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***Research Paper***

# Efficient uniform digital filter bank with linear phase and FRM technique for hearing aids

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**Abstract**

This paper attempts to present an uniform digital filter bank based on linear phase FIR and IIR filters applied for Frequency Response Masking (FRM) technique in hearing aid applications. In the proposed filter bank, nine uniformly spaced sub-bands are formed with the help of half band filters and masking filters. These nine channel FIR filter bank is realized using an interpolated half band linear phase FIR filter and an appropriate number of masking FIR filters. The nine channel IIR filter bank is realized using an interpolated half band approximately linear phase IIR filter and an appropriate number of masking filters. The proposed approximately linear phase IIR half band filter bank is compared with filter bank based on linear phase FIR half band filters in terms of area, power, memory and number of gates needed for implementation. The experiment was carried on various hearing loss cases and the results obtained from these tests proves that, the proposed filter bank achieved the required matching between audiograms and magnitude response of the filter bank at very reasonable range with less computational complexity.

***Keywords***: *FIR and IIR Filter Banks; Frequency Response Masking; Approximately Linear Phase IIR Filter*.

## 1. Introduction

Human hearing loss is one of the major and common problems in the word. As per the survey, 12% of the world population suffering from this cause. An electronic portable digital hearing aid that used by the an impaired person makes sound louder so that hearing loss person can listen, communicate with people around him and also improves day to day activities. Hearing thresholds are defined as the softest sounds one can hear and are represented by audiograms [1]. In figure 1, a typical pure tone audiogram measured on a normal person with normal hearing is illustrated. Here, all hearing thresholds are below the intensities consonant and vowels. However, for peoples with impaired hearing, the hearing thresholds become high at certain frequencies causing hearing loss. Using an audiogram the hearing test is marked on the graph that shows the softest sound a person can hear at different frequencies. In addition, the loudness level and frequencies of different speech sound were also presented by the audiogram. It is known fact that, using hearing aid an impaired person sound signals can be amplified selectively such that the processed and amplified sound matches one's audiogram [2]. To demonstrate this, the hearing aid should be able to adjust the sound levels at arbitrarily frequencies of sound signals. In practice, the process can be carried out passing the sound signal through filter bank that splits them into different frequency bands and the gains at each sub bands are adjusted to suit the needs of hearing impaired.


**Fig. 1:** A Typical Audiogram.

Digital FIR filters are most popular because of their linear phase. The complexity of linear phase FIR filter can be significantly reduced by using Interpolated FIR and frequency response masking (FRM) technique [3], [4], [5]. Uniform and non-uniform linear phase digital filter banks are common choice for hearing aid devices. More effort was invested in the development of uniform digital filter bank for hearing aid and audio applications [5], [6], [7], [8]. Digital Filter banks are used widely in digital hearing aids [9] in order to decompose a wide band signal in to sub-bands. Digital filter banks directly designed using linear phase FIR filters requires high order filters which increases the computational complexity and overall delay of filter bank, [10]. The increase in computational complexity increases the power consumption by the filter bank structure. The complexity of linear phase FIR filter can be significantly reduced by using frequency response masking (FRM) technique. The FRM technique can be used to design sharp FIR or IIR filters.

The complexity of linear phase FIR filter can be considerably reduced by using cascade connection of an interpolated FIR filter and properly designed masking filters. The realization of uniform filter bank is presented in [11], [12] Here Prototype filters are low order linear phase FIR half band filters. By interpolating the prototype filter, multichannel filters are obtained. Then channels are separated by using one or more masking filters. All the filters in the design are linear phase FIR half-band filters. First low pass filters are designed according to the requirements. The complementary property is used to derive high pass filters from the low pass filters which in turn reduce the complexity of filter bank structure.

