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UNIFORM POLYPHASE FILTER BANKS FOR USE IN HEARING AIDS: DESIGN AND CONSTRAINTS

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ABSTRACT

Signal processing in hearing aids for purposes such as hearing loss compensation or noise reduction requires transforming the input signal into frequency bands. Additional audiological constraints need to be considered to accurately model the hearing loss compensation as well as to prevent unintended artefacts. This necessitates highly efficient solutions with low delay, strong alias suppression and sufficient frequency resolution. After discussing requirements from several aspects, we show how uniform polyphase filter banks can be designed to meet these requirements. One important topic is a prototype filter design, for which we present numerically optimized solutions. Finally, we show results for quantized filter coefficients.

1. INTRODUCTION

Filter banks for signal processing applications are widely applied in technical systems and provide a means for frequency selective processing of audio and video. Each application provides its own set of requirements which needs to be met. For the important field of speech and audio compression, for example, it is of high priority to provide means for critical subsampling to keep the resulting amount of data as low as possible. On the other hand, the requirements on pure stop band attenuation may be relaxed due to aliasing cancellation properties of the filter bank (as provided by quadrature mirror filters, for example). Of course, this effect cannot be utilized if the signal undergoes significant manipulation as in hearing aids.

For the utilization of filter banks in hearing aids, several aspects of the human auditory system need to be considered in order to provide accurate hearing loss compensation. Former analogue devices had very limited capabilities in terms of frequency shaping. With the advent of hearing aids with digital signal processing, which emerged during the 90's, the configurability improved drasticly. Now it was possible to implement state-of-the-art digital signal processing algorithms which, under certain constraints, provide a new quality of hearing to the wearer.

Besides accurate frequency shaping and dynamic compression adapted to the auditory needs of patients with hearing loss, many more algorithms benefit from frequency selective processing [1]. Candidates are for example directional microphones, feedback cancellation, single channel noise reduction or advanced binaural algorithms. To facilitate implementation of such computationally expensive algorithms in hearing aids, subsampled processing of the frequency bands is highly desirable.

On the other hand, with complex signal processing, it is necessary to keep artefacts of the processing system unnoticable to the hearing aid wearer. Unfortunately, the sensitivity to artefact conception by hearing impaired people can not be modelled easily due to the broad variation of hearing losses, and thus, conservative assumptions may be applied here.

In Section 2 we discuss general requirements for filter banks in hearing aids. In Section 3 several well known concepts for filterbanks are presented and reviewed under the given constraints. An optimized design of a prototype filter is derived in Section 4, where we focus on stop band attenuation with special attention to quantization of the filter coefficients. In Section 5 final conclusions are drawn.

2. FILTER BANKS FOR HEARING AIDS

For the design of filter banks for hearing aids requirements from several aspects need to be considered, which are partly even contrary. From an algorithmic point of view, several constraints have to be met:

- It is desirable to have uniformly spaced frequency bands, which simplifies subsampling of the bands and thus helps saving power. Especially with more and more processing inside the frequency bands, subsampling is one of the most effective ways to reduce computational complexity by a far amount.
- For frequency selective algorithms, most prominently single-channel noise reduction, it is usually desirable to have narrow bands with little overlap between the bands. Thereby, desired signal and noise can be estimated and modelled accurately.
- Hearing loss compensation often requires a wide range of different gains applied to the different frequency bands. For example, a so-called ski-slope hearing loss needs very little amplification in the lower frequencies, where the high frequencies require a high amplification. To avoid aliasing components to enter the hearing area of the hearing impaired, aliasing and imaging suppression need to be considerably high. Therefore, the stop band attenuation needs to be at least 60 dB in typical hearing aid application, preferably higher.

