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Digital Signal Processing Algorithms for Noise Reduction, Dynamic Range Compression, and Feedback Cancellation in Hearing Aids

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Abstract: **Digital signal processing (DSP) is widely used to manipulate, modify, enhance or filter signals such as speech, audio, image and telecommunication signals. These signals can be processed in the analog domain but the digital domain offers high speed, better accuracy, greater flexibility, increased storing capabilities, and simpler implementation. DSP has become a fundamental area of research for many real-world applications, e.g., mobile phones, digital cameras, GPS, video/tele-conference, radar, MP3 players and many more. The work presented here is focused on DSP for hearing aids which is important for hearing impaired people in order to communicate and interact with other people in the daily life. It should be mentioned that some of the algorithms developed here can be applied to, e.g., hands-free telephony, in-vehicle communication, and public address systems. The two types of technology for hearing aids are analog and digital. Current state-of-the-art hearing aids are exploiting various aspects of DSP and according to percent of all hearing aids sold in 2005 were digital. The core function of traditional hearing aids is mainly based on signal amplification. However, digital hearing aids allow for more advanced signal processing since the purpose of modern hearing aids is not only to amplify sounds.**

Keywords: **Hearing, impairment, MP3, DSP, AFC, DRC, NIHL**

1. INTRODUCTION

Digital signal processing (DSP) is widely used to manipulate, modify, enhance or filter signals such as speech, audio, image and telecommunication signals. These signals can be processed in the analog domain but the digital domain offers high speed, better accuracy, greater flexibility, increased storing capabilities, and simpler implementation. DSP has become a fundamental area of research for many real-world applications, e.g., mobile phones, digital cameras, GPS, video/tele-conference, radar, MP3 players and many more. The work presented here is focused on DSP for hearing aids which is important for hearing impaired people in order to communicate and interact with other people in the daily life. It should be mentioned that some of the algorithms developed here can be applied to, e.g., hands-free telephony, in-vehicle communication, and public address systems. The two types of technology for hearing aids are analog and digital. Current state-of-the-art hearing aids are exploiting various aspects of DSP and according to 93 percent of all hearing aids sold in 2005 were digital. The core function of traditional hearing aids is mainly based on signal amplification. However, digital hearing aids allow for more advanced signal processing since the purpose of modern hearing aids is not only to amplify sounds.

This dissertation addresses several topics in DSP for hearing aids, namely noise reduction (NR), dynamic range compression (DRC), and adaptive feedback cancellation (AFC) which is only a subset of DSP algorithms that are used to build a digital hearing aid. The design of NR, DRC and AFC is closely related and equally important. Reducing acoustic feedback increases the available gain and allows the hearing aid to get closer to the prescribed gain. Making speech audible does not mean that hearing aid users can understand the speech without enhancement of, e.g., spectral or spatial signal information. This of course becomes more crucial when the hearing aid user is listening in the presence of background noise which makes NR an important component as well.

In this introduction, we will briefly motivate and explain the problems related to hearing aids and hearing impairment. An overview of open problems and state-of- the-art DSP algorithms in the areas of NR, DRC and AFC will also be discussed. At the end of the introduction we will explain how this work fits within the current open problems in hearing aids and point out the main contributions of this work together with a chapter-by-chapter outline.

Hearing impairment

Hearing impairment is becoming more common and can be caused by many reasons. Other reasons are daily exposure to excessive noise in the work environment (construction site, factory etc.) and listening to loud music (MP3 players, iPod, concerts, night clubs etc.). In general two factors can be mentioned as the primary reasons that can cause hearing loss, i.e., the level of the sound and the duration that people are exposed to this sound. The function of these hair cells is to convert sound energy into electrical signals that are sent to the brain by the auditory nerve.

