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Filter Design and Amplitude Shaping Methodologies for De-Noising Digital Hearing Aids

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Abstract: ***Uses of the digital hearing aids have been increased in last decades due to heavy noise pollution. The efficiency of the hearing aids is challenging under the presence of highly noisier environments. Therefore, it is highly required to design an efficient de noising method capable of reducing the excessive amplitude present in environment. It is also desirable to design the compact tiny digital herring aids easy to wear. This paper is focused to preset the de-noising methodologies for speech enhancement in digital hearing aids. Various methodologies based on FIR and IIR filter design are presented and performance is evaluated. It is observed that for short term speech samples the amplitude shaping methods become challenging task due to significant amplitude reduction during de-noising. Thus paper presets the adaptive scaling method for amplitude enhancement. The overall goal of paper to design efficient methodology for digital hearing aids.***

Keywords: **FIR, IIR, Adaptive.**

I. INTRODUCTION

Hearing aids are typically used for handicapped or deaf people to offset hearing loss and to recreate good voice quality. These devices can fit dynamic speaking ranges into a limited dynamic range of impaired ore. This is seen as one of the key functions of an auditory aid. Some of the noises are completely inaudible, some can be detected since their portion of the spectrum is audible but especially in higher frequencies they are not identified properly. Filtering is therefore necessary. The voice signal collected is mostly more noisy and must be filtered at the front end. Usually used FIR/IIR filter. The most prevalent type of hearing loss is neural hearing loss sensor.

A. Human Speech Production system

The speech is essentially produced with three functions, including engine control, motion articulation and sound generation. The motor control function is provided by the human brain that develops speakers and sends signals to producing bodies via sensory nerves. Signals are then received from the control unit and the word or sound to be created will take form. This is called "articulatory movement" Speech is finally produced by throwing air through the mouth and nasal cavities and sounding.

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A hearing aid loudness the sound and helps speaking for the disabled individual. It is designed to capture the audible signals by means of a microphone, turns weak signals into powerful signals and sends the speaker to the ear. The main task of digital hearing aids is signal processing algorithms for speech quality reproduction. Various speech processing sequentially required to be performed by the hearing aids are as follows.Amplifying

- Filtering for noise cancelation
- Peak-clipping or amplitude shaping
- Compression: output limiting, WDRC, etc
- Frequency shifting

Fig.1. Stages of processing for hearing aids

The most essential part is noise cancelation. The hearing aids pre-processing is implemented in the three stages. The processing stages are shown in the Figure 1. The initially for generating the synthetic signal the noise is added to the signal. The noise filtering is the main processing block for the process. The prime concern of the paper is to design the methodology for hearing aids de- noising. Hearing Aids (HA) aretiny electroacoustic gadget primarily designed for selective amplification of sounds or speech signals. The objective of HA is .to enhance speech quality to design an intelligible HA for impaired person with hearing disability.The major tasks to be performed by the digital hearing aids are shown in the Figure 2.

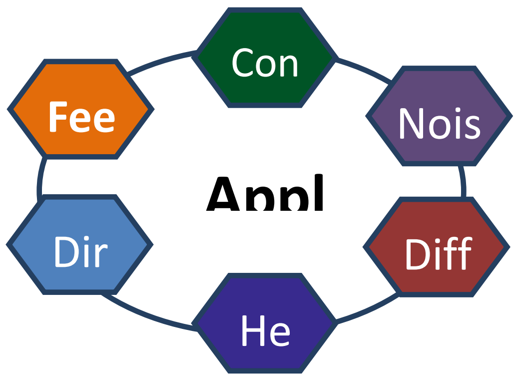

Fig.2. Applications 'tasks of the Digital Hearing aids

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Filtering is an essential processing step for HA. Filter banks of digital HAcan be adaptively tunes for setting the sub band gains therefore end up compensating against the hearing loss pattern of every individuals. This process may also require filtering the noisy speech before intensifying it to improve the nature of sound.

B. Various Design Challenges in Hearing Aids

Some of the major challenges identified for digital hearing aids designing are addressed in this section.Most of the existing designs were opted for FIR filters, but filter order was higher.It is observed that better performance is achieved by designing filter using an optimization technique may enhance performance of Hearing Aid.

It is required to reduce the order of the filter to optimally use IIR filter design is an open challenge.The speech processing algorithms are executed in the DSP block contains the digital processor. The binarisation is done by the A\D converter block at the front end. The digital speech filtering hardware is available on DSP board on the processor. Thus it is highly required 8to design the optimum filter with lower reduced order. Thus dissertation proposed to reduce the order of IIR Filter. So that number of taps can be minimized to reduce hardware area on chip.

Tuning IIR filter for Speech de noising is a great challenge. Amplitude reduction is essential part of the processing of hearing aids for loss control a changing problem.Evaluating the FFT analysis of the filtered is required.

