FIR SYNTHESIS OF THE HUMAN HEARING MECHANISM RESPONSE

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Abstract: The paper deals with the synthesis of a model for the nonlinear human hearing mechanism response. The non-uniform frequency sampling is combined with the interpolation methods to design a FIR filter to approximate the psychophysical data (audiogram). The analysis of the designed filters is achieved in LabVIEW. To evaluate the FIR synthesis the magnitude characteristic and the impulse response for different interpolation methods are compared.
Keywords: *Audiogram, approximate FIR filter, interpolation, impulse response.*

I. INTRODUCTION

The human ear is not equally sensitive to all frequencies, particularly in the low and high frequency ranges. Fletcher has charted the frequency response over the entire audio range, originally and Munson in 1933, as a set of curves showing the sound pressure levels of pure tones that are perceived as being equally loud [1].

Human hearing ranges from 20 Hz to 20 kHz. There is little speech information above 8 kHz. Perception of frequencies below 100 Hz is increasingly tactile in nature, making them difficult to assess. The loss of hearing sensitivity is observed first at high frequencies (8 kHz) and later on as the loss progresses, its effect is observed in the mid-frequency region (1-2 kHz). When there is a loss in the low frequency region, the subject nears to deafness. Hence, audiometric tests carried out in the low frequency region do not give any significant information about hearing loss. Therefore, audiologists routinely test only in the range of 125-8000 Hz, often in octave steps. Standardized tested frequencies include 125, 250, 500, 1k, 1.5k, 2k, 3k, 4k, 6k and 8k Hz. They are at octave intervals. Intervening frequencies may also be tested.

When a hearing loss occurs a hearing aid is recommended, which is an electro acoustic device that makes listening easier. Today's digital hearing instruments amplify soft sounds to make them audible just like hearing aids of the past. They provide many advantages over past hearing aids: ear protection to sudden loud, sound classification with automatic characteristics adjustment of the hearing aid based on the sounds around and even allowing wirelessly talk on mobile phone and have the sound amplified to meet the listening needs.

The aim of this paper is to model with a linear FIR filter the nonlinear human hearing mechanism using the auditory data provided by audiologists. This work will continue with the correction of the system's response to enhance the processed sound quality and the methods going to provide good results will be used for hearing aids development.

The paper is organized as follows: section II deals with psychophysical measurements regarding the auditory analysis; section III presents the approximate filter design; section IV details the implementation of the design algorithm; section V presents the validation of the results, and section VI contains the main conclusions of the paper.

II. PSYCHOPHYSICAL MEASUREMENTS

In the absence of complete knowledge of the physical processes of auditory analysis, it is useful to measure the functional performance of the hearing system by means of psychophysical experiments. In such experiments, pure tones instead of speech or other sound stimulus are given to human subjects, who are asked to respond in various ways according to what they hear. Audiometry is the technique to identify and quantitatively determine the degree of hearing loss of a person by measuring his hearing sensitivity. When the results are plotted graphically, they are named as audiograms (frequency response of the human hearing mechanism). The electronic instrument used for measuring the hearing threshold level is called an audiometer. One of the most basic types of auditory measurement is known as an audiogram, which displays the root mean square pressure of sound, which is just audible as a function of frequency of sinusoidal stimulation. This display can be extended to include plots of subjective judgments of equal loudness at different frequencies, for levels well above the threshold of detection.

The outer ear has a damped resonance that enhances the response of the tympanic membrane at around 3.5 kHz. The ossicles of the middle ear couple the vibrations of the tympanic membrane to the spiral-shaped cochlea of the inner ear. Psychophysical tuning curves are derived using masking techniques to show human ability to separate the responses to individual frequency components of a complex signal. Separation is not effective for components closer than the bandwidth of the auditory filter, which is about 10% of centre frequency above 1 kHz and a somewhat larger percentage for lower centre frequencies. Human ability to judge the pitch of

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tones and the frequency of resonances is much better than indicated by the width of critical bands. It is believed to be the result of analyzing the timing pattern of neural firings.

A simple digital filter can provide a good model of the outer and middle ears. The filtering of the cochlea can be modeled as a series of transmission line sections or as a set of discrete filters.