In our daily-life we are often exposed to sounds with high intensity without realizing the danger to our hearing abilities. The consequence of NIHL can typically not be reversed by surgical or medical procedures, i.e, once the hair cells are damaged they cannot grow back again. Typically the damage is done when people realize that they have a NIHL. Sound levels are typically measured in decibels (dB) which is not necessarily something that we think about when we are in various environments. To give a perspective on the different sound levels that we can be exposed to some examples are

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shown in Figure shows hazardous exposure limits for various sound levels. This shows that the louder the sound is the shorter the time is before NIHL occurs. Sounds less than 75dB are unlikely to cause NIHL even after a long exposure time. Another factor that can play a role is of course the distance to the sound source(s).

For a perspective the degree of hearing loss can be compared to the level and frequency of average speech which is shown in Figure For hearing impaired people with mild to moderate hearing loss a hearing aid is needed in specific situations or at least on a frequent basis. For severe hearing loss a hearing aid is needed for all communications and for profound hearing loss the use of a hearing aid may be combined with speech-reading (lip-reading) or sign language. Furthermore there exist three distinct types of hearing loss.

Noise reduction in hearing aids Background noise tends to decrease the speech intelligibility especially for people suffering from hearing loss. The topic of NR in hearing aids is therefore of great importance and many different DSP strategies have been addressed in the past. The goal of NR algorithms is to reduce the background noise and enhance the desired speech signal in complex acoustical environments in order to improve speech intelligibility and/or listening comfort by increasing the SNR without introducing any signal distortion.

NR algorithms can be classified as either fixed filters or adaptive filters. Adaptive filters on the other hand are more flexible and can adapt the filter characteristics automatically depending on the input signals. The general trade-off in NR is the amount of noise that can be removed versus speech distortion. NR algorithms can also be categorized into single-channel NR and multi-channel NR and here we will provide a broad overview of different NR algorithms.

Voice activity detection

The fundamental component of any NR algorithm is the voice activity detector (VAD). Typically a stationarity assumption is made such that the noise characteristics can be estimated and updated during noise-only periods. The purpose of a VAD is to distinguish between speech dominant segments and noise dominant segments (silence) which can be a challenging problem. . In the past various VAD algorithms have been proposed which all are aimed at improving the robustness, accuracy and reliability. A VAD algorithm can be divided into two separate blocks, i.e., feature extraction and a decision module. The objective of feature extraction is to find discriminative speech features that can be used in the decision module. In this section we will give a brief overview of existing VAD algorithms.

Non-parametric noise reduction

Non-parametric NR relies on an estimate of the noise characteristic estimated during noise-only periods, e.g. using a VAD or a minimum statistics algorithm, which then is applied during speech-plus-noise periods to extract the clean speech signal. Many single-channel NR algorithms.The analysis and synthesis part of the different NR algorithms is commonly performed using the short-time Fourier transform (STFT) with overlap-add or overlap-save procedure. Spectral subtraction

Features used in VAD algorithms have been based on: energy levels, pitch, and zero crossingrates, the LPC distance measure, cepstral features, adaptive noise modeling of voiced signals, the periodicity measure, and high order statistics (HOS). The problem with these approaches is the robustness at low input SNR and with non-stationary noise sources since the VAD is typically based on a fixed threshold.

Recent approaches to improve the VAD performance are based on using statistical models with the decision rule derived from the statistical likelihood ratio test (LRT) applied to a set of hypotheses. Other decision rules have been based on: Euclidean distance , Itakura-Saito and Kullback-Leibler divergence, fuzzy logic, and support vector machines (SVM). Various statistical models have been proposed to improve the VAD performance such as Gaussian, Laplacian, and Gamma models. VADs can also be distinguished based on whether a hard decision (binary) or a soft decision (value between 0 and 1) is used.

It is obvious that an accurate estimate of the noise spectrum is the key to an improved estimate of the original speech. A common noise estimation technique is based on a recursive averaging procedure during periods where the speech is absent and then keeping the noise estimate fixed during periods where speech is present. This approach requires a VAD which in itself suffers from reliability at low input SNR. An interesting approach has therefore been proposed in [35] called improved minima-controlled recursive averaging (IMCRA). Basically the smoothing parameter is now adapted over time and frequency based on the conditional speech presence probability (SPP). The advantage of IMCRA is the continuous update of the noise spectrum and the fact that a binary VAD is not required.