II. LITERATURE REVIEW

There is also certain approach [4, 9, 14], designed based on thresolding for removing the unwanted noise from the noisy speech signals. The threshold based technique may de noise speech signals efficiently. The basic classification of filtering techniques used for the digital hearing aids is shown in the Figure 3. Below.

Table 1. Summary of the Literature Review

Authors	Filter Algorithm	Description	Parameters
IG. Mota et al [1]	frequency transposing	Designed the Hearing aid based on frequency transposing..	Frequency responses analysis
J. I. Marin-at al [2]	Wiener filter	Low bandwidth and complexity multichannel wiener filter for binaural hearing aids	Reduce order filleting
A. Martinez et al [4]	IIR Filtered	Acoustic feedback reduction based on FIR and IIR adaptive filters	Filtered signal quality. And frequency response
K. N. Parvin et al [9]	FIR Filter	FIR fitter design for hearing aids.	Frequency response of the filter.
Z. Shang et al [11]	FIR filter	Low power Fir filter using frequency masking.	Frequency masking
H. Qi et al [9]	Partial fraction IIR	Designed IIR filter based on partial fraction minimization approach	Filter order,
Proposed	reduced IIR by pass and stop band	Reduced order IIR filter designing using transfer functions 6for signal demising,	Amplitude response, filter band response

III. FILTER DESIGN METHODOLOGIES

The filter design methodologies are primarily classified in the below figure. The human perception relies on the logarithmic scale; non-unified filter banks exceed uniform filter banks. The fundamental advantage of non-uniform filer banks is that they require fewer sub-band filters, and consequently minimal hardware complexity and cost.

Non-uniform FIR digital rank filters [2] are used for Hearing Aids processing for the noise signal. The frequency response masking is commonly used in filter design methods in the application of hearing aids.

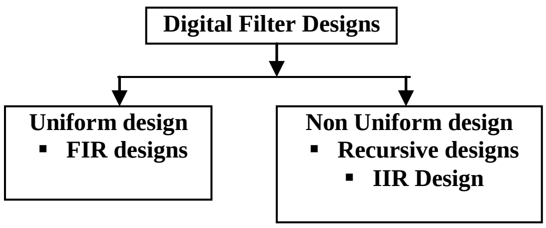

Fig.3. Classification of filters for Digital Hearing Aids

A. Adaptive Filter Design for HA

Background noise has a significant influence on speech intelligibility. Over many years, the issue of noise loss in hearing aids has been dealt with in numerous ways. The methods employed vary from simple filters to complex signal processing systems. This paper provides a method of negating the internal audio effect (error signal) in digital additive support induced by the microphone's acoustic connection to the speaker with the adaptive filter. For the eradication of a digital audio signal and hence for the production of the original signal, we examined the Normalized Last Medium Square Algorithms and the Recursive Least Mittel Square (RLMS).

B. Digital Filter Banks

A lot was done to build consistent and uniform filter banks for audio aid. But it never addresses major variances in intermediate-frequency hearing losses in most cases. This work proposes a low-complicated architecture for the digital auditory assistance application of the uniformly spaced FIR filter bank. For fabrication of

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8 non-uniformly spaced, one-half-strand filters as a prototype filter; the frequency response masking technology (FRM) is employed. With both the FRM and half-band filter approaches, the number of multipliers and adders in the linear FIR filter is drastically decreased. In addition, masking filters can be made complex and effective from the filter prototype. The FRM technique is accomplished by cascading various filter prototype combinations and their interpolating filters for the construction of sub bands. The results demonstrate that the suggested filter bank attenuates 120 dB with only 13 multipliers. The suggested FRM Filter Bank can be used to match audiograms in the mid-frequency audio threshold with large variations. The approach of selecting the band boundaries of each band optimizes the match between the audiograms and the filter banks.

C. IIR Filter Design

The process of IIR filter design methodologies for designing hearing aids is shown in the process of system as presented in Figure 2.