III. APPROXIMATE FIR FILTER DESIGN

In this section, we present a computational algorithm to model the human hearing mechanism response (audiogram) as a FIR filter. It is similar to gamma-tone filter banks [2] or dual resonance nonlinear filter unit [3] or a novel methodology based on restoring normal neural representation [4] that are in current use. The main difference is that there is only one filter to approximate the measured audiogram instead of a filter bank. It may be shown that the algorithm can be used to model any kind of audiograms even those in the animal kingdom.

To approximate any continuous frequency response, one could sample the frequency response at points along the interval of normalized frequencies (0, π) and evaluate the continuous frequency response as an interpolation of the collected samples [5], [6], [7]. The approximation error would then be exactly zero at the sampling frequencies and be finite between them. The smoother the approximated frequency response is, the smaller the interpolation error between the sample points [8].

There are four FIR filter types with respect to the symmetry of the coefficients and the parity of the filter order N. The design equations for the approximate FIR filter are as follows:

I. symmetric, N even

$$h(n)=\frac{1}{N}\cdot\left[A(0)+2\sum_{k=0}^{\frac{N}{2}-1}(-1)^{k+1}A_{k+1}\cdot\cos\left(\omega_{k,n}\right)\right]$$

II. symmetric, N odd

$$h(n)=\frac{1}{N}\cdot\left[A(0)+2\sum_{k=0}^{\frac{N-1}{2}}(-1)^{k+1}A_{k+1}\cdot\cos\left(\omega_{k,n}\right)\right] \tag{1}$$

III. antisymmetric, N even

$$h(n)=\frac{1}{N}\cdot\left[(-1)^{n+1}A\left(\frac{N}{2}\right)+2\sum_{k=0}^{\frac{N}{2}-1}(-1)^{k+2}A_{k+1}\cdot\sin\left(\omega_{k,n}\right)\right]$$

IV. antisymmetric, N odd

$$h(n)=\frac{1}{N}\cdot\left[0+2\sum_{k=0}^{\frac{N-1}{2}}(-1)^{k+2}A_{k+1}\cdot\sin\left(\omega_{k,n}\right)\right]$$

where N is the number of samples, A_{k+1} is the real-valued frequency response, $\omega_{k,n}=2\pi(k+1)n/N$ and $h(n)$ is the filter coefficient at instant n.

IV. IMPLEMENTATION

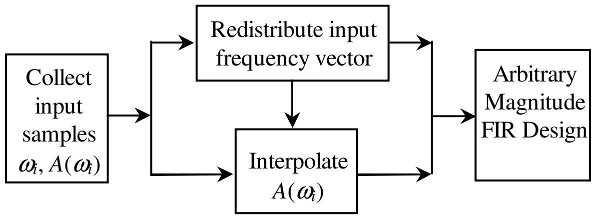

Figure 1. The auditory data processing path.

Figure 1 shows the main stages to process the auditory data (audiogram). The first step is to collect and place into vectors the values of an audiogram at specific frequencies. In the second stage, the interpolation procedure of the audiogram is applied and the frequency vector due to the unequal spacing between the values is redistributed. In the last stage an arbitrary magnitude FIR filter is designed to approximate the given audiogram according to (1).

The redistribution of the frequency vector is necessary for two reasons: there are few numbers of samples in the audiogram and the length between the samples is non-uniform. Therefore, the greatest common divisor gives the uniform length between the samples.

As for the interpolation procedure, the following types were tested: linear, cosine, Spline, Hermite, rational, polynomial and nearest interpolations.

In the coefficient computation procedure, there are three main stages: order parity test, symmetry test of the given samples and generation of the coefficients vector. Figure 2 presents the block diagram of the filter coefficients computation developed in LabVIEW. The computation starts with reading the samples of the audiogram; next, the samples are interpolated and then the parity, the symmetry of these samples are tested. The logical responses of these tests contributes to the computation of the maximum values of the *for loops* indexes and to the selection of some non-common parts from the coefficient expressions. The filter coefficients are computed in the smaller *for loop* and in the greater one, the first, respectively the last element from the given samples is selected to compute the coefficients. Finally the vector with these values is generated and provided. When some of the obtained filter coefficients are too small, a threshold test may be included in the smaller *for loop* arithmetic and as a result, those coefficients will be set to zero. After funding these coefficients, they are sent to the testing block to be analyzed and validated.

