A Computationally Efficient Half band FIR Filter Bank for Hearing Aids

V.D.M.Jabez Daniel[1] M.Kanishka[2]
[1]Assistant Professor [2]PG Student
[1,2]Department of Electronics & Communication Engineering
[1,2]Dr. Sivanthi Aditanar College of Engineering, Tamil Nadu, India

Abstract— Digital Finite Impulse Response (FIR) Filter bank is one of the most significant parts in Digital Signal Processing (DSP) applications. Designing and implementation of filter bank is a challenging task in Very Large Scale Integrated (VLSI) Circuits in terms of power, area and speed. The multipliers and adders are the major key blocks of the filter bank which consumes high power, more area and needs high computation time. In this project, an efficient FIR filter bank is designed based on the Sub-band coding technique. In this proposed filter bank design, there are 10 filters which are designed using two prototype halfband FIR filters and each filter has 5 taps, so that the order of each filter in the filter bank structure is 4. This proposed filter bank will split the given input audio signal into six sub-band signals based on its frequency which is mostly used in hearing aids for audiogram matching. In hearing aids, the frequencies of the audio signal are split into different bands and then amplification is provided according to the different levels of hearing threshold which is called audiogram matching. To improve the performance of this filter bank, Pipelined Floating Arithmetic has been used.

Key words: Audiogram matching; Finite Impulse Response (FIR); Hearing aid; Sub-band signals; Pipelined Floating Arithmetic

I. INTRODUCTION

VLSI (Very Large Scale Integration) is the field which involves more number of logic devices in small area. At present for the implementation of VLSI, modern Digital Signal Processing (DSP) systems are well-matched. If DSP systems are employed with VLSI technologies, it is technically practicable or economically feasible. Many DSP systems are produced in very large amount and so, they require high performance circuits with respect to throughput and power. The main task of the hearing aid device is to selectively amplify the audio sounds and the processed sound should match audiogram of the patient. The softest sounds a person can hear is shown by Audiogram graph. Digital filter banks are very important parts of DSP system. In DSP, a Filter bank is a cluster of band pass filters which divides or decomposes an input signal into multiple sub-band signals, each one shipping a single frequency sub-band of the original signal. The process of decomposition of a given input signal into many sub-band signals based on its frequency is called as analysis and the filter bank used for this analysis process is generally called as analysis filter bank. The synthesis is the process of reconstructing the complete original signal from the sub-band signals of the analysis filter bank and for this synthesis process filter bank is used which is generally called as synthesis filter bank. Digital filter banks are extensively used in hearing aids for audiogram matching purposes. To strengthen the sound for the wearer, amplifying device called hearing aid is used. More generally the aim is making the speech more understandable, and to correct impaired hearing as dignified by audiometry. Hearing thresholds are defined as the softest sounds one can hear and they are plotted using a graph called audiogram. However for people with impaired hearing, the hearing thresholds become high at certain frequencies causing hearing loss i.e. they have a low sensitivity toward certain frequencies. Such hearing loss is caused by aging on the inner ear or due to the related structure. It is necessary to selectively amplify sounds to balance this type of hearing loss at required frequencies or at different pitches and from 250Hz to 8000Hz along the y-axis and from -10 to 120 dB along the x-axis is plotted. The hearing aid must able to alter the sound levels at random frequencies within a given spectrum for attaining the payment of this hearing loss. This is accomplished in practice through filter banks by passing the input audio signals that divides the input audio signals into different frequency bands. To suit the desires of hearing impaired each sub-band gains are modifiable that is the amplitude response of filter bank's must match the audiogram. In 2016 Nisha Haridas and Elizabeth Elias designed an efficient bank of filters derived from the variable bandwidth filter structure, for audiogram matching in a digital hearing aid, using Farrow structure. The magnitude and band edge frequencies of the filters were be independently adjusted. However, even if the difficulty of the Farrow structure was comparatively low, it would develop quickly as the bandwidth of the Fractional delay filter lines π. Also in decimation the direct Farrow structure often performs poorly, and the coefficients need to be optimized separately for each decimation factor. For the synthesis process, the most effective way used is Frequency Response Masking (FRM). FRM method is used with the very high sparse coefficients with random pass band bandwidth for digital filters. The filter has very high sparse coefficient vector, which is the main benefit of FRM method. So for the filter implementation, the requirement of arithmetic hardware is very low. But, there is no closed-form analytic expression for discovering the value of optimum M is the weakness of FRM methods. For each M (nonzero multipliers), by assessing the complexity of the filter good choice of M can be attained and then selecting the M which corresponds to the lowest estimate. This project focus on constructing an efficient FIR filter bank structure based on decomposition process of Sub-band coding technique using Pipelined Floating point arithmetic for refining its efficiency in terms of area, power and speed.