The main advantage of IIR filter is there low complexity, order of sharp IIR filter are small when compared to that of FIR filter of same specification. But IIR filters have a disadvantage that they are non-linear which makes them unsuitable for filter banks. The disadvantage of non-linearity can be overcome by using approximately linear phase IIR filters. The uniform filter bank structure presented in [13], [14] uses approximately linear phase IIR filters. The overall delay and complexity of filter bank can be reduced by use of IIR filters. FRM technique can be used to design sharp IIR filters. Filter bank based on approximately linear phase IIR half-band filters is also similar to filter bank proposed in and uses FRM technique. Filters in the filter bank are either interpolated approximately linear phase IIR half-band or approximately linear phase IIR half-band. The number of coefficient required for IIR filters are less compared to that of FIR, so that it reduces the number of multiplications required which makes them efficient for implementation. Both the filter bank structures proposed are based on half-band filters. One of the important benefits of using half-band filter is that every second coefficient in the transfer function is zero which makes them suitable for efficient VLSI implementation [15].

This paper is organized as follows: in section II filter bank based on linear phase FIR half-band filters are explained, in section III filter bank based on approximately linear phase IIR half-band filter is explained, in section IV results of simulation and synthesize of both filter bank structures are compared and section V concludes the paper.

## 2. Linear phase fir filter bank based on phase half band filters

The Z-transform function of linear phase FIR half band filter is defined as the power series

$$F(z) = \sum_{n=0}^{2k} f(n) z^{-1} \qquad (1)$$

Where, $k$ is odd and coefficients are symmetric in respect to the central coefficient $f(k)$ which is always equal to the $1/2$. The length $N$ is also an odd number given by $N = 2k + 1$.

The stop band and pass band of linear phase FIR half-band filters are symmetric with respect to the middle of baseband $\omega_c = \frac{\pi}{2}$. Thus pass band and stop band edges are symmetric and peak pass band and stop band ripples are equal. The basic element of the filter bank is complementary linear phase FIR filters $[F(z), F_c(z)]$. First low pass filter of desired transfer function is designed and high pass filter transfer function is derived using complementary property. The transfer function of complementary high pass filter $F_c(z)$ defined as,

$$F_c(z) = z^{-(N-1)/2} - F(z) \qquad (2)$$

Where, $N$ is the length of the low pass filter $F(z)$. the high pass filter is implemented from low pass filter using a delay branch .The realization of filter pair $[F(z), F_c(z)]$. is shown in Figure 2.

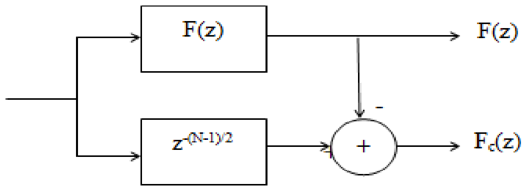


**Fig. 2:** Realization of Linear Phase FIR Filter Pair $[F(z), F_c(z)]$.

A uniform filter bank structure consist of a connection of interpolated prototype filter $F^I(z)$ and several masking filters. The realization of a nine-channel filter bank is shown in figure 3. By interpolating a low order prototype filter $F(z)$ by a factor of 8 multiband filter is obtained. The bands of multiband filter $F^I(z)$ represents the odd channels. The even channel is given by the filter $F_c^I(z)$ which is obtained by interpolating the complementary prototype filter $F_c(z)$. The stopband frequency of prototype filter $F(z)$ is $\omega_p = 0.8\pi$. By interpolating the filter by a factor of 8, the stopband frequency of first filter $F^I(z)$ is obtained as 0.8π/8= 0.1π and images appears at frequencies 0.3π, 0.5π,0.7π and π. The magnitude response of prototype filter pair $[F(z), F_c(z)]$. is shown in figure 4 and magnitude response of interpolated prototype filter pair $[F^I(z), F_c^I(z)]$ is shown in figure 5. The subbands of filter bank are separated from interpolated prototype filter $F^I(z)$ and $F_c^I(z)$ by appropriate connection of masking filters. The making filters are designed so that it masks the required subband of preceding filter. The masking filters are either interpolated half-band filters or half-band filters according to the requirement. Interpolation of filters is performed by replacing each delay elements with a number of delay elements that is equal to the interpolation factor [16], [17].