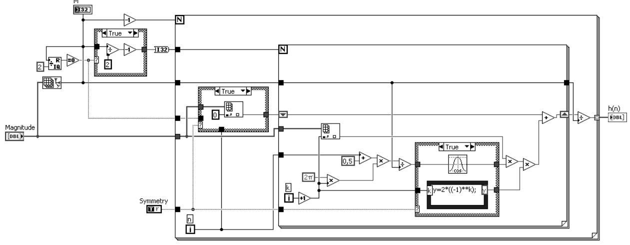

Figure 2. The block diagram of the coefficient vector computation (N even and FIR symmetric).

V. EXPERIMENTAL VALIDATION

The FIR design and analysis were done in LabVIEW. Several audiograms and interpolation methods were used to test the approximate FIR design. To present and validate the synthesis let us consider the audiogram from Table 1.

To ease the interpolation task we inserted some samples between the values of the audiogram. Thus, there are two audiograms: the audiologist's (Table 1) – 10 measurements and the expanded one – 20 samples. The design algorithm processes both of them and the obtained results are compared. The audiologist's audiogram and the expanded one are depicted in Figure 3.

f [Hz]	125	250	500	750	1k	1.5k	2k	3k	4k	6k
A [dB]	20	25	22	23	25	30	37	38	40	32

Table 1. Audiologist's audiogram (experimental).

In Figures 4÷8 the approximate FIR magnitude plots with different interpolation methods (linear, cosine, Spline, Hermite and rational) are presented. In each figure, there are two plots: the interpolation for the audiologist's audiogram (solid line) and the one for the expanded audiogram (dotted line). The computational effort increases for the expanded audiogram, but the approximation is smoother. Figure 7 shows the Hermite interpolated audiogram where the difference between the two plots is very little and the magnitude plot is very smooth. The worst approximation is obtained with the rational interpolation filter (Figure 8).

Figures 9÷11 show the impulse responses for the approximate FIR designed with the non-expanded audiogram and the linear, Hermite and rational interpolations. These responses have a typical shape for a symmetrical and even length band-pass FIR filter. Figure 10 shows the case when the impulse response is the smoothest. In Figure 11 the impulse response presents oscillations having significant amplitudes in the region expected to have a zero response. It is obvious that the rational interpolation is the worst (Fig. 8 and 10) and the Hermite one is the best method (Fig. 7 and 11). The Hermite interpolation provides the best results, but requires complex computations.

The length of the approximate FIR filter was *N=104* for the non-expanded audiogram and *N=520* for the expanded one. Even if the expanded audiogram is closer to the FIR plot than the audiologist's, the increase of FIR order is not acceptable taking into account the higher cost of a hardware implementation and subjective nature of the audiograms' samples.

