AN EFFICIENT 18-BAND QUASI-ANSI 1/3-OCTAVE FILTER BANK USING RE-SAMPLING METHOD FOR DIGITAL HEARING AIDS

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ABSTRACT

This paper presents the multirate and re-sampling techniques to realize a low-delay, 18-band quasi-ANSI filter bank for digital hearing aids, which not only achieves a rather low computation complexity without a significant increase in the latency, but reduces greatly the total computation complexity for sub-band signal processing followed by the filter bank, such as noise reduction as well as wide dynamic range compression (WDRC). Researches done in the literature all focused on how to reduce the computation complexity of the filter bank. In particular, with the efficient multirate and interpolated FIR (IFIR) approaches for a 10-ms, 18-band quasi-ANSI filter bank, approximately 93% of the multiplications are saved, compared that with a straightforward parallel FIR filters architecture. However, they did not consider the computation complexity of the sub-band signal processing. In this paper, we first investigate realizing the FIR filter bank efficiently by using the multirate re-sampling techniques. To reduce the complexity, the optimized re-sampling factor for each filter band is explored carefully. Then, with the re-sampling technique, an efficient multirate quasi-ANSI FIR filter bank architecture is proposed. Compare to the state-of-the-art quasi-ANSI filter bank, approximately 17.7% of multiplicative complexity is reduced further and, up to 25% of the total computation complexity for sub-band signal processing followed by the filter bank is saved, but with only a slight increase in latency, i.e. 13.6 ms.

***Index Terms*—** Filter bank, ANSI S1.11, real-time application, hearing-aids

1. INTRODUCTION

Digital hearing aids become an emerged way to help hearing loss people regain their listening ability. Hearing aids compensate the hearing loss with auditory compensation algorithm and improve the speech intelligibility with the echo cancellation, the noise reduction, and the speech enhancement algorithms. One common approach to realize the auditory compensation algorithm, which makes the sound audible for hearing-aid wearer, is to employ an analysis filter bank followed by sub-band amplification and multi-channel wide dynamic-range compression (WDRC) and a synthesis filter bank. A low power Mandarin-specific hearing aid test chip was recently implemented in UMC 90 nm CMOS technology with high-VT standard cells [1]. The test chip contains an 18-band filter bank and a 3-channel WDRC auditory compensator and a multi-bank noise reduction with entropy enhanced voice activity detection. The power consumed by the analysis filter bank is approximately 27% of the total power. That is, the workload of the signal processing followed by the analysis filter bank dominates the power consumption of the chip.

As it is well known, the filter bank can be considered as a frequency decomposer in digital systems, not only in hearing-aids. Among various types of filter banks, a detailed comparison of the uniform [2], critical-like [3], symmetric [4], and 1/3-octave [5] filter banks has been discussed in [6]. The simulation results revealed that the 1/3-octave filter bank can fit the famous NAL-LN1 prescription formula [7] with zero matching error and preserve the acoustic property well, such as harmonics. Indeed, the matching performance of the prescription formula for digital hearing aids relies heavily on the choice of the filter bank. And, filters usually cause delays in the datapath. Although the 1/3-octave filter bank obtains the best matching capability, it suffers from large delay (up to 27 ms for parallel minimum-phase infinite-impulse response (IIR) filters architecture, while 78 ms for multirate FIR architecture in [5]). To reduce the long latency, a 10-ms and 18-band quasi-ANSI 1/3-octave filter bank is proposed by Liu *et al.* [6] successfully. With the proposed multirate interpolated FIR (IFIR) filter bank, approximately 93% of the multiplications and up to 74% of the storage elements are saved, compared that with a parallel FIR filters architecture. Nevertheless, researches done in the literature all focused on how to reduce the computation complexity of the filter bank. They did not consider the computation complexity of the sub-band signal processing followed by the analysis filter bank, such as multi-band monaural [8] or biaural [9] noise reduction as well as the multi-channel WDRC.

