A LOW-NOISE DIRECTIONAL MICROPHONE SYSTEM

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1. INTRODUCTION

First-order directional microphones are commonly used in hearing aids. The conventional method for implementing a directional response using two non-directional microphones is depicted in Fig. 1. The microphones are arranged such that there is a front microphone and a rear microphone. A directional response pattern, with its main beam pointing toward the front microphone, is formed by subtracting the delayed rear microphone signal from the front microphone signal. The optional equalization filter (EQ) is provided to equalize the directional microphone on-axis frequency response to that of a single, omnidirectional microphone. A variety of directional patterns can be implemented by varying the delay [1].

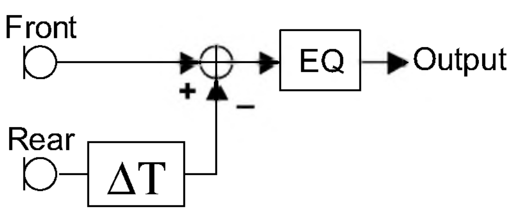

Fig. 1. Block diagram of a conventional directional microphone system.

While such microphones do provide significant directional gain and patient benefit it is a well-known fact that the unequalized frequency response possesses a first-order high-pass characteristic [2]. The reduction in signal level, particularly acute at low frequencies, leads to a reduction in signal-to-noise ratio (SNR) due to microphone self noise, preamp noise and wind. While the equalization filter overcomes the high-pass characteristic, it does not improve the SNR.

The increase in audible noise due to directional processing is considered to be one of the major drawbacks of a directional hearing instrument. In many cases it causes the wearer to deactivate the directional microphone.

Existing automatic directional systems handle the low-frequency noise problem in one of two ways: 1) they automatically switch the hearing aid to omnidirectional mode when the wearer enters a quiet environment, or 2) they automatically adjust the amount of low-frequency equalization applied to the microphone signal to reduce noise amplification. The first approach is disadvantageous in situations where a small amount of directional performance would benefit the wearer but the low-frequency noise is still a problem. By switching the hearing aid to omnidirectional mode, the wearer loses the benefit of the directional microphone. While the second approach does maintain the operation of the directional microphone, the lack of low-frequency equalization reduces the audio quality of the hearing aid and can impair audibility for wearers with a low-frequency hearing loss.

This paper describes a simple method for improving the SNR in directional microphone applications. The method provides the maximum directional gain for a prescribed allowable degradation in SNR. This is accomplished using a frequency-specific phase shift to create a controlled loss in directional gain over a frequency band of interest. In this way a minimal amount of directional performance is lost while maintaining a targeted amount of low frequency sensitivity or SNR. As will be shown, the signal processing requirements are simple enough to permit deployment in low-power digital hearing aids.

2. LOW-NOISE DIRECTIONAL SYSTEM

The proposed system limits the amount of noise amplification at low frequencies by replacing the pure time delay of the conventional system with a frequency-specific phase shift. This phase shift is implemented using specially designed filters to process both the front and rear microphone signals. The filter outputs are then summed to provide the directional microphone output. The equalization (EQ) filter is provided to equalize the directional microphone on-axis frequency response to that of a single, omnidirectional microphone. The block diagram is depicted in Fig. 2.

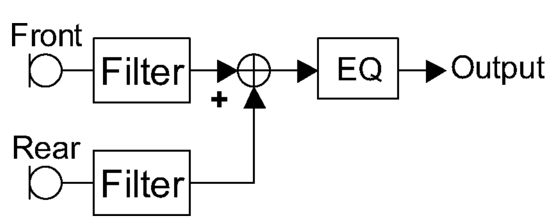

Fig. 2. Block diagram of a low-noise directional microphone system.

The phase shift required to implement a low-noise directional system is determined by the maximum SNR loss that is acceptable. If a high loss of SNR can be tolerated, then the phase shift approaches that of a conventional directional system. On the other hand, if there is a desire to limit SNR loss, then the phase shift deviates significantly from that of a conventional system.

As an example, consider the design of a directional microphone system with a front to rear microphone spacing of 10.7mm. The SNR loss as a function of frequency for a conventional system is shown in Fig. 3 (curve 'Conv'). Superimposed on this are curves representing maximum desired levels of SNR loss (20dB, 15dB, 10dB, 5dB and 0dB) for the same microphone. Fig. 4 shows the corresponding directivity index (DI). The main effect of the SNR loss limit is the decrease in low-frequency DI compared to a conventional system. The lower the desired SNR loss, the lower the resulting DI.