Figure 6 shows the amplitude response of odd channels. Since length of masking filters are different, delay provided by each masking filter pair is different.

The delay provided by each channel is different, so in order to compensate additional delay is provided to all channels except that of channels with longest delay (channels 6 and channel 4). Here filter $F_{m7}(z)$ is of highest length since it separates channel 4 and 6 which closer to frequency 0.5π. The overall delay of filter bank depends on delay of critical masking filter $F_{m7}(z)$ and is equal to 171 samples.

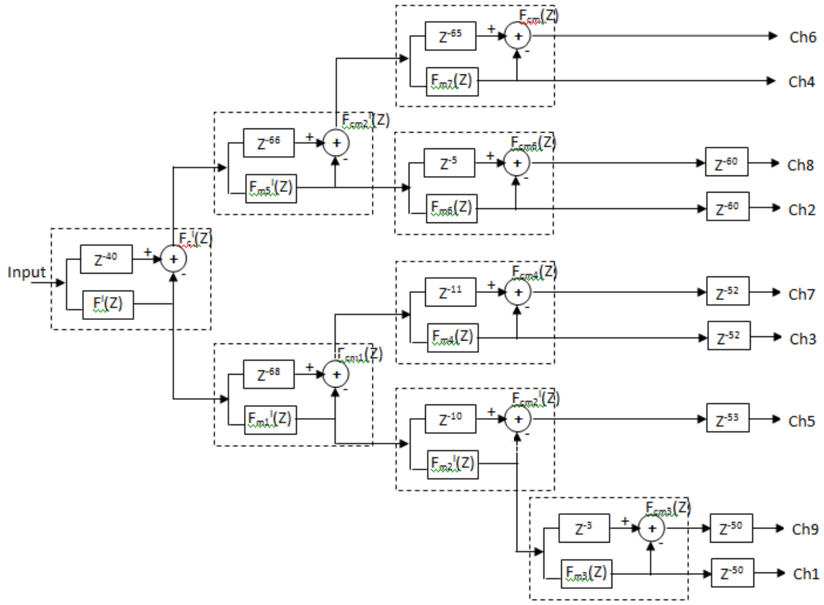


**Fig. 3:** Nine Channel Uniform Digital FIR Filter Bank.

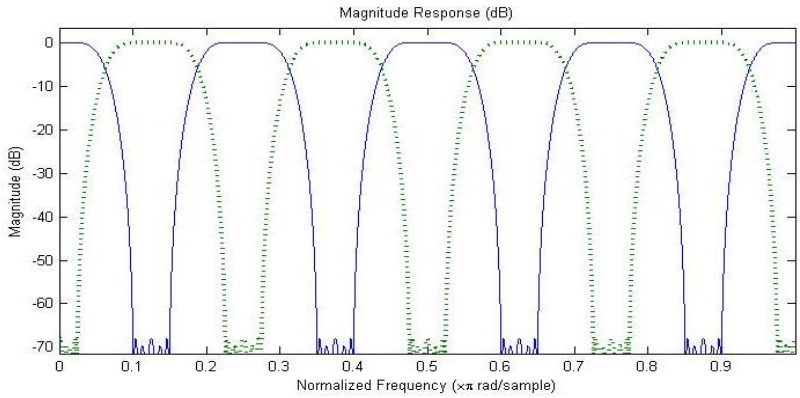


**Fig. 4:** Magnitude Response of Prototype Interpolated Filter Pair $[F^{I}(z), F_c^{I}(z)]$ .

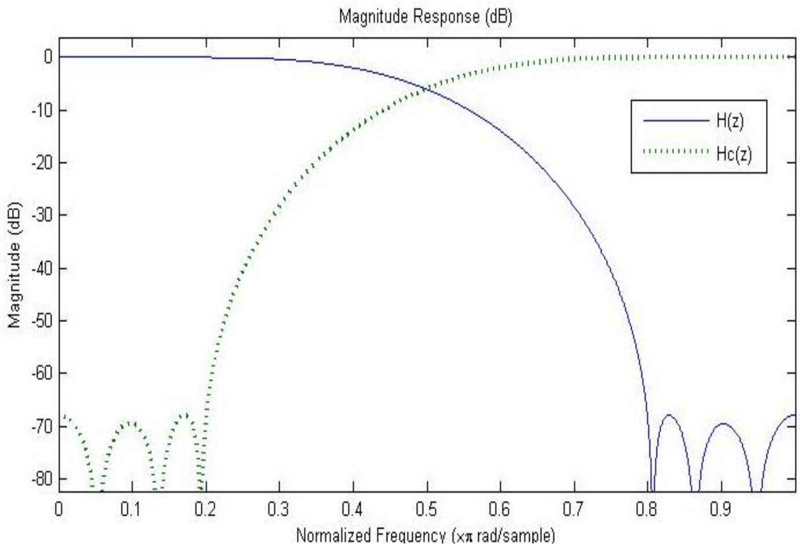


**Fig. 5:** Magnitude Response of Prototype Filter Pair $[F(z), F_c(z)]$.

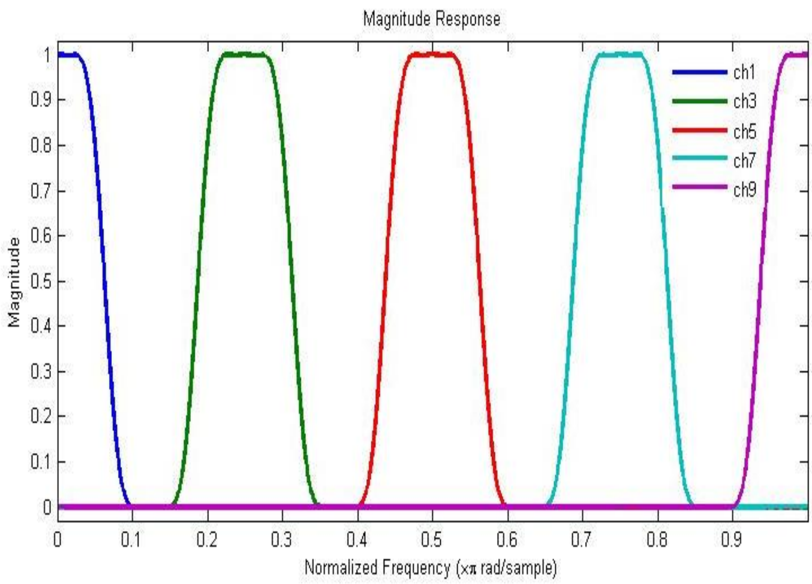


**Fig. 6:** Amplitude Response of Add Channels.

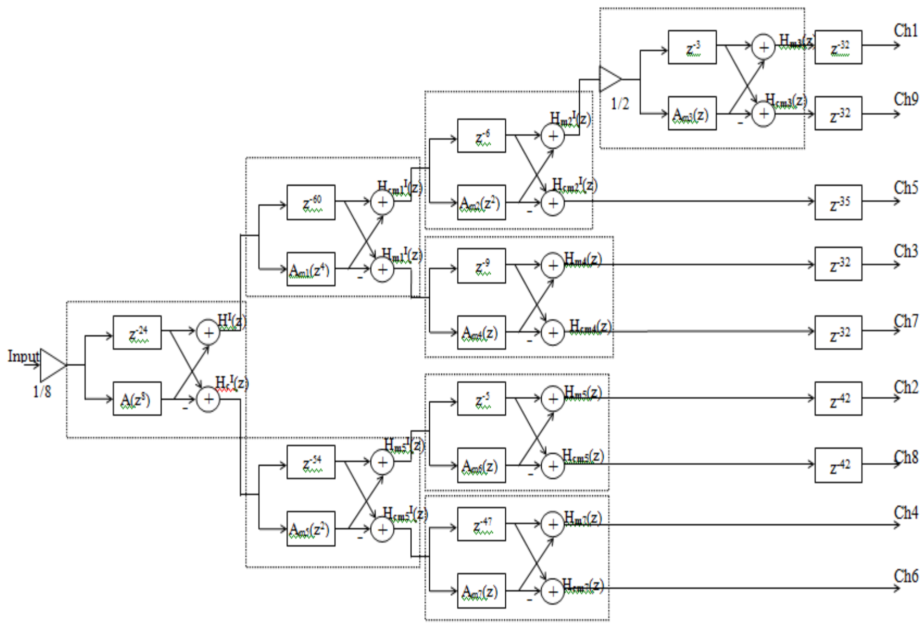


**Fig. 7:** Nine Channels Uniform Filter Banks Based on Approximately Linear Phase IIR Half Band Filters.

**Table 1:** Parameters of Prototype FIR Filter Bank

| Filter | Pass band edge Frequency | Length of filter | No. of multiplications required |
|---|---|---|---|
| F(z) | 0.2π | 11 | 4 |
| $F_{m1}(z)$ | 0.4π | 35 | 10 |
| $F_{m2}(z)$ | 0.2 π | 11 | 4 |
| $F_{m3}(z)$ | 0.1 π | 7 | 3 |
| $F_{m4}(z)$ | 0.35 π | 23 | 7 |
| $F_{m5}(z)$ | 0.45 π | 67 | 18 |