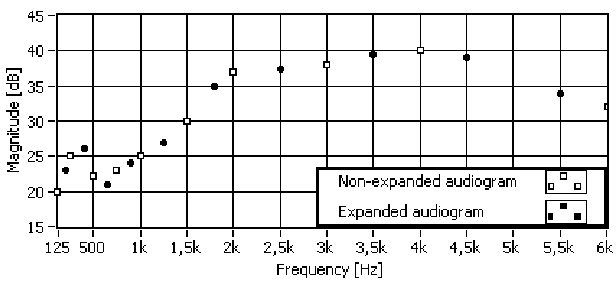

Figure 3. The audiologist's and the expanded audiograms

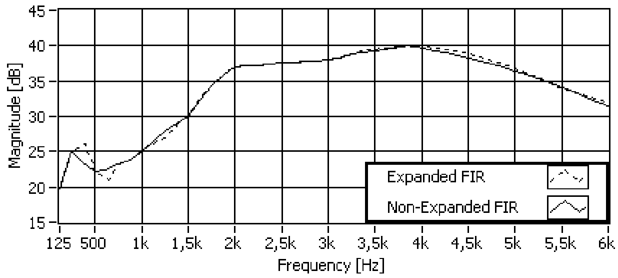

Figure 4. The audiogram approximation with linear interpolation FIR filter

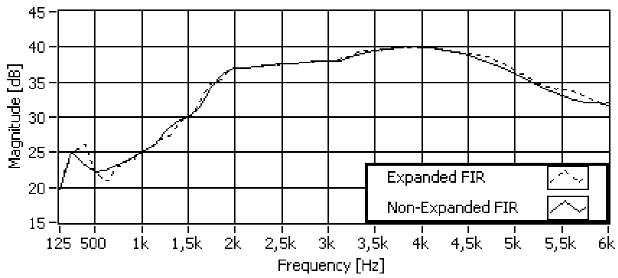

Figure 5. Audiogram approximation with cosine interpolation FIR filter

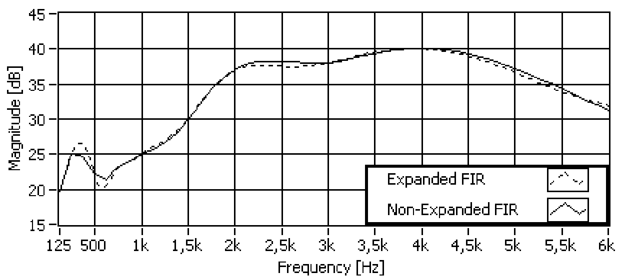

Figure 6. The audiogram approximation with Spline interpolation FIR filter

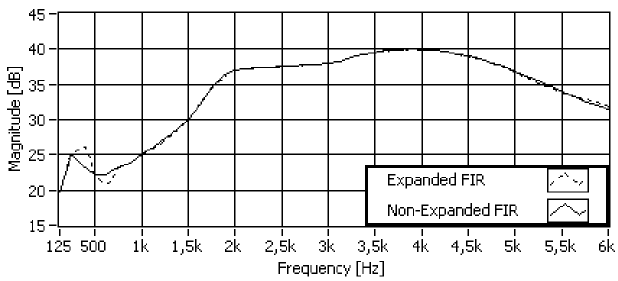

Figure 7. The audiogram approximation with Hermite interpolation FIR filter

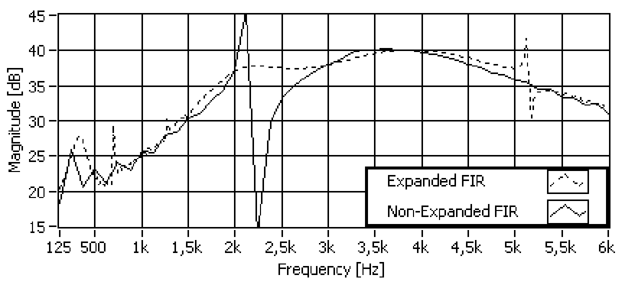

Figure 8. The audiogram approximation with rational

interpolation FIR filter

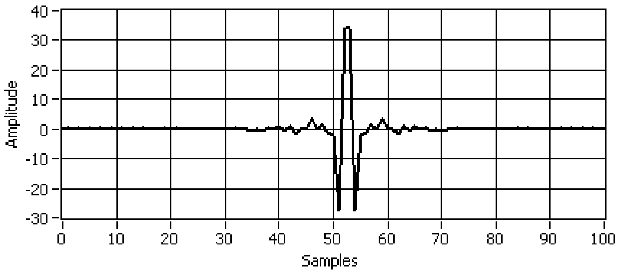

Figure 9. The impulse response of the linear interpolation FIR filter

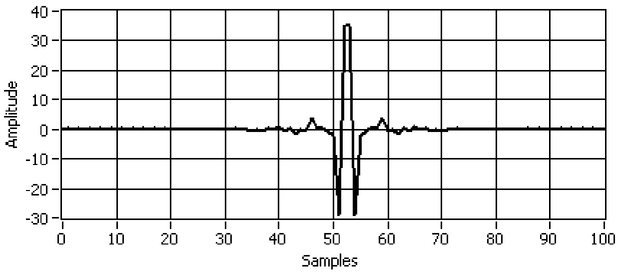

Figure 10. The impulse response of the Hermite interpolation FIR filter

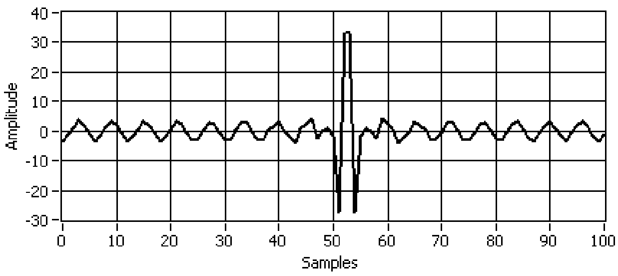

Figure 11. The impulse response of the rational interpolation FIR filter

VI. CONCLUSIONS

The aim of this paper is to model with a linear FIR filter the nonlinear human hearing mechanism using the auditory data provided by audiologists. The algorithm is rather simple and renders good results. First, an interpolation procedure is applied to the input data and then the obtained data is redistributed due to the unequal spacing between the frequencies. The uniform distance between the frequencies is given by the greatest common divisor of the samples' frequencies. In the last stage, an arbitrary magnitude FIR filter is designed to approximate the given audiogram.

The method was tested in LabVIEW on several audiograms. The efficiency of the method is closely linked to the interpolation method. The following interpolation types were tested: linear, cosine, Spline, Hermite, rational, polynomial and nearest interpolations. The obtained results prove that the best interpolation procedure is the Hermite; it renders smooth magnitude approximations, with low differences with respect to the interpolated curve, but needs rather complex computations. Acceptable results are obtained with the linear interpolation at a lower computational complexity.