Especially in hearing aid applications, the overall delay which is imposed onto the processed signal is crucial to the subjective listening quality. For lip-synchronicity, a delay of $\approx$ 70msec is sufficient to keep the sound subjectively synchronous to lip movement. Therefore, lip-synchronicity is no hard limitation compared to other contraints:

- Comb filter effects occur when sound processed by the hearing aid is overlaid at the eardrum with the sound which directly travels through the hearing aids vent. Figure 1 depicts the 2 paths, where the path $H_{\mathrm{HA}}(f,t)$ through the hearing aid usually has a much larger group delay than the direct path $H_{\mathrm{vent}}(f)$ through the vent. When these two signals arrive at approximately the same power density in a certain frequency band, a comb filter effect can be observed in this frequency band. The venting itself can range from closed earpieces, which provide a high attenuation of the sound, through earpieces with a venting bore from 1mm to 3mm, which typically have some kind of lowpass characteristic, to completely open fitted devices, which impose almost no attenuation to the sound. The worst case effect is depicted in figure 2. It is obvious that even for a very short processing time of 3msec, the combfilter effect can be observed clearly. With a 10msec delay, a clear increase of the notch density is obvious, though the notch width shrinks accordingly. These effects have no significant effect on the subjective listening quality at least up to a signal delay of $\approx$ 10msec [2]. At delays of significantly more than 10msec, the time resolution of the human ear becomes of interest [2], and the subjective impression resembles an echo.

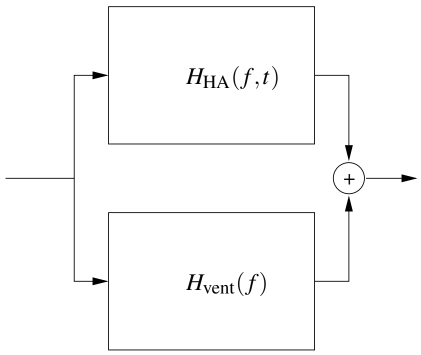

Figure 1: Comb filter effects with hearing aids.

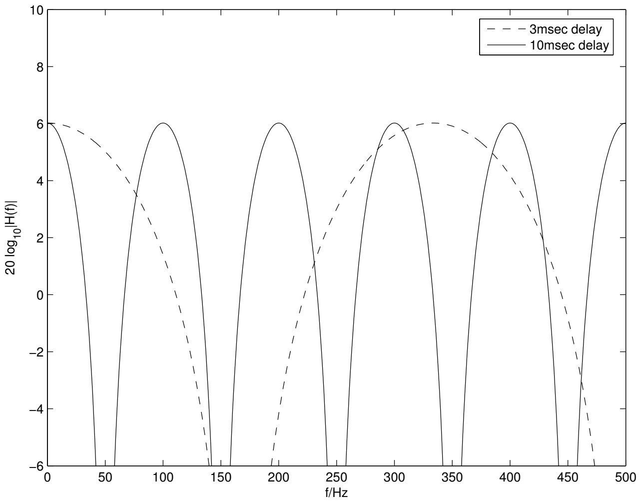

Figure 2: Comb filter effect for a signal delay of 3msec and 10msec.

- As mentioned above, strong aliasing suppression is crucial if subsampling is applied to the frequency bands. There are two effects in digital hearing aids which tighten the requirements on aliasing suppression: first, potentially significantly different gains are applied to the different frequency bands. This may lead to a situation, where the unwanted aliasing term is amplified significantly more than the original desired signal. And second, the compression system in digital hearing aids, which, on purpose, gives a higher gain onto signal bands with lower power than onto signal bands with high power. This may lead to a high amplification of the undesired aliasing term and a low amplification of the desired signal.
Of course, one has to consider that hearing loss compensation tries to restore the hearing capability to the level of a normal hearing person, which means that signals below the hearing threshold of a normal hearing will still be below the hearing threshold of a hearing impaired, even after amplification. The hearing threshold here combines the absolute hearing threshold and the masking threshold.
Attractive filter banks with perfect reconstruction which utilize aliasing cancellation, e.g. QMFs, can not be used for hearing aid applications due to the heavy signal manipulation in the different bands, which does not aim at a *reconstruction* of the signal, per se. In this case the aliasing cancellation property of the filter bank is lost.
- The processing of signals in digital hearing aids result in some form of delay in the signal and can be modelled as a kind of group delay, which is well known from LTI systems. With sufficiently high aliasing and imaging suppression and adaptive processing switched off, the hearing aid can be treated as an LTI system, and the group delay can be evaluated. The human ear is in general rather robust to moderate phase distortion. Though, if there is a rapid change in the group delay over frequency, the effect can be noticable to the listener [3].