By inspecting the previous work [6], we find that most of sub-bands in quasi-ANSI filter bank did not operate at their lowest rates. As one tries to decimate the filter's rate near to its Nyquist rate, it concludes that a complicated fractional sampling rate conversion (SRC) is required to accomplish the

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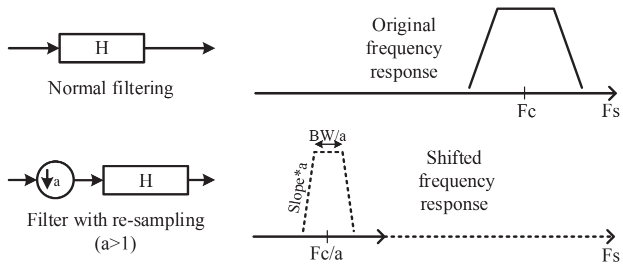

Fig. 1. Effect of sampling rate alteration

necessary rate alteration. An efficient way to implement the fractional rate conversion is by using a low-pass anti-aliasing filter to filter out the image part of the interpolated signal, then the decimator lowers the signal down to the desired sampling rate. Let's consider the SRC with *L/M* rate alteration and the tap-length of the anti-aliasing filter to be *N*. By direct implementation, the multiplicative complexity (MPYs) is *L*N*. Instead, applied by the efficient polyphase matrix decomposition [10], the MPYs is reduced to *N/M* greatly. Recently, a further decomposition on polyphase matrix was proposed [11] by utilizing coefficient symmetry of linear-phase property of the FIR filter and thereby, the MPYs is halved. According to these works, we observe that it is possible to design an efficient filter bank with fractional rate conversion, which not only achieves a low-complexity FIR filter bank but reduces the total computation complexity for sub-band signal processing followed by the filter bank.

The rest of this paper is organized as follows. Section 2 briefly introduces the concept of filtering with re-sampling. With the multirate re-sampling technique, section 3 presents the proposed 13.6 ms, 18-band quasi-ANSI 1/3-octave filter bank for digital hearing aids. Simulation results are also provided there which reveal that, compared to the state-of-the-art quasi-ANSI filter bank [6], approximately 17.7% of multiplicative complexity is reduced further and, up to 25% of the total computation complexity for sub-band signal processing followed by the filter bank is saved, but with only a slight increase in latency. Finally, section 4 concludes this paper.

2. LOW COMPLEXITY FILTERING WITH RE-SAMPLING

The main idea of filtering with re-sampling method is that changing sampling rate of filter input signal before filtering can result in frequency shift of the filter's response. Actually, it not only causes filter to haul on the frequency axis but also make the frequency response expanded or shrunken horizontally.

Fig. 1 illustrates the effect of filtering with sampling rate alteration. If we decimate the input signal by any factor a, $a > 0$, which supposes that it does not violate the Nyquist rate and no image occurs, it will bring on left shift of filter's frequency response, bandwidth reduction, and increase of the attenuation slope. Similarly, increasing sampling rate will have reverse effects on the frequency response of filter. According to ANSI S1.11 specification, the specification of every sub-band has $2^{1/3}$ times smaller than the adjacent left-side sub-band. Therefore, the Nyquist rate of each sub-band can be obtained in Table 1. Based on this property and re-sampling method, we can change the sampling rate to the desired one by rate converters and begin to filter out with the bandpass filter at its lowest rate. Fig. 2 illustrates a filter bank can be generated with single filter with various rate converters. Those re-sampling factors in Fig. 2 are obtained from the fractions or integers that are closest to the reciprocal of normalized Nyquist rate of the sub-band signal, which are shown in Table 1.