Another common technique to estimate the noise characteristics is known as the minimum statistics algorithm. This approach differs from the traditional VAD methods since the minimum statistics algorithm does not need to distinguish between speech activity and speech pauses. Instead the minimum statistics algorithm is based on the fact that during speech pauses the speech energy is close to zero which means that by tracking the minimum power the noise floor can be estimated [140][141]. In [86] an approach for noise tracking is proposed where the noise PSD can be updated in the presence of both speech and noise. This method is based on the eigenvalue decomposition such that the noisy speech can be decomposed into a signal-plus-noise subspace and a noise-only subspace. Other techniques that can be mentioned are histogram [188] and quantile based noise estimation techniques.

depends on a VAD such that the noise power can be kept fixed during speech segments and updated during noise-only segments which requires a stationarity assumption on the background noise.

2. SPEECH DISTORTION WEIGHTED MULTI-CHANNEL WIENER FILTER (SDW-MWF^)

In this chapter we will introduce the multi-channel Wiener filter (MWF) and establish a baseline for the research and

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development related to the MWF based NR algorithm. Over the years many modifications and formulations have been introduced in the MWF either to improve the performance or the robustness. The benefit of using an MWF based NR compared to, e.g. an GSC or an LCMV beamformer, is the reduced sensitivity against signal model errors such as microphone mismatch. For multi-channel NR algorithms like the LCMV or the GSC beamformer a priori assumptions regarding the desired signal model are required, e.g., location of the desired speaker, calibrated microphones, low reverberation etc. The performance of the MWF has been evaluated in which showed that the MWF outperformed the GSC in adverse listening environments. Furthermore perceptual evaluation with normal hearing and hearing impaired subjects performed in indicated a significant SRT improvement for the MWF algorithm.

In this work we will focus on two main extensions of the MWF which have shown to have certain interesting features but at the same time we will show that there is still room for improvements. The first extension of the MWF is based on introducing a weighting factor that allows for a trade-off between NR and speech distortion referred to as the speech distortion weighted MWF (SDW-MWF). This is an interesting approach however it is not clear how to actually select this weighting factor and therefore in the past this weighting factor has simply been a scalar valuethat is fixed for all frequencies and for all frames. The second extension is based on a rank-1 formulation of the SDW-MWF which has been shown to be more robust against estimation errors in the correlation matrices [38]. The interesting feature of the rank-1 formulation is that the SDW-MWF is now decomposed into a spatial filter and a single-channel postfilter. However it still remains an open problem if the single-channel postfilter is optimal in terms of spectral tracking since it is based on correlation matrices that are adapted slowly over time.

3. SDW-MWF^ BASED ON SPEECH PRESENCE PROBABILITY (SPP)

This chapter address the issue of using a weighting factor to trade-off between NR and speech distortion that is kept fixed for each frequency and for each frame which potentially can limit the spectral tracking of the non-stationarity of the speech and the noise. Combined with the fact that the correlation matrices are kept fixed at different time instants all together with a long averaging time the fixed weighting factor certainly does not help the spectral tracking.

To tackle this problem we have been inspired by an interesting technique which has primarily been used in single-channel NR algorithms, referred to as the conditional speech presence probability (SPP). This technique is based on a two-state speech model, i.e, a Ho(k, l) state represents speech absence and a H\(k, l) state represents speech presence which is defined for each frequency and for each frame. This model is based on the fact that the noise can be assumed to be continuously present whereas speech typically contains many pauses. For this reason the conditional SPP is estimated and updated for each frequency and for each frame. Since single-channel NR algorithms are not able to exploit spatial signal information extensive research has been conducted to obtain a spectral distinction between the speech and the noise which is something that has not received a lot of attention in multi-channel NR algorithms. In multi-channel NR the concern has primarily been to improve the spatial filtering. For this reason we propose that the conditional SPP is incorporated in the SDW-MWF based NR such that the speech dominant segments and the noise dominant segments can be weighted differently.

