

CONTACTLESS HEARING AID DESIGNED FOR INFANTS

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It is a well known fact that language development through home intervention for a hearing-impaired infant should start in the early months of a newborn baby's life. The aim of this paper is to present a concept of a contactless digital hearing aid designed especially for infants. In contrast to all typical wearable hearing aid solutions (ITC, ITE, BTE), the proposed device is mounted in the infant's bed with any parts of its set-up contacting the infant's body. A processed speech signal is emitted by low-power loudspeakers placed near the infant's head. The hearing aid architecture employs a digital signal processor based on Texas Instruments technology. Since one of the main problems is the acoustic feedback between the microphone and the loudspeakers, the methods of its elimination are also briefly reviewed in this article. The first of the discussed methods employs an adaptive algorithm, the second alters the frequency response of the entire instrumentation through the use of notch filter banks, and the third incorporates a microphone array and beam-forming techniques. The paper also includes descriptions of some algorithmic solutions engineered by the authors in purpose to eliminate acoustic feedbacks. All the conclusions introduced in this article have been derived based on the simulations of an experimental contactless hearing aid set-up.

Key words: hearing aids, digital signal processors, acoustic feedback elimination.

1. Introduction

In the last few years the number of digital hearing aids introduced by the manufacturers has increased significantly. This tendency is reasonable as the flexibility of digital over analogue signal processing enables to engineer hearing aids that not only compensate the hearing loss but also increase the comfort of hearing. Unfortunately, all newly developed wearable ITE (in-the-ear), ITC (in-the-canal) and CIC (completely-in-canal) compact digital hearing aids are unsuitable for infants due to small dimensions of the infants' ear canals and the danger of being swallowed. An important issue is that language development is dependent on infants' ability to perceive the speech. In order to

allow proper language development through home intervention, the hearing loss should be identified and compensated in the early months of a newborn baby's life. As standard wearable devices are not suitable for infants, a dedicated novel digital hearing aid is presented in this paper.

In this approach a hearing aid set-up is mounted in the infant's bed, and none of its parts contact the infant's body. In contrast to all standard hearing aid solutions, where speech signal is emitted directly to the ear canal, this newly engineered device operates in an acoustical field. Consequently, the processed speech signal is emitted through low-power loudspeakers placed near the infant's head. Because the distance between the microphone and the receivers is significantly greater comparing to the compact devices, the reduction of the acoustic feedback is a major issue. Various techniques are commonly employed in digital hearing aids in order to prevent the occurrence of the acoustic feedback [9, 11]. In the presented device, two different methods are combined. The first is based on the beamforming technique that allows for selecting only a desired acoustic signal while attenuating any adverse signals. The beamforming algorithm is based on artificial neural network, which is used as a nonlinear filter in frequency domain [8]. Consequently, the possibility of the acoustic feedback occurrence is significantly reduced. One can notice that a contactless hearing aid set-up can be viewed as a modified public addressed system. Thus, an algorithm dedicated to this kind of systems is also employed [10, 14].

This paper is structured as follows. In Sec. 2, a brief description of the contactless hearing aid set-up is introduced. In Sec. 3, the methods for the acoustic feedback elimination are discussed and compared, and in Sec. 4, the experimental results for the automatic notch filtering technique are presented. Finally, in Sec. 5 the authors summarize the results obtained and outline future research related to the issue of acoustic feedback elimination.

2. Contactless hearing aid

The contactless hearing aid set-up comprises of the following three main modules: microphone array with preamplifiers mounted in front of the infant's bed, DSP unit responsible for signal processing, and loudspeakers mounted near the infant's head. In Fig. 1, the infant's bed equipped with a contactless hearing aid is presented.



Fig. 1. Contactless hearing aid application.

As the hearing aid is not a simple sound reinforcement system, advanced signal processing techniques have to be applied in order to meet specific requirements of signal capture and hearing loss compensation. The functional diagram illustrating the processing modules of the hearing aid is shown in Fig. 2.

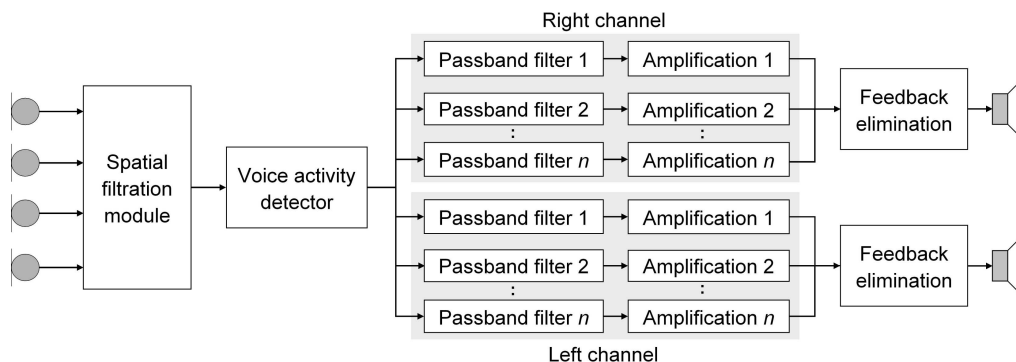


Fig. 2. Block diagram of signal processing in the contactless hearing aid.

