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A Survey of Filter Bank Algorithms for Biomedical Applications

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Abstract— **The Digital signal processing algorithms are promising techniques, which are used to alleviate the filter bank designs in various applications. Filter bank is an enabling technique for numerous capabilities such as speech coding, Noise reduction, Sub band coding and auditory compensation. The design procedure mainly concentrates on the core part of the system which is the type of the filter. However, it imposes several solutions to many problems high computation complexity, area and power consumption are most critical concern. In digital hearing aid application, the filter bank improves the sound ability for hearing-impaired people. The scope of this work is to give an overview of the filter bank algorithms under various traits and the challenges that they face, along with the current state-of-the-art. To enhance the feasibility of the design by applying the characteristics of the filter banks for suitable applications. This paper covers wide range of issues in the design of filter bank. The contribution of this paper is threefold. First, we show the functional role of filter bank. Second, the classifications of filter bank algorithms for the hearing instruments. Third, merits, demerits and further design challenges of the filter banks are discussed.**

Keywords-component; Filter bank, Digital hearing aid, DSP algorithms

I. INTRODUCTION

With the rapid proliferation of new techniques, Digital Signal Processing (DSP) algorithms such as Sub band coding, Noise Reduction, Echo cancellation, auditory compensation and Speech Enhancement are executed repetitively. Digital hearing aid devices employ a large number of DSP algorithms to enhance speech intelligibility in the existence of noise. Filter bank is popular in the sphere of signal processing, and it has variety of applications [11]. Auditory compensation is a significant part since it compensates the hearing loss of the hearing impaired people who usually suffer from the loss of audibility and hearing dynamic range reduction. Fig.1 shows the block diagram for the hearing aid [11].

Recent hearing aid has a filter bank with few bands and a dynamic range compressor to compensate the hearing loss. A lot of low power techniques that are used in each algorithm. A person with hearing impaired tends to have a low sensitivity hence hearing thresholds for high frequency sounds is higher than the normal speech intensities.

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The main role of a hearing aid is to amplify the input sounds to match with the audiogram. The sound levels are adjusted at different frequencies to achieve the matched sounds.

Practically, this is done by passing the input signals through a filter bank that separates them into several number of frequency bands in the analysis bank. The insertion gains are applied to improve the hearing ability and given to the compressor which reduces the dynamic range reduction at the synthesis bank. To measure the hearing thresholds the audiologists use signals over octave frequency ranges from 250Hz to 8 KHz. This method is Pure Tone Audiogram (PTA) test to identify the hearing loss [6]. An example of moderate-to-severe hearing loss is shown in Fig.2. Fitting process consists of a prescription formula NAL-NL1. It produces various electro acoustic responses for various inputs.

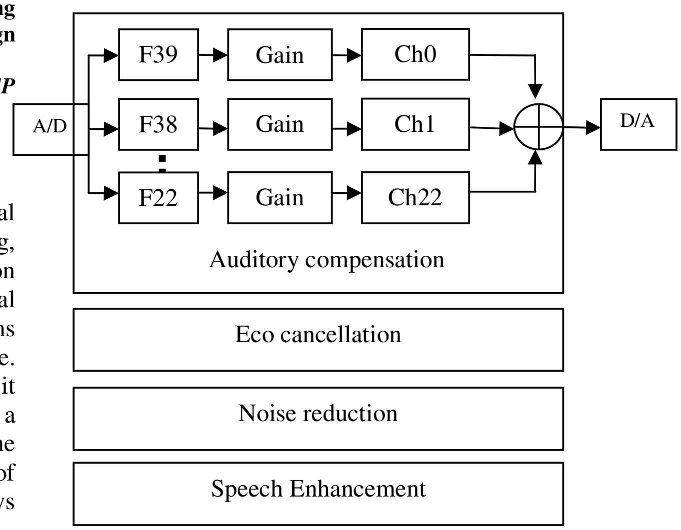

Fig.1 Block diagram for the Hearing Aid

Hearing losses present in both ears are known as a binaural hearing loss. Binaural hearing aid [16] which means on each ear a hearing instrument is placed along with a wireless link to exchange the information between them. The Power Spectral Density has produced a binaural noise reduction scheme that can operate in complex acoustic environments that are included diffuse noise, transient noises, and reverberant conditions. However, the lack of synchronization between both hearing aids may lead to a loss of localization cues, which has been recognized as perceptually annoying.