3. FILTER BANK SELECTION

There are numerous concepts available for filter bank design. Due to the tight power constraints in hearing aids, we focus on filter bank concepts here for which an efficient implementation is feasible, but also meet the requirements stated above [6, 9, 5, 4].

3.1 Filter bank concepts

- Cascaded IIR Filter banks: cascades of lowpass-highpass combinations lead to a flexible design where each cut-off frequency can be individually adjusted. There are efficient implementations based on wave digital filters available, but the filter bank itself needs to run on a full input clock. In general, phase distortions occur and need to be taken care of by appropriate filter design.
- STFT/windowed overlapped FFT: Based on an FFT, an efficient implementation is available, but subsampling performance is mediocre at the targeted aliasing suppression due to the limited window length.
- Uniform polyphase filter banks: Closely related to the STFT, the subsampling performance can be adjusted arbitrarily by a flexible prototype design. It can be efficiently implemented based on an FFT of appropriate length and subsampling can always be performed *before* the filter bank when using FIR prototype filters. With an IIR prototype filter, tight restrictions are placed upon the prototype design when the filter bank shall be subsampled which make IIR prototype filters less attractive.
- Warped polyphase filter banks: In contrast to the uniform filter bank, the individual bandwidths are adjustible which would allow direct adaption to human auditory system. As a downside, the possibilities for subsampling prior to the filtering operation are very limited and increase complexity due to different sampling rates in the different bands.

3.2 Uniform polyphase filter banks for use in hearing aids

Due to its flexibility and efficiency, in the following we focus on uniform polyphase filter banks with FIR-prototype filters. For a hearing aid application, an audio bandwidth of up to 8kHz is assumed. In conjunction with an FFT of length $M = 64$, this results in a channel spacing of $f_c = 250$Hz of the 33 usable bands and a sampling rate of $f_s = 16$kHz. The achievable decimation factor and the filter bank delay is dependent on the prototype filter length. Without loss of generality, we fix subsampling to a decimation factor of $R = 16$, i.e. quarter critical decimation unless otherwise noted. This leads to a decent trade-off between aliasing suppression and filter bank delay as we will see in the remainder of this paper.

The overall output signal of the filter bank analysis and synthesis system is given by

$$Y(e^{j\Omega}) = \sum_{r=0}^{R-1} X(e^{j(\Omega+\frac{2\pi}{R}r)}) \sum_{m=0}^{M-1} H(e^{j(\Omega+\frac{2\pi}{M}m+\frac{2\pi}{R}r)})G(e^{j(\Omega+\frac{2\pi}{M}m)}) \tag{1}$$

where X is the input signal, H the analysis and G the synthesis filter prototype frequency respone. Figure 3 illustrates how the overall flat frequency response is composed by superposition of the channels.

Figure 4 shows overall frequency responses of the filter bank fitted to 4 exemplary linear hearing loss fitting targets. We get over-

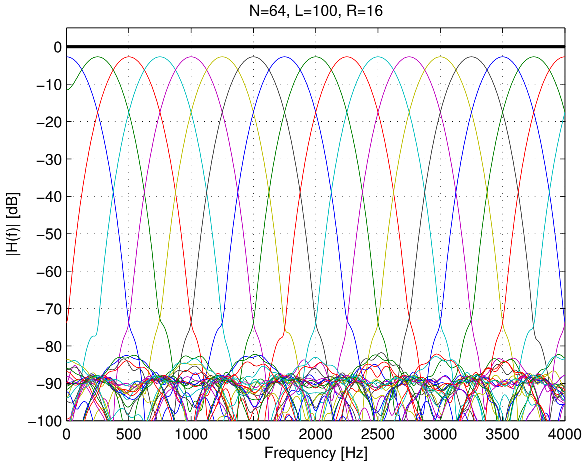

Figure 3: Overall frequency response and shape of the single channels.