Four omni-directional microphones capture sounds coming from all directions. A spatial filtration module amplifies the voice signal coming from a particular direction (e.g. from a speaker) and attenuates signals from other directions. In this case, all undesired signals which do not come from the position of a speaker can be reduced. Beamforming takes the advantage of the fact that the distance from the source to each microphone in the array is different, which means that the signals captured by the microphones will be phase-shifted replicas of each other. Knowing the amount of the phase-shift at each microphone in the array is sufficient to calculate the direction [1]. Although various approaches to spatial filtration can be found in literature, the technique employing artificial neural network is used in contactless hearing aid. Because spatial filter works in the frequency domain, it is assumed that each spectral component representing signals coming from unwanted directions is to be attenuated by at least 40 dB [2–4, 6–8]. In order to prevent the amplification of loud unexpected sounds as well as any other acoustic signals that should not be presented to the infant, the voice activity detector is employed. This algorithm operates in the frequency domain and takes the properties of the captured signal spectrum into account to decide if the signal should be further processed [13].

The third module implements signal processing algorithms that are typically employed in digital hearing aids. The signal is divided into subbands. Either constant amplification or dynamic processing with compression characteristic is applied to the signal in every subband basing on the characteristics of an infant's hearing impairment. Hearing loss is compensated for each ear independently. As a sequel, two separated automatic notch filters banks must be used to eliminate the acoustic feedback in the signals corresponding to the left and right channel respectively. All of described algorithms are implemented using TMS320VC5503 fixed-point digital signal processor [12].

3. Acoustic Feedback Elimination Methods

All standard methods of dealing with feedbacks (e.g. passband equalizing) are static and unable to adapt to changes occurring in the system itself (e.g. microphones and loudspeakers movements) or in the acoustic environment. Thus, two dynamic methods are often used to limit feedbacks: adaptive notch filtering and adaptive feedback cancellation.

The goal of notch filters deployed in the electroacoustic forward path between the microphone and the speaker is to eliminate frequency components resulting from the acoustic feedback. In this method, local maxima of the signal amplitude spectrum are detected and classified as representing feedback components or not. If a maximum representing a feedback component is identified, a notch is deployed with a centre frequency equal to the frequency of the local maximum [10].

Feedback discrimination is the most important part of the adaptive notch filtering algorithm. Feedback components have a few properties that are very useful in separating them from natural sound features. A feedback component amplitude rises monotonically and exponentially, while its frequency is constant, which is illustrated in Fig. 4. There are usually no harmonics of a feedback component, however nonlinearity of electroacoustic devices working with high-level signals can create them. After a notch is placed on a potential feedback frequency, the feedback component amplitude not only decreases by some value but continues to decrease at an exponential rate [14].

The computational complexity of the adaptive notch filtering method is rather low while its effectiveness in feedback elimination is high. Furthermore, sound distortion introduced by this method is insignificant. The only disadvantage is the fact that in order to be detected and eliminated, the feedback components must first appear in an audio signal and be audible for a short time.

The adaptive feedback cancellation method is very similar to algorithms used in acoustic echo cancellation for teleconferencing systems. The idea is to accurately model the loudspeaker to microphone transfer function F and then use this model to remove all of the audio sent out the local loudspeaker from the microphone signal. An illustration of the method is presented in Fig. 3.

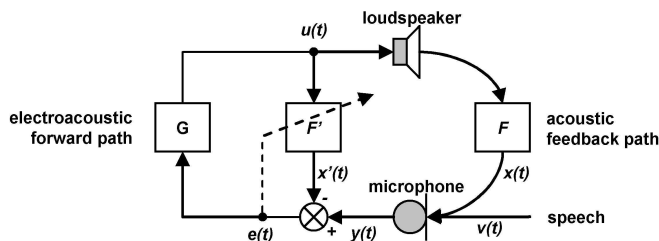


Fig. 3. Diagram of the adaptive feedback cancellation method.