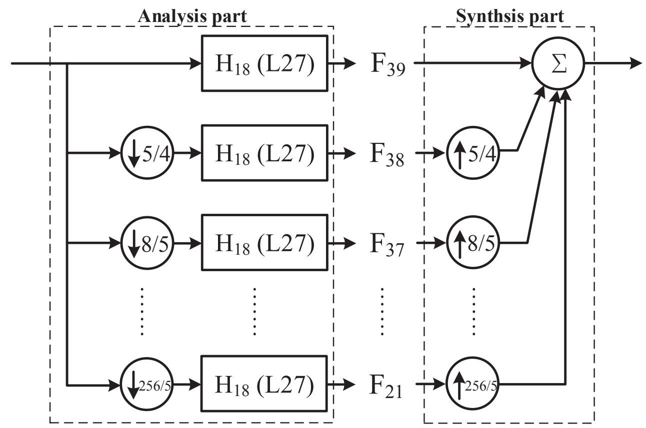

Fig. 2. Demonstration of re-sampling method using H_{18}

But rate alteration usually requires a low-pass filter to suppress aliasing part. The tap length of this anti-aliasing filter is proportional to the rate alteration factor, and the complexity can grow quickly when the rate is large. A more efficient way is to use multi-level rate alteration to avoid long filter processing. Therefore, we pick up the fractions or integers that can easily factorize into a set of small integer elements.

Note that using shorter filter and operating at lower rate can greatly decrease the complexity. Let's take 6^{th} octave band as an example. The operating rate of H_{17} and H_{16} are reduced to 4/5 and 5/8 of highest sampling rate, respectively. The multiplication per sample (MPY) of those three sub-filters is $\{14, 16, 21\}$. The total MPYs of 6^{th} octave band is 51 as operating at highest rate. By using this method, the MPYs of each sub-filter becomes $\{14, 14*4/5, 14*5/8\}$, and the total MPYs decreases to 34.

The exploration results of multi-rate filter implementation [6] indicated that the optimized value of decimation factor is 4 for minimum multiplications per sample, and the complexity grows when the factor is greater than 8. Based on this result and ease of implementation, we choose 6.4 (4,8/5) as *max-*

Table 1. Nyquist rate of each sub-band signal of quasi-ANSI one-third octave band filter bank and selected factors for filter resampling method

Sub-filter	Nyquist Rate (Normalized to Fs)	Rational form of decimation factors	Practical decimation factors D_i or M_i/L_i or $(D_i,M_i/L_i)$	Decimation factors of previous work
h_{18}	$2^{-0/3} = 1$	1	1	1
h_{17}	$2^{-1/3} = 0.7937$	5/4	5/4	1
h_{16}	$2^{-2/3} = 0.6299$	8/5	8/5	1
h_{15}	$2^{-3/3} = 0.5$	2	2	2
h_{14}	$2^{-4/3} = 0.3968$	10/4	(2,5/4)	2
h_{13}	$2^{-5/3} = 0.3149$	16/5	(2,8/5)	2
h_{12}	$2^{-6/3} = 0.25$	4	4	4
h_{11}	$2^{-7/3} = 0.1984$	20/4	(4,5/4)	4
h_{10}	$2^{-8/3} = 0.1574$	32/5	(4,8/5)	4
h_9	0.1093	8	(4,8/5)	4
h_8	0.0874	40/4	(4,8/5)	4
h_7	0.0706	64/5	(4,8/5)	4
h_6	0.0592	16	(4,8/5)	4
h_5	0.0497	80/4	(4,8/5)	4
h_4	0.0417	128/5	(4,8/5)	4
h_3	0.0341	32	(4,8/5)	4
h_2	0.0292	160/4	(4,8/5)	4
h_1	0.0247	256/5	(4,8/5)	4

imum re-sampling factor. So, the factors which are greater than the maximum value will be set to 6.4, instead. The factorized practical decimation factors of 1st to 6th octave bands are exposed in Table 1.

2.1. Architecture exploration

An architecture exploration with different orders of decimation factors about complexity and latency constraints should be concluded before entering the final design. Consider the *i*-th branch sub-filter h_i with a set of an integer decimation factor D_i and a fractional decimation factor M_i/L_i, there are two different approaches to achieve the same magnitude specification of quasi-ANSI S1.11 one-third filter bank, as shown in Fig. 3. Note that both are not fully equivalent and can not be derived from each other.

For the complexity-aware case, we desire the filter which has longer length to operate at lower sampling rate. The Fig. 3(a) has displayed the low complexity approach where the integer decimation D_i is before the fractional rate decimation M_i/L_i. Down-sampling is beneficial to keep the SRC operated at low rate as possible, but the anti-aliasing filter of SRC could contribute an unacceptable latency to our design constraints.

