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Transducer Approach for Sensorial Hearing Loss by Sound Bite Hearing System

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Abstract: The demand for personalized hearing aids needs the filter bank of a hearing aid system, capable of decomposing the sound waves in accordance with the patient's hearing loss. In this paper an efficient adjustable filter bank is proposed and the modulated signals are implemented in sound bite hearing system. The number of the subbands and the exact location of the subbands can be adjusted by changing control signal. The proposed filter bank has extremely low complexity compared to other filters. It is because of the adoption of fractional interpolation and the technique of symmetric filters and also due to complementary filters. The proposed filter bank can meet different needs of hearing loss cases with acceptable delay. This modulated signals are implemented in newly proposed sound bite hearing aid.

Key words: A/D converter · LWDFB · Highpass filter · SoundBite Hearing System

INTRODUCTION

The auditory system is a very sensitive network. Diseases, noise and aging may have resulted in common reason for sensory disturbances. To compensate the hearing loss the hearing aid system must have integration of voice amplification, noise reduction, environmental adaption, automated power switching etc. A schematic diagram of digital hearing aids is shown in Fig.1. The analog signals are sent into digital signals by an A/D converter the digital signal is divided into various subband signals within different frequency bands by a filter bank [2] [3]. The subband signals are then synthesized. They are send to D/A converter. Filterbank-based algorithms makes a easy adjustment of speech amplification. The systems are fully programmable and suits patient's comforts

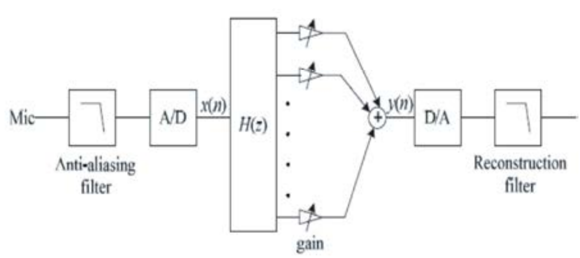

Fig. 1: A schematic diagram for digital hearing aid

Uniform filter banks are the most widely used filter banks in hearing aids. Researchers have done lots of works to reduce the complexity of the filter. Lattice wave digital filter bank (LWDFB) was introduced with much reduced complexity when compared to FIR filter bank.

These filters are also not sensitive to coefficients. MDLNS (Multi dimensional logarithmic number system) was used to reduce the complexity. DFT filter bank was used in this model filter. EMQF (Elliptic minimal Q-factor) filters was used later as filter bank. A tree structured filter bank made was introduced based on all pass compensatory filter. To reduce the complexity more, 8 band filter bank based on frequency response masking was used to implement the hearing aids. In this paper, a linear phase reconfigurable digital FIR filter bank with small complexity is proposed. This filter bank also has and acceptable delay. The proposed filter bank is based on fractional interpolation. It allows us to build the system using small number of prototype filters. The whole frequency range is divided into three regions and each region has three different subband decomposition schemes. By changing the control signal, different band decomposition schemes are chosen without changing the structure of the filter bank. These modulated signals are sent to the

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soundbite hearing system. SoundBite Hearing System is a non-surgical bone conduction device, that transmits sound via the teeth. SoundBite uses the tooth instead of the implanted component and eliminates the need for surgery.The simple circuit diagram is shown in Fig. 2.

The Fundamental Ideas of the Design: The proposed filter bank is based on fractional interpolation. For a fractional interpolated filter $H(z^{M/D})$, every D^{th} coefficients of the prototype filter $H(z)$ are grouped together, discarding the in between coefficients. Then (M -1) zeros are inserted in between two coefficients. N_f is determined by interpolation factor M,

$$N_f=[(M+1)/2] \tag{1}$$

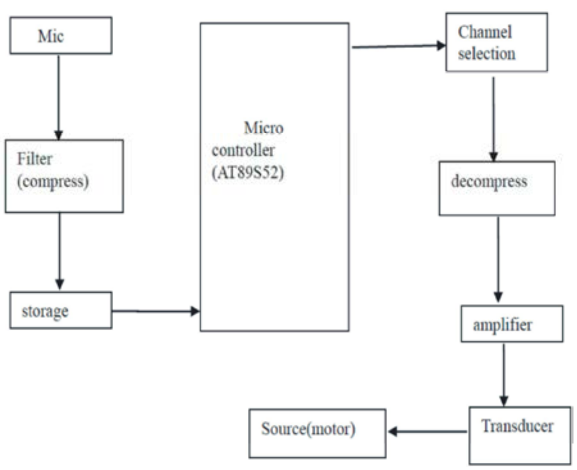

Fig. 2: Block diagram

The bandwidth of the i^{th}passband,$B_w(i)$,is determined by

$$B_w(i) = \begin{cases} B_o \times D/M & i=1 \text{ and } N_f \\ 2B_o \times D/M & \text{otherwise} \end{cases}$$

