DISTORTION OF INTERAURAL TIME CUES BY DIRECTIONAL NOISE REDUCTION SYSTEMS IN MODERN DIGITAL HEARING AIDS

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ABSTRACT

In this paper we analyze the distortion of interaural time information by modern digital hearing aids. It is shown that the directionality in dual microphone hearing aids, created by a software directional microphone or an adaptive directional microphone, are very sensitive to intermicrophone mismatch, which in particular results in severe distortion of interaural time information. This interaural information is crucial for sound localization as well as for speech perception in noise.

1. INTRODUCTION

Hearing impaired persons often localize sounds better without hearing aids than with their hearing aids [1, 2]. This is not surprising, since noise reduction algorithms currently used in hearing aids are not designed to preserve localization cues. This puts the hearing aid user at a disadvantage as well as at risk. Sound localization is important in speech segregation in noisy environments (a.k.a 'the cocktail party effect'), and in certain situations, such as traffic, incorrect localization of sounds could even endanger the user. When designing monaural noise reduction systems, the property of phase deafness of the ear can be exploited. This means that the phase characteristic of the noise reduction system has no influence on the resulting monaural speech perception. However, the human auditory system uses interaural information (or 'binaural cues') to localize sounds [3, 4, 5, 6, 7] and to segregate speech in noisy environments [8]. Interaural time delay (ITD) is the time delay between the arrival of the sound signal at the left and the right ear and is one of the most important binaural cues for horizontal sound localization and speech perception in noise. Interaural time differences between 10 and 700μs are used by the binaural centra in the human auditory system. Therefore the phase characteristic of noise suppression systems becomes an important issue when using bilateral hearing systems (one on each ear). Important to note is that only the low-frequency part of the sound spectrum (f<1000Hz) is used by the binaural centra to determine ITD's.
This paper focusses specifically on the impact of a software directional microphone and an adaptive directional microphone on ITD cues. Both noise reduction techniques discussed in this paper are commonly used in hearing aids due to the fact that they combine a good noise suppression performance with a low complexity. A disadvantage of these techniques is that they generally assume that the location of the speech source is known and fixed (0°, in front of the listener). It will be shown that these noise reduction techniques are very sensitive to intermicrophone mismatch which in particular results in severe distortion of interaural time information.

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2. A SOFTWARE DIRECTIONAL MICROPHONE

One of the basic building blocks of directional noise reduction systems in dual microphone hearing aids is the software directional microphone (see Figure 1). This can be used as a stand alone noise reduction system or it can be integrated in more enhanced noise reduction schemes (e.g. an adaptive directional microphone, 2 stage GSC techniques, ... [9, 10, 11]).

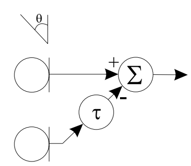

Figure 1: A directional microphone. The sound is captured by two omnidirectional microphone ports. Both signals are combined to create a directional pattern. Different patterns can be created by using a different time delay τ.

The transfer function of a software directional microphone with time delay τ and inter-microphone distance d equals

$$H_{dm}(w,\theta) = (1 - Ae^{-jwt_0}) \tag{1}$$

with A the amplitude of the sound at the second microphone relative to the amplitude of the sound at the first microphone and t_0 the time delay of the sound due to the inter-microphone distance d and the time delay τ ($t_0 = \tau + \frac{d}{c}\cos\Theta$ with Θ the angle of the sound source relative to the hearing aid and c the speed of sound). The transfer function can be written as:

$$H_{dm}(w,\theta) = B_{dm}(w,\theta)e^{j\Phi_{dm}(w,\theta)} \tag{2}$$

with

$$B_{dm}(w,\theta) = \sqrt{1 + A^2 - 2A\cos(wt_0)} \tag{3}$$

$$\Phi_{dm}(w,\theta) = \arctan\left(\frac{A\sin(wt_0)}{1 - A\cos(wt_0)}\right) \tag{4}$$

In an ideal system, i.e. in the absence of reflections and with ideal amplitude characteristics of both microphones, so that A is equal to 1, the response of the system becomes:

$$B_{dm}(w,\theta) = \sqrt{2-2\cos(wt_0)} \quad (5)$$

$$\Phi_{dm}(w,\theta) = \frac{\pi}{2} - \frac{wt_0}{2} \quad (6)$$

If $\cos\Theta = -\frac{\tau c}{d}$ then the amplitude of the output, $B(w)$, equals 0 and so the sound arriving from angle Θ will be cancelled. When used in hearing aids, the assumption is usually made that the speech signal arrives from 0° (in front of the listener) and that the noise signals arrive from the rear horizontal plane. Therefore the factor τ is usually set at $\frac{d}{c}$ which gives a zero output for $\Theta = 180^\circ$. This results in a forward facing cardioid polar pattern (Figure 2).