4. SDW-MWF^ BASED ON ROBUST ESTIMATION OF THE CORRELATION MATRICES

This chapter addresses the issue of using correlation matrices that are kept fixed during speech-plus-noise periods and are updated during noise-only periods or vice versa. As mentioned this can limit the tracking both spectrally and spatially. The robustness of the correlation matrices can also be compromised especially when estimating the clean speech correlation matrix which requires an accurate estimation of the noise-only correlation matrix.

For this reason we once again turn our attention to single-channel NR where several attempts have been made to continuously track and update the noise power during periods of speech-plus-noise [133]. One of the interesting approaches is referred to as the improved minima controlled recursive averaging (IMCRA) noise estimation approach where the conditional SPP is used as a time-varying smoothing factor [35][36]. Inspired by this approach we propose a robust method to estimate and update the correlation matrices that exploits prior knowledge of the correlation matrices combined with a continuous update approach based on the conditional SPP.

In this chapter we have introduced an SDW-MWF-based NR that incorporates a robust method to estimate and update the correlation matrices. The robust estimation of the correlation matrices is based on introducing prior knowledge of the correlation matrices together with a continuous updating approach based on the conditional SPP. Combining this method to estimate and update the correlation matrices with a weighting factor to trade-off between NR and speech distortion that also varies for each frequency and for each frame, resulted in a novel SDW-MWF based NR that improves the robustness and the tracking capabilities.

Experimental results show that the proposed algorithm improves the SNR and the signal distortion compared to the traditional method with a perfect VAD used to estimate and update the correlation matrices for both the SDW-MWF and the rank-1 SDW-MWF approaches. Analysis has shown that the estimated correlation matrices using a perfect VAD results in negative power in the estimated speech correlation matrix which in practice should not happen since the speech correlation matrix is estimated by subtracting the noise-only correlation matrix from the speech-plus-noise correlation matrix.

The SDW-MWF based NR proposed here has solved the problems of estimating and updating the correlation matrices

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in a robust way such that the speech correlation matrix can be reliably estimated. This is achieved by continuously estimating the noise-level in the speech-plus-noise and the noise-only correlation matrix during both speech-plus-noise and noise-only periods. Is has also been shown how the conditional SPP can be used to further improve the single-channel postfilter by exploiting the proposed correlation matrices. Furthermore, the proposed correlation matrices also alleviate the sensitivity of the traditional SDW- MWF which was the reason to use the rank-1 SDW-MWF in the first place.

5. ROBUST CAPON BEAMFORMING FOR SMALL ARRAYS

This chapter presents a different multi-channel NR algorithm based on a standard Capon beamformer (SCB) also referred to as an MVDR beamformer. The main difference between the SCB and the MWF is that the SCB relies on a correct estimation of the steering vector of the target speech signal whereas the MWF is uniquely based on the estimated correlation matrices. This means that the estimated correlation matrices are not mainly responsible for the SCB performance but the target now is to find a robust method to estimate the steering vector. Therefore a robust Capon beamformer (RCB) is presented where the target is to adaptively estimate the steering vector in the presence of reverberation and noise. The proposed RCB is based on using prior knowledge of the steering vector combined with a steering vector uncertainty principle.

The SCB suffers from a substantial performance degradation when there is a mismatch between the presumed and the actual steering vector of the target signal. Therefore many approaches have been proposed to improve the robustness of the SCB. A variation of the SCB is known as the linearly constrained minimum variance (LCMV) beamformer [93] where a set of linear constraints is added. These constraints broaden the main beam by imposing a set of unity-gain constraints for steering vectors close to the presumed steering vector of the target signal such that robustness against a steering vector mismatch is achieved. A drawback with the LCMV is that each constraint removes one degree of freedom for interference suppression. Other robust extensions of the SCB have been based on diagonal loading of the sample correlation matrix. The main problem with these approaches is to find the optimal value of the diagonal loading factor and that it reduces performance and the beam sharpness. Recent approaches estimate the diagonal loading factor based on the uncertainty region of the presumed steering vector of the target signal. These methods are robust against target signal suppression when the actual steering vector is within the predefined uncertainty region. Spherical, ellipsoid and polyhedron uncertainty regions have all been studied.

