HEARING AID HOWLING SUPPRESSION BY ADAPTIVE FEEDBACK CANCELLATION WITH FREQUENCY COMPRESSION

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ABSTRACT The use of adaptive feedback cancellation to prevent howling requires a reference signal that is correlated with the feedback signal but is not correlated with the input signal. Such a signal is hard to obtain in hearing aids. In this paper, the use of frequency compression to decorrelate the output signal with the input signal for use as reference is presented. Performance evaluation results indicate that with the proper choice of system parameters, the use of this system can provide a significant increase in howling margin with minimal deterioration in output signal quality.

1 INTRODUCTION

As with public address systems, howling is not an uncommon problem in hearing aids. A good way of preventing howling is by canceling the feedback signal which also reduces the signal deterioration caused by the feedback signal. Here, we consider adaptive feedback cancellation due to its resistance to changes in the operating environment. In adaptive feedback cancellation, an adaptive canceler is used to estimate and subtract the feedback signal from the microphone signal. The adaptive canceler, however, requires a reference signal that is correlated with the feedback signal but not with the input signal.

In this paper, we describe the use of a frequency compressor to remove the correlation between the output and input signals to allow its use as reference signal for the adaptive canceler. Results of computer simulations to verify system performance are also presented.

2 ADAPTIVE FEEDBACK CANCELLATION WITH FREQUENCY COMPRESSION

2.1 DECORRELATION BY FREQUENCY COMPRESSION

Since two pure-tone signals of different frequencies are uncorrelated then shifting each frequency component of a signal by an amount proportional to the frequency of the component (frequency compression/expansion) would remove its correlation with the original signal without introducing harmonic distortion. Since it is more common to have greater residual hearing in the low-frequency range, shifting frequency components towards the low

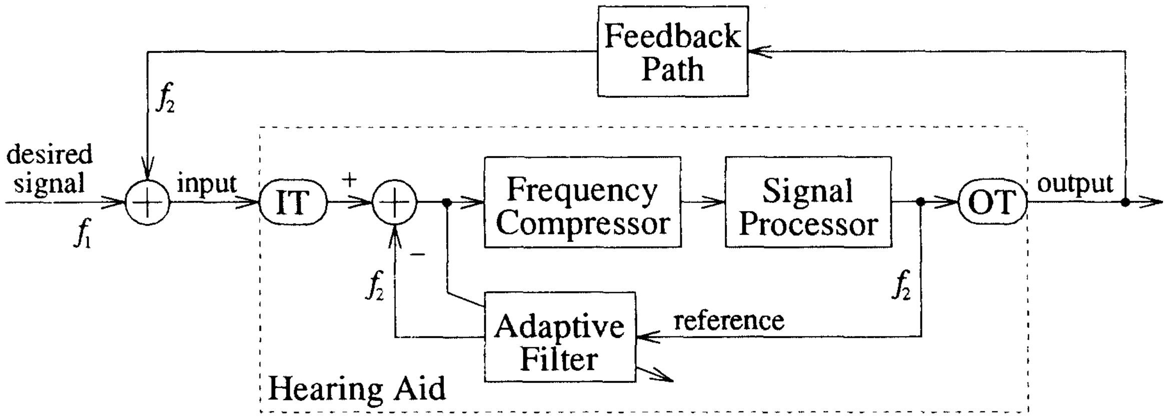


Figure 1 Adaptive feedback canceler with frequency compression.

frequencies (frequency compression) would be of more use than frequency expansion. Also, frequency expansion would produce frequency components above the Nyquist limit, causing loss of information/cues.