Besides the interpolation type, another analyzed aspect was the influence of the samples number. The larger the number of samples is, the smoother the approximate magnitude plot is and fewer differences in comparison with the interpolated curve are. The Hermite interpolation is found to be the best being less sensitive to the number of samples. The increase of the FIR order is not acceptable because of the subjective determination of the audiogram. Furthermore, a high value of the FIR order is not suitable for hardware implementation.

Further work will focus on FIR length truncation and audiogram equalization for hardware implementation to enhance signal processing in hearing aids.

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REFERENCES

[1] M. K. Nalamwar, "Advanced Audiometer: A Novel Signal Generator Technique," *M. Tech. Credit Seminar Report, Electronic Systems Group,* Bombay, November 2004

[2] A. G. Katsiamis, E. M. Drakakis, R. F. Lyon, "A 4.5uW, 120+dB CMOS Analogue Cochlea Channel with AGC", *IEEE Journal of Solid-State Circuits*, vol. 44, 2009, pp. 1006 – 1022.

[3] E. A. Lopez-Poveda, R. Meddis, "A human nonlinear cochlear filterbank", *Acoustical Society of America Journal*, vol. 110, no. 6, 2001, pp. 3107-3118.

[4] J. Bondy, S. Becker, I. Bruce, L. Trainor, S Haykin, "A novel signal-processing strategy for hearing-aid design: neurocompensation", *Signal Processing*, *Elsevier North-Holland*, vol. 84, 2004, pp. 1239-1253.

[5] T. W. Parks, C. S. Burrus, *Digital Filter Design*, John Wiley & Sons Inc., 1987.

[6] N.D. Black, M. Lydon, N. Waterman, M. Powderly, "Programmable audiogram matching using a frequency sampling filter implemented on the Texas TMS 320C30", *Proceeding of IEEE International Conference Acoustics, Speech, and Signal Processing (ICASSP '92)*, 1992, vol. 2, pp. 225-228.

[7] H.G. McAllister, N.D. Black, N. Waterman, M. Li, "Audiogram matching using frequency sampling filters", *Proceedings of the 16th Annual International Conference of the IEEE Engineering in Medicine and Biology Society*, vol. 1, 1994

[8] A. Zaknich, G. E. Lee, "Arbitrary Audio FIR Filter Design by Bode Plot Smoothing using Tunable Approximate Piecewise Linear Regression", *Proceedings of Acoustics*, November 2006, New Zealand.