where B_o is the bandwidth of the prototype filter $H(z)$. The center frequencies of the passbands, ω_k, can be calculated by,

$$\omega_k=2\pi k/M, k=0,1,2\ldots.,N_f - 1 \tag{2}$$

From equations (1) and (2), changing the interpolation factor M, the number and the positions of the passbands of $H(z^{M/D})$ can be changed. Further, from equation above, by changing the interpolation factor M and the decimation factor D, the bandwidths of the passbands of $H(z^{M/D})$ can be changed. The above particularities form the basis of the configurability of the proposed filter bank. Also, the operation of subtraction and operation of complementation is used and more passband can be generated. The prototype filter must be designed by smaller passband and stopband ripples. Masking filters are used to extract the variable passbands generated by fractional interpolation. To generate uniform subbands, the center frequencies of the passbands should be in consistent with the cutoff frequencies of the masking filters except for the first and the last ones. By masking a whole passband is divided into two parts. Each part is extracted to be an individual subband. We can assign different masking filters for different fractional interpolated filters but the complexity of the system will be huge. To reduce complexity, masking filter must be reusable. The passbands generated by different filters by same interpolation factor can be extracted by same masking filter. The two fractional interpolated filters share the same set of masking filters. It together cover the whole frequency range. The number of the masking filters is M.

The Structure of the Proposed Filter Bank: Based on the above discussions, we propose a adjustable filter bank for hearing aid. The frequency range is divided uniformly as three regions, the low frequency region $(0, \pi/3)$, the middle frequency region $(\pi/3, 2\pi/3)$ and the high frequency region $(2\pi/3, \pi)$. Three band schemes are selected as,

Scheme 1: 1 subbands with bandwidth $\pi/3$,
Scheme 2: 2 subbands with bandwidth $\pi/6$,
Scheme 3: 4 subbands with bandwidth $\pi/12$

the structure of the proposed filter bank is illustrated in Fig. 2. There are two functional blocks for the proposed filter bank. They are the multiple passbands generation block and the masking block. To reduce the delay, the masking block is kept in front of the multiple passbands generation block. Proper delays should be added to the branches to balance the group delay, which are not shown in the figure.

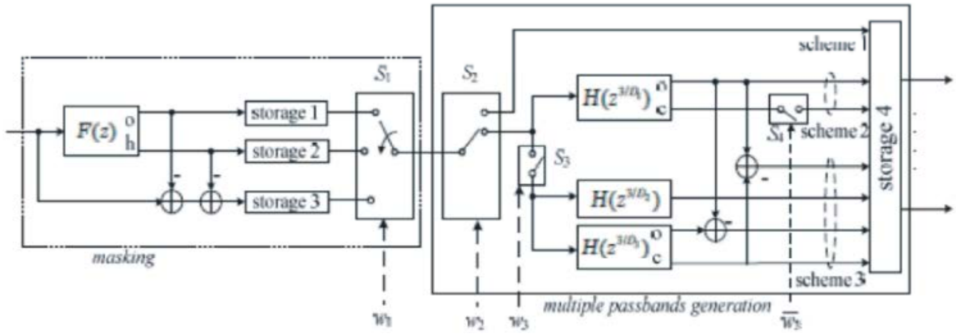

Fig. 3: The structure of the proposed filter bank

The Masking Block: The desired passbands are extracted by masking filters. The subfilter F(z) is used to produce the the masking filters.It divides the whole frequency range into 3 uniform regions naturally. $\pi/2$ is the cutoff frequency of F(z).The "o" port and "h"port provides required output of original filter and output of the highpass filter respectively, symmetric with $F(z)$ at $\pi/2$, denoted as $F_h(z)$. $F(z)$ at $\pi/2$, $F_h(z)$ I=1,2,3, are

$$F_1(z) = F(z) \tag{3}$$

$$F_2(z)=z^{-\square}-F(z)-F_h(z) \tag{4}$$

$$F_3(z)=F_h(z) \tag{5}$$

The outputs are stored and sent to multiple passband generation block in accordance with the 2-bit region-selection signal $w1$ of switch $S1$.If w_1=00 low frequency is selected, if w_1=01 middle frequency is selected, high frequency region is selected if w_1=10. Switches $S2$, $S3$ and $S4$ together are responsible for selecting the scheme.

