—	0	0
−2	0	0	—	5	0	0	0	—	1	0
−1	0	0	—	6	0	0	0	—	1	0
0	0	1	—	6	1	0	0	—	1	1
1	0	2	—	6	2	0	0	—	1	1
2	1	2	—	7	2	0	0	—	1	1
3	1	3	—	5	1	0	0	—	2	1

stability criterion based on the obtained TVF in real-time. However, it should be noted that the input signal for hearing aids should always be a white noise signal.

5. Comparison between PCR and TVC methods

PCR is a simple method of detecting the feedback signal and is generally used in the evaluation of hearing aids. However, there are several problems associated with its use as a performance standard as the result of applying PCR is based on the shape of the output spectrum. If the power of input white noise signal varies, the shape of the output spectrum changes causing the PCR threshold to arbitrarily change due to the fact that the ratio between highest five bins and the total power of the output signal spectrum is easily changeable with respect to the input white noise signal level. In addition, the power of the five bins cannot be predicted when there are feedback oscillations. As such, the PCR approach sometimes misses feedback oscillations because of this dependency on the input white noise signal level.

The proposed TVC method is based on a transfer function that is independent of the input white noise signal power as the transfer function is the system characteristic regardless of the input signal level. Furthermore, TVC can be used as a classification method of the three stability stages in hearing aids, stable, unstable and sub-oscillatory stages as well as a feedback detector.

Fig. 8 represents a comparison between PCR and TVC when feedback oscillation has occurred. In this case, the output spectrum and TVF are obtained to check whether the PCR and TVC can detect the feedback oscillation or not. The PCR value is described in Section 2 and calculated by Eq. (2) based on the hearing aid output spectrum in Fig. 8(a). As shown in Fig. 8(b), TVC detects the feedback oscillation with a 10 dB criterion for the unstable stage, while the PCR value 0.44 obtained for the output signal was less than the intended threshold 0.5 for the feedback detection. This result suggests that the TVC method is considered a more accurate and reasonable feedback detection scheme than the PCR method. Fig. 9 clearly summarizes the proposed TVC feedback detection method.

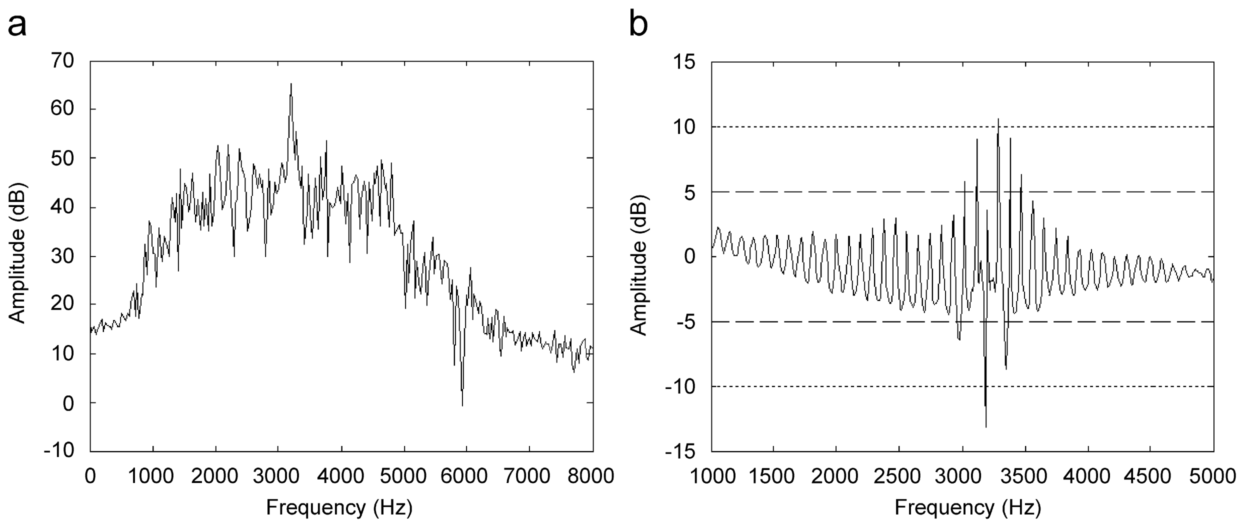

Fig. 8. Comparison between PCR and TVC when feedback oscillation occurs: (a) output spectrum (PCR value is 0.44), and (b) transfer function variation function.

Proposed stability classification method

Transfer function Variation Criterion (TVC)

• **Definition of TVC**

$$TVC = \max(|TVF|)$$

• **Transfer function Variation Function (TVF)**

① Estimation of the transfer function in its initial stable stage.
② Estimation of transfer functions with increasing the gain by 1dB step size.
③ Subtracting the gain applied transfer function in initial stabel stabe from that in unstable state.
④ Limiting the subtracted transfer function into frequency boundary from 1kHz to 5kHz.

• **Thresholds for each stage**

$$\begin{cases} TVC < 5dB & \text{stable stage} \\ 5dB \le TVC < 10dB & \text{sub - oscillatory stage} \\ TVC \ge 10dB & \text{unstable stage} \end{cases}$$

Fig. 9. Summary of proposed transfer function variation criterion (TVC) stability classification method.

6. Conclusion

This paper proposes a new feedback detection method based on transfer function variation. Additionally, an experimental setup design for obtaining the consistent evaluation results was proposed. In this design, TVF is introduced to effectively observe the variation of transfer function. Two hearing aids produced by two different companies were evaluated for the verification of the proposed method. It was determined that this proposed method, TVC, functioned not only as a feedback detector but classifier of the three hearing aid stability stages: stable, unstable, and sub-oscillatory stage. In particular, the sub-oscillatory stage was clearly classified by the proposed TVC method; it provided a fair and precise evaluation of hearing aid performance. The TVC method gave more reliable evaluation results of feedback detection than the conventional PCR method for detecting hearing aid oscillations. However, the TVC method is based on the condition that the hearing aid under test has no nonlinear operating element whereas the PCR has no such constraints as it operates on the output signal itself. This method was implemented off-line. Despite these limits, the possibility for real-time implementation was discussed. Potential uses of the TVC method could be a feedback for detecting switches in unattended sound systems, or any acoustic system, due to its ability to detect sub-oscillatory stages at which sound quality distortion occurs, as well as the onset stage of acoustic feedback.

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