A study on the Adaptive Suppression of Howling in Hearing Aids(補聴器における適応ハウリング抑圧手法に関する研究)

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論 文 内 容 要 旨

1 Introduction

Effective communication requires the proper interaction between the information transmitter and the receiver. Damage to either the receiver or the transmitter will greatly impair the communication process. Hence, it is important not only that the transmitter transmits the information well, but that the receiver can receive and understand the information.

Hearing corresponds to the reception half of the acoustic communication process. Unfortunately, impairment to the hearing mechanism is not an uncommon phenomenon, especially in these times when the advancements in technology had also increased environmental noise. Hearing aids offer a means of alleviating this problem when medical treatment is no longer possible.

With the support of technology and a better understanding of the hearing loss phenomenon, better hearing aids can now be designed to give the hearing aid user better hearing comprehension. However, a lot of problems still remain in the modern hearing aids. One of these, howling, is a common problem particularly in high-gain ear-level and ear-tube type hearing aids. Howling is not only annoying, but more importantly, it reduces the maximum usable gain of the hearing instrument.

In the recent decades, much excitement was brought about by the introduction of digital hearing aids. This promises a lot of benefits, including better control over the hearing aid response. Among this is the capability of applying powerful signal processing techniques to solve the signal processing problems in the hearing aid. One of these is the concept of adaptive systems where the system adjusts itself, and thus making it robust, to the changes in the operating environment.

In consideration of this, the use of adaptive digital signal processing for howling suppression in hearing aids was considered in this study. Using this, this study aimed at developing an adaptive howling suppression method that can provide a high stable gain, cause minimal signal degradation, and work for all sound audible and significant to normal listeners.

2 Problems in Howling Suppression in Hearing Aids

In this chapter, a survey of conventional methods for preventing/suppressing howling by signal processing, proposed in different fields of study, and the problems underlying their use and implementation were presented. Among these, the use of the feedback canceler was singled out since aside from suppressing howling, it also has the advantage of reducing background noise due to the presence of the feedback signal. This method is based on the idea that howling is caused by the presence of the feedback signal. Hence, howling can be prevented if the feedback signal can be cancelled. In this method, an estimate of the feedback signal is obtained by filtering the output signal through a fixed filter that approximates the feedback response. This is then subtracted from the signal at the input terminal to cancel the feedback signal. This method, however, depends on how much the fixed filter approximates the feedback path. Being a fixed filter, a significant problem with this method is then its vulnerability to the changes in the feedback response.

The problem with the use of a fixed filter for feedback cancellation can be corrected with the use of an adaptive scheme. This leads to the method generally referred to as adaptive feedback

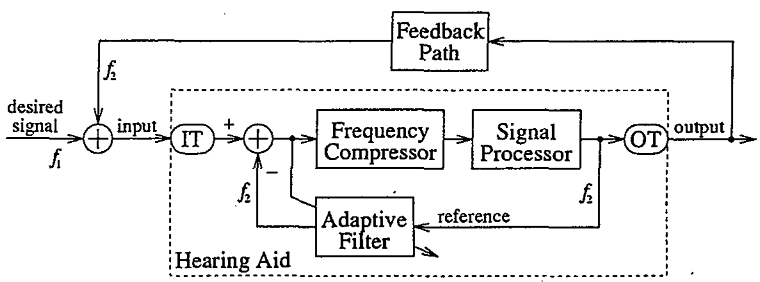

Fig. 1 Block diagram of the proposed adaptive feedback canceler with frequency compression AFC/CP (IT - Input Transducer, OT - Output Transducer).

cancellation (AFC). In adaptive feedback cancellation, an adaptive filter is used in the place of the fixed filter. In this method, the adaptive feedback canceler adjusts its coefficients to eliminate all components in the noisy input signal that are correlated with a reference signal. Hence, a necessary requirement for the optimum use of adaptive feedback canceler is for a referrence signal that is correlated with the feedback signal but not with desired components of the input signal. Such a signal is hard to obtain in hearing aids. A possible candidate would be the output signal which is correlated with the feedback signal. However, for linear systems, the output signal is also correlated with the input signal. The use of the unprocessed output signal as reference signal would then cause the unnecessary attenuation of certain components, particularly relatively steady-state tonal components, of the input signal. Thus, to use the output signal as reference signal requires a method for sufficiently removing its correlation with the input signal while retaining its correlation with the feedback signal, and without introduction of very large degradations in the output signal.