The Multiple Passbands Generation Block: $H(z)$ and its fractional interpolated versions forms a block which function to generate multiple passbands. The "o" ports of the interpolated filters provide the original outputs and "c" ports provide the outputs of the complementary filters. the number of the masking filters is the same with the interpolation factor, M equals to 3. The generation of the subbands is shown in Fig. 3.

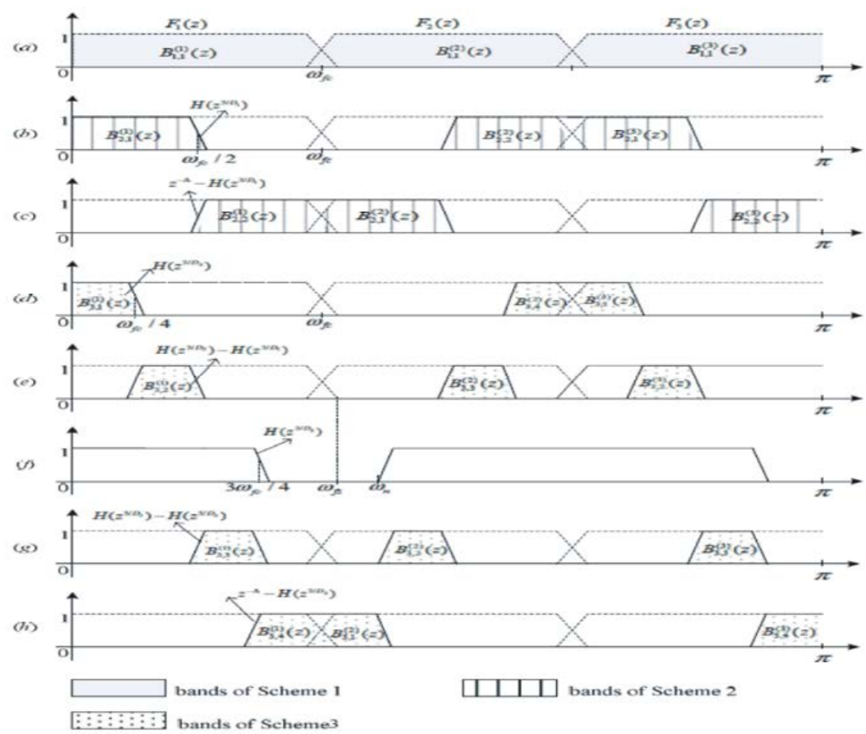

Fig. 4: The generation of the subbands.

For scheme 1, the masking filters produce three uniform subbands with bandwidth $\pi/3$. For scheme 2, half of the passbands is generated by $H(z3/D1)$. The other half of the passbands is generated by the complementary filter of $H(z3/D1)$. To generate uniform passband with $\pi/6$ as bandwidth, in accordance with equation above and Fig. 3 (*b*), we have,

$$\omega_{hc}D_1/3 = \omega_{fc}/2 \quad (6)$$

For scheme 3, $H(z^{3/D1})$, $H(z^{3/D2})$ and $H(z^{3/D3})$ together are used to generate passbands with bandwidth $\pi/12$ as shown in Fig. 3 (*d*) to (*h*).

$$P_1(z)=H(z^{3/D2}) \quad (7)$$

$$P_2(z)=H(z^{3/D1})-H(z^{3/D2}) \quad (8)$$

$$P_3(z)=H(z^{3/D3})-H(z^{3/D2}) \quad (9)$$

$$P_4(z)=z^{\square}-H(^{z3/D3}) \quad (10)$$

To obtain uniform subbands,

$$\omega_{hc}D_2/3 = \omega_{fc}/4 \quad (11)$$

$$\omega_{hc}D_3/3 = 3\omega_{fc}/4 \quad (12)$$

Set D_2 =1 and substitute ω_{fc}=p/3 into the equations (6), (11) and (12),

$$\omega_{hc}=p/4 \quad (13)$$

$$D1=2 \quad (14)$$

$$D2=3 \quad (15)$$

The cut-off frequencies of the prototype filters $F(z)$ and $H(z)$ are $\pi/3$ and $\pi/4$, respectively.As shown in Fig.3(f).