II. PROPOSED WORK

A. Filter Bank Based on Sub-Band Coding Method

The speech signal is partitioned into several frequency bands and each band is digitally coded. But according to hearing aid application, an audio signal of 8000Hz which has a sampling frequency of 16000Hz is split into 6 frequency sub-bands. The first frequency subdivision splits the given input audio signal into two equal-width segments, a low pass signal ($0 < F \leq 4000$Hz) and a high pass signal ($4000\text{Hz} \leq F \leq 8000$Hz). The second frequency subdivision splits the low pass signal from the first stage into two equal bands, a low pass signal ($0 < F \leq 2000$Hz) and a high pass signal ($2000\text{Hz} \leq F \leq 4000$Hz). The third frequency subdivision splits the low pass signal from the second stage into two equal bands, a low pass signal ($0 < F \leq 1000$Hz) and a high pass signal ($1000\text{Hz} \leq F \leq 2000$Hz). The fourth frequency subdivision splits the low pass signal from the third stage into two equal bands, a low pass signal ($0 < F \leq 500$Hz) and high pass signal ($500\text{Hz} \leq F \leq 1000$Hz). The fifth frequency subdivision splits the low pass signal from the fourth stage into two equal bands, a low pass signal ($0 < F \leq 250$Hz) and high pass signal with ($250\text{Hz} \leq F \leq 500$Hz). Thus the audio signal of 8000Hz is divided into six frequency sub-bands. Then down sampling by factor 2 is performed after frequency subdivision to reduce the bit rates of the sub-band signals. The halband low pass filter can be taken as the combination of low pass filter and a downsampling by 2. And also the combination of low high pass filter and a downsampling by 2 is considered as a halband high pass filter. So, in figure.1, the halfband low pass filter is denoted as HLP and the halfband high pass filter is denoted as by HHP. Then, these halfband low pass filter and half band high pass filter are used as prototype filters to construct this entire Filter bank structure.

