International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395 -0056
Volume: 03 Issue: 12 | Dec -2016 www.irjet.net p-ISSN: 2395-0072

FIR filter bank design for Audiogram Matching

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Abstract: *A computationally efficient non-uniform digital FIR filter bank is proposed for hearing aid applications. Filter bank for digital hearing uses significantly different criteria than those designed for coding applications. In this paper we are designing a computationally efficient non-uniform FIR filter bank for hearing aid applications. In this proposed work we are to match the Audiogram of a patient with the significant loss over the wide range of frequencies. The audiogram hearing threshold are matched with our designed filter coefficients so that the high SNR is generated and noise is minimized. In this way we are adjusting the filter bank channel gains over a large dynamic range to compensate for the hearing loss in digital hearing aids.*

Key words: *Audiogram, FIR Filter Bank, DWT, RLS algorithm, LMS algorithm*

1.INTRODUCTION:

The importance of a hearing aid is to provide hearing capability to a person having suffering from hearing disability. The process of hearing aid is to amplify the required input signals to the levels that the hearing impaired can hear. An audiogram is a graph plotted between frequencies and hearing threshold indicates the hearing capability of a person for various frequencies. An ideal hearing aid device includes several important features as adjustable magnitude response on different frequencies, low processing delay, linear phase to prevent the audio signal from distortion, noise reduction, low power consumption, small in size, programmability etc. All these required parameters can be achieved by using the digital signal processing approach compared to the analog signal processing for implementing the hearing aid. the digital signal processing algorithms have seen significant improvement in recent years, common hearing aid fitting process are still focusing on frequency band and corresponding gain of the filters to match the audiogram of the specific user with individual hearing loss (Kuo et. al., 2008). hence the basic requirement of any hearing aid is to provide correct adjustment of the gain versus frequency, which is also called audiogram matching/fitting. Sometimes, there are small errors may have a significant influence on performance, in particular with regard to speech intelligibility.

2.HEARING AMPLIFICATION:

The auditory system is a sensitive and complex network which transfers sound waves to neuroelectrical signals towards the brain. Problems in auditory system will cause hearing difficulties or deafness. Tohelp the hearing-impaired peopleimprove the quality of life, assistive technology has been developed for quite a long time. The main types of hearing loss are categorized as conductive, sensor neural, mixed and central [82]. The damage to outer and middle ear results in conductive hearing loss. The degree of hearing impairment can be classified using the hearing thresholds.

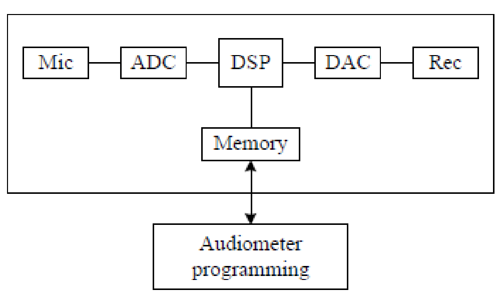

Figure (1) block diagram of digital hearing aid

2.1 Audiogram:

An audiogram is a graph plotted during the hearing test shows the softest sound a person can hear at different pitches or frequencies. The Y axis represents the intensity or response of the ear measured in decibels (dB) corresponding to the frequency in hertz (Hz) marked on the X axis. An `O' generally used to represent the responses for the right ear and an `X' represents the responses for the left ear. Curves displayed in decibels generally describe the individual hearing threshold of a person compared to the normal hearing average, which lies around 0 dB. Due to individual differences, all thresholds up to 20 dB are considered as normal. Figure 1 shows the audiogram of a normal hearing person. The threshold between 21 to 40 dB is considered as mild hearing loss, 41 to 55 dB is considered as moderate hearing loss, 56 to 70 dB as moderately severe, 71 to 90 dB is considered as severe hearing loss and greater than 90dB is considered as profound hearing loss.