In order to minimize the delay introduced by the rate alteration process, the solution is to change the order of rate alteration. Let the filters with longer tap length operate at higher rate, as shown in Fig. 3(b). Obviously, the latency is greatly reduced but the overall computational complexity is increased. Moreover, most of intermediate nodes do not have the same operating rate in this case. Compared to previous

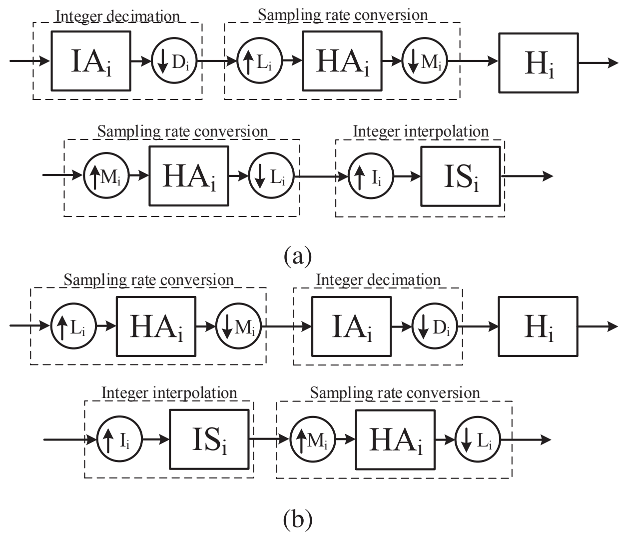

Fig. 3. Architecture exploration for analysis and synthesis bank (a) Complexity-aware approach (b) Latency-constrained approach

case, only *1st* to *7th* branch can be shared with the same rate alteration processing including decimation filter IA_i, SRC, and interpolation filter IS_i.

3. PROPOSED FILTER BANK DESIGN

In the previous section, we have exposed a filter re-sampling concept to lower the operating rate of sub-bands. Then, a careful architecture exploration about the concern of complexity and latency issue have been made. In this section, we apply with re-sampling concept and all of the design issues to the proposed filter bank design.

3.1. Sub-filters design

Now that the re-sampling factors and the desired operating frequency for each sub-band have been determined. The rest of all is to design band-pass filters according to the quasi-ANSI specification. An efficient filter design procedure [6] to optimize the coefficients of quasi-ANSI filters was applied with our sub-filter design.

There are two different situations of sub-filter design. One is that all of the sub-filters completely satisfy the ANSI S1.11 specification, such as 1st, 2nd and 3rd octave band. Only the highest frequency sub-band H_{18} is required. The other situation is that we can not further use H_{18} and large re-sampling factors to generate sub-filters of 4th to 6th octave bands due to latency reason. Thus, we should design the sub-filter for each band at specified sampling frequency. And the modification on the low delay quasi-ANSI specification with respect to the re-sampling factors is essential for low delay requirement, too. In order to counteract the effect of filter with re-

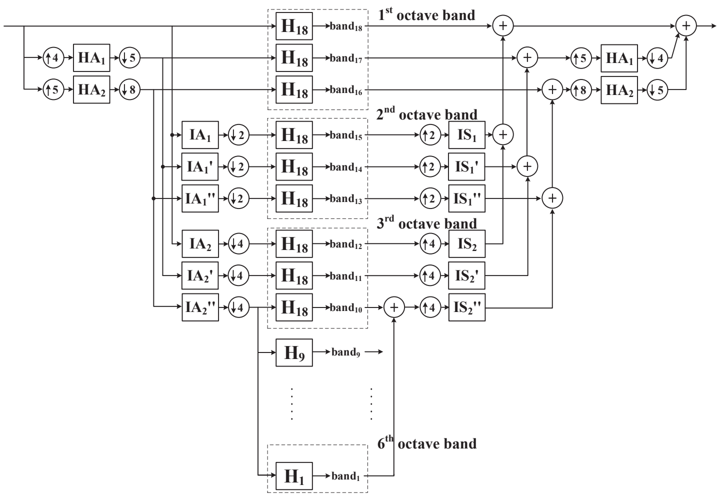

Fig. 4. Proposed design

Table 2. The tap length information of all sub-filters required in the proposed filter bank design

