OPPORTUNITIES FOR ACTIVE NOISE CONTROL IN COMMUNICATION HEADSETS

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1. Introduction

Almost all the current generation of commercial circumaural headsets incorporating active noise reduction (ANR) employ feedback control, with fixed filters and analogue signal processing. This combination provides, at best, a conditionally stable device. Good performance can be obtained at the lower speech frequencies, with some headsets reducing environmental noise at the ear at frequencies of up to 1 kHz. Inconsistent control of environmental noise results, however, when air leaks occur in the seal between the earmuff and the head.[1] These can occur, for example, with poor fitting of the device, or head rotation when wearing the headset.

The variation in coupling between the ear and the secondary source located within the ear cup when the device is displaced relative to the ear suggests an adaptive control system is required that is capable of optimizing its performance while the headset is used.

The signal processing algorithm and device that best meet the requirements of a communication headset remain to be established. The purpose of this paper is to describe the potential for improved speech communication and reduced noise-induced hearing loss in an ANR headset by employing adaptive feedforward control. Details of the ANR produced by the device have been described elsewhere [2,3].

2. Adaptive Feedforward ANR Headset

Earmuff. The earmuff consists of a rigid ear cup lined with sound absorbing material, together with a circumaural cushion that provides an acoustic seal to the head. A microphone is attached to the outside of the muff, senses the sound field surrounding the ANR headset and provides a reference input signal X to the digital controller (see Fig. 1). A miniature loudspeaker is used to generate the secondary sound field in the volume enclosed by the earmuff, and is driven by signal U, which is derived from signal X. A communication signal may also be reproduced by the loudspeaker. A second microphone, located close to the ear canal entrance, provides the error signal, E, used by the algorithm to optimize the control filter.

Control System. A block diagram of the control system is shown in Fig. 1. Environmental noise is transmitted through the acoustic plant, consisting of the earmuff with air leaks, to become the primary noise in the volume enclosed by the muff. Adaptation of the control filter is performed using the filtered-X LMS algorithm.

A dual-rate sampling structure is used to provide short control system latency.[3] This involves over-sampling the acoustic signals at the reference and error microphones, and updating the control signal to the secondary source at a decimated rate of the sampling frequency. The technique effectively reduces the total signal delay in the control path and, at the same time, permits the low-frequency performance of the digital control filter to be improved.

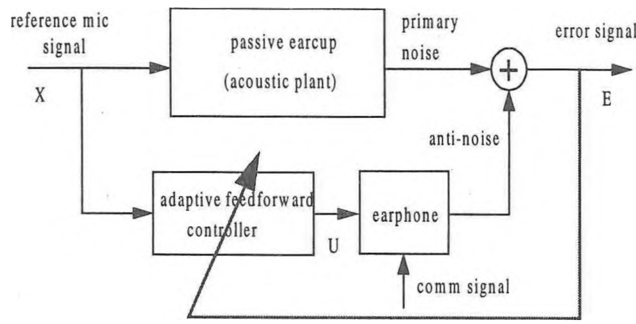

Fig. 1: Control structure.

A separate control system has been constructed for each ear using TMS320C32 floating-point digital signal processors and 16-bit A/D and D/A converters.

3. Speech Communication

In contrast to feedback control, a feedforward approach does not derive the control signal directly from the sound in the volume enclosed by the earmuff (i.e., from the error microphone), which contains both the undesired environmental noise and speech reproduced by the secondary source. In the former approach, the speech component sensed by the microphone close to the ear must be removed from the control signal, which can be an imprecise process with a tendency to introduce distortion. It is evident from the feedforward control structure (Fig. 1) that only the component of the sound under the earmuff correlated with the reference signal can be reduced. The extent to which reproduction of a speech signal by the secondary source at the same time as it is controlling environmental noise degrades ANR performance has been demonstrated in the noise environment of a Leopard tank. For this experiment, the subject was immersed in a simulated diffuse sound field. The ANR systems were configured to control environmental noise at frequencies below 400 Hz, with input/output signals digitized at a frequency of 33 kHz, a control frequency of 3 kHz, and an adaptive filter containing 400 coefficients.