3 Howling Suppression by Adaptive Feedback Cancellation with Frequency Compression

Theoretically, two pure-tone signals of different frequencies are uncorrelated. Hence, shifting each frequency component of a signal would remove its correlation with the original signal. To avoid destroying the harmonic structure of the signal, the shift in frequency should be made proportional to the frequency of the component. Such a process is commonly referred to as frequency compression or expansion depending on the direction of the shift in frequency. Based on this, the idea of frequency compressing the output signal to remove its correlation with the input signal was proposed. The block diagram of the proposed adaptive howling suppression system is shown in Fig. 1 . As can be seen, the frequency compressor was used as a preprocessor to the hearing aid signal processor. Doing this would decorrelate the output signal with the input signal without affecting its correlation with the feedback signal. This would allow the use of the output signal as reference signal for the adaptive feedback canceler.

In the proposed system, the component of the input signal of frequency f_1 will be converted to another frequency f_2 by the frequency compressor (cf. Fig. 1). The shifted frequency component will then be output by the system and part of which will return as feedback signal. The next segment of input will then contain two components, the original f_1 and the feedback f_2. The adaptive filter will then adjust its coefficients to cancel the components of the input that are correlated with the reference signal, *i.e.*, of frequency f_2. Since the original signal has a frequency f_1, it is uncorrelated with the reference signal. Thus, this signal component will pass through the system without attenuation. Hence, it can be seen that the inclusion of a frequency compressor in the AFC/CP would allow the system to process signals without attenuating the input signal even if it is periodic.

The performance of the proposed system was then investigated via computer simulation. Results showed that when the reference signal is tonal, the adaptive canceler operates as an adaptive notch filter and cancels frequency components near the reference signal frequency. From an analysis of this condition, it was shown that the attenuation of tonal signal components can be limited by the proper choice of adaptation step size and compression ratio. Performance evaluation results showed that with the introduction of the frequency compressor into the adaptive feedback cancellation system, tonal signal components no longer get unnecessarily attenuated.

Since the frequency compressor was primarily introduced to decorrelate the output signal with the input signal, as expected, no significant difference in howling margin increase was found between the use of adaptive feedback cancellation with and without frequency compression. Both methods showed an increse in howling margin of about 18 dB. This limitation in the howling margin was considered to be due to problems inherent in the adaptive feedback cancellation scheme that include the signal-to-noise ratio at the adaptive canceler input and modeling limitations.

4 Adaptive Howling Suppression by Notch Filtering

Theoretically, if the feedback signal was completely canceled by the adaptive feedback canceler, the system should have an infinite howling margin. However, it was shown in the previous chapter that due to limitations in the practical implementation, the howling margin increase delivered by the adaptive feedback canceler under stable operation is limited to about 18 dB.

For cases when higher gain is needed, the use of an adaptive implementation of notch filtering in conjunction with the adaptive feedback canceler was considered. In this system, the adaptive feedback canceler is first made to converge. This would reduce the number of 0-phase frequencies, and thus, the number of possible howling frequencies. An adaptive notch filter is then used to detect and identify the remaining howling frequencies. Notch filters are inserted to reduce the system gain at these frequencies, thereby suppressing the howl. The insertion of notches in the system would cause some loss in information. However, with the use of narrow notches, the loss in information is considered to be negligible.

Fig. 2 shows the block diagram of the adaptive howling suppression system using notch filters. In the proposed system, a howl detector-identifier is used to monitor the incoming signal. From this, it checks the onset of howling by comparing the signal power of the largest frequency component. Howling is judged to have occurred when this exceeds a certain threshold level, set near the maximum output capability of the hearing aid. Furthermore, another purpose of the howl detector-identifier is to identify the howling frequency. Once the howling frequency has