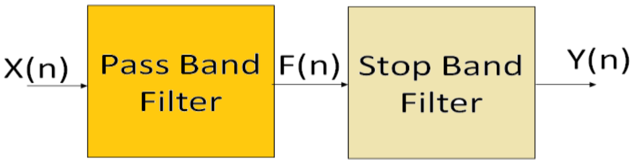


Fig.4. Process of Digital IIR Filter Design

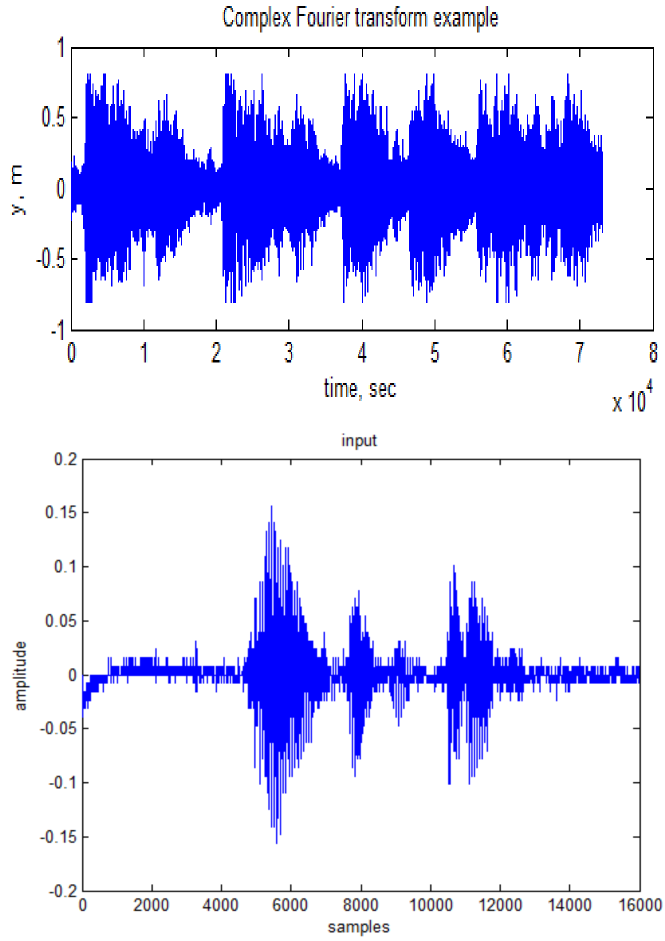
$$h = \frac{0.2079\, s^{\wedge}4 - 0.4158\, s^{\wedge}2 + 0.2079}{s^{\wedge}4 - 2.157\, s^{\wedge}3 + 1.662\, s^{\wedge}2 - 0.6878\, s + 0.195}$$

$$h1 = \frac{\begin{array}{c} 0.02426\, s^{16} + 0.2165\, s^{15} + 1.039\, s^{14} + 3.401\, s^{13} \\ + 8.38\, s^{12} + 16.32\, s^{11} + 25.86\, 8s^{10} + 33.89\, s^{9} + 37.05\, s^{8} \\ + 33.89\, s^{7} + 25.86\, s^{6} + 16.32\, s^{5} + 8.38\, s^{4} + 3.401\, s^{3} \\ + 1.039\, s^{\wedge}2 + 0.2165\, s + 0.02426 \end{array}}{\begin{array}{c} s^{16} + 5.194\, s^{15} + 13.43\, s^{14} + 23.61\, s^{13} + 32.38\, s^{12} + 36.66\, s^{11} \\ + 34.85\, s^{10} + 28.07\, s^{9} + 19.36\, s^{8} + 11.44 s^{7} + 5.75\, s^{6} + 2.427\, s^{5} \\ + 0.8482\, s^{4} + 0.2382\, s^{3} + 0.05069\, s^{2} + 0.007401\, s + 0.0005974 \end{array}}$$

IV. PROPOSED IIR FILTER DESIGN

The Long term and short term Speech samples are proposed to Evaluate as specific case. A short term alphabet speech samples are recorded. It is proposed to design optimization method for reduced order IIR filter de-noising of herring aid signal. Proposed IIR filters have to be designed using the pass band and stop band filter for de nosing.Adoptive amplitude and threshold scaling is proposed for short term speech enhancement. It is also proposed to evaluate the FFT of the filtered responses for data preservation.The transfer unction comparison is used for evaluating the performance of filtering.

The database of sound waves is created for the study distinct set are shown in the Figure 2. The proposed approach is implemented in MATLAB and results are analyzed in terms of optimum sound quality and 8filter orders. Designed IIR filter is a combination of the band pass and he stop band based cascaded filter design approach.



a) Short term speech recorded as alphabet A.
Fig.5. Speech Data Examples used for study

