Enriched Pole Positioning Technique for Audio Applications

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Abstract— In audio signal processing parallel filters are highly efficient to reduce noise in the signal and also power consumption. In this proposed design parallel filter is designed in the following method: firstly the target response is smoothened, then warped IIR filters are designed to the smoothed target response which was divided into bands, and finally the united pole set is used to design the parallel filter. Warped FIR is not preferable because it is an all zero filter so poles cannot be selected to reduce the noise in audio signals, and also effective results could be achieved only at higher orders. Due to this computational cost and complexity increases. So by using warped IIR filter the problems of warped FIR can be reduced.

Key words: Warped IIR filters, logarithmic frequency resolution, smoothing

I. INTRODUCTION

One of the common tasks in audio signal processing is to design a digital filter for modelling or equalizing an audio system. The frequency resolution of the parallel filter is controlled by the suitable choice of the filter poles. While the use of traditional FIR and IIR design algorithms results in a linear frequency resolution, audio applications usually require a logarithmic (or logarithmic-like) frequency resolution that better fits the properties of human hearing. In order to obtain logarithmic frequency resolution we are going for warped digital filters. The resolution of warped filters with the Bark scale is very interesting, because it follows the resolution bandwidth of the human auditory system.

In the proposed design warped IIR digital filter are used. Since these filters are more advantageous when compared with warped FIR filters in audio signal processing. In general, if the filter order is in the same range as the system order, the filter poles should correspond to system poles for best accuracy. This is achieved by designing a warped IIR filter to the target response and using the poles of this warped IIR filter as the poles of the parallel filter.

II. LITERATURE SURVEY

G. Ramos, J. J. L´opez, and B. Pueo, designed a, “Cascaded warped-FIR and FIR filter structure for loudspeaker equalization with low computational cost requirements [1]”. In this design Cascaded structure of a finite impulse response (FIR) filter and a warped-FIR filter are used in order to obtain the best performance. In the task of loudspeaker equalization, FIR filters achieve excellent resolution and equalization at high frequencies, but at low frequencies the resolution obtained is too poor when evaluated in logarithmic frequency axis, that could only be improved using high order filters. To solve this lack of resolution at low frequencies, warped-FIR filters have been employed, but at the expense of decreasing the resolution of the filter at high frequencies and increasing the complexity of the filter structure and its computational cost. The proposed combination of both types of filters, combined with the correct selection of their orders, and the λ value for the warped-FIR filter, allows the FIR filter to maintain its good resolution at high frequencies and achieve enough resolution at low frequencies with the warped-FIR filter. In this way, lower order filters with lower computational cost could be obtained than when using FIR or warped-FIR only. This approximation attains a more uniform resolution of the filter when evaluated in octaves, behaving much more like human hearing, than the linear frequency resolution obtained when employing only FIR filters.

III. POLE POSITIONING TECHNIQUE USING WARPED IIR FILTER

In the proposed design firstly, the target response is smoothened, then warped IIR filters are designed to the smoothed target response which was divided into bands, and finally the united pole set is used to design the parallel filter.

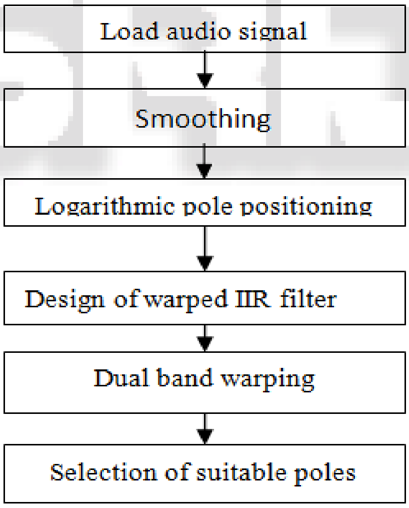

Fig. 1: Process Sequence

First load the input audio signal which is in time domain. Then convert this time domain signal into frequency domain by using FFT .During this conversion some leakages may occur which leads to signal energy smearing over a wide range in FFT when it should be in narrow frequency range. So in order to avoid the leakage windowing technique is used. Now the obtained frequency domain signal is smoothened by using smoothening technique.

A. *Smoothing:*

Smoothing is nothing but equalization. In smoothing, the data points of a signal are modified so that individual points that are higher than the immediately adjacent points (presumably because of noise) are reduced, and points that are lower than the adjacent points are increased. This naturally leads to a smoother signal [3].

B. Logarithmic Frequency Resolution:

The smoothed signal is divided into bands and represented in logarithmic scale. Audio applications usually require a logarithmic frequency resolution that better fits the properties of human hearing. The logarithmic frequency resolution is more demanding than linear frequency resolution because the frequency is unevenly distributed which is suitable for plotting high range of frequencies in an audio signal [4].

C. Design of Warped IIR Filter:

We are designing warped IIR filters on logarithmic frequency scale. Warping is process in which unit delays of traditional IIR and FIR filters are replaced by frequency dependent delays.

$$z^{-1} \leftarrow D(z) = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}}.$$

The smoothed-flattened target responses are used as the specification for warped IIR filter design. In this design warping parameters λ = 0.986 and λ = 0.65 for the low- and high-frequency parts, respectively are used. These were chosen so that the warped filters have the maximal logarithmic resolution (minimal Δf /f) in the middle of their respective bands by finding such λ values where the minimum of

$$\frac{\Delta f}{f} = \frac{1 + \lambda^2 - 2\lambda\cos(2\pi f / f_s)}{(1 - \lambda^2) f}$$

is at f = 100 Hz for the low-frequency band and f = 3160 Hz for the high-frequency band. the transfer function of warped IIR filter is[2]

$$H_{\mathrm{WIIR}}(z) = \frac{\sum_{i=0}^{M} \beta_i [D(z)]^i}{1 + \sum_{i=1}^{R} \alpha_i [D(z)]^i}$$

In this proposed design smoothened target response is divided into dual bands.