With respect to the current surveys, the contribution of this work is of three sections. In Section II, we present the filter bank Classifications. Section III identifies present existing algorithms for various applications. Section IV focuses on the way ahead in this research area: it summarizes the outstanding challenges and identifies actions in the context of the future evolution of low power filter bank through experimental results in terms of frequency responses for some of the filter banks. Finally, Section V concludes the paper by discussing some important open issues in Filter bank design.

II. Classifications of Filter Bank

The design of the digital filter bank for the hearing instruments can be classified into two categories; namely,

(i)Uniform filter banks

(ii)Non-uniform filter banks

A uniform flexible filter bank method is designed in [2], and an 8 band FIR filter bank is implemented in [3]. The uniform filter bank can be categorized into Quadrature Mirror Filter bank (QMF), orthogonal [26] and biorthogonal [23] Filter banks. The orthogonal wavelet filter bank was proposed to reduce the interferences. In [10], each sub filter was designed individually. The orthogonal filter banks, though the complexity is comparatively high, the hearing loss at different frequencies was well compensated. Orthogonal and biorthogonal filter banks can be designed to have either the perfect-reconstruction (PR) or nearly perfect-reconstruction (NPR) property, whereas QMF banks have the NPR property due to their characteristics. The uniform filter does not meet the resolution requirement of human hearing system.

The non-uniform filter bank has different types of filter banks. First, octave bands of computationally efficient non uniform FIR filter bank [4]. With the help of the Frequency Response Masking (FRM) technique, the eight sub bands are produced which can be implemented using 15 multipliers to reduce the computational complexity [5]. The 16 channel filter bank [6] has proposed pre-computational unit (PCU) to obtain micro power and small IC area. The one third octave filter bank is popular in acoustic applications [9]-[12]. In [12], a complexity effective multi rate filter bank algorithm has been proposed to achieve low power. A systematic coefficient design flow has been designed to diminish the order of the filter in [12].

Most of the designs produce filter bank with fixed sub bands. Different hearing impaired people cannot proceed specific case to improve their auditive performances. Hence the design can be customized for each patient very attractive, but the main drawback of this approach is that it has less number of sub-bands. To address this issue, filter bank with few adjustable sub-bands are proposed [12] and [13]. In [14], a programmable spectrum cut-up has been proposed to adjust filter bands according to patient's pathology. In a three channel filter bank, variable low pass, high pass and band pass filter has been designed from a prototype filter [14].

ANSI S1.11 standard defines 43 1/3 octave bands which covers the frequency range of 0-20kHz. The mid –frequency (f_{mid}) specifies each 1/3 octave band [11].

$$f_{mid}(i) = 2^{(i-30)/3} f_{ref}$$

$f_{mid}(i)$ is the mid band frequency of the i[th] band and f_{ref} is the reference frequency which is always set as 1KHz. The frequency between the lower band (f_L) and the upper band (f_U) frequencies define the bandwidth (Δf).

$$f(i) = f_L(i) - f_U(i)$$

Multi rate Architecture provides area efficient structure by using a recursive structure that covers 18 bands from 22[nd] to 39[th] one third octave frequencies [26]. The structure contains 6 octave frequencies arranged from high frequencies to low frequencies. Each octave consists of three sub bands. In the conventional design, parallel architecture is used as shown in the Fig.2. In this structure, input *x(n)* is splitted into18 bands and amplified separately. The output *y(n)* is obtained after that the processing is completed.