| $F_{m6}(z)$ | 0.225 π | 11 | 4 |
| $F_{m7}(z)$ | 0.475 π | 131 | 34 |

All the filters in the FIR filter bank are designed as linear phase FIR half-band filters with stop band attenuation of 60 dB. The pass band edge frequency of all filters along with length of filters before interpolation is shown in Table I. The nine channels of filter bank pass different range of frequency. The figure 5 shows the amplitude response of odd channels [18]. Since length of masking filters are different, delay provided by each masking filter pair is different. The delay provided by each channel is different, so in order to compensate additional delay is provided to all channels except that of channels with longest delay (channels 6 and channel 4). Here filter $F_{m7}(z)$ is of highest length since it separates channel 4 and 6 which closer to frequency 0.5π. The overall delay of filter bank depends on delay of critical masking filter $F_{m7}(z)$ and is equal to 171 samples.

## 3. Filter bank based on approximately linear phase IIR half-band filters

The filter bank overall delay can be reduced replacing linear phase FIR half-band filters with approximately linear phase IIR half-band filters. The basic element of the filter bank is complementary approximately linear phase IIR filters $[H(z), H_c(z)]$ defined as,

$$H(z) = \frac{z^{-D} + A(z)}{z} \tag{3}$$

$$H_c(z) = \frac{z^{-D} - A(z)}{z} \tag{4}$$