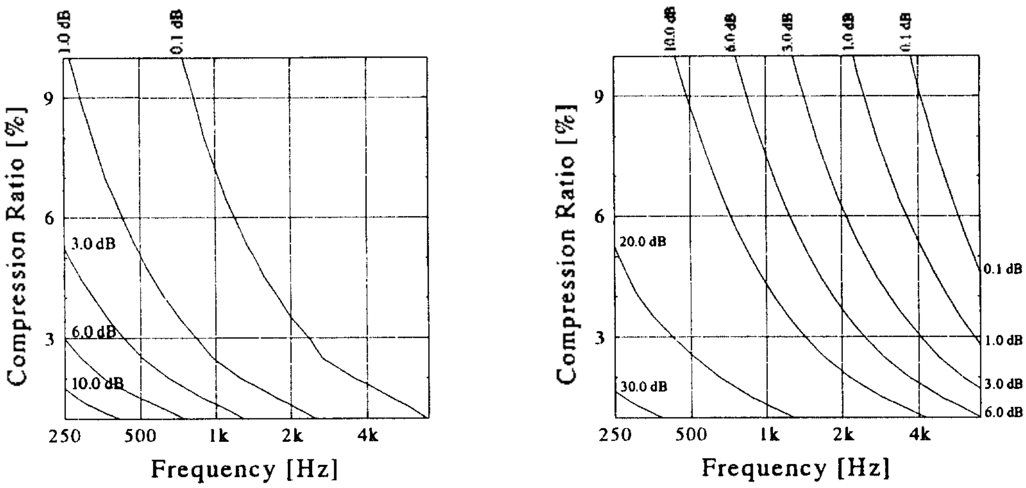
2.2 HOWLING SUPPRESSION SYSTEM

The block diagram of the adaptive feedback canceler with frequency compression (AFC/CP) is shown in Fig. 1. The hearing aid is composed primarily of the input (microphone IT) and output (receiver OT) transducers, and a signal processor for hearing-loss compensation. In the proposed system, the frequency compressor is used as a pre-processor to the signal processor to remove the correlation between the output and input signals. Also, being a pre-processor, it would not affect the hearing-loss compensation processing.

In this study, frequency compression was performed in the time domain using the "sampling method"[1, 2] where the input signal is sectioned into segments and resampled at $f_s/(1+r)$. The compression ratio r $(0<r<1)$ governs the amount by which the frequency components will be shifted. The resampled segments are then stretched to obtain the original sampling interval and extra data points are overlapped with the next segment. Listening tests on three overlap functions (discard, linear, and half-cosine) indicated very little recognizable differences for low compression ratios.

3 SYSTEM PERFORMANCE EVALUATION

The performance of the proposed system was then investigated theoretically and by computer simulation. In the simulation, all operations were performed with floating-point arithmetic. For simplicity, the hearing aid signal processor used in the study had a flat gain and a fixed group delay of 5 ms. The input and output transducers were modeled using FIR filters with transfer functions measured from actual hearing aid microphone (RION EU 08) and receiver (Knowles ED-3012). The feedback transfer function used was based on Bustamante *et al.*[4]. The adaptive filter used had a tap-length of 32 and was implemented using the normalized least-mean square algorithm[3]. A delay was inserted before the adaptive filter to compensate for the delay in the feedback path. Frequency compression was performed with a segment length of 250 ms (4000 points at 16-kHz sampling).



(a) stepsize = 0.005 (b) stepsize = 0.050

Figure 2 Attenuation level vs compression ratio and frequency.

3.1 TONAL SIGNAL ATTENUATION

Preliminary tests showed that even after frequency compression, there were cases when tonal input signals still get canceled. Investigation showed that when the reference signal to the adaptive canceler is tonal, the adaptive filter weights converge to a dynamic rather than a static solution, causing the adaptive canceler to operate like an adaptive notch filter.

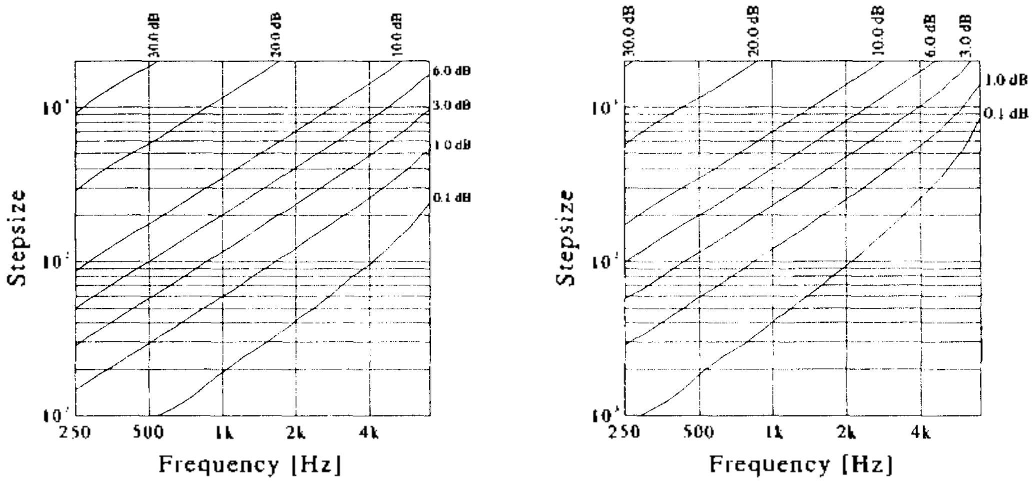
It was shown by Glover that for a periodic reference signal, the adaptive notch filter modulates the reference signal such that frequency components near the reference signal get attenuated[5]. Following Glover's analysis, it can be shown that with the NLMS algorithm, the transfer function $H(z)$ of the adaptive canceler is approximated by