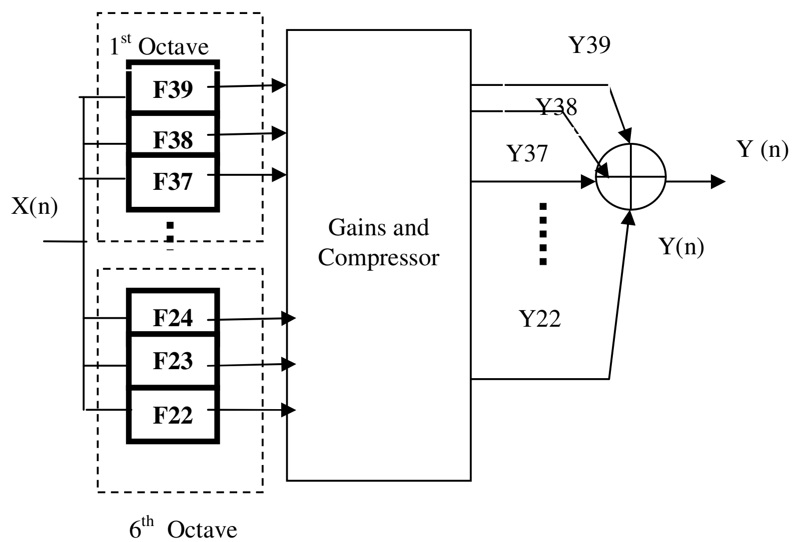

Fig.2 18 Parallel band structure

The Interpolated FIR (IFIR) filter bank [5] has tree structured architecture with seven bands. This filter bank addresses the design of asynchronous blocks for one of the applications of hearing aid achieve low power. The asynchronous design re-implements the synchronous design. The design of IFIR filter bank is complex to bring out some of the key characteristics of the asynchronous circuit. Much effort has been taken to reduce the number of multiplication in order to save the power. The algorithm can be implemented using processor structure with single Add Multiply and Accumulate data path, Random Access Memory (RAM) and Read Only Memory (ROM). Due to the folded structure of the IFIR filters it is convenient to use a dual port RAM.

Most of the Hearing aid systems have fixed sounds due to fixed sub bands, and could not provide feasible solution to compensate various types of hearing impairment. All the sub-bands were adjustable based on some control parameters. Two blocks were used to design the filter bank. First, Multi band generation block has been proposed [12] to produce the desired frequency responses with some pass bands. The BW_P is the pass band width and BW_s is the stop band width. The filters were interpolated by the factor M_i and then decimated by the factors D_1 and D_2.

$$BW_P = \frac{2 f_c D_j}{M_i}$$

$$BW_S = \frac{2(\pi - f_c) D_j}{M_i}$$

Second, Sub band selection block whose function is to extract the desired sub bands. The output signals of the multiband generation block stored one of its memory elements with the help of interpolation factor. Only three prototype filters were used in order to reduce the computation complexity of the entire circuit. The branch selection signal is selecting the input of the masking Filter bank. This masking filter bank includes several number of sub filters and switches. Control signal is provided to both blocks so that both blocks can be reconfigured.

The delay calculation of the complete block with the assistance of the following equation:

$$D_{Total} = D_{Multiband} + \sum_{k=1}^{Ni} D_{i,k}$$

Total delay is the sum of both multiband generation block delay and k th selection block delay. In [14], a filter bank was proposed whose output contains 27 different sub bands. The entire system has sub divided into three sections and each with three schemes. The design mainly concentrates on Decimation, Interpolation and Frequency Response Masking. By replacing each unit delay of a digital filter with an all-pass filter, the warped filters are obtained which can be used for various audio processing applications. The Quasi ANSI S1.11 Standard Filter bank filter bank [18] has been designed by using the high performance of ANSI one third octave filter bank with a group delay bank for the application of the hearing aid device. The group delay can be significantly reduced with the support of the optimization algorithm which has two recursive structures: one satisfies the 10ms group delay and other controls the matching error. The algorithm proposed in Quasi ANSI algorithm is as same as the optimization steps in the ANSI S1.11 for the design of filter coefficients with minimal order, except that it changes the transition Band width.

III. ALGORITHMS

In recursive pyramid algorithm, instead of using all the 18 bands the recursive structure has three frequency bands i.e. first octave frequencies with decimator at the analysis bank and Interpolator at the synthesis bank. This recursive structure was proposed in [12].