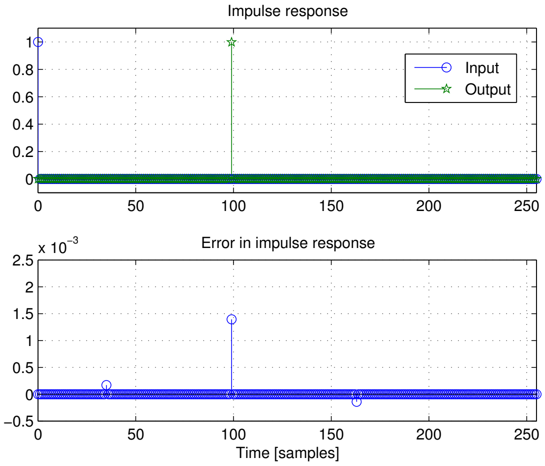

Figure 5: Impulse response of filter bank with prototype filter of lenght $L = 100$.

all very good adherence to the targets, with at most 2.7 dB deviation at single frequency points.

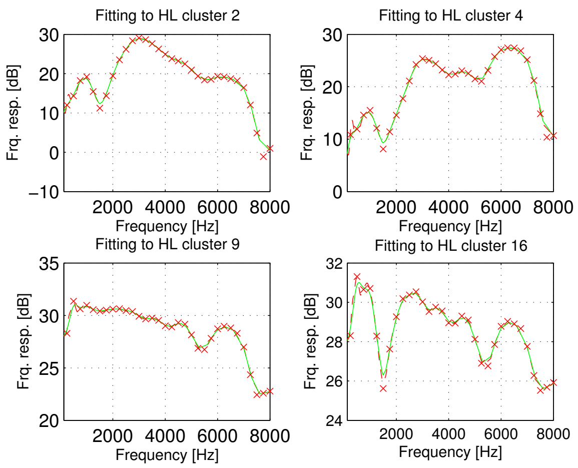

Figure 4: Overall frequency response fitted to example hearing losses: Achived frequency response (solid), target (dashed), discrete gains(x).

4. PROTOTYPE DESIGN FOR UNIFORM POLYPHASE FILTER BANKS

The choice and design of the prototype filters has significant impact on the properties of a polyphase analysis-synthesis filter bank system for subband processing. Since we desire very low overall delay we concentrate on FIR prototype filters of arbitrary length L. By appropriate zero-padding to an integer multiple of M, the filter can be efficiently implemented in the subsampled domain while overall group delay is directly given by the number of taps in the prototype filters as $t_d = L/f_s$. Figure 5 shows the reaction of the filter bank with $L = 100$ to excitation with a single impulse and the deviation from ideal behavior.

For extensive signal manipulation in the subbands between analysis and synthesis filter banks, as required for hearing aid applications, only sufficiently low aliasing and imaging in the reconstructed signal can be tolerated. A worst case estimation, assuming constructive superposition of all undesired aliasing components can be obtained by

$$A(e^{j\Omega}) = \frac{\sum_{r=1}^{R-1}\sum_{m=0}^{M-1} |H(e^{j(\Omega+\frac{2\pi}{M}m+\frac{2\pi}{R}r)})G(e^{j(\Omega+\frac{2\pi}{M}m)})|^2}{\sum_{r=0}^{R-1}\sum_{m=0}^{M-1} |H(e^{j(\Omega+\frac{2\pi}{M}m+\frac{2\pi}{R}r)})G(e^{j(\Omega+\frac{2\pi}{M}m)})|^2} \quad (2)$$

which depends on the decimation factor R and the frequency selectivity of the prototype filters. Longer prototype filters allow better frequency selectivity and hence more subsampling but obviously contradict low-delay requirements and increase computational complexity of the filter bank itself, so tradeoffs have to be found. We present some results with filters designed according to [7]: We start with linear-phase Dolph-Chebyshev windows with prescribed stop band attenuation U_s which is nearly constant over frequency. The autocorrelation function of this filter is then multiplied with a $sinc()$ function to enforce perfect reconstruction, and the result is split into a minimum-phase analysis filter $h(k)$ and maximum-phase synthesis filter $g(k) = h(L-k-1)$. Other design approaches, for example for obtaining linear phase prototype filters [8], are possible. Figure 6 shows the impulse response of such a filter of length $L = 100$.