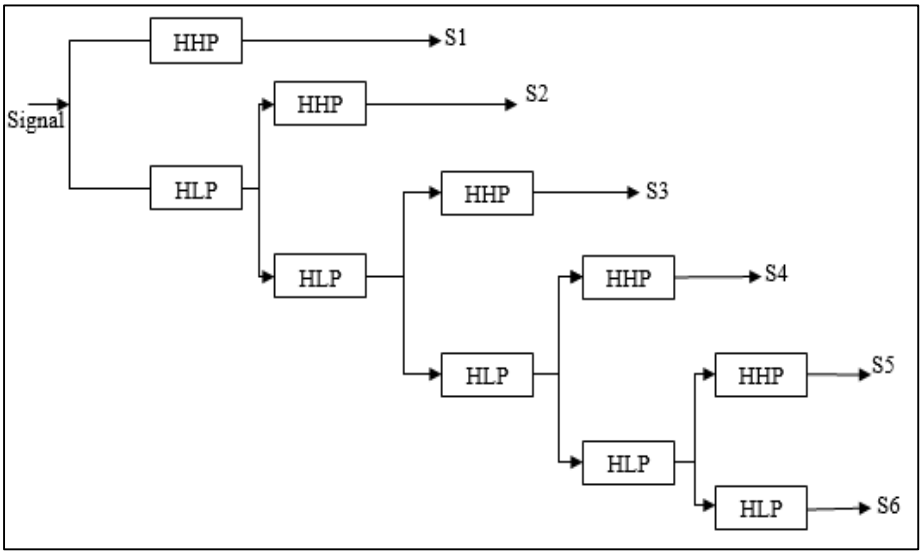

Fig. 1: Filter Bank based on Sub-band coding method

B. Floating Point Arithmetic Operation in Pipelining

The pipelining technique is usually used for reducing the time complexity. It always processes the steps periodically without delay. Once it completed the sign bit operation it ready to do the next operation simultaneously. It calculates the exponent value, at the same time it process the second step sign bit. The block shows the storage and processing of the sign, exponent and mantissa.

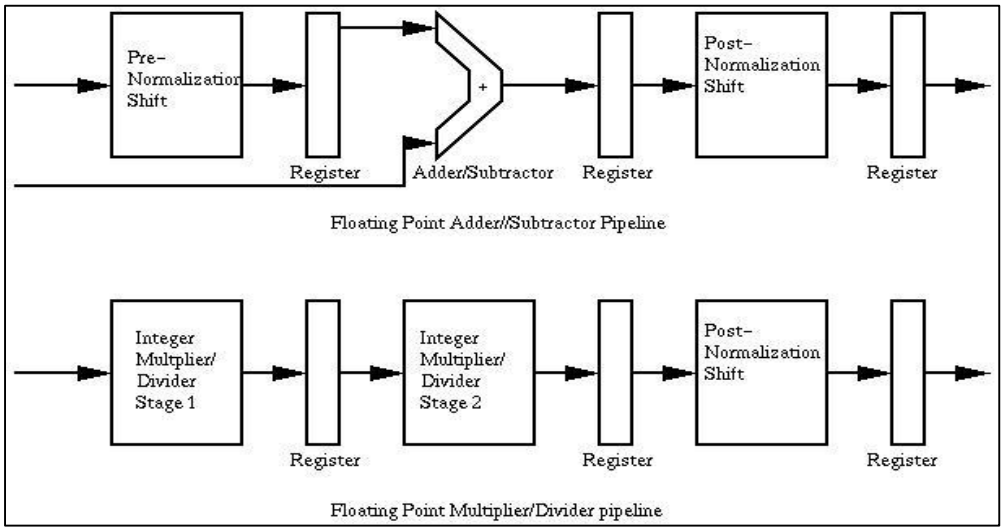

Fig. 2: Pipelined Floating Point Adder and Multiplier

1) Floating Point Addition

The IEEE 754 Single precision Floating Point Adder has 32 bit. The following steps are performed to add two floating point numbers:

1) The exponents of the two numbers (A and B) are compared and the smaller number is shifted to the right until its exponents matches the larger exponents
2) Add the Mantissa of two numbers.
3) Normalize the result: to obtain the 1 at the MSB of the results Mantissa.