While the control systems were operating, a pre-recorded speech signal was fed into the ANR system for the right ear, and replayed by the miniature loudspeaker at a level that could be clearly understood by the subject. The speech signal was not introduced to the ANR system for the left ear, which consequently continued to function as an unperturbed active noise control system. The speech consisted of a male voice repeating sentences with a brief pause between each sentence. The pause was not of sufficient duration for the active control system to re-adapt to the sound under the earmuff without speech, but was long enough for the ANR to be measured.

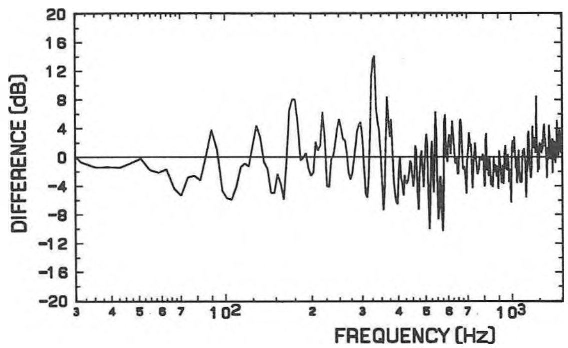

Fig. 2: Difference in ANR.

The difference between the ANR recorded at the error microphones for the left and right ears under these conditions is shown in Fig. 2. It is evident from the difference spectrum recorded with the two ANR systems operating [(right ANR, with speech) - (left ANR, no speech)] that the noise reduction displays little systematic degradation with the presence of the communication signal (i.e., the difference is not generally negative). There is an indication of a minor reduction in ANR at frequencies below 80 Hz, and a cyclical variation in ANR with frequency which may reflect the tonal nature of the noise source. Thus, the expected lack of dependence of the ANR performance on the presence, or absence, of a communication signal is confirmed for this adaptive controller. In addition, the feedforward control structure does not perturb the communication signal, and so offers the potential for higher fidelity reproduction.

4. Intelligibility and Hearing Loss

It is well known that the risk of noise-induced hearing loss is related to the sound level at the ear, when the latter is expressed in terms of the A–weighted sound pressure level. It is also possible to relate speech intelligibility to the speech signal to noise ratio, when both are expressed as A–weighted sound pressure levels.[4] In view of these considerations, the possibility of operating the adaptive feedforward ANR headset with a frequency-dependent target convergence function has been explored.

A critical feature of this application of feedforward control is the short distance between the reference and error microphones, and the correspondingly short time delay in which the controller must function to maintain causality. Once this condition has been achieved, as in the present feedforward system,[2] introducing additional phase shift with a frequency-dependent target convergence function is possible. In contrast, feedback control systems do not lend themselves to the introduction of additional phase shifts.

A demonstration of the potential for controlling band-limited white noise (150-700 Hz) so as to produce an A-weighted noise spectrum at the ear has been conducted by introducing an analogue filter in the error signal path with approximately this frequency response. The experiment employed one earmuff mounted on a flat-plate coupler in a small enclosure.[5] The control system employed an input/output frequency of 40 kHz, a control frequency of 10 kHz, and an adaptive FIR filter containing 200 coefficients.

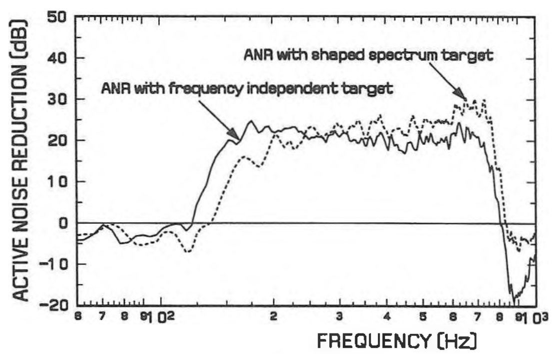

Fig. 3: ANR with different target functions.

The difference between the ANR recorded at the error microphone when the system was operating with, and without, a frequency-selective target convergence function can be seen from Fig. 3. It is evident from these results that the ANR is not compromised by selecting the A-weighted frequency network as the target convergence function. Thus it is believed that the adaptive feedforward ANR headset may be operated at close to optimum for maintaining speech intelligibility and the preservation of hearing, at least for environmental noise at frequencies above 150 Hz.

5. Acknowledgment

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6. References

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