There are many methods for estimating the coefficients of an adaptive filter F' , for example NLMS (Normalized Least Mean Squares), RLS (Recursive Least Squares) [5].



However, the resulting estimators are biased because the source signal v and the loudspeaker signal u are correlated. The bias can be eliminated by reducing the correlation. This can be achieved directly in the signal loop (by delaying or nonlinearly distorting the loudspeaker signal u) or in an additional identification loop (by means of prefiltering the input signal y and the output signal u which assures that both the source and the loudspeaker signals v and u are whitened) [11]. The latter variant is computationally more complex.

Adaptive feedback cancellation requires a significantly more powerful digital signal processor than adaptive notch filtering, but it is capable to eliminate any audible signs of feedback at the cost of some minor sound distortions. However, because of very strict power consumption constraints placed upon the hardware used in embedded hearing aid devices, the adaptive notch filtering method is favoured.

4. Experiments

Initial experiments were focused on the implementation of a feedback component discrimination algorithm on a PC platform. The algorithm is the most important part of the adaptive notch filtering method. During the experiments, pieces containing both speech and music fragments were played through a computer speaker and recorded with a microphone. The placement of the speaker and microphone and their output and input gains were altered in order to produce a large variety of feedbacks. The recorded sequences were then processed to identify all local maxima in the signal spectrum and to recognize any feedback components.

Feedback components can be divided into three groups. The first group contains components with amplitudes rising slower than approx. 6 dB per one time frame of the signal (the frame length of 46 ms was used). Such components are considered to result from feedback if their amplitudes rise monotonically for 6 time frames in the row. The second group is formed by the potential feedbacks with amplitudes rising faster than approx. 6 dB per time frame. Such a component is illustrated in Fig. 4. These components are cancelled if their amplitudes rise for 3 frames. The last group contains the feedbacks impossible to be directly tracked because their amplitudes increase from the background level to the maximum level allowed in the system almost instantly (during the length of one to two time frames). There is only one way of dealing with such

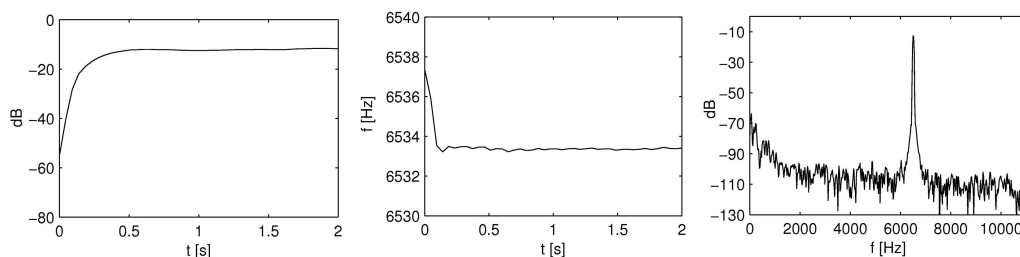


Fig. 4. Fast feedback component amplitude (left) and frequency change in time (middle); signal spectrum with strong feedback present (right).

feedbacks: all components with amplitudes higher than a given threshold are unconditionally considered as feedbacks. All components classified as resulting from feedback are eliminated by a notch filter. The algorithm guarantees that the louder and faster a component rises, the shorter is the time needed to detect and eliminate it.

The results of experiments show that the algorithm is able to detect and identify feedback components with great accuracy. In a test sequence that was infected with feedbacks and lasted 1.5 min., the algorithm detected 37 frequencies on which feedbacks appeared. In order to determine the effectiveness of classification, the algorithm was used to detect feedbacks in the original sequence – the one without feedbacks. As a result, only 3 false detections were obtained. The amount of false positives can be further reduced through observing a new component for a few frames after a notch filter has been deployed. If its amplitude decreases at an exponential rate then the classification is correct.

5. Conclusions

In this paper the novel contactless hearing aid dedicated to infants is presented and its structure is thoroughly described. The reduction of the acoustic feedback is a major issue in the device, thus two different approaches to the issue – adaptive notch filtering and adaptive feedback cancellation – are shown. The former method has been chosen for a preliminary implementation on a PC, as it is more suitable for digital signal processors.

During the experiments the nature of feedbacks has been examined and the algorithm for feedback component discrimination has been implemented. The results of the experiments prove that the algorithm is able to detect feedbacks with great accuracy and it may be implemented on a digital signal processor.

Further work will be focused on building a dedicated extension card that together with Texas Instruments DSP development kit will form a complex environment for the four channel audio processing. It will be used for implementing, testing, and evaluating all the algorithms utilized in the hearing aid devices, including the ones intended for feedback implementation.

Acknowledgments

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