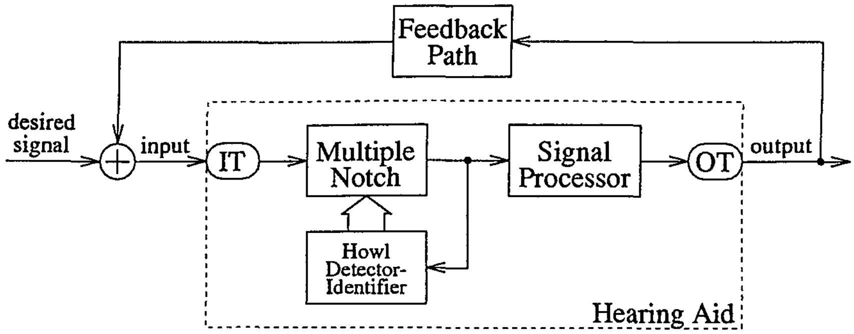

Fig. 2 Block diagram of adaptive howling suppression system using notch filtering.

been identified by the detector-identifier, this information is sent to the multiple notch block where a notch filter is then created at the identified howling frequency. This process is repeated, with the notch filters cascaded in the multiple notch block, until all the howling frequencies have been identified, and howling suppressed.

Three types of notch filters were investigated for use in this method: the 2-pole 2-zero notch filter, the frequency-sampling structure transversal notch filter, and the quasilinear-phase notch filters. Careful investigation showed that when among the different notch filters considered, the quasilinear-phase notch filter, IIR and FIR, designed using a rectangular window, can provide the highest howling margin when applied to the suppression of howling. For example, at a sampling frequency of 16 kHz, a quasilinear-phase IIR notch filter having a group delay of 10 ms and a bandwidth of 50 Hz can be designed to provide a howling margin of around 11 dB, while one having a bandwidth of 100 Hz can provide a howling margin of around 26 dB.

5 Conclusion

This chapter gives the conclusions of the study along with some suggestions on further works that can be done to improve the performance of the adaptive howling suppression methods proposed.

審　査　結　果　の　要　旨

高齢化社会の進行に伴って，老人性難聴者の急増が予想されている。このような状況の下で，快適な補聴器の開発は極めて重要な課題である。現在の補聴器の快適性を損なっている大きな問題のひとつにハウリングがある。ハウリングは，音質を損ない，不快であるばかりか最大利得を制限するため，十分な補聴が実現できない原因ともなる。著者は，補聴器におけるハウリングを抑圧するのに有効なディジタル信号処理手法を開発するための研究を行ってきた。本論文は，その研究の成果をまとめたもので，全編5章からなる。

第1章は序論である。

第2章では，まず，補聴器の音響伝達系が短いため，これに音場用のハウリング抑圧手法を用いることは困難であることを指摘している。次いで，最も有望な信号処理手法として，適応エコーキャンセラに着目し，これを補聴器に適用する場合には，帰還信号とは高い相関を持ち，入力信号とは無相関な参照信号が必要となることを指摘している。

第3章では，上記の問題を解決するため，補聴器出力信号を周波数圧縮することによって無相関化し，これを適応エコーキャンセラの参照信号とするという新しい方式の提案を行っている。実際の補聴器の音響特性を測定した上で，詳細な計算機シミュレーションを行い，6％程度の周波数圧縮によって，音質を損なうことなく，有効な無相関化が可能となること，また，この無相関化による適応エコーキャンセラを用いることによって，典型的な補聴器の場合でハウリングマージンを約18dB増加することができることを述べている。これは，実際に補聴器のハウリングを抑圧することができる有効な方式の提案を行ったもので，高く評価できる。

第4章では，前章で提案した適応エコーキャンセラだけではハウリング抑圧効果が不十分な場合に，更に適応ノッチフィルタを用いることによってハウリング成分を取り除くという処理方式の提案を行っている。各種のノッチフィルタについて検討した結果，零位相ノッチフィルタの時間軸を移動させた上で窓掛けを行って作製した準直線位相フィルタを用いた場合には，帯域幅100Hz，群遅延10msの場合に，利得余裕を更に26dB改善できることを示している。これは，極めて高い利得を必要とする補聴器を実現する上で，有用な知見である。

第5章は結論である。

以上要するに本論文は，補聴器のハウリングを効果的に抑圧するためのディジタル信号処理手法の提案を行って，その有効性を示したもので，音響情報工学，通信工学の進展に寄与するところが少なくない。

よって，本論文は博士（工学）の学位論文として合格と認める。