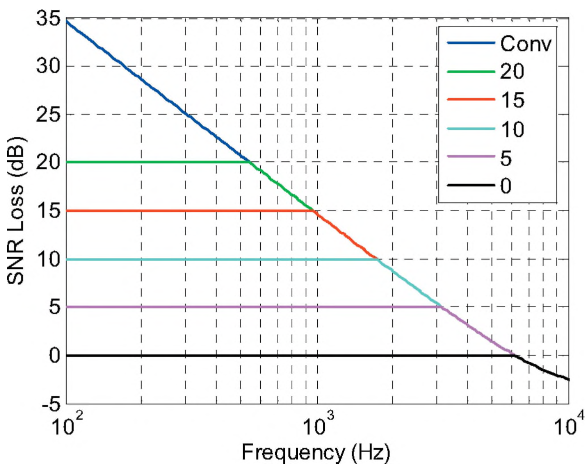

Fig. 3. SNR loss for directional microphone system with a front to rear microphone spacing of 10.7mm.

Since hearing aids are primarily intended for speech directional performance is often stated as AI-DI, which is a weighted average of the DI at frequencies 500,1000,2000 and 4000 Hz [3]. The points used in the AI-DI calculation are indicated by the circular markers on the curves in Fig. 4. The DI at the frequencies used in the AI-DI calculation is much less affected by the limit on SNR loss. For example, if the SNR loss is limited to 10dB, a reduction of 25 dB from the conventional system, the AI-DI only drops by 1.3 dB.

The inter-microphone phase shift required to implement the systems described in Fig. 3 and 4 is shown in Fig. 5. Strictly speaking, any type of filter could be used as long as the magnitude responses of both filters are identical. It is a much simpler design task, however, if allpass filters are used. By using allpass filters, the inter-microphone phase response from Fig. 5 is divided equally between the front and rear filters. This phase is then added to the phase of a known realizable allpass filter to obtain a realistic phase target. The required allpass filters can then be designed using any known allpass filter design procedure [4].

This approach has been used to obtain low-noise directional microphones using only one second-order allpass filter in each of the front and rear microphone signal paths.

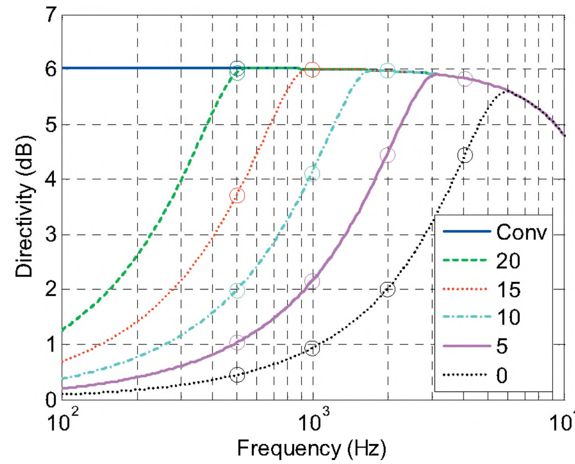

Fig. 4. Directivity Index achievable for the SNR loss curves shown in Fig. 3. The circles indicate the points used to calculate AI-DI. AI-DI values for these systems are: 'Conv' = 6 dB, '20' = 5.9 dB, '15' = 5.5dB, '10' = 4.7 dB, '5' = 3.6 dB and '0' = 2dB.

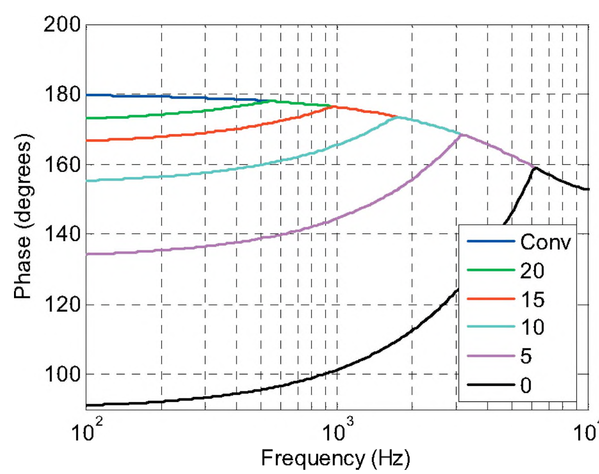

Fig. 5. Inter-microphone phase shift that is required to achieve the directional gains indicated in Fig. 4.

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