Where, $D$ is an integer and $A(z)$ is an all-pass filter of order $D+1$ gives the delay introduced by the filter. Similar to FIR half-band filters pass band and stop band of IIR half-band filters are also symmetric with respect to the middle of baseband $\omega_c=\pi/2$. A nine-cannel filter bank based on approximately linear phase IIR half-band filter is shown in figure 7. The structure is similar to the filter bank structure described in section II, the difference is that filter pairs are realized using approximately linear phase IIR filters. For VLSI implementation the number of multiplications needs to be reduced so the structure is modified by multiplying the input by 1/8 initially instead of multiplying with 1/2 at each stage.

The pass band edge frequencies of all filters are same as that of FIR filter bank. Since IIR filter order is less compared to FIR for same parameters, the numbers of multiplications required are less. The entire all-pass branch in IIR filters are designed as connection of second and forth order sections. The number of multiplications required for implementation of all-pass branch is the half of order of all-pass branch. Table II shows the order of all-pass branch and number of multiplications needed. The function of interpolated prototype filter and masking filters are also same as that of FIR filter bank but the phase of filters are approximately linear. The Figure 8 shows magnitude response of prototype filter $H(z)$ along with the phase, here the phase linear except at transition width. Since the delays introduced by IIR filters are less, the delay of overall filter bank is less.

