Bark Scale Equalizer Design Using Warped Filter

Wang Peng, Wee Ser, Ming Zhang

Center for Signal Processing
School of Electrical and Electronic Engineering
Nanyang Technological University
Singapore 639798

ABSTRACT

Bark scale (also called critical band rate in literature) has attracted increasing attention among audio engineers as a good measurement of the frequency resolving ability of human auditory system. In this paper, warped filter technique has been applied on Bark scale to construct an equalizer for a low-cost loudspeaker system. The resulted equalizer displays overall better equalization effect than the conventional deconvolution-based equalizer in the conducted simulation.

1. INTRODUCTION

The concept of critical band was first introduced by Fletcher in [1] to deal with the masking of a narrowband stimulus in wideband noise. Critical bands can be simply interpreted as a series of bandpass filters located in the auditory system [2]. Bark scale, which is also called critical band rate in literature, has been defined such that one Bark is right equal to the bandwidth of a critical band. Bark scale is very popular among audio engineers for its ability to represent the frequency resolving power of human auditory system [3, 4].

Loudspeakers are the most nonideal components in audio systems. The magnitude distortion introduced by a low-cost loudspeaker can dramatically spoil the output audio. Loudspeaker equalization is a technique that has been developed to overcome this problem. As is demonstrated by psychoacoustical experiments [2], human ears are more sensitive to low frequency distortion. However, in conventional deconvolution-based equalization scheme [5], a uniform degree of equalization is provided for all frequency components. It is not able to provide sufficient equalization for low frequency distortion when being implemented with limited taps. Comparatively, warped filter scheme can provide a variant degree of equalization with a single filter, and thus has the possibility to distribute equalization according to the characteristics of human auditory system.

Warped filter equalization has been studied in literature as a good substitute for the deconvolution-based scheme [6-10]. In this paper, a Bark scale equalizer design using warped filter is presented. It tries to maximize the bandwidth where the frequency resolving ability of the proposed equalizer exceeds that of human auditory system by a prescribed factor. The maximization is achieved by optimizing the warped filter using a method similar to that appeared in [9]. Computer simulation has been conducted and the result shows the efficiency of the proposed scheme.

2. PRINCIPLE OF WARPED FILTER EQUALIZATION

In conventional equalization schemes, the measured impulse response of the target loudspeaker is used directly to obtain the coefficient vector of the equalizer. E.g., in deconvolution-based scheme, the coefficient vector is derived by the deconvolution of the measured impulse response with the ideal impulse response. While in warped filter scheme, the warped representation of the measured response is used to derive the equalizer that works on warped frequency domain.

Through warping process, the input distorted signal where enhanced equalization is needed will be expanded, at the expense of other parts of the signal being compressed. Any conventional equalization method, e.g., deconolution-based method, can be applied on the warped distorted signal to produce a warped equalization effect. The generated equalized warped signal need to be unwarped to linear frequency domain.

Warping and unwarping are achieved by the use of a special filter structure, warped filter. To construct a warped filter, the unit delays in a normal FIR or IIR filter are replaced with first-order all-pass function, $D_1(z)$, and

$$\mathrm{D}_1(z) = z'^{-1} = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}} \qquad (1)$$

where λ is a warping parameter ranging between -1 and 1, $z' = e^{j\omega'}$, $z = e^{j\omega}$, ω' and ω are respectively the warped circular frequency and the linear circular frequency. Fig. 1 shows a warped FIR structure.

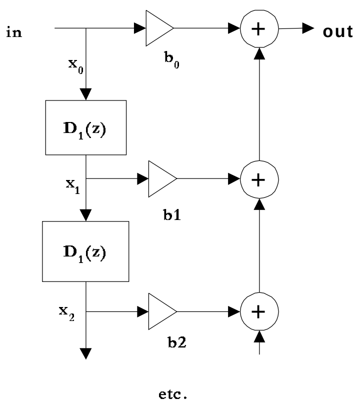

Fig. 1. Warped FIR structure

Positive values of λ enhance the resolution (equalization) at low frequencies while negative values of λ enhance the resolution at high frequencies.

Detailed introductions of warped filter scheme can be found in [6-9]. In [6], a thorough comparison with other schemes was presented; In [8], this scheme was combined with deconvolution-based equalization to construct a dual-band equalizer; In [9], a warping optimization method was introduced, and a technique that enables warped filter equalization effective for middle frequency distortion was also presented.

