# Complexity-Effective Auditory Compensation with a Controllable Filter for Digital Hearing Aids

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**Abstract - Auditory compensation consumes significant power due to the computation-intensive operations in the filter bank. To reduce the complexity, a controllable filter was designed to replace the filter bank. Filter order was designed to match prescriptions within a specific error constraint with minimum computational cost. An interpolation scheme according to the variation of signal intensity was implemented to reduce the overhead of coefficients calculations. The proposed auditory compensation reduces 80% of multiplications and 30% of power consumption compared to the complexity-effective multi-rate filter bank architecture [4]. Moreover, the group delay was also reduced from 10 ms to 2.4 ms.**

## I. INTRODUCTION

Auditory compensation of hearing aids compensates hearing losses by applying proper gains to input signal according to the level of hearing loss in both frequencies and the signal intensity aspects. One of the most well-known prescribing strategies is NAL-NL1 [1], which determines prescriptive gains on eighteen 1/3-octave frequencies from 125 Hz to 8000 Hz and 7 different sound pressure levels (SPL). To realize the prescriptions and minimize the computational complexity, a hearing aid should satisfy the following two constraints. First, delay should be less than 10 ms [2] to reduce the interference from the user's own voice. Second, matching error should be smaller than just-noticeable level, which is ±1.5 dB SPL according to [6].

Auditory compensation is usually implemented with a filter bank and a dynamic range compressor. The former decomposes the input signal into different frequency bands. The later determines target gains of each frequency band according to the intensity of the input signal. A 1/3-octave filter bank which follows ANSI S1.11 specification was designed in [3]. This design largely reduced the computational complexity by multi-rate signal processing, but the delay was too large. Another design presented a quasi-ANSI S1.11 filter bank [4], which reduced group delay, but increased the computational complexity.

The other category of design compensates hearing losses by using a controllable filter [5]. This design provided smooth matching result, but the overhead of coefficients calculation of the controllable filter was too large and therefore, higher order approximation was impractical. In this work, we proposed an auditory compensation based on the controllable filter. Filter order optimization according to prescriptions was applied to minimize the computational complexity of the controllable filter. Furthermore, the run-time calculation of filter coefficients was realized by coefficients interpolation according to the SPL variation to reduce the computational overhead.

## II. PROPOSED CONTROLLABLE FILTER DESIGN

### A. *Compensate with a Controllable Filter*

Assume the filter bank of an auditory compensation includes $n$ FIR filters to decompose input signal into $n$ frequency bands. The coefficient sets of each filter are denoted as $h_1 \sim h_n$. The target gains $g_1 \sim g_n$ were applied to each band signal to observe the output signal.

$$y[m] = \sum_{i=1}^{n} g_i \left( \sum_{k=0}^{l-1} h_i[k] x[m-k] \right) = \sum_{k=0}^{l-1} x[m-k] \left( \sum_{i=1}^{n} g_i h_i[k] \right) \quad (1)$$

Equation (1) shows the amplification of each frequency band, where $x[m]$ denotes the input sample, $n$ denotes the number of analysis bands, and $l$ denotes the length of filter. The equation shows that the filter bank architecture can be implemented with a single filter when the coefficients were generated from the filter bank's coefficients and prescriptive gains. Since the prescriptive gains were specified at seven input SPLs by NAL-NL1 (40, 50, 60, 65, 70, 80, 90 dB), the target gains $g_1 \sim g_n$ of controllable filter needs to be real-time changed by the input level. The controllable filter architecture is shown in Fig. 1, where seven sets of coefficients for the respective levels were stored ($h_{40dB} \sim h_{90dB}$) and the run-time coefficients were interpolated by the stored coefficients.

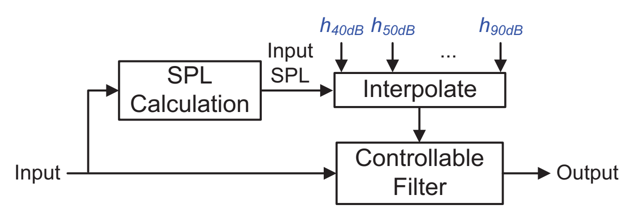


Figure 1. The concept of controllable filter.

The filter order of the original filter bank was generally fixed number; this implies overdesign issue for some hearing losses. For example, high frequency hearing loss needs only to amplify the high-frequency bands; however, the filter bank decomposes input signals into all the frequency bands. To optimize the complexity of each prescription, the stored seven sets of coefficients should be reduced. The optimization of filter order can be realized by Parks-McClellan algorithm with arbitrary magnitude response. Since the NAL-NL1 only prescribes gains at 18 1/3-octave bands, gains of all the other frequencies are not defined. If the magnitude response vector describes only the 18 gains, the gains of inter-prescription frequency may vary largely. A reasonable magnitude response is one specified frequency increases or decreases smoothly to another

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defined frequency. Therefore, we defined the maximum error of gains of inter-prescription frequency to be ±0.5 dB compared to the linear interpolation between the two defined frequencies. By adjusting the filter order and columns of magnitude response vector, the error of each prescribed gain-frequency response can be observed and find out the optimal filter length that satisfies the error tolerance.