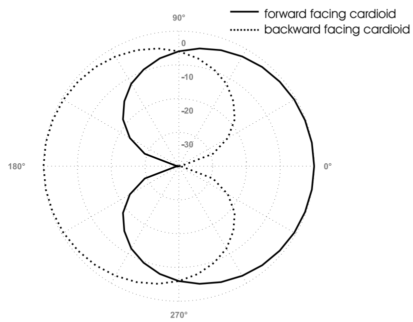

Figure 2: A forward and backward facing cardioid polar pattern generated by a first order directional microphone with $\tau = \frac{d}{c}$ and A=1

The interaural time information generated by two hearing systems with ideal software directional microphones can be calculated using Formula (6) for both devices. The interaural time information becomes:

$$\widetilde{\Delta_{ITD}} = (\Delta_{ITD} - \frac{\pi}{2w} + \frac{\tau_2}{2} + \frac{d}{2c}\cos\Theta) - (-\frac{\pi}{2w} + \frac{\tau_1}{2} + \frac{d}{2c}\cos\Theta) \quad (7)$$

where Δ_{ITD} is the true ITD between the front microphone inputs of the hearing devices. When using identical devices, τ_1 will be equal to τ_2 and the angle Θ is equal for both hearing devices. Therefore, hearing aids with an ideal directional microphone do not distort interaural time information.

In a non-ideal system, A will not be equal to 1 due to the presence of the pinnae and due to the differences in microphone characteristic between the two microphones on the hearing aid. Therefore the phase characteristic of the directional microphone is dependent on the factor A. An important aspect is that the phase response, given in Formula (4), has a high sensitivity to small changes close to A=1 when wt_0 is small. This area happens to be the area with the best noise reduction performance. The partial derivative of the phase transfer function (4) to A equals:

$$\frac{\delta\Phi(w)}{\delta A} = \frac{sin(wt_0)}{1 - 2A\cos(wt_0) + A^2} \quad (8)$$

This derivative is dependent on the factors A,ω,τ and Θ. When wt_0 decreases, the shape of the partial derivative becomes more and more peaked around A=1. The typical range of values (for frequencies from 0 to 1000Hz) for wt_0 are 0 (around the null-angle) to 0.78 (opposite to the null-angle and with f the maximum frequency of interest, f=1000Hz) for a hearing aid with an inter-microphone distance of 2cm. An example is given in Figure 3 showing the partial derivative of the phase response of a directional microphone with $\tau = \frac{d}{c} = 59\mu s$ to the parameter A for a 500Hz sinusoid arriving from $\Theta = 115^\circ$ and $\Theta = 85^\circ$. For $\Theta = 115^\circ$ a change in amplitude of a 500Hz sinusoid at the second microphone from 1 to 0.9 creates a phase change of 0.81 rad which generates a time distortion of approximately 258μs. For $\Theta = 85^\circ$ the same change in magnitude creates a phase change of approximately 0.48 rad or a time shift of 154μs. This shows that the amount of time distortion that is introduced by a non-ideal directional microphone is dependent on the angle of arrival Θ and that the amount of introduced time distortion is well within the range used by the auditory system. Moreover, the amount of reflections and microphone mismatch can differ per frequency, which means that broadband stimuli can generate not only distorted but also conflicting interaural time delay cues.

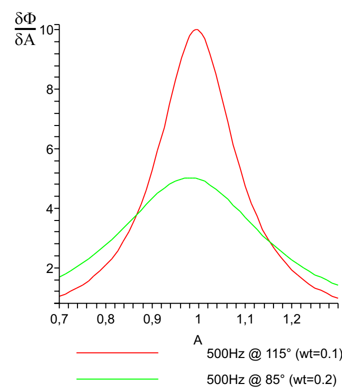

Figure 3: Plot of the partial derivative of the phase response of a first order directional microphone formula to the parameter A for a 500Hz sinusoid arriving from $\Theta = 115^\circ$ and $\Theta = 85^\circ$.