3. BARK SCALE BASED WARPING SCHEME

Warping optimization is equivalent to the searching for the optimum λ value. In [9], it has been carried out by maximizing a function, WWEA (Weighted Warping Effective Area). WWEA is defined as

$$\mathrm{WWEA}(\lambda) = \int \mathrm{W}(\omega, \mathrm{F}) \times \mathrm{F}(\omega, \lambda)\, \mathrm{d}\,\omega \qquad (2)$$

where $\mathrm{W}(\omega, \mathrm{F})$ is a weighting function, and $\mathrm{F}(\omega, \lambda)$ is termed as warping effect function

$$\mathrm{F}(\omega, \lambda) = \frac{\partial \omega'}{\partial \omega} \qquad (3)$$

Bark scale based warping can be achieved by choosing a suitable weighting function that takes into account psychoacoustic characteristics. The weighting function used in this study is

$$\mathrm{W}(\omega, \mathrm{F}) = \begin{cases} \mathrm{F}(\omega, \lambda)^{-a} & , \text{ when } \mathrm{F}(\omega, \lambda) > \mathrm{F_B} \\ 0 & , \text{ when } \mathrm{F}(\omega, \lambda) < \mathrm{F_B} \end{cases} \qquad (4)$$

where a is a weighting parameter, $\mathrm{F_B}$ is given by

$$\mathrm{F_B} = k \times \left(\frac{\pi}{\mathrm{T_r}(f_s / 2)} \times \frac{f_s}{2\pi} \times \frac{\mathrm{d}}{\mathrm{d}\, f} \mathrm{T_r}(f_\mathrm{B}) \right) \qquad (5)$$

In this expression, f_s is the sampling frequency, k is another optimization parameter to adjust the optimization, $\mathrm{T_r}(f)$ is the mapping function from Hertz to Bark, $\frac{\mathrm{d}}{\mathrm{d}f}\mathrm{T_r}(f)$ can be regarded as a measure of the frequency resolving ability of the human auditory system, and f_B is the frequency value that satisfies

$$\mathrm{F}\left(\frac{2\pi}{f_s} \times f, \lambda\right) = k \times \left(\frac{\pi}{\mathrm{T_r}(f_s / 2)} \times \frac{f_s}{2\pi} \times \frac{\mathrm{d}}{\mathrm{d}\, f} \mathrm{T_r}(f) \right) \qquad (6)$$

Clearly, f_B is a function of λ. The factors inside the outermost parentheses are used to unify the scales of $\mathrm{F}(\omega, \lambda)$ and $\frac{\mathrm{d}}{\mathrm{d}f}\mathrm{T_r}(f)$.

When parameter a is set to 1 in Eq. (4), the maximization of WWEA is equivalent to the maximization of f_B. The resulted warped filter has the maximum bandwidth where the frequency resolving ability exceeds that of human auditory system by a factor of k . Such an excessive frequency resolving ability is valuable because of two reasons. First, critical bands only depict part of the characteristics of human auditory system. The ‘excessive’ frequency resolution is probably useful for some veiled functions. Secondly, an exact expression of $\mathrm{T_r}(f)$ is not available at present.

4. SIMULATION RESULTS

In order to test the effectiveness of the proposed scheme, computer simulation has been conducted on a low-cost desktop loudspeaker.

The mapping function from Hertz to Bark used in this simulation is as follows.

$$\mathrm{T_r}(f) = \begin{cases} \dfrac{f}{100} & \text{when } f < 421.6 \text{ Hz} \\ \dfrac{26.81}{1 + (1960 / f)} - 0.53 & \text{when } f > 421.6 \text{ Hz} \end{cases} \qquad (7)$$

At frequencies above 421.6 Hz, Traunmüller’s formula [10] is applied to approximate the transform. While at frequencies below 421.6 Hz, linear transform is assumed. The value of 421.6 is calculated by equating $\frac{f}{100}$ with $\frac{26.81}{1+(1960/f)} - 0.53$. The resulted mapping curve is shown in Fig. 2, where 'x' denotes rounded Barks obtained from empirical observations.