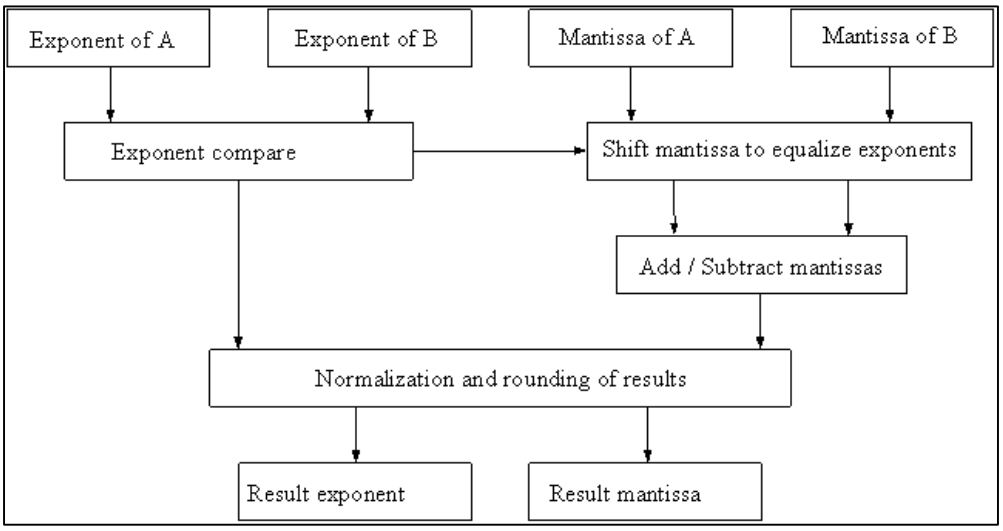

Fig. 3: Flow chart of Floating point addition

2) Floating Point Multiplication

The IEEE 754 Single precision Floating Multiplier has 32 Bit. The following steps are performed, to multiply two floating point numbers:

1) Obtain the Sign: S1 XOR S2
2) Add the exponents & subtract the bias value: (E1 + E2Bias).
3) Place the decimal point in the Mantissa Result.
4) To obtain the 1 at the MSB of the results Mantissa, Normalize the result.