The delay provided by channels are different, so additional delay are provide to all channels except to 4 and 6 (channels with highest delay) to compensate. The overall delay is 125 samples for filter bank based on approximately linear phase IIR half-band filters. For VLSI implementation of filter bank all the filter coefficients are represented by 32 bit fixed point. For efficient implementation of filter bank the number of multiplications need to be reduced which reduces the power consumption, area and number of gates. The use of half-band filters has benefit of efficient implantation since every second coefficient in the transfer function is zero. The main significance of the proposed filter bank with FRM is to increase the clarity of the audiogram.

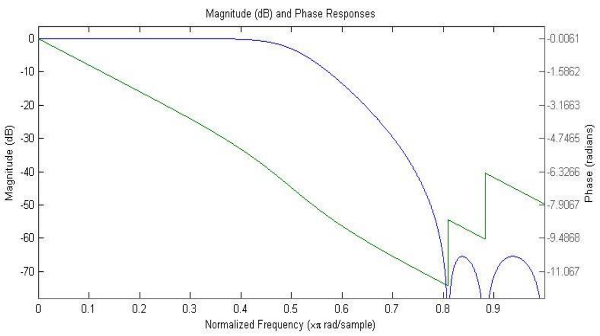


**Fig. 8:** Magnitude and Phase Response of Prototype Filter $H(z)$.

**Table 2:** Parameters of Prototype Filters of IIR Filter Bank

| Filter | Pass band edge Frequency | Order of All Pass branch | No. of multiplications required |
|---|---|---|---|
| H(z) | 0.2π | 4 | 2 |
| $H_{m1}(z)$ | 0.4π | 16 | 8 |
| $H_{m2}(z)$ | 0.2 π | 4 | 2 |
| $H_{m3}(z)$ | 0.1 π | 4 | 2 |
| $H_{m4}(z)$ | 0.35 π | 10 | 5 |
| $H_{m5}(z)$ | 0.45 π | 28 | 14 |
| $H_{m6}(z)$ | 0.225 π | 6 | 3 |
| $H_{m7}(z)$ | 0.475 π | 48 | 24 |

## 4. Results and discussion

The filter bank structures are modeled using MATLAB Simulink. A speech signal is inputted to the filter banks model to analyze the output of all the channels and finally speech signal is reconstructed. The proposed filter bank performance is verified for hearing aid application. Audiogram matching test for impaired person hearing loss is simulated and verified. The audiogram is shown in figure 9 shows the matching result and matching error.

The simulation results of filter bank in Modelsim are shown in figures 10, 11 and 12 respectively. The input samples to the filter bank changes at each positive edge of clock cycle i.e. one sample occurs at one clock cycle, this is shown in Fig.10. Initially the reconstructed output will be zero since filter bank provides a delay. The overall delay provided by filter bank based on FIR filter is 171 samples and that of IIR is 125 samples, so input samples are reconstructed from 171th clock cycles after applying input in case FIR filter bank and 125th clock cycles for IIR filter bank. Figure 11 shows the reconstructed output for filter bank based on FIR filter. Figure 12 shows the reconstructed output of IIR filter bank. The reconstructed output for FIR filter bank is perfect where as reconstructed output of IIR filter bank has small distortion. The overall delay provided by IIR filter bank is 125 samples which is lesser than FIR filter bank overall delay. Both the designs are synthesize using Cadence encounter RTL compiler and compared. Table III gives the comparison of two filter banks.

