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Recent Development of Bone-Conductive Microphone in DSP

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Abstract
This article presents recent developments on ongoing challenges in bone conduction hearing devices. In view of the large number of problems and challenges in electronics for medical devices, this article focuses on areas that will advance bone conductive microphone devices for hearing aids. It will be described in greater details in the upcoming that the bone conduction microphone need to receive increasing attention and have found uses in many aspects of voice signal processing . This device can improve the quality of life for patients suffering from different hearing aids and make it feasible to use this technology all over the world.

Keywords: Bone Conductive Microphone (BCM)

Introduction

Digital signal processing is one of the core technologies, in rapidly growing application areas, such as wireless communications, audio and video processing and industrial control. The number and variety of products that include some form of digital signal processing has grown dramatically over the last few years. DSP has become a key component, in many of the consumer, communications, medical and industrial products which implement the signal processing using microprocessors etc. Due to increasing popularity of the above mentioned applications, the variety of the DSP-capable processors has expanded greatly. DSPs are processors or microcomputers whose hardware, software, and instruction sets are optimized for high-speed numeric processing applications, an essential for processing digital data, representing analog signals in real time. The DSP processors have gained increased popularity because of the various advantages like reprogram ability in the field, cost-effectiveness, speed, energy efficiency etc. One of the most recent developments in equipment for medical application.

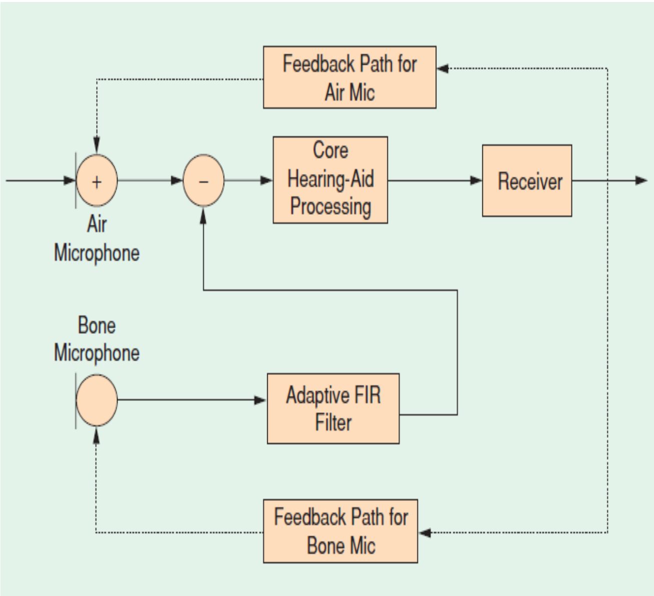
Signal processing applications includes medical applications, where signal analysis has been widely applied in patient monitoring, diagnosis and prognosis, as well as physiological investigation and in some therapeutic settings (e.g. muscle and sensory stimulation, hearing aids).

Approaches in signal processing research on digital hearing aids fall into expanding hearing aid technology into devices that are also able to perform other functions, such as mobile phones and music players. In this area, issues such as echo cancellation, bone conductive microphones, and wireless voice link are of interest. Because of the limitations imposed by the hardware requirements, computational, speed, power consumption, and other practical factors, the development and implementation of signal processing techniques for digital hearing aids has been a challenging and active research area over the past decade.

This article provides an overview on recent developments in these areas from engineering and applications point of view and outlines topics in signal Processing techniques for hearing aids like as Bone Conductive Microphones (BCM). In turn, their solutions are potentially the most beneficial for meeting the increased demands placed on the performance of hearing aids.

BONE-CONDUCTIVE MICROPHONES

Recently, bone conductive microphones (BCM) have received increasing attention and have found uses in many aspects of voice signal processing. Unlike regular air conductive microphones, bone-conductive microphones receive vibrations only from the bones of the human talker and from nearby air. In other words, the output of the bone-conductive microphone is only the voice of the human talker. Taking this property of bone-conductive microphones, a novel hearing-aid system has been proposed and is shown in Figure below. In comparison with traditional air-conductive microphone- based hearing-aid systems, the proposed scheme offers significant advantages in feedback cancellation and the reduction of the hearing-aid wearer's own voice, which is unnecessarily amplified by compression amplification units. Also with this proposed scheme processing delay and its impact on speech quality can be reduced. It should be noted that the air-conductive microphone in Fig. is either one single microphone or a microphone array.