$$\omega_{fc}=\omega_n \quad (16)$$

Suppose the transition bandwidths of $F(z)$ and $H(z)$ are d_f and d_n, respectively. Using the variables to replace ω_{fs} and ω_n,

$$\omega_{fc}+ (d_f/2) = (2p/3) – ((\omega_{hc}+d_h/2)D_3)/3) \quad (17)$$

$$d_h+d_f = p/6 \quad (18)$$

To avoid aliasing,

$$\omega_{hs}\bullet \max(D_i)<p, I=1,2,3 \quad (19)$$

substituting ω_{hc}=p/4 and dh to replace ω_{hs},

$$d_h<(2p/\max(Di)) – p, I=1,2,3 \quad (20)$$

Switches s_2,s_3 and s_4 are controlled by w_2 and w_3.For switch s_2 the upper and lower branch is connected when w2=0 and w2=1 respectively. For switch *S*3, it is open when *w*3=0 and it is close when *w*3=1. S4is opposite state of s3. to get all the outputs in accordance with the structure three rounds are needed for three regions.The outputs are stored and wait for their outputs.

Working Procedure: The working procedure of the proposed reconfigurable filterbank is shown in Fig. 5.

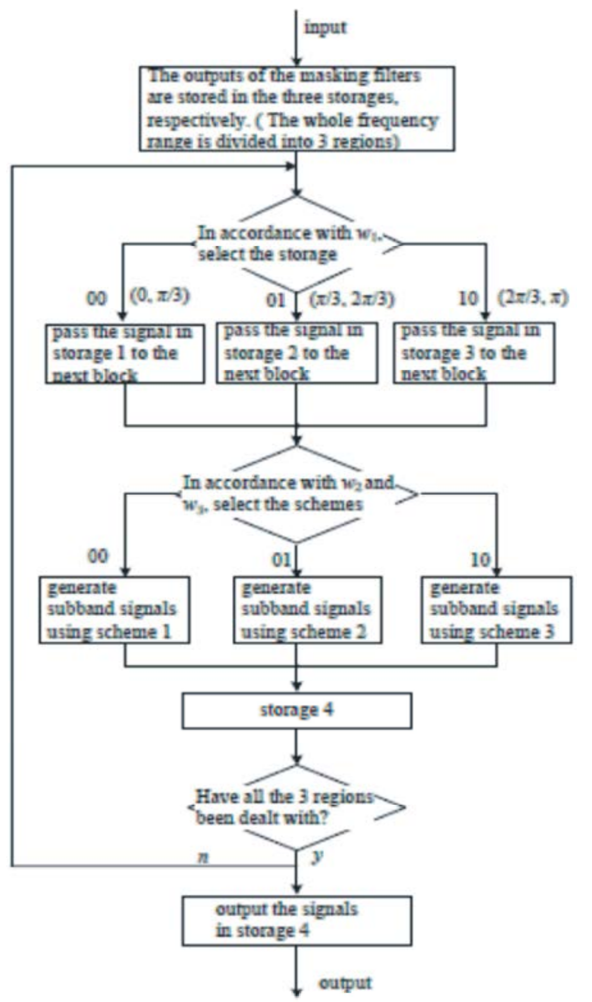

Fig. 5: Working procedure

Implementation: The modulations are implemented in the soundbite hearing system. SoundBite Hearing System is a non-surgical bone conduction device, that transmits sound via the teeth. SoundBite uses the tooth instead of the implanted component and eliminates the need for surgery.

Voice input given to mic after that we use filters to compress the voice. Voice input stored in storage device of controller, give the another section depends upon the channel selection. The decompress section in the system retrieve the given input voice. The signals are less amplitude and hence we need to amplify the signal. The amplified signals uses the transducer for converting sound signal into vibrating signal. The signals are sent to transducer and the vibrations are sensed in the teeth. This vibrations makes sound to be felt in the cochlea. The vibration in cochlea sends signals to the brain and senses the sound.

Experiment Result and Analysis: In this paper a simple and very efficient adjustable filterbank is proposed and implemented in sound bite hearing system. This adjustable filter bank with new improved bone conduction method, can be the future hearing aid model. This method makes the hearing difficulty patients, effective adjustable filter bank and easily adjustable environmental friendly.

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