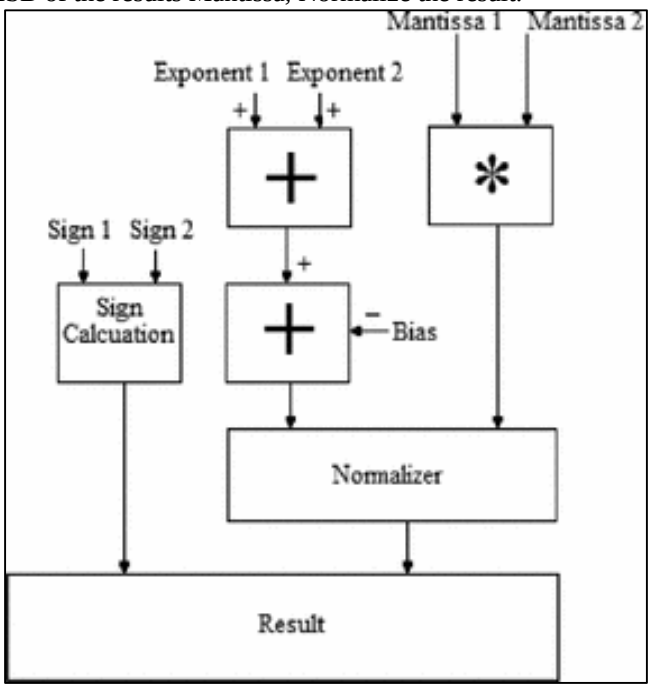

Fig. 4: Flowchart of Floating Point Multiplication

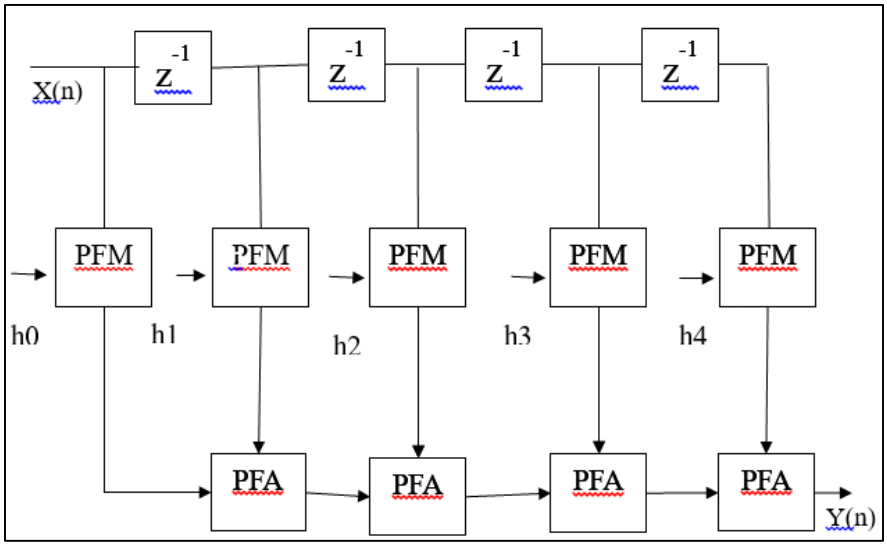

Fig. 5: Diagram of one of the 5 tap FIR filters present in Filter bank

In Figure.5, PFA represents the Pipelined Floating Point Adder, then PFM represents the Pipelined Floating Point Multiplier while h0, h1, h2, h3, h4 are the coefficients of 5 tap FIR filter. Then each FIR filters are designed using Kaiser Window technique in MATLAB software tool.

III. SIMULATION RESULTS

The implementation of Sub-band coding based FIR filter bank includes the simulation results of Sub-band coding based FIR filter bank and Analysis filter bank of 6 bandpass filters. Simulation of the design is done by Modelsim 6.5b. The memory usage and delay of above mentioned filter banks are synthesized using Xilinx 13.4 software and their power is synthesized using Quartus II 9.1 software.