D. Pole Positioning:

The frequency resolution of the parallel filter is controlled by the suitable choice of the filter poles. In general poles of warped IIR filters are found by using below equation

$$p_k = \frac{\widetilde{p_k} + \lambda}{1 + \lambda \widetilde{p_k}}$$

Finally suitable poles are selected at the point where the noise is reduced effectively that is taking the minimum difference obtained between output signal and input audio signal.

The poles obtained at different bands are united and are used to design a parallel filter.

IV. EXPERIMENTAL RESULTS

By increasing the order of the filter the effective results in terms of equalization can be achieved with high order parallel filters. But they require high computational cost, in order to be implemented in real-time. The experimental results of the proposed design are

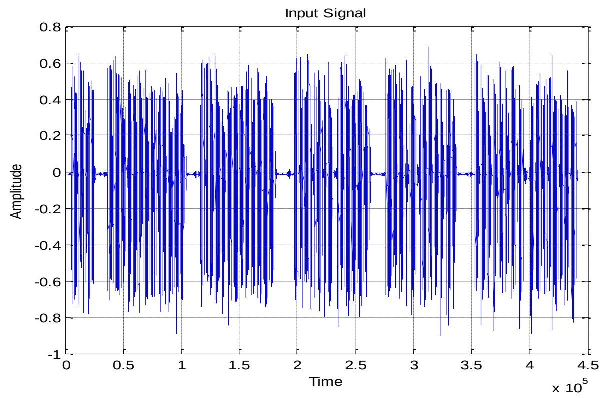

Fig. 2: Audio signal

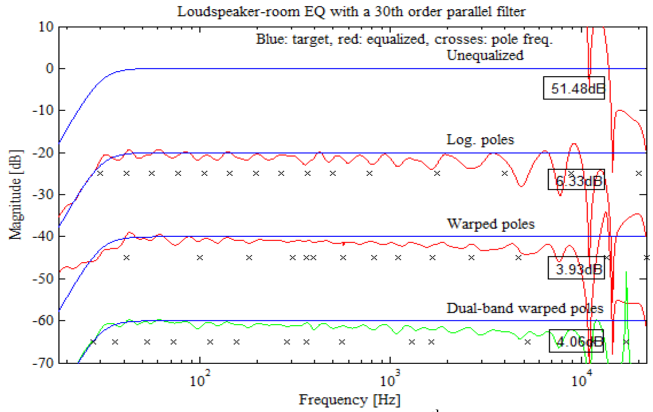

Fig. 3: Loudspeaker room EQ with a 30^{th} order parallel filter

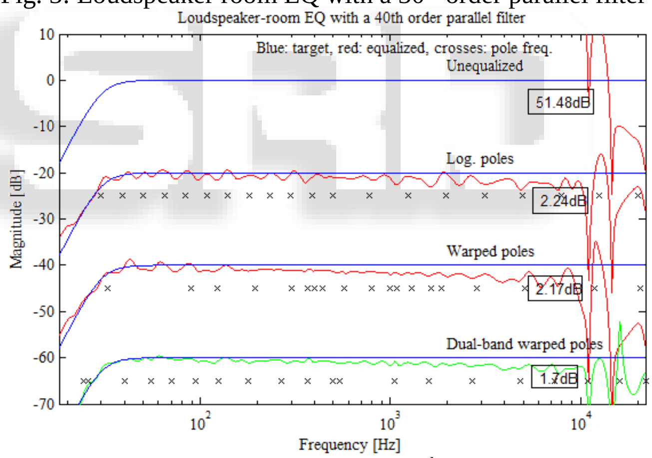

Fig. 4: Loudspeaker room EQ with a 40^{th} order parallel filter

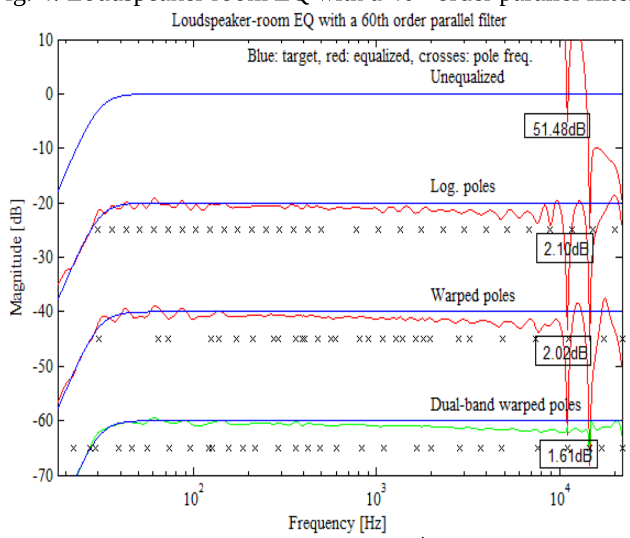

Fig. 5: Loudspeaker room EQ with a 60^{th} order parallel filter

A. *Comparisons of Noise:*

Technique	Low frequency(dB)	Mid frequency (dB)	High frequency(dB)
Un equalized	24.58	37.49	51.48
Log poles	1.19	1.68	2.24
Warped poles	1.16	0.92	2.17
Dual band warped poles	0.26	0.32	1.7

Fig. 6: Comparisons of Noise

V. CONCLUSION

This proposed design has presented an improved pole positioning method using warped IIR filter design for audio equalization. By using this design the noise in the original signal that is 51.48dB was reduced to 1.7dB.

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