### B. *Coefficient Interpolation*

The prescriptions of NAL-NL1 only defined gains at seven SPLs. However, the practical SPL may fluctuate according to the input signal and is possible to be undefined SPL. For this situation, interpolation of coefficients is one of the most efficient methods to generate the coefficients at run time. From Fig. 2, the amount of multiplications for coefficient interpolation is twice as many as that for the controllable filter calculation. The interpolation for each input sample costs large computational overhead. Considering that the speech level would not change abruptly, the coefficients interpolation can be performed only when the accumulated variation of input level is greater than a specified threshold to reduce the computational overhead. A dynamic SPL variation detector is implemented to determine the level change of input signal. When the variation of level is greater than just noticeable level (±1.5 dB), the coefficient interpolation is performed. The architecture of the design is presented in fig. 2.

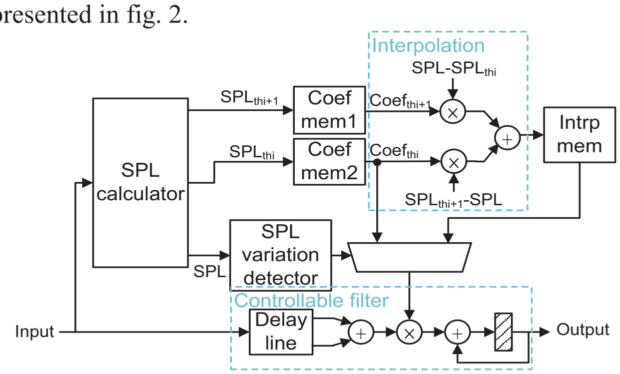


Figure 2. The controllable filter architecture with SPL detector

## III. RESULTS

To verify the effectiveness of the proposed design, three common prescriptions were selected as follows. Case 1 - high-frequency hearing loss, case 2 - low-frequency hearing loss, and case 3 - flat & severe hearing loss. A one-minute mandarin speech served as input signal. Figure 3 shows the matching result of case 3, which contributes maximal error among the three cases. The red solid line is the overall hearing aid magnitude response and the blue point is the prescriptive gains by using the filter bank architecture of auditory compensation with 18-channel and 7-segment wide dynamic range compression [3].

The design was implemented using cell-based design flow with HVT cell library in UMC 90nm technology. The simulation results are shown in TABLE I. The proposed design optimizes filter order for each prescription, therefore, the computational complexity and group delay varies. The proposed design saves 80% of multiplications and 30% of power consumption compared to the design in [4]. The group delay was also reduced from 10 ms to 2.4 ms for the same test case.

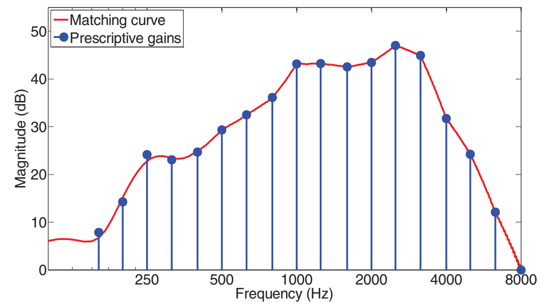


Figure 3. Matching result of common type hearing loss

TABLE I. COMPARISON OF AUDITORY COMPENSATION

| | | Max. error (dB) | Group delay (ms) | Mult per sample | Mem. size (KB) | Power (µW) |
|---|---|---|---|---|---|---|
| [4] | | 1.5 | 10 | 296 | 1.5 | 271.3 |
| Proposed | Case1 | 1.4 | 2.4 | 60 | 2.3 | 189.6 |
| | Case2 | 1.3 | 2.5 | 62 | | 190.3 |
| | Case3 | 1.5 | 7.9 | 191 | | 198.4 |

## IV. CONCLUSIONS

This paper proposes a complexity-effective auditory compensation for digital hearing aids. Controllable filter architecture and filter order optimization were combined to compensate both frequency and level dependent hearing losses. A coefficient interpolation scheme with SPL detector is also presented to adaptively adjust coefficients. The proposed auditory compensation saved 80% of multiplications, and 30% of power consumption. The group delay was also reduced from 10 ms to 2.4 ms while compared to the quasi-ANSI S1.11 filter bank [4].

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