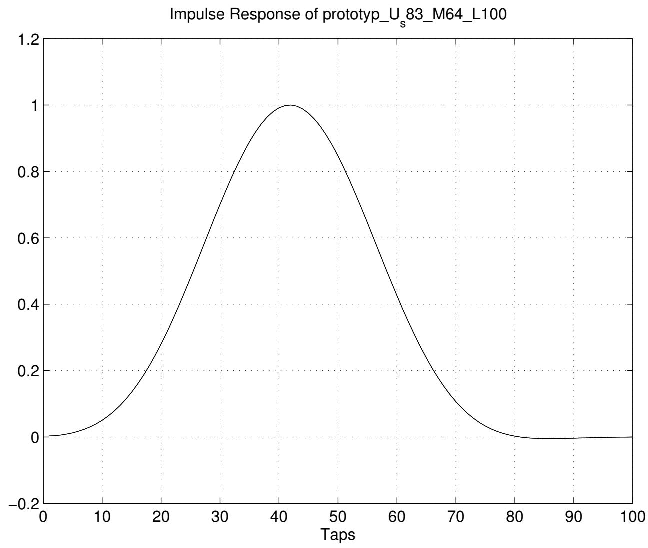

Figure 6: Impulse response of minimum phase protoype with $L = 100$.

Figure 7 shows aliasing for filters of length $L = 100$ and a filter bank with quarter critical subsampling ($M = 64$, $R = 16$) designed for various initial filter stop band attenuations U_s. For small U_s and hence prototypes with narrow pass band, the overall aliasing is determined directly by the superposition of the stop band components of the prototypes. For large U_s, the performance is determined by the overlap of the slopes of neighboring filters in the subsampled domain. A compromise can be found, in the example corresponding to $U_s = 83$ dB if maximal aliasing is considered as criterion or to $U_s = 85$ dB if mean aliasing over frequency is considered.

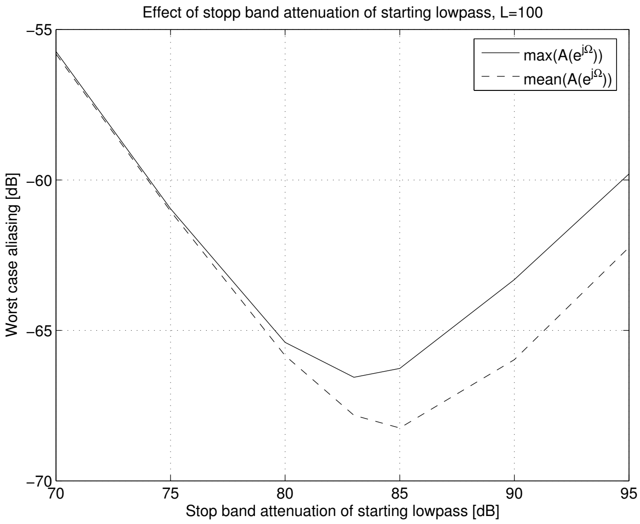

Figure 7: Worst case aliasing for prototype filters with $L = 100$ designed from starting low passes with different stop band attenuation for $R = 16$.

The final choice of the prototype filter can now be made as a compromise between achievable decimation factor and required filter length. Figure 8 illustrates this by plotting achievable maximal alias suppression over various decimation factors for filter prototypes of different lengths. Stop band attenuation of the starting window of each filter was designed to achieve minimal $\max_\Omega(A(e^{j\Omega}))$, as illustrated in figure 7 for each R. If for example we require $\max_\Omega(A(e^{j\Omega})) < -60$ dB we see that none of the filters of length up $L = 256$ satisfies this criterion if we desire half-critical subsampling ($R = 32$). For $R = 12$ on the other hand any filter, down to $L = M = 64$ is sufficient. Since implementation is eased by choosing an integer decimation factor M/R a possible choice is $R = 16$ and $L = 100$, corresponding to a delay of t_d =6.25msec.