[FIG] Block diagram of the use of bone-conductive microphone in hearing aids.

The configuration in Figure provides two physical feedback paths, one for the air-conductive microphone (which could be a directional microphone, an omni-microphone, or a combination of both), the other for the bone-conductive microphone. The output of the air-conductive microphone is the summation of the target signal and the signal through one feedback path. The output of the bone-conductive microphone is the signal from only the other feedback path, and this feedback signal is highly correlated with the feedback signal in the air-conductive microphone. Following the bone-conductive microphone, an adaptive FIR filter is added. The input of the core hearing-aid processing unit is the difference signal between the output of the adaptive FIR filter (the reference signal) and the output of the air-conductive microphone (the primary signal). If the coefficients of the adaptive FIR filter are updated by minimizing the power of the difference between the primary signal and the reference signal, the output of the adaptive filter can track the part of the signal provided by the feedback path through the air-conductive microphone, and the effect incurred by the feedback can be minimized. In the ideal case, the output of the adaptive filter approximates the feedback signal through the air conductive microphone so that the difference signal can approximate the real target signal without any feedback signal in the receiver. If we analyze this scheme in the frequency domain by denoting the frequency responses of the physical paths in the air-conductive microphone, the bone-conductive microphone, and the adaptive filter as A(w), B(w), and W(w), respectively, we see that the frequency response of the adaptive filter after the convergence is the ratio of A(w) to B(w). In contrast, the optimized frequency response F(w) of the adaptive FIR filter of the traditional feedback cancellation scheme is equal to A(w). It is easy to see that W(w) will have a simpler response than F(w). Hence it would be easier to implement W(w) than to implement F(w). In other words, the adaptive FIR filter following the bone-conductive microphone is used to model the decibel difference between the time-varying acoustical feedback paths of the air conductive microphone and the bone conductive microphone. Because the variation of this difference is much smaller than that of the acoustical feedback path itself in the air-conductive microphone, the length of the adaptive FIR filter [W(w)] used in the proposed scheme of Figure can be much shorter than that of the adaptive filter [F(w)] used in the traditional scheme. In the ideal case, W(w) can simply be a gain factor. With these, this proposed scheme can provide us with a simple alternative tool in feedback cancellation. As a matter of fact, the physical feedback path of the bone-conductive microphone in Figure serves as the IIR filtering in the two-stage filtering. The difference is that the physical feedback path of the bone conductive microphone is no longer fixed but varying as the physical feedback path of the air-conductive microphone. In other words, the processing in Figure can be considered to include two-stage adaptive processing: adaptive FIR filtering and the adaptive IIR filtering provided by the bone-conductive microphone. As a result, the problems listed previously will no longer exist in this bone-conductive microphone-based scheme. For further illustration, let us discuss the placement of a telephone headset to the aided ear. In this situation, the feedback signal picked up in the air-conductive microphone will increase about 10 dB, and the scheme with the adaptive FIR filtering plus the fixed IIR filtering would fail. Since the variation is too large. However, the feedback signal picked up in the bone conductive microphone could also increase about 10 dB, and this would make the decibel difference variation of the two feedback path signals very small. The adaptive FIR filter following the bone-conductive microphone can still easily track this small variation. With traditional hearings aids, the wearer's own voice is picked up by the air-conductive microphone and is unnecessarily amplified. It is highly desirable to effectively reduce this amplified voice. The scheme in Figure very powerfully attacks this problem. The wearer's voice is picked up by both the air-conductive microphone and the bone-conductive microphone; hence, the voice to be amplified by the compression amplification processing is the difference of two microphone channels instead of the part picked up in the air-conductive microphone of traditional hearing aids alone. More importantly, the optimized coefficients of the adaptive filter are obtained by minimizing the difference signal between the output of the air-conductive microphone and the output of the adaptive filter following the bone-conductive microphone. This means that the wearer's own voice inputted to the amplification processing unit can also be minimized, and its impact on the target signal can be correspondingly minimized as well. Ideally, the wearer's own voice through the microphone channel can be reduced to zero.

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