$$H(z) \approx \frac{z^2 - 2z\cos(\omega_2 T) + 1}{z^2 - 2(1-\mu)z\cos(\omega_2 T) + (1-\mu)}, \tag{1}$$

where T is the sampling interval $(1/f_s)$, ω_2 $(= 2\pi f_2)$ is the angular frequency of the reference signal and f_2 is the center frequency of the notch related to the input signal frequency f_1 (cf. Fig. 1) by the compression ratio r in $f_2 = (1-r)f_1$. From this, the 3–dB bandwidth of the adaptive notch filter is found to be directly proportional to the adaptation step size, as approximated by $(\mu/\pi)f_s$.

The amount of tonal signal attenuation when using the proposed system was then investigated. Samples of the results are shown in Figs. 2 and 3. From these, it can be seen that: 1) attenuation is less at high frequencies since the absolute shift in frequency is large there, 2) smaller adaptation step sizes causes less signal attenuation due to the narrowing of the notch filter, and 3) large compression causes less attenuation since the original frequency component is farther from the notch center frequency.

These results indicate that a slow convergence (small step size) and a large compression are desirable. However, in the choice of the adaptation step size, system adaptation would not be able to keep up with the onset of howling if the step size chosen is too small. On



(a) 3% compression

(b) 6% compression

Figure 3 Attenuation level vs adaptation step size and frequency.

the other hand, a large step size would produce a faster convergence at the expense of a larger misadjustment. Though a large compression ratio would lead to less attenuation, it would also cause degradation in signal quality. Initial results of simple listening tests showed barely noticeable effects for compression ratios up to 6%.

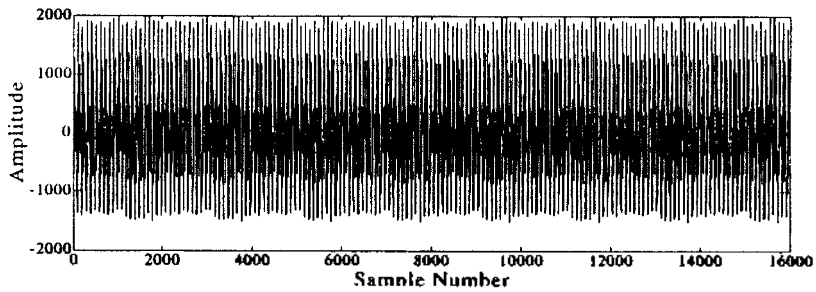
3.2 SIGNAL QUALITY

Figure 4 shows sample sections of the output signals, for the sustained vowel input, with and without frequency compression. The compression ratio used was 6%, while the adaptation step size was set at 0.005. Without frequency compression (AFC), tonal signal components retain their correlation and the initial segment is attenuated. This attenuated signal then comes back as reference signal causing the next segment to pass with little attenuation. This process is repeated causing a warbling quality in the output signal (see Fig. 4(b)). Due to the decorrelation of the output signal, this cannot be seen in the AFC/CP output.

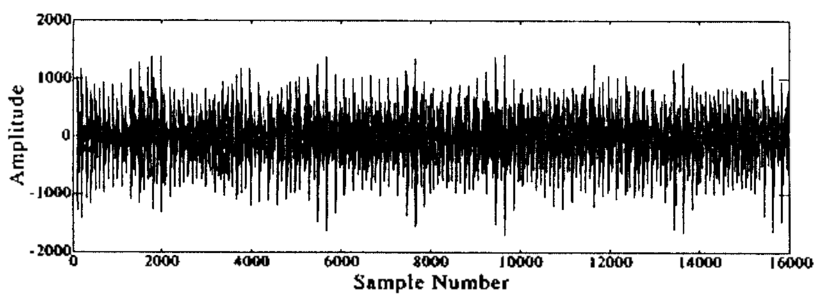
3.3 HOWLING SUPPRESSION

The increase in howling margin for different system parameters, as well as for AFC without compression, was measured using white noise as input signal. Since the frequency compressor is used primarily to prevent tonal signal components from being attenuated, and not for howling margin increase, a similar performance in howling margin is expected for both the AFC and AFC/CP.