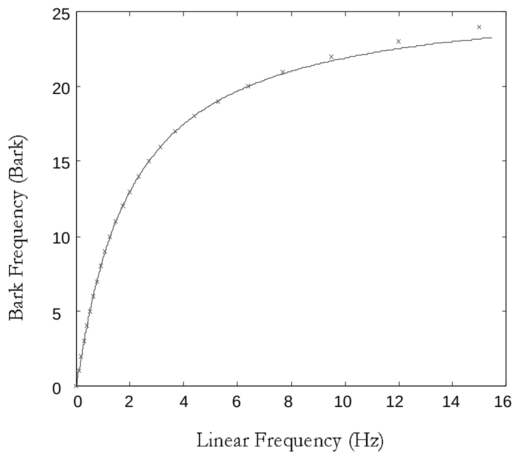

Fig. 2. Mapping from Hertz to Bark

Weighting parameters *a* and *k* are both set to 1 in this design, which implies the resulted optimal warped filter has maximum bandwidth with 'excessive' frequency resolving ability. The optimum λ can be obtained by iterative searching for the maximum f_B.. The bandwidth with excessive frequency resolving ability, abbreviated as ERBW, is depicted as a function of λ in Fig. 3, where the sampling rate f_s is 44.1 kHz. The optimum value of λ at this sampling rate corresponding to the peak value of ERBW as shown in Fig. 3 is 0.7405.

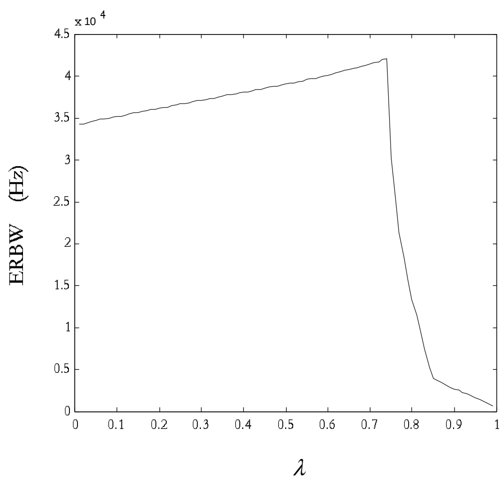

Fig. 3. ERBW shown as a function of λ

The optimum value of λ changes with regard to the sampling rate. This is illustrated in Fig. 4. The optimum λ calculated with the following equation [3] is also displayed for comparison.

$$\lambda_{opt} = 0.8517 \times \sqrt{\arctan(0.06583 \times F_s)} - 0.1916 \qquad (8)$$

where F_s denotes sampling rate in kHz. λ_{opt} is optimum in that the warping function (mapping from linear frequency to warped frequency) best approximates the mapping from linear frequency to Bark frequency at a given sampling rate.

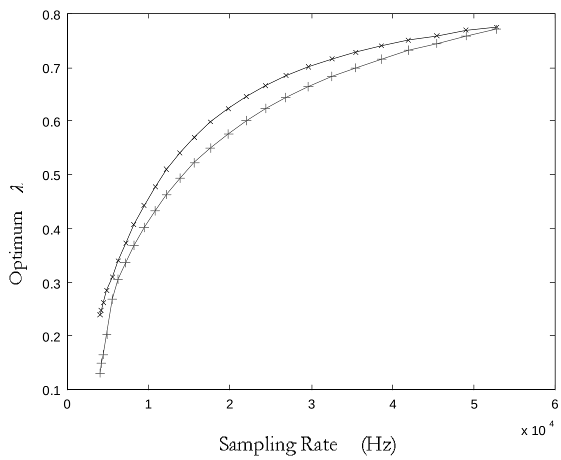

Fig. 4. Optimum λ shown as a function of sampling rate
-----+-----: Optimum values that can maximize ERBW
-----x-----: Optimum values obtained with Eq. (8)

The deviation of the two curves shown in the above figure is due to the employment of different optimization criteria. By selecting a suitable form of weighting function instead of the one defined in Eq. (4), optimum λ values obtained from Eq. (8) can be approximated by maximizing WWEA given in Eq. (2). In this sense, the Bark scale warping optimization scheme presented in this paper can be regarded as an expansion of that introduced in [3].