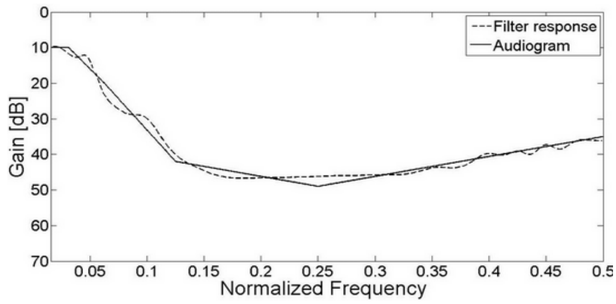


**Fig. 9:** Comparison of Matching between Filter Response and Audiogram.

From the synthesize comparison of two filter bank structures, filter bank based on approximately linear phase IIR filters more efficient in terms of area, number of gates and memory compared to filter bank based on linear phase FIR filters. But the power consumption by IIR filter bank is higher compared to the FIR filter bank.

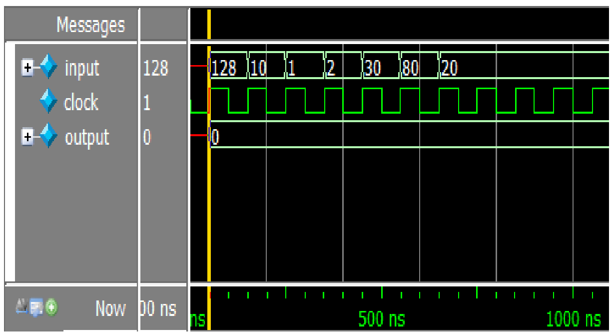


**Fig. 10:** Simulation Result of Filter Bank (Shows the Input Applied).

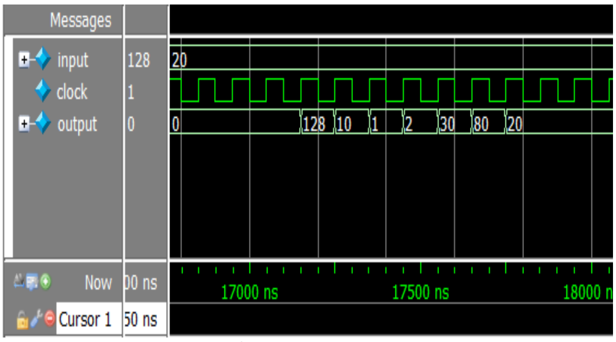


**Fig. 11:** Simulation Result of FIR Filter Bank (Reconstructed Outputs are Shown).

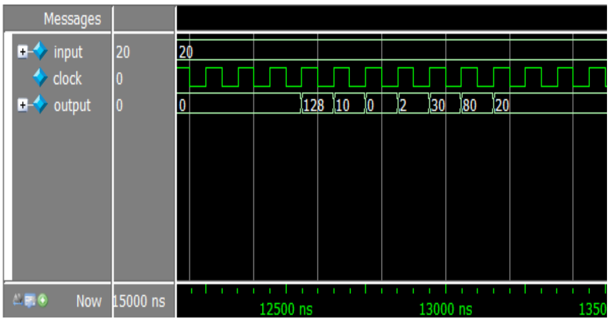


**Fig. 12:** Simulation Result of IIR Filter Bank (Reconstructed Outputs are Shown).

**Table 3:** Synthesize Result Comparison

| Synthesis parameter | FIR filter bank | IIR filter bank |
|---|---|---|
| Power | 12874373 nw | 30477026 nw |
| Area | 766559 nm | 592528 nm |
| Number of gates | 76602 | 58400 |
| Memory | 347400K | 277740K |

## 5. Conclusion

Uniform nine-channel filter bank based on linear phase FIR half-band filters and approximately linear phase IIR half-band filters were designed and implemented successfully. The input samples were split into nine channels and reconstructed in the both filter banks. The overall delay introduced by filter bank based on approximately linear phase IIR filters is small compared to filter bank based on linear phase FIR filters. Both the designs are synthesized and compared in terms of power, area, number of gates and memory. The performance of the proposed filter bank has more significance in the audiogram as compared to the existing models.

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