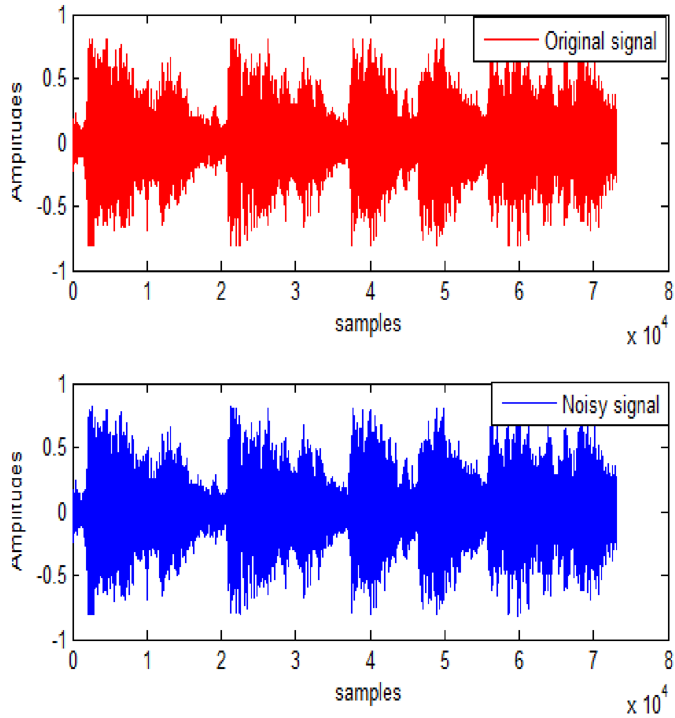


Fig.6. Comparison of the original and noisy speech

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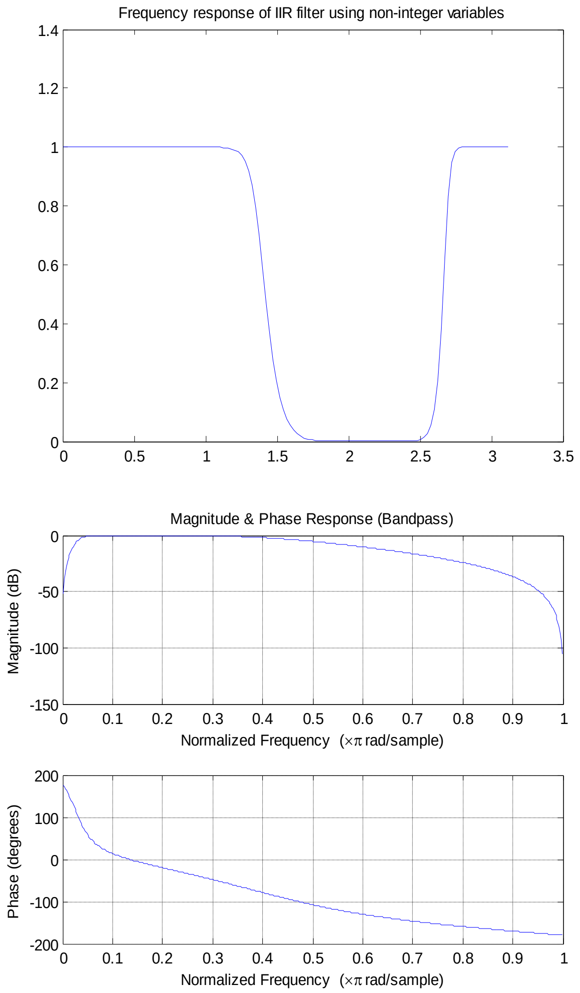

Fig.7. Two step IIR Filter Responses, a) frequency response of the proposed IIR filter with optimum parameter tuning. b) Magnitude and phase responses

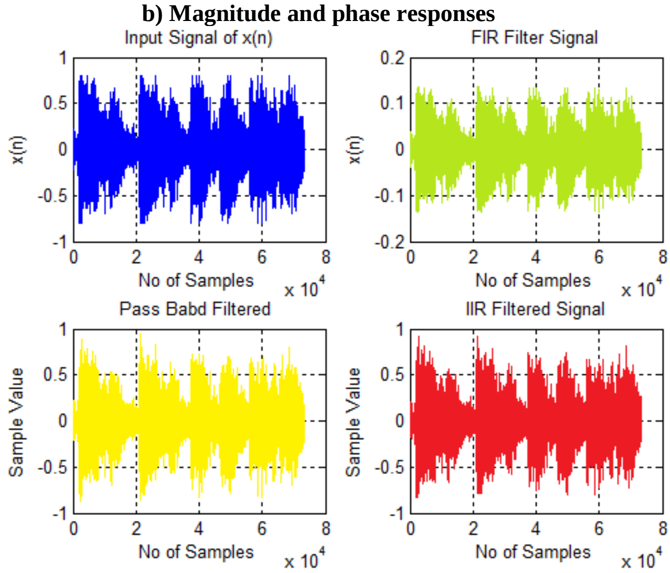

Fig.8.Comparison of the Filtering performance FIR Vs IIR

The major challenge is to select the proper lower and upper cutoff frequencies for pass bad ad stop band filets.

V. CONCLUSION

In this paper Min-Max optimization based IIR filter at reduced order is presented for de-noising the speech for digital Hearing aids. The major conclusions drawn from the study are:

- It is proposed to design the optimization based reduced order IIR filter
- Three case of speech are evaluated as Long term, short term, and real time speech.
- It is concluded that filter order is reduced from 16 to 2 with proposed approach.
- Compared to FIR filter proposed IIR filter performs better in terms of voice quality.
- Adaptive scaling based amplitude shaping and Frequency shaping methods are designed.

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