In it is shown that a frequency-domain SCB outperforms a time-domain Frost beamformer and a generalized sidelobe canceler for a scenario with two or more nonstationary interfering speech sources and an array with two microphones. A frequency-domain SCB exploits the time-frequency sparseness of the sources better than a time-domain implementation.

In a fixed steering vector is used for the target signal; the goal in this chapter is to extend the frequency-domain SCB to an adaptive frequency-domain RCB. The RCB proposed here is based on a gradient approach where the steering vector is adaptively estimated based on a predefined level of uncertainty in the steering vector. The proposed RCB offers a low complexity, simple implementation and suffers no loss of degrees of freedom for interference suppression.

6. DYNAMIC RANGE COMPRESSION(DRC)

This chapter introduces the DRC algorithm used in this dissertation. DRC is a basic component in digital hearing aids and the use of DRC in hearing aids has increased over the years. DRC is a signal processing strategy that makes speech audible over a wide range of sound levels and reduces the dynamic range of speech signals. Basically, a DRC is an automatic gain control, where the gain is automatically adjusted based on the intensity level of the input signal. Typically the design of DRC is based on clean speech scenarios without considering the presence of background noise. Therefore the work here is focussed on the design of DRC operating in the presence of background noise, i.e., to analyze how the DRC reacts to the background noise compared to clean speech scenarios.

Design of DRC algorithms

Reduced audibility and reduced dynamic range between the hearing threshold and and the uncomfortable level are some of the problems that people with a sensorineural hearing loss are dealing with [45][100]. The role of dynamic range compression (DRC) algorithms in hearing aids is to map the wide dynamic range of speech signals into the reduced dynamic range of hearing impaired listeners. Hearing aids or more specifically DRC should enhance the speech signal such that all of the important features of the speech signal are above the hearing threshold but at the same time below the discomfort level [149]. This is achieved by allowing more gain at low input levels and less at higher input levels which means that the DRC provides comfort for loud sounds and audibility for soft sounds.

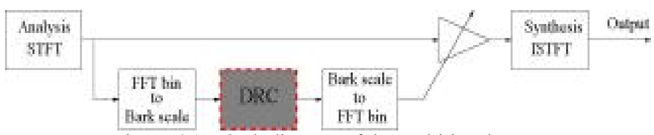

Figure 6.1: Block diagram of the multi-band DRC

Even though DRC is a main component in hearing aids there is still a disagreement about the best way to incorporate DRC in hearing aids [45][100]. In the past extensive work has analyzed the challenges and difficulties in the design of DRC algorithms. The general design of different DRC algorithms can be found in. The aim here is to show the effect that background noise has on DRC, and to discuss the problems and challenges when designing DRC algorithms in the presence of background noise.

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Multi-band compression

The developments on DRC have mainly been focussed on multi-band DRC since the hearing loss and the dynamic range of speech varies markedly with frequency. This can be achieved by using filter banks or an FFT approach. In this work the critical bands [247] are realized using an FFT approach such that the FFT bins are combined to produce a critical band spectrum. A block diagram of the multi-band DRC is shown in Figure 6.1. First the input signal is divided into frames using either an overlap-add or an overlap-save procedure with a window function. Then an FFT is performed on each frame and as input to the DRC the FFT bins are combined to produce a critical band spectrum. The DRC block then estimates

7. PREDICTION ERROR METHOD-BASED ADAPTIVE FEEDBACK CANCELLATION

This chapter introduces the prediction error method-based adaptive feedback cancellation (PEM-based AFC) together with the idea of using a near-end signal model. In four commercial hearing aids were evaluated and compared to the PEM-based AFC. It was shown that the PEM-based AFC offered a high added stable gain (ASG) compared to certain commercial hearing aids. However the PEM-based AFC was more sensitive towards tonal input signals. This is mainly due to the near-end signal model used which in this case was based on linear prediction (LP). For this reason a cascaded near-end signal model is introduced.