3. AN ADAPTIVE DIRECTIONAL MICROPHONE

A commonly used, more advanced, noise reduction technique in hearing aids is an adaptive directional microphone (Figure 4) which consists of two software directional microphones and an adaptive part. One directional microphone looks forward (with a zero in its polar pattern at 180° to create a speech reference and one directional microphone looks backwards (with a zero in its polar pattern at 0°) to create a noise reference. Then the adaptive part, which is often implemented as a one tap adaptive filter, subtracts the noisy part of the signal from the speech reference. The transfer function

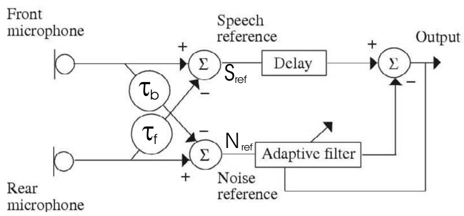

Figure 4: An adaptive directional microphone.

of the system equals

$$H_{adm}(w,\theta) = 1 - Ae^{-jw(\frac{d}{c}\cos(\theta)+\tau_f)} - \beta Ae^{-jw(\frac{d}{c}\cos(\theta))} + \beta e^{-jw\tau_b} \quad (9)$$

The output of the system when a speech signal $S(w)$ and a noise signal $N(w)$ are present at the microphone inputs equals:

$$O_{adm}(w,\theta) = S(w)H_{adm}(w,\theta_s) + N(w)H_{adm}(w,\theta_n) \quad (10)$$

where the adaptive factor β minimizes the noise part of the signal at the output by combining the forward and backward cardioid in the correct way. β will converge to:

$$\beta = \frac{(1 - Ae^{-jw(\frac{d}{c}\cos(\theta_n)+\tau_f)})}{(Ae^{-jw(\frac{d}{c}\cos(\theta_n))} + e^{-jw\tau_b})} \quad (11)$$

When we assume $\tau_f = \tau_b = \tau$ (usually set at $\frac{d}{c}$), then the phase of the signal at the speech reference and the noise reference equals:

$$\Phi_{Sref} = \arctan(\frac{A\sin w(t_x+\tau)}{1 - A\cos w(t_x+\tau)}) \quad (12)$$

$$\Phi_{Nref} = -\arctan(\frac{\sin wt_x - A\sin w\tau}{A\cos wt_x - \cos w\tau}) \quad (13)$$

with t_x the delay between the front and back microphone of the hearing aid ($t_x = \frac{d}{c}cos\Theta$) and τ the delay used in the directional microphones.

In an ideal one-tap system, A equals 1 and $\tau_f = \tau_b = \tau$. The transfer function of the system can be written as

$$H_{adm}(w,\theta) = B_{adm}(w,\theta)e^{-j\Phi_{adm}(w,\theta)} \quad (14)$$

with

$$B_{adm}(w,\theta) = [sin\frac{wd(1+\cos(\Theta))}{2c} - \beta\sin\frac{wd(1-\cos(\Theta))}{2c}] \quad (15)$$

$$\Phi_{adm}(w,\theta) = \frac{\pi}{2} - \frac{w\tau}{2} - \frac{wd}{2c} \quad (16)$$

The adaptive factor β will create a single independent null angle Θ_0 in the direction of the dominant noise source. The relation between Θ_0 and β can be written as:

$$\Theta_0 = \arccos(\frac{2c}{wd}\arctan(\frac{\beta-1}{\beta+1}\tan\frac{wd}{2c})) \quad (17)$$

In figure 5 some polar diagrams are shown created by using different noise angles Θ and corresponding factors β. Formula (16) shows that the adaptive part (β) of the system has no influence on the phase transfer function of the noise reduction system, therefore it will have no influence on the ITD perception of a bilateral hearing aid user. This can also be derived from formulas (12) and (13) which are identical under the assumption that $A = 1$. This means that the signal at the noise reference is in phase with the signal at the speech reference which makes the 1-tap adaptive filter a scaling factor with no influence on the phase relationship between in-and output. Therefore, hearing aids with an ideal adaptive directional microphone (A=1) with $\tau_f = \tau_b$ do not distort interaural time information.