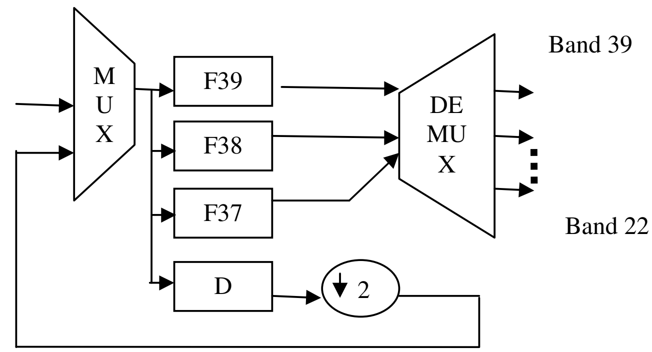

Fig.3 Recursive Structure

The Frequency Response Masking (FRM) is the technique has been widely used in numerous applications [19], [20] where the FIR filters with extremely narrow transition band are needed. A novel two stage FRM structure was proposed to increase the efficiency of the FRM algorithm. It provides a low complexity design. The multipliers of the sub filters were shared. Compared with the single filter based frequency response masking, both complexity of the design and the group delay are significantly reduced. The optimal design of one and two dimensional FIR filters using the FRM. The low power is achieved because minimum utilization of multipliers and memories. Time multiplexed FRM filters achieve lower complexity compared with the multiplexed single stage filters. The prototype filter is of non-linear phase in [20]. The main reason for considering non-linear phase masking filters to constant group delay. In [20], an FRM technique has been proposed a filter structure to reduce the complexity of the band-edge shaping filter and one of the mask filters. Normally, the background noises produce distortion in the speech signals. To address this issue, the generalized Side lobe Canceller (GSC) has been proposed [22]. This GSC includes a blocking matrix and an Adaptive Noise Canceller (ANC). The blocking matrix was used for removing the unwanted signal or feedback signal from the micro phone. The combination of GSC and standard adaptive echo cancellation has been proposed, called the generalized echo and interference canceller (GEIC). A discrete-time filter with length K_S is represented as the polynomial function of d shown in the below equation.

$$F(d) = f_0 d^0 + \ldots + f_{Ks-1} d^{-Ks-1}$$

The micro phone signals are $m_i[x]$, and the loud speaker signals are $u[x]$. Every microphone signals were decomposed based on the following equation.

$$m_i[x] = m_i^s[x] + m_i^n[x] + F_i(d)\, u[x]$$

Where $m_i^s[x$, $m_i^n[x]$ the ith component of speech signal and noise. The third part represents the feedback component in the ith microphone signal. The feedback suppression filter is denoted by the (M+1) x 1 vector $W(d\,)$that works on micro phone signals and the loud speaker signal. The output $e[x]$
as shown below.

$$e[x]= W(d) \begin{pmatrix} m[x] \\ u[x] \end{pmatrix}$$

$$e[x]= W(d) \begin{pmatrix} m^s[x]+m^n[x] \\ 0 \end{pmatrix} +W(d) \begin{pmatrix} m[x] \\ u[x] \end{pmatrix}$$

The first term is the algorithm’s output in the absence of internal and acoustic feedback. It reduces the noise as much as possible while preserving the speech signal. The second term is the residual feedback measured at the output of the algorithm and the more feedback is cancelled by $W(d)$. In [24], the Coefficient decimation technique provides a decimated version of the original frequency response whose pass band width is M times that of prototype filter. M is the integer decimation factor. Every Mth coefficients are grouped together and in between coefficients are discarded in the prototype filter [25]. The frequency response of a filter is obtained using CDM from the prototype filter for different values of M. A computationally efficient Variable Digital Filter with warped filter and coefficient decimation was proposed. Consider Nth FIR filter with its impulse response. The warped FIR version was obtained by replacing every delay with first-order all pass structure. The warping coefficient controls the warped frequency response.

IV. COMPARISIONS

The multi-rate Filter bank: In the design of ANSI S1.11 Filter bank [12] the low power optimization techniques has been applied to get efficient area and low power. The Fig. 4 states that it consumes power in the memory, Multiply & Accumulator and other blocks of the design by applying clock gating and operand isolation.

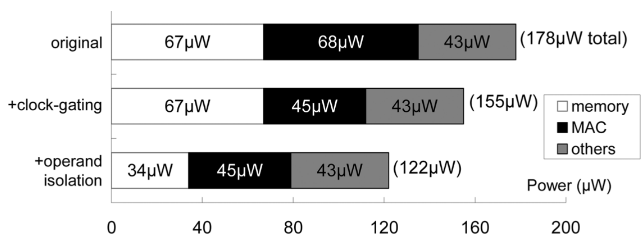

Fig.4 Low power design techniques with multi-rate ANSI

Here, two examples have been discussed in [12] and [13]. An audiogram output for Mild hearing loss for high frequencies and all frequencies is shown in Fig.4 & Fig.5. In these examples, the comparison of reconfigurable filter bank output along with the non-uniform and uniform output filter bank as showed in Table I.