Filter	IA_1	IA_1'	IA_1"	IA_2	IA_2'	IA_2"	HA_1	HA_2
Tap length	35	29	23	49	41	31	51	81

Filter	H_{18}	H_9	H_8	H_7	H_6	H_5	H_4	H_3	H_2	H_1
Tap length	27	43	53	61	63	63	63	63	63	63

sampling method, a right shifted and horizontally expanded version can be obtained easily by multiplying the pass-band and stop-band specification with the re-sampling factors.

3.2. Final design

On the basis of previous exploration, the proposed design of low delay quasi-ANSI one-third octave band filter bank with re-sampling method is shown in Fig. 4. H_i is quasi-ANSI sub-filter. The remained filters are used for rate alteration. IA_i, IA_i', and IA_i" are integer decimation filters. IS_i, IS_i', and IS_i" are integer interpolation filters. HA_i is an anti-aliasing filter for SRC. All of rate alteration process is using polyphase decomposition to lower the complexity, and we adopt the efficient SRC design method [11] for our SRC implementation.

The magnitude response of proposed design is plotted in Fig. 5, where the dash line represents the response of original design. A series of design comparisons with original quasi-ANSI filter bank is shown in Table 3. The complexity ratio of proposed versus original design is 82.3%. Overall, 24.67% of sub-band sampling rate has been saved by using this design method. The latency becomes 13.58ms which is less than 15ms and still considered to be acceptable for hearing-aids application.

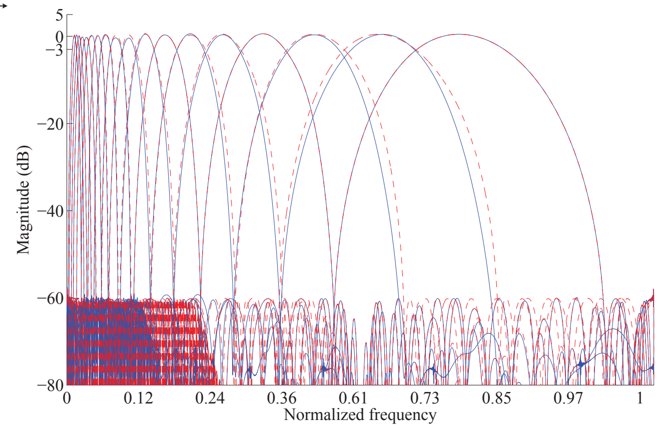

Fig. 5. Magnitude response of proposed filter bank

Table 3. Design comparisons with original quasi-ANSI filter bank

Design	Complexity (MPY/sample)	Number of filters required	Total coefficients	Latency (ms)	Sub-band sampling rate reduction (%)
original	226	14	506	10.25	100
proposed	186	16	451	13.58	75.33

4. CONCLUSION

Based on previous design [6], an optimization method of reducing sub-band sampling rate results in both complexity-efficient filter bank and relaxation of the power-budget burden of overall hearing-aid DSP system. Using re-sampling method to generate 1st to 3rd octave bands is very beneficial for VLSI implementation. Consider the FIR processor in [6], a context-switch for next filter includes replacement of coefficients and delay line storage. Only the swap of delay line elements is needed in this proposed design. Also the booth encoding information can be preserved for next filter. Thus, we can expect that the dynamic power will be further reduced using this method. Besides, this method is applicable to DSP implementation, too. A DSP processor with single instruction multiple data (SIMD) can execute vector multiplication simultaneously. Especially for 1st to 3rd octave band sub-filters, it can process sub-filters at the same time due to the same coefficients. This will greatly reduce the burden of DSP chip by using this method because the computation load of those three octave bands occupy 58.4% of entire filter bank. Finally, we believe that our proposed design will be efficient for both VLSI fabrication and DSP implementation in hearing-aid applications.

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