4.1 Improved prototype filters

We have seen that filter bank performance largely depends on the chosen prototype. Following established design procedures, we arrive at some minimal filter length and thus delay for fulfilling given aliasing and subsampling requirements. Equivalently we may wish to reduce $A(e^{j\Omega})$ as given by (2) for given filter length and deciamtion factor. We formulate a distortion criterion

$$d = \frac{1}{\pi}\int_0^\pi A(e^{j\Omega})d\Omega + \lambda \max_\Omega(A(e^{j\Omega})) \qquad (3)$$

which expresses a trade-off between mean and maximal aliasing over frequency as determined by the parameter λ. We now seek a set of prototype filters with reduced distortion d compared to an initial set of filters designed by traditional methods. We note that the coarse form of a filter impulse response mainly determines the low-frequency parts of the transfer function and thus imaging of a filter bank while the fine structure determines the high frequency behavior. We start with a filter which satisfies our constraints w.r.t. possible subsampling of the filter bank, but is chosen shorter than

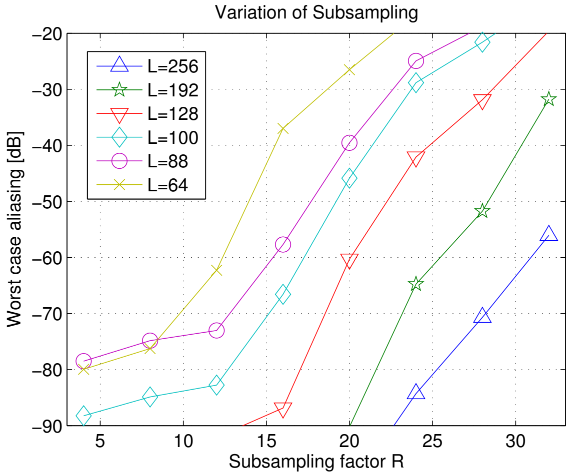

Figure 8: Aliasing over decimation factor for filters of different lengh L.

required for satisfactory alias suppression. The coefficients of this filter are then modified in small steps to locally minimize our distortion criterion (3). This can be done by numerical optimization techniques such as simulated annealing which also allows to incorporate further constraints on the filter coefficients such as minimal or linear phase. The choice of (3) improves convergence compared to directly minimizing maximal aliasing. Other optimization criteria and approaches would also be possible [10].

Figure 9 shows the frequency response before and after this optimization procedure for a minimum phase filter of length $L = 92$ and for a filter bank with FFT-length $M = 64$ and decimation by $R = 16$. We see that the behavior in the pass band is virtually identical while the stop band attenuation reflects the structure of the filter bank: Attenuation is high, where undesired superposition is possible, while requirements are relaxed where undesired signals fall within the stop band of both analysis and synthesis filter.

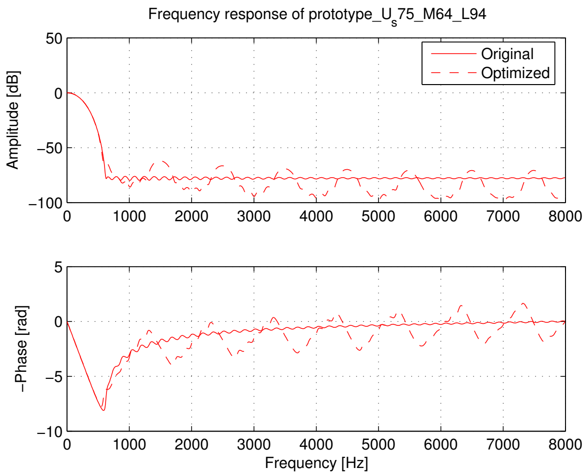

Figure 9: Optimized prototype filter.