TABLE I [13]

Type of filter bank	Maximum matching error (dB)
Uniform	6.39
Non-Uniform	9.61
Reconfigurable	4.82

A future work on [16] could include the extension of the binaural PSD estimators to binaural hearing aids with multiple sensors on each ear and the resulting application of those estimators in binaural noise reduction schemes.

Table III summarizes the multiplicative complexity of three different architectures of the 18-band 1/3-octave filter bank, which are the parallel low-delay quasi-ANSI FIR filters, the iterative standard ANSI filter bank [14], and the proposed architecture. Comparing with the parallel FIR filters, this design saves approximately 93% of multiplications.

TABLE III Quasi –ANSI [18]

Filter Bank	Group delay (ms)
Parallel Filters	7.7
ANSI S1.11	78
Quasi-ANSI	10

V. CONCLUSIONS

This paper has provided an overview on various types of the filter banks. Though, filter bank design is of recent interest to research community, many contributions have already been proposed, and a number of protection techniques are identified. The group delay has been reduced by using Frequency Response Masking technique in the filter bank application. The Quasi –ANSI Filter bank provides the right solution with the help of optimization algorithm to control the maximum delay of the circuit. Adjustable Filter bank gives the entire frequency bands at the output using some control parameters. Estimation of the filter bank usage in multiple dimensions and applications and identifying opportunities in these dimensions and developing algorithms for prediction into the future using past information can be considered as some of the open research problems.

REFERENCES

[1] P. P. Vaidyanathan, Multirate Systems and Filter Banks. , New Jersey: Prentice Hall, 1993.

[2] R.Brennan and T. Schneider, “A Flexible Filter Bank Structure for Extensive Signal Manipulations in Digital Hearing Aids,” in *Proceedings of IEEE International Symposium Circuits Systems, CA, 1998, Page Number: 569-572.*

[3] H. Li, G. A. Jullien, V. S. Dimitrov, M. Ahmadi, and W. Miller, " A 2-Digit Mulltidimensional Logarithmic Number System Filter Bank for a Digital Hearing Aid Architecture," in *Proceedings of IEEE International Symposium Circuits Systems*, AZ, 2002, Page Number: II 271–350.

[4] Y. Lian and Y.Wei, "A computationally Efficient Nonuniform FIR Digital Filter Bank for Hearing Aids," *IEEE International Symposium Circuits Systems I Volume 52 No:12, Page Number:2754-2762, December 2005.*

[5] L.S. Nielsen and J. Sparso, "Designing Asynchronous Circuits for Low Power: An IFIR Filter Bank for a Digital Hearing Aid," *Proeeiding of IEEE* , Volume 87, No.2, Page Number: 268-281, February 1999.

[6] K. S. Chong, B. H. Gwee, and J. S. Chang, "A 16-Channel Low-Power Nonuniform Spaced Filter Bank Core for Digital Hearing Aid," IEEE Transactions on *Circuits Systems I*, Volume 53, No. 9, Page Number: 853–857, September 2006.

[7] Y. Wei and Y. Lian, "A 16-Band Nonuniform FIR Digital Filterbank for Hearing Aid," in *Proeedins of IEEE Biomedical Circuits System, Conference.*, Page Number:186-189,2006.

[8] A. Lozano and A. Carlosena, "DSP-based Iimplementation of an ‚ANSI S1.11 Acoustic Analyzer," *IEEE* Transactions on Instruments and Measurements, Volume 52, No.4, Page Number: 1213-1219, August 2003.

[9] S. B. Davis, "Octave and Fractional Octave Band Digital Filtering Based on the Proposed ANSI Standard, " in *Proceedings of ICASSP Conference*, Page Number: 945-948, 1986.

[10] *Specification for Octave-Band and Fractional-Octave-Band Analog and Digital Filters*, ANSI Standard S1.11-2004.