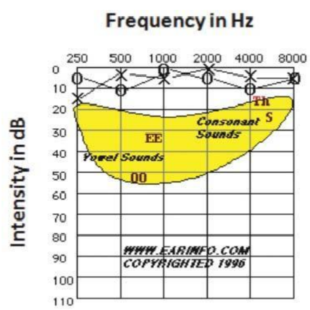

Figure 2 (a) Audiogram for normal hearing

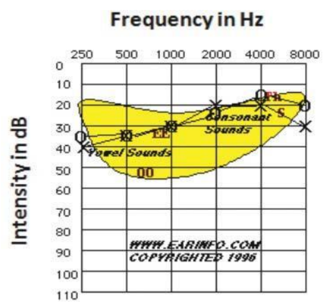

Figure 2(b) Audiogram for mild to moderate hearing loss

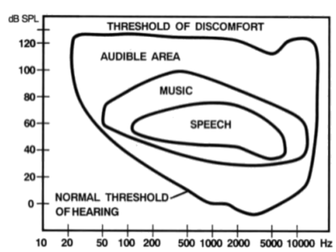

Figure 3(a): Hearing threshold of:
(a) normal hearing person;

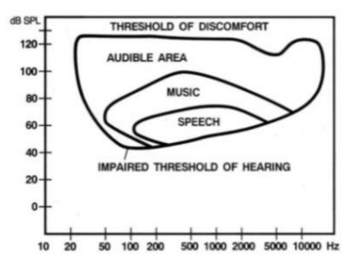

Figure 3(b) hearing-impaired person. Pictures were originally published in [21]. Used with permission from Widex A/S.

2.2 Filter bank for hearing amplification:

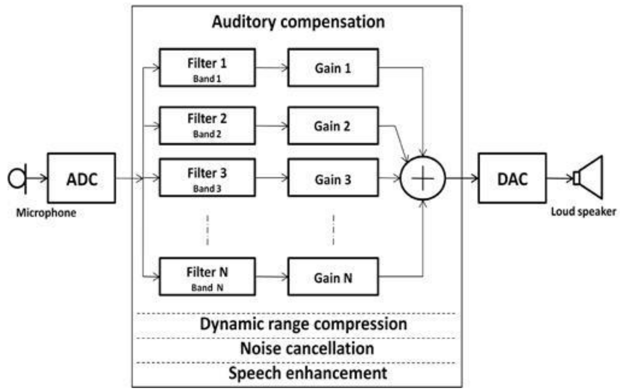

Figure (4) filter bank for digital Hearing aid

2.3 Digital Filter:

In DSP, the main function of digital filter is to extract the desired components or to remove the undesired components of the input signal. From an mathematical view, a digital filter performs the convolution operation between the sampled input and the weighting function of the filter. There are two types of digital filter, namely, finite impulse response (FIR) filter and infinite impulse response (IIR) filter. FIR filter structure is non recursive while IIR filter structure is recursive. The advantage of FIR filter is high stability, low coefficient sensitivity and linear phase response. However there are certain disadvantages of FIR filters as they increase the computation cost as number of multipliers are added. For Non-linear phase filter the number of multipliers is equal to the length of the filter and for linear phase filter number of multipliers are half of the filter length.

2.4 Filter Bank review:

A filter bank is an array of bandpass filters. Based on their operations they are classified as analysis filter bank and synthesis filter bank. A filter analysis bank separates the input signal into several components, with each one of the sub-filters carries a single frequency sub band of the original signal. On the contrary, a synthesis filter bank combines the outputs of sub bands to recover the original input signal. In most applications certain frequencies are more important than other. Filter bank can isolate different frequency components in a signal. Therefore we can put more efforts to process more important components and less efforts for less important component to be proceed.

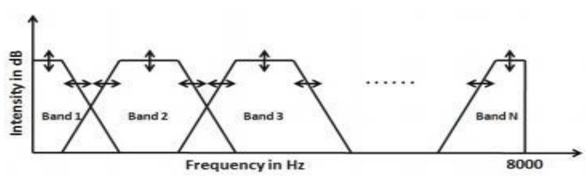

Figure (5) General filter bank structure

3. PROPOSED WORK:

The proposed paper will use the Discrete Wavelet Transform along with adaptive RLS filter algorithm to implement the FIR filter Bank for efficient audiogram matching. In the proposed work we will provide a minimized error in the audiogram matching of a hearing impaired person and then we will improve the signal to noise ratio, so the hearing threshold will get reduced as a result a hearing disabled person will become able to hear the speech sound below its actual threshold level. The cited method of adaptive RLS algorithm will provide us required filter coefficients in order to achieve the suitable gain on different speech frequencies.