The optimized filter achieves maximal worst case aliasing of -66.28 dB, compared to -60.18 dB for the initial filter with $U_s = 75$ dB and -60.72 dB for $U_s = 77$ dB. This performance is comparable to the best prototype filter of length $L = 100$ shown in figure 7 achieving -66.56 dB alias suppression. We thus see that local optimization of prototype filter coefficients lets us achieve better alias suppression or allows the use of shorter prototype filters, resulting in less filter bank delay and less computational complexity for achieving the same performance.

While optimization may result in filters violating the necessary conditions for perfect reconstructing filter banks, we still arrive at *nearly* perfect reconstruction which is good enough for our purposes. By further relaxing the perfect reconstruction requirement, it is possible to achieve even better aliasing properties or shorter filters.

4.2 Quantization of filter coefficients

Battery power and available chip size are still very limited resources in hearing aids. For efficient implementations we thus strive to use fixed-point arithmetic with word lengths as short as possible. In this section we demonstrate the effect of quantization of prototype filter coefficients on achievable aliasing and imaging suppression and show that incorporation of quantization into the filter design procedure can improve results. In the following we assume that filter coefficients are quantized with b bits, resulting in a quantization step size of 2^{-b+1} and a value range of $\{-1\ldots 1-2^{-b+1}\}$ and a scaling which uses this available range. Quantization of the input signal and the FFT twiddle factors are not considered here.

Figure 10 shows some results. The solid curve gives worst case aliasing for a prototype of length $L = 100$ designed from a Chebyshev starting window with 83 dB stop band attenuation according to section 4. We see that quantization down to 16 bit has virtually no effect while further reducing word length results in decreasing filter performance: Word lengths of 14 and 12 bits may still be considered acceptable while 10 bits or less are clearly not feasible.

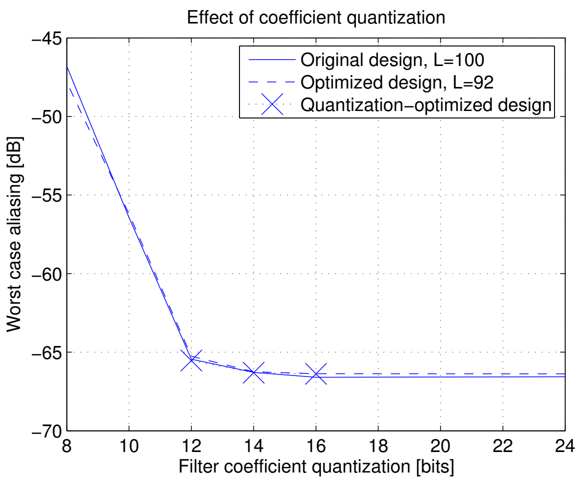

Figure 10: Effect of prototype filter coefficient quantization.

The dashed curve repeats the experiment for a filter prototype of length $L = 92$ which has been numerically optimized according to section 4.1 and quantized afterwards. We observe similar performance as in the case above, where numerical optimization does not increase quantization sensitivity of the filter coefficients.

The points marked with x and connected by a dotted curve finally show results for $L = 92$ where the coefficient quantization to 16, 14 and 12 bits has been taken into acount for the optimization procedure. The result is slightly improved filter performance, for 12 bit we get -65.53 dB compared to -65.26 dB for direct quantization of the optimized filter. We note that the extend of this improvement depends on filter length and filter bank parameters such as decimation and can be larger in some cases.

5. CONCLUSIONS

We have discussed filter bank design for digital hearing aids. In the area of hearing aids specific conditions arise, which place tight constraints on the filter bank design. With a sensible selection of the filter bank type and decent prototype filter design, these constraints can be met even with significant decimation of the bandpass signal. With standard design approaches, protoype filters can be created which build a base for further numerical optimization. We have seen that a prototype filter optimization with respect to the aliasing suppression adapts the filter to the underlying filter bank structure, especially to the decimation factor. This optimization can be performed directly on the quantized coefficients and shows robustness up to a specific degree of coefficient quantization.

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