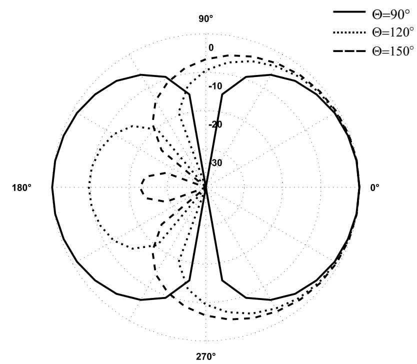

Figure 5: Theoretical polar diagrams for an ideal adaptive directional microphone for three different dominant noise source angles Θ.

In a non-ideal one-tap system, A will not be equal to 1 due to the presence of the head and pinnae and due to the difference in microphone characteristic between the two microphones on the hearing aid. Two effects are present. Firstly, the interaural time distortion discussed in the section concerning the non-ideal directional microphone is present between the speech references of the left and right noise reduction system. Secondly, the speech reference is no longer in phase with the noise reference (formula 12 and 13) which makes the phase at the output of the system dependent on the adaptive part (β) of the system. This adaptive part is operating independently in current bilateral hearing aids which will lead to extra interaural time distortion. Subband implementations of the adaptive part of the algorithm have a better noise reduction performance because they can adapt to the angle with the dominant noise source for each frequency bin. However different factors β_f per frequency bin will create independent time delays per frequency bin. This will not only produce distortion of interaural time delays but it can produce conflicting interaural time delays over the different frequency bins.

Figure 6 shows a simulation of the time distortion generated by 2 systems with similar amplitude inputs A (A=0.9) but with a different adaptive factor β. This was done for a 500Hz sinusoid. Simulations are given for two different sound source angles. Figure 7 shows the combined effect of β and A on the interaural time information between two hearing aids with an adaptive directional system. One hearing aid has fixed parameters (A=1 and β=0.9) and the parameters A and β are varied for the other hearing aid.

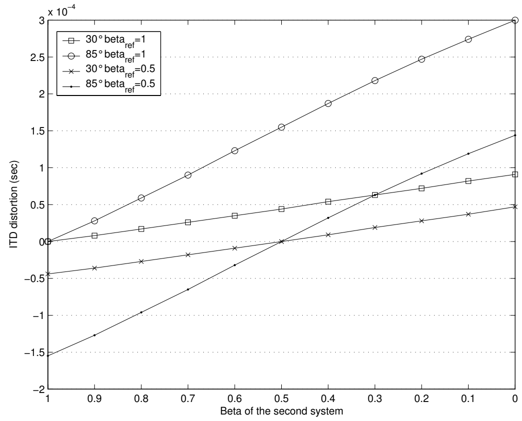

Figure 6: Interaural time distortion generated by an unequal β-factor at the left and right hearing aid for a 500Hz sinussoid. Both systems have A=0.9 and τ=59μs. Interaural time distortion is shown for two sound source angles (85° and 30°) generated between a system with fixed β (β_{ref}) and a variable β (X-axis)

It can be seen that both A and β have a significant effect on the interaural time information, well within the audible range of 20 to 1000μs.

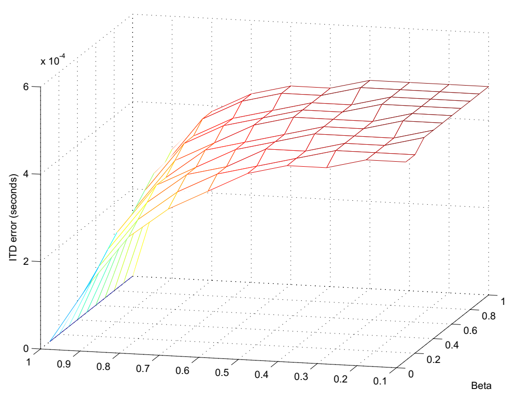

Figure 7: The influence of the factor A and the adaptive factor β of a first order directional microphone on the interaural time information. The deviation on the interaural time information for is shown for a bilateral system. One hearing aid has fixed settings (A=1 and $\beta = 0.9$), while these parameters on the second devices are changed. The simulation was performed with a 500Hz pure tone.

4. CONCLUSIONS

We conclude that an ideal directional microphone and an ideal adaptive directional microphone have no significant effect on interaural time information. However the phase characteristic of both widespread directional noise reduction systems is very sensitive to deviations in amplitude between the microphones on one hearing aid. This causes distortion of interaural time information between two hearing devices, this interaural information is crucial for a correct horizontal sound localization and for speech perception in noisy environments.

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