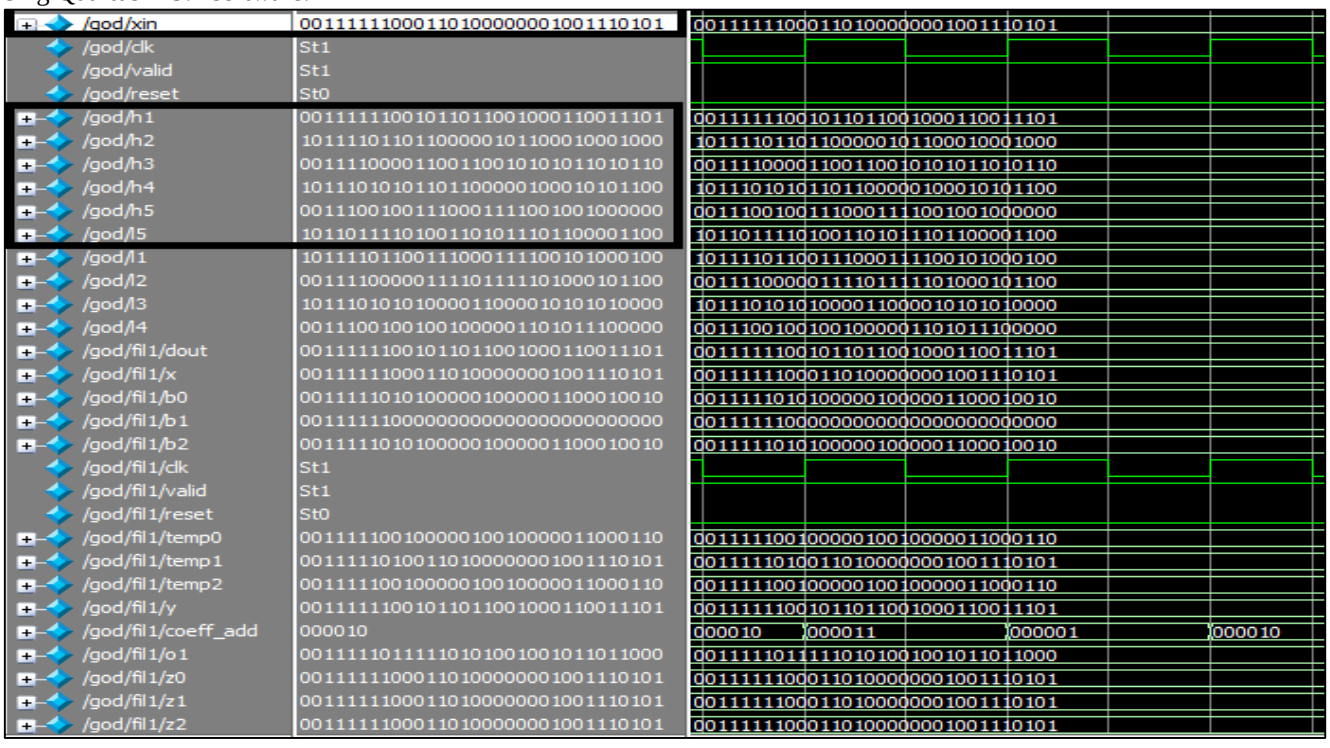

Fig. 6: Simulation results for Sub-band coding based FIR filter bank (I)

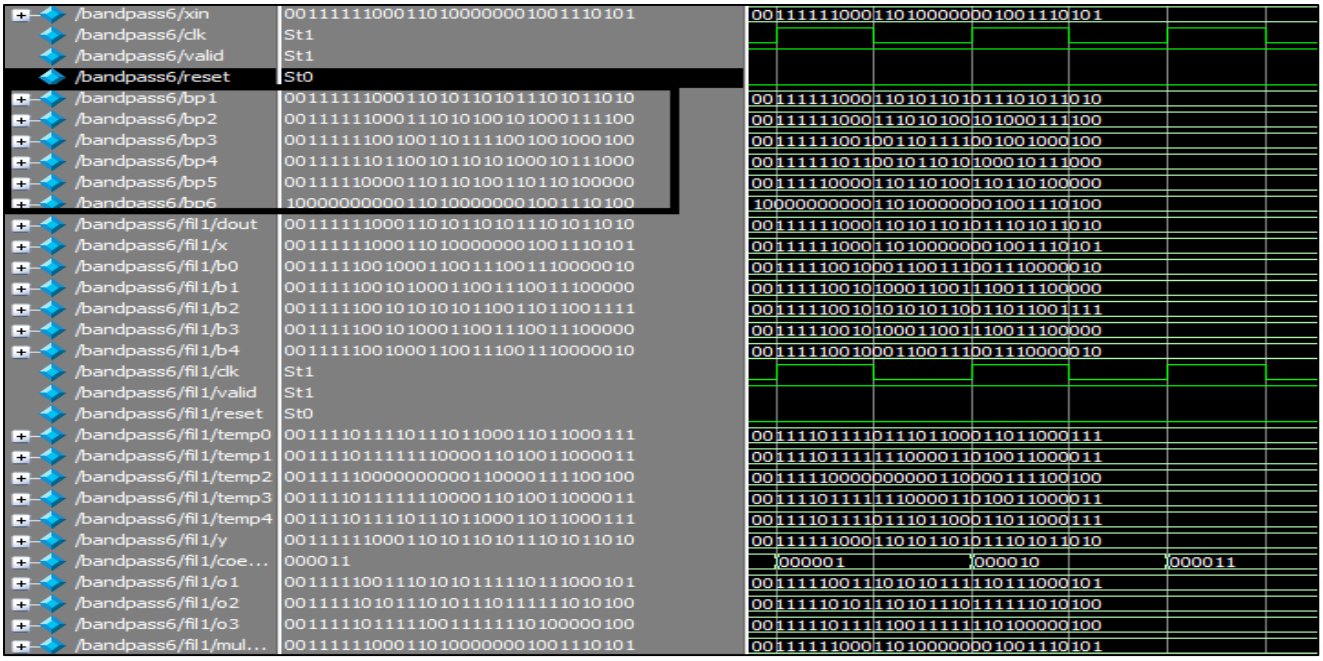

Fig. 7: Simulation result of Analysis filter bank of Bandpass FIR filters (II)

A. Comparison

The Filter bank based on Sub-band coding technique is compared with the Analysis Bandpass FIR Filter bank. The performance parameters such as power, delay and total memory usage of the filter are analyzed. The parameters are essential for determining the efficiency of FIR filter bank.

