

Noise Reduction Schemes for Digital Hearing Aids and their Use for the Hearing Impaired

Vom Fachbereich Physik der Universität Oldenburg
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The ear was not fashioned with the prospect of industrial revolution in mind. Its superlative sensitivity and scope of action have made it victim to the culmination of the last few hundred years of industrial and social development. Much of what we now hear is, in one sense or another, unwanted, and it is this element of unwantedness which defines a sound as noise.

Dylan M. Jones

Abstract

The aim of this thesis is to improve both the assessment methods and the available algorithms for noise reduction in hearing aids. In particular, the whole development chain from the construction of algorithms, subjective assessment of algorithmic performance by normal-hearing and hearing-impaired listeners as well as objective assessment methods is considered.

The speech pause detection algorithm proposed in Chapter 2 detects speech pauses by tracking minima in a noisy signal's power envelope, specifically in its low-pass and high-pass power envelopes. It maintains a low false-alarm rate over a wide range of signal-to-noise ratios. This facilitates its application for noise estimation in noise reduction algorithms.

Chapter 3 shows that the musical noise phenomenon, one widely reported artifact of most single-microphone noise reduction schemes based on spectral subtraction, can to a high degree be overcome by the Ephraim-Malah noise reduction algorithms (Ephraim and Malah, 1984, 1985). If combined with the procedure for automatically adjusting the noise spectrum estimate during speech pauses (Chapter 2), a self-adaptive noise reduction scheme is obtained.

Comprehensive evaluations of the Ephraim-Malah noise reduction algorithms with hearing-impaired subjects show that besides better "sound quality" (Chapter 5), most obvious benefits are reductions in the mental effort needed to listen to speech in noise and hence in listener fatigue over longer periods of time. To assess this feature, a new listening effort test is developed (Chapter 4).

Although a significant amount of noise reduction is obtained with the Ephraim-Malah algorithms for various noise conditions, an increase in speech intelligibility measured with a sentence test is not found. Only the binaural directional filter and dereverberation algorithm (Wittkop, 2000) is found to provide speech intelligibility improvements. On the other hand, differences in terms of listening effort are found for different algorithms which did not show up in word recognition scores. These findings indicate that conventional speech recognition tests and tests of listening effort measure different aspects of the effect of noise reduction schemes in speech perception.

The method of paired comparisons in combination with the Bradley-Terry scaling model is suggested for subjective quality assessment of the algorithms in Chapter 5. The results show that noise reduction is worthwhile in all of the different noises that were investigated. The Ephraim-Malah single-microphone noise reduction algorithms can be recommended for use in rather stationary noises. They fail in strongly fluctuating noises where the binaural directional filter and dereverberation algorithm may be used, particularly at lower SNRs.

In Chapter 6, the predictive power of several "objective" speech quality measures is investigated with respect to the subjective noise reduction effect for hearing-impaired listeners. Particularly the PMF and LAR objective quality measures reflect different subjective results.

It is demonstrated how objective measures can be employed to assess the often large parameter space in the development of noise reduction algorithms aiming at a preselection of noise reduction algorithms and parameter settings which are worthwhile a comprehensive subjective evaluation.

Finally, it is hoped that the proposed methods might be used in the future to provide further benefit to hearing-impaired patients from "intelligent" digital hearing aids.

Kurzfassung

Das Ziel dieser Dissertation ist die Entwicklung bzw. Verbesserung von existierenden Störgeräuschunterdrückungsalgorithmen für digitale Hörgeräte sowie von Methoden zur Evaluation derartiger Algorithmen. Dabei wird die gesamte Entwicklungskette von der Entwicklung der Algorithmen über die Erfassung ihrer Fähigkeiten und Unzulänglichkeiten mit normalhörenden und schwerhörigen Versuchspersonen bis hin zur objektiven Qualitätsvorhersage mit technischen und psychoakustischen Maßen berücksichtigt.

In Kapitel 2 wird ein Algorithmus zur Sprachpausenerkennung entwickelt. Dieser Algorithmus erkennt Sprachpausen, indem er Minima in den Leistungshüllkurven des Signals sowie des tiefpaß- und hochpaßgefilterten Signals verfolgt. Er zeichnet sich insbesondere durch eine geringe Falsch-Alarm-Rate aus, die er über einen großen Bereich an Signal-Rausch-Verhältnissen bewahrt. Dadurch eignet sich der Algorithmus insbesondere für eine Anwendung zur Schätzung von Störgeräuschspektren, die von vielen Algorithmen zur Störgeräuschunterdrückung benötigt werden. Der Sprachpausalgorithmus wird kombiniert mit den Störgeräuschunterdrückungsalgorithmen, die von Ephraim und Malah (1984, 1985) vorgeschlagen wurden. Wie in Kapitel 3 gezeigt wird, zeichnen sich diese durch besonders geringe Verarbeitungsartefakte aus.

Neben einer Verbesserung der Klangqualität, die in Kapitel 5 untersucht wird, verringern die Algorithmen insbesondere die mentale Anstrengung, die nötig ist, einem Sprecher in stark störrauschbehafteter Umgebung zuzuhören. Zur Erfassung dieses Aspektes wird ein neuartiger Zuhöranstrengungstest in Kapitel 4 entwickelt und angewendet. Obwohl eine starke Störgeräuschunterdrückung mit den monauralen Ephraim-Malah-Algorithmen erreicht wird, schlägt sich dies nicht in verbesserter Sprachverständlichkeit, wie sie mit einem Satztest erfaßt wird, nieder (Kapitel 4). Mit einem binauralen Störgeräuschunterdrückungsalgorithmus (Richtungsfilter und Enthüllung; Wittkop, 2000) konnten dagegen Verbesserungen der Sprachverständlichkeit nachgewiesen werden. Andererseits wurden bezüglich der Zuhöranstrengung Unterschiede zwischen Algorithmen gefunden, die sich nicht in den Ergebnissen der Sprachverständlichkeitsmessungen abbilden. Dies kann als ein Hinweis darauf verstanden werden, daß der entwickelte Test auf Zuhöranstrengung tatsächlich andere Aspekte der Störgeräuschunterdrückung erfaßt als konventionelle Sprachverständlichkeitstests.

Zur Erfassung von verschiedenen subjektiv wahrgenommenen Qualitätsaspekten der Algorithmen wird in Kapitel 5 die Paarvergleichsmethode in Verbindung mit dem Bradley-Terry-Skalierungsmodell vorgeschlagen. Die Ergebnisse zeigen, daß die Ephraim-Malah-Algorithmen von den schwerhörigen Versuchspersonen insbesondere in stationären Störgeräuschen bevorzugt werden, während der binaurale Algorithmus in fluktuierenden Geräuschen (besonders bei niedrigen Signal-Rausch-Verhältnissen) besser beurteilt wird.

In Kapitel 6 werden verschiedene "objektive" Sprachqualitätsmaße auf ihre Fähigkeit hin untersucht, die subjektiv erfaßten Qualitätsurteile widerzuspiegeln und damit in gewisser Weise vorhersagen zu können. Insbesondere die Maße PMF und LAR erweisen sich dabei als erfolgreich.

Es bleibt zu hoffen, daß die in dieser Arbeit eingeführten und vorgestellten Methoden zukünftig angewandt werden mögen, um damit schwerhörigen Patienten zu besserer Lebensqualität durch "intelligente" digitale Hörgeräte zu verhelfen.

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