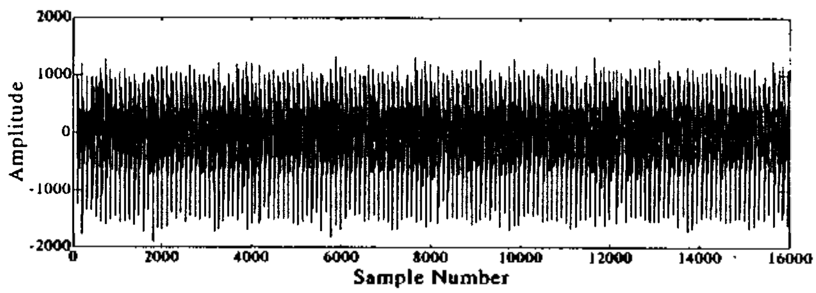
The results are shown in Fig. 5. From this figure, no significant difference in howling margin increase can be seen between the AFC and AFC/CP. Both showed an increase in howling margin of about 18 dB for step sizes between 0.001 and 0.1. Smaller step sizes showed little improvement in howling margin since the system adaptation is too slow to match the onset of howling. On the other hand, too large step sizes makes the misadjustment dominant, causing unstable performance.



(a) input signal



(b) AFC output



(c) AFC/CP output

Figure 4 Processed signal segments for a 5–sec sustained vowel /e/.

4 ADDITIONAL HOWLING SUPPRESSION BY NOTCH FILTERING

Theoretically, the adaptive filter should model the feedback path and thus eliminate the feedback signal. Due to physical limitations, exact modelling is not achieveable. However, we note that with the use of the AFC, the zero-phase crossings of the open-loop response, and thus the number of possible howling frequencies, is reduced. Hence, additional howling margin can be obtained by introducing notches at the howling frequencies not suppressed by the AFC. With narrow notches, the loss in information due to notch filtering is assumed to be unnoticeable.

This method is currently being investigated and more detailed discussion and experimental results are left to a later publication.

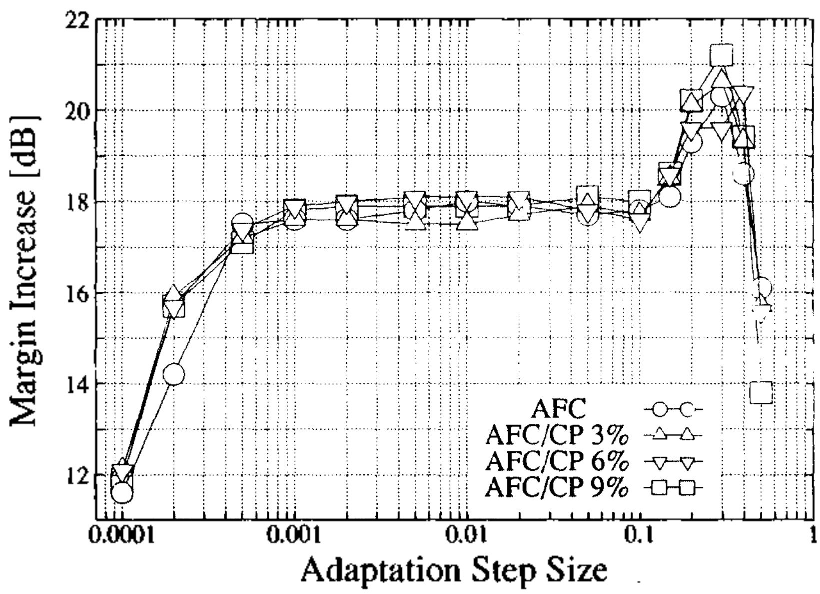


Figure 5 Increase in howling margin introduced by howling suppression as a function of the adaptation step size.

5 CONCLUSION

In this paper, we presented the use of frequency compression to decorrelate the output signal with the input signal to allow its use as the reference signal for the adaptive canceler. Computer simulation results indicated that with the proper choice of system parameters, the use of this system can provide a significant increase in howling margin with minimal deterioration in output signal quality. As an example, 6% frequency compression with an adaptation step size of 0.005 could provide a howling margin increase of about 18 dB with tonal-signal attenuation levels of below 3 dB for frequencies above 250 Hz. The possibility of increasing the howling margin with the use of notch filters was proposed.

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