The notation related to PEM-based AFC will be given together with the evaluation using objective measures such as maximum stable gain (MSG) and filter misadjustment which differ from the objective measures used to evaluate the NR algorithms. Furthermore the advantage and disadvantage of the current PEM-based AFC will be discussed and the motivation for further research on the PEM-based AFC will be explained.

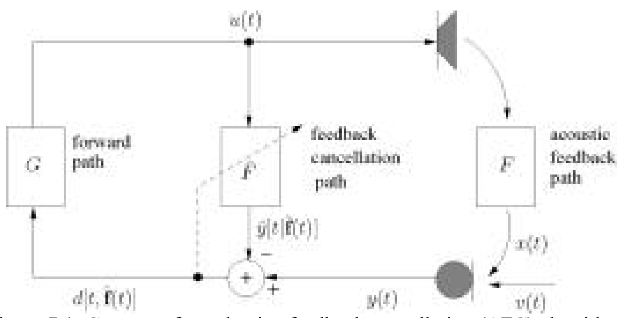

Figure 7.1: Concept of an adaptive feedback cancellation (AFC) algorithm.

pole zero LP (PZLP) model. The idea is that the LP can model the noise component whereas the PZLP can model the tonal components. In this way a better decorrelation effect is achieved with tonal input signals. Section 8.4 presents the experimental results.

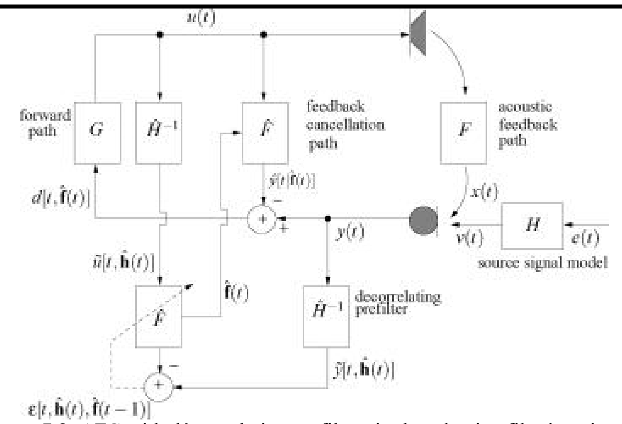

Figure 7.2: AFC with décor relating prefilters in the adaptive filtering circuit.

from the microphone signal y(t). The feedback-compensated signal is given by

$d(t) = v(t) + [F(q,t) - F(q,t)]u(t).$

8. PEM-BASED AFC USING A HARMONIC SINUSOIDAL NEAR-END SIGNAL MODEL

This chapter addresses the issue of designing improved prediction error filters for PEM-based AFC. To this aim a harmonic sinusoidal near-end signal model is introduced together with various pitch estimation techniques. Basically the idea is to find an improved method to represent the near-end signal by using the knowledge that we have regarding speech signals. For this purpose we have turned our attention to the research area of speech and audio coding based on harmonic sinusoidal based pitch estimation techniques. The reproduction of speech signals highly depends on an accurate estimation of parameters such as pitch, amplitude, model order, and the selection of voiced-unvoiced frames. It is therefore interesting to analyze if a more accurate estimation of the near-end signal model would result in improved PEM-based AFC performance.

9. CONCLUSION

Hearing impairment can be caused by many factors, e.g., daily exposure to excessive noise in the work environment and listening to loud music which are scenarios that we all can be exposed to in our daily life. Hearing loss can also be age-related which makes the research on hearing aids very important. For hearing aid users background noise and acoustic feedback imposes a major problem in terms of speech understanding and listening comfort. If these problems are not resolved some hearing aid users may even choose not to use their hearing aids. The overall objective of this dissertation is the design of DSP algorithms for hearing aids. The focus is on three main areas such as NR, AFC, and DRC. The DSP algorithms considered here are all adaptive approaches which is important when dealing with time-varying acoustic environments, reverberation, and nonstationary signals such as speech and multi-talker babble.

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