[11] D. Byrne, H. Dillon, T. Ching, R. Katsch, and G. Keidser, "NAL-NL1 Procedure for Fitting Nonlinear Hearing Aids: Characteristics and Comparisons with Other Procedures," *J. Amer. Acad. of Audiology*, volume 12, No. 1, Page Number. 37–54, January 2001

[12] Y. T. Kuo, T. J. Lin, Y. T. Li, and C. W. Liu, "Design and Implementation of Low Power ANSI S1.11 Filter Bank for Digital Hearing aids," *IEEE* Transaction on Circuits and Systems, Volume 57, No.7, Page number 1684-1696, July 2010.

[13] Ying Wei and Debao Liu , "A Reconfigurable Digital Filter Bank for Digital Hearing Aid systems with Variety of Sound Wave Decomposition Plans", IEEE transactions on Biomedical Engineering,Volume 60, Page number 6, June 2013

[14] Ying Wei and Debao Liu , " Digital FIR Filter Banks with Adjustable Sub-band Distribution for Hearing Aid Systems", in Proceedings of 8th International Conference on Information and Communication Signal Processing, Singapore, December 2011

[15] T.B. Deng, "Three-channel variable filter-bank for digital hearing aids" IEEE Transaction on Signal Processing, Volume. 4, Issue: 2 , page Number. 181 – 196, 2010

[16] A. Homayoun Kamkar-Parsi and Martin Bouchard, " Instantaneous Binaural Target PSD Estimation for Hearing Aid Noise Reduction in Complex Acoustic Environments", IEEE Transactions on Instrumentation and Measurement, Volume. 60, No. 4, Page Number: 1141-1154, April 2011

[17] N. Ito and T.-L. Deng, "Variable-Bandwidth Filter-bank for Low-Power hearing Aids," in Proceedings of 3rd International Conference on Image Signal Processing , Page Number. 3207–3201, 2010.

[18] Chih-Wei Liu, Kuo-Chiang Chang, Ming-Hsun Chuang, and Ching-Hao Lin, "10-ms 18-Band Quasi-ANSI S1.11 1/3-Octave Filter Bank for Digital Hearing Aids." IEEE Transactions on Circuits And Systems—I: Regular Papers, Volume 60, No. 3, PP: 638-649, March 2013

[19] Y. C. Lim, "Frequency-Response Masking Approach for the Synthesis of Sharp Linear Phase Digital Filters," *IEEE Transaction on Circuits and Systems*, Volume. CAS-33, PP: 357–364, April 1986.

[20] Tian Shen and Yong Ching Lim, "Low complexity Frequency-Response Masking Filters Using Modified Structure Based On Serial Masking", 9th European Signal Processing Conference (EUSIPCO 2011) Barcelona, Spain, Page Number: 1400-1404, August 29 - September 2, 2011

[21] Yongzhi Liu, and Zhiping Lin, "Optimal Design of Frequency-Response Masking Filters With Reduced Group Delays", IEEE Transactions on Circuits And Systems—I: Regular Papers, Volume. 55, No. 6, Page Number: 1560-1570,JULY 2008

[22] Ann Spriet, Geert Rombouts, Marc Moonen, and Jan Wouters," Combined Feedback and Noise Suppression in Hearing Aids", IEEE Transactions On Audio, Speech, and Language Processing, Volume 15, No. 6, , Page Number:1777-1790, August 2007

[23] Fran¸cois Moreau de Saint-Martin, Pierre Siohan, *Member, IEEE,* and Albert Cohen," Biorthogonal Filterbanks and Energy Preservation Property in Image Compression." IEEE Transactions On Image Processing, Volume 8, No. 2, PP: 168-178, February 1999.

[24] R. Mahesh andA. P.Vinod, "Coefficient Decimation Approach for Realizing Reconfigurable Finite Impulse Response Filters," in *Proceedings of IEEE Inernational Symposium Circuits System*, Seattle, USA, Page Number:81-84,May 2008.

[25] Sumit J. Darak, Vinod A. Prasad, and Edmund M.-K. Lai , " Efficient Implementation of Reconfigurable Warped Digital Filters With Variable Low-Pass, High-Pass, Bandpass, and Bandstop Responses", IEEE Transactions On Very Large Scale Integration (VLSI) Systems, PP: 1-5.

[26] P.P.Vaidyanathan, "Multirate Digital Filters, Filter Banks, Polyphase Networks, and Applications A Tutorial", Proceedings of the IEEE, Volume. 78, No. 1, PP: 56-93, January 1990.