3.1 DWT:

DWT decomposes the signal into mutually orthogonal set of wavelets. Discrete wavelet transform can be used to denoising of a noisy signal. If we take only a limited number of highest coefficients of the discrete wavelet transform spectrum, and we perform an inverse transform (with the same wavelet basis) we can obtain more or less denoised signal. There are several ways how to choose the coefficients that will be kept. Within Gwyddion, the universal thresholding, scale adaptive thresholding [12] and scale and space adaptive thresholding [13] is implemented. For threshold determination within these methods we first determine the noise variance guess given by

$$\hat{\sigma} = \frac{\text{Median}\,|Y_{ij}|}{0.6745}$$

where Y_{ij} corresponds to all the coefficients of the highest scale subband of the decomposition (where most of the noise is assumed to be present). Alternatively, the noise variance can be obtained in an independent way, for example from the AFM signal variance while not scanning. For the highest frequency sub band (universal thresholding) or for each subband (for scale adaptive thresholding) the variance is computed as

$$\hat{\sigma}_Y^2 = \frac{1}{n^2} \sum_{i,j=1}^{n} Y_{ij}^2$$

Threshold value is finally computed as

$$T(\hat{\sigma}_X) = \hat{\sigma}^2 / \hat{\sigma}_X$$

where

$$\hat{\sigma}_X = \sqrt{\max(\hat{\sigma}_Y^2 - \hat{\sigma}^2, 0)}$$

When threshold for given scale is known, we can remove all the coefficients smaller than threshold value (hard thresholding) or we can lower the absolute value of these coefficients by threshold value (soft thresholding).

DWT denoising can be accessed with *Data Process* → *Integral Transforms* → *DWT Denoise*.

4.ADAPTIVE FILTER:

An adaptive filter is designed in such a way which the transfer function adjusts itself which is driven by an error signal. The adaptive filters uses of feedback in the form of an error signal to modify its transfer function to be in accordance with changing parameters. An adaptive filter is a non-linear filter because its characteristics are fully dependent on the input signal [14]. The aim of an adaptive filter is to determine the difference between desired output and an adaptive filter output. The error signal is again fed back to an adaptive filter and its coefficients are changed according to the algorithm in order to minimize this difference.

4.1Use of Adaptive Filters for Noise Cancellation in Speech Signals:

Use of adaptive filters for noise cancellation is shown in figure 4.1 [15]. When the audio signals are to be transmitted in a noisy atmosphere, adaptive filters are used. Two inputs are required for noise cancellation with the help of adaptive filters. One of the input contains the noise contaminated signal and the other signal contains the reference signal which is a noise related to the input signal in some way. The system is so designed that in filters the reference noise signals to make it similar to the signal provided at main input and now this filtered output is subtracted from the main input. This process removes the noise and leaves behind the exact speech signal [16]. However in practical implementations of the system, the noise is not fully removed but it is reduced considerably.

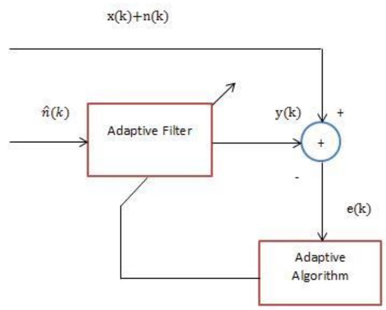

Figure (6) Adaptive filter used for noise cancellation

5. TOOL USED: MATLAB 2015(a)

We have utilized the various DSP algorithms used in binaural hearing aids and showed the simulations for the same in MATLAB version 2015a. The individual algorithms cited above should be performed in a sequential order to correctly simulate binaural hearing aids. First, Adaptive noise cancellation should be performed followed by frequency shifting and controlled amplification

6.CONCLUSION:

We propose the application of subband adaptive filtering to equalization problem as in case of fullband. The proposed subband filtering gives a significant reduction in computations and improved convergence rate over the full-band scheme. Use of adaptive filters in a filter bank can reduce the noise in the signal and thereby we increase SNR. In this paper we are providing oversampled subband system that offers much reduction in noise and hence it is the best method of adaptive filtering, both in terms of convergence and computational efficiency. In addition we are going to prefer Affine Projection Algorithm (APA) as Adaptive filtering algorithm at the place of Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) algorithm. since it provides better convergence rate and less computational complexity as compared to other adaptive filtering algorithm.

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