Filter bank type	Power mW	Delay ns	Memory kb	adder
I	220.17	10.52	567080	690
II	220.19	13.36	1022740	810

Table 1: Performance analysis of Filter Banks

IV. CONCLUSION

An efficient FIR filter bank is designed based on Sub-band coding technique using Pipelined Floating Point Arithmetic. It is observed that, the power dissipation of the Filter bank based on Sub-band coding technique is slightly reduced by 0.02mW, when compared with that of the Analysis Filter bank. The delay of Filter bank based on Sub-band coding technique is reduced to 10.324ns when compared with the delay of Analysis Filter bank which is 13.367ns. So, the delay of Filter bank based on Sub-band coding technique is reduced by 3.043ns.The total memory required for Filter bank based on Sub-band coding technique is well reduced by approximately 2 times of memory requirement of the Analysis Filter bank of bandpass FIR filters. The total number of adders and registers required for Filter bank based on Sub-band coding technique are lesser when compared with that of the Analysis Filter bank of bandpass FIR filters. Thus the Filter bank based on Sub-band coding technique is efficient in terms of power, delay and memory, when compared to the Analysis Filter bank of bandpass FIR filters.

REFERENCES

[1] Raghu and Nisha (2018), "High Performance Continuous Variable Bandwidth Digital Filter Design for Hearing aid Application", International Journal of Electronics and Communications, Vol.92, pp 36-53.
[2] Nisha Haridas and Elizabeth Elias (2016), "Efficient Farrow Structure Based Bank of Variable Bandwidth Filters for Digital Hearing Aids", IEEE International Conference on Signal Processing, Informatics, Communication and Energy Systems (SPICES), pp 255-262.
[3] James George and Elizabeth Elias (2014), "A 16-Band Reconfigurable Hearing Aid using Variable Bandwidth Filters", Global Journals of Researchers in Engineering, 1 December, Vol. 14, No.
[4] Ying Wei and Debao Liu (2013), " A Reconfigurable Digital Filter bank for Hearing Aid Systems with a Variety of Sound Wave Decomposition Plans", IEEE Transaction on Biomedical Engineering, June, Vol. 60, No. 6.
[5] James George and Elizabeth Elias (2012), "Continuously Variable Bandwidth Sharp FIR Filters with Low Complexity", Journal of Signal and Information Processing, No.3, pp 308-315.
[6] Ying Wei (2012), "A Design of Digital FIR Filter Banks with Adjustable Sub-band Distribution for Hearing Aids", IEEE Transaction on Biomedical Engineering, Vol. 67, No. 6.
[7] Y. Wei and D. Liu (2011), "A design of digital FIR filter banks with adjustable sub-band distribution for hearing aids", IEEE Conference on Information, Communications and Signal Processing (ICICS).

[8] Noboru Ito and Tian-Liang Deng(2012),"Variable-Bandwidth Filter-Bank for Low Power Hearing Aids", IEEE International Congress on Image and Signal Processing (CISP), Vol.45, No.6.
[9] Ying Wei and Yong Lian (2012), "A 16-Band Non-uniform FIR Digital Filter bank for Hearing Aid", IEEE Biomedical Circuits and Systems Conference, Vol. 67, No. 6.
[10] Ya Jun Yu, Yong Ching Lim, Dong Shi(2009), "Low-Complexity Design of Variable Band edge Linear Phase FIR Filters with Sharp Transition Band", IEEE Transactions On Signal Processing, Vol. 57, No. 4.