Shown in Fig. 5a is the measured frequency response of the target loudspeaker. A warped filter equalizer has been constructed in the simulation, where λ is set to 0.7405 to maximize ERBW at the given sampling rate of 44.1 kHz. Fig. 5c is the frequency response that has been equalized by this equalizer. The frequency response equalized by a deconvolution-based equalizer with the same number of taps is also shown in the figure, and it is Fig. 5b. Compared with the deconvolution-based equalizer, the proposed Bark scale equalizer displays better capability to compensate for low frequency (0~20 Bark) distortion. However, the tradeoff is the loss of high frequency compensation capability.

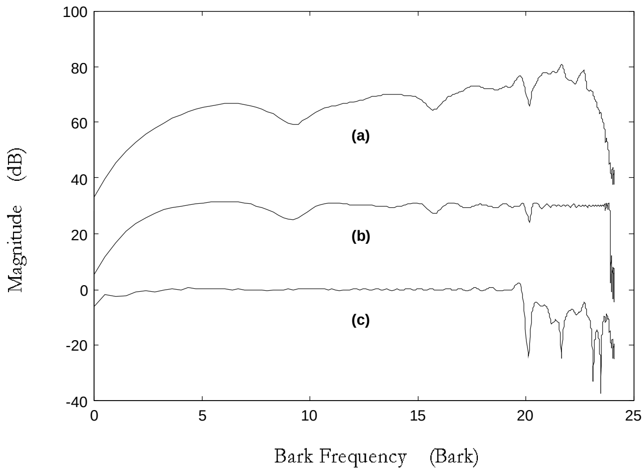

Fig. 5. Representations on Bark scale of the frequency response: (a) of the target loudspeaker (b) equalized by deconvolution-based method (c) equalized by warped filter method (λ = 0.7405)

5. CONCLUSIONS

In this paper, a Bark scale loudspeaker equalization scheme is proposed. Warped filter is applied to construct an equalizer that can provide different equalization effect at different frequencies according to the frequency resolving ability of human auditory system. Computer simulation has been conducted and the result shows the effectiveness of the proposed scheme.

The proposed scheme can be applied in practical audio system design, especially where the implementation cost is a crucial consideration and the energy of the target signal concentrates at relatively low frequencies. This scheme can also be combined with other equalization schemes to construct a multiband equalizer. The resulted equalizer is expected to be able to provide good equalization effect on the whole frequency band.

Bark scale signal processing is a prevalent working tool in audio engineering. The method introduced in this paper has the potential to be used to solve other problems.

6. REFERENCES

[1] H. Fletcher, “Auditory patterns”, *Revs Modern Physics*, 12, pp. 47-65, 1940.

[2] J. O. Pickles, *An Introduction to the Physiology of Hearing*, Academic Press Inc. (London) Ltd., 1982.

[3] J. O. Smith and J. S. Abel, “The Bark and ERB bilinear transforms”, *IEEE Trans. Speech and Audio Processing*, vol. 76, pp. 697-708, Nov. 1999.

[4] T. Sporer and K. Brandenburg, “Constraints of filter banks used for perceptual measurement”, *J. Audio Eng. Soc.*, vol. 43, no. 3, pp.107-116, Mar. 1995.

[5] J. Mourjopoulos, “Digital equalization methods for audio systems”, presented at the 84th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (Abstracts), vol. 36, p.384 (1988 May), preprint 2598.

[6] M. Karjalainen, E. Piirilä, A. Järvinen and J. Huopaniemi, “Comparison of loudspeaker equalization methods based on DSP techniques”, *J. Audio Eng. Soc.*, vol. 47, no.1/2, pp.14-30, Jan./Feb. 1999.

[7] M. Karjalainen, E. Piirilä and A. Järvinene, “Loudspeaker response equalization using warped digital filters”, in *Proc. NorSig-96*, (Espoo, Finland), pp.367-370, Sept. 1996.

[8] P. Wang, W. Ser and M. Zhang, “A dual-band equalizer for loudspeakers”, *J. Audio Eng. Soc.*, vol. 48, no. 10, pp. 917-921, Oct. 2000.

[9] P. Wang, W. Ser and M. Zhang, “Multiband warped filter equalizer design for loudspeaker systems”, in *Proc. ICASSP 2000* (Istanbul, Turkey), pp. 913-916, June 2000.

[10] H. Traunmüller, “Analytical Expressions for the Tonotopic Sensory Scale”, *J. Acoust. Soc. Am.* 88, 1990, pp. 97-100.