Filters for Audio Equalizing and Artificial Reverberation

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1 Derivation of Linear-Phase FIR Filter for Audio Equalizer

A full derivation on the design of linear-phase FIR filters is given in [1]. Since we have chosen to shape our equalizer by the desired frequency response in terms of Hz. We will index the desired response $H_r(\delta_f k)$ in which δ_f is the frequency sampling resolution.

$$\delta_f = \frac{Fs}{N} \tag{1}$$

Fs is the sampling frequency in Hz, and N is the number of taps or samples used to implement the filter. We chose to use a symmetric linear-phase FIR filter. We highlight the key equations from [1] (see Table 8.3), with our desired frequency response. We chose to use an $\alpha = 0$, and an N which was even. The resulting desired linear-phase FIR filter response $h_d(n)$ is shown in (3).

$$G(k) = (-1)^k H_r(\delta_f k), k = 0, 1, ..., N-1 \tag{2}$$

$$h_d(n) = h_d(N-1-n) = \frac{1}{N}\{G(0) + 2\sum_{k=1}^{\frac{N}{2}-1} G(k)\cos\frac{2\pi k}{N}(n+\frac{1}{2})\} \tag{3}$$

Lastly we chose to use a Hamming window, $w(n)$ (4), for sampling the audio data in order to minimize ripples in the resulting frequency response of our FIR filter.

$$w(n) = 0.54 - 0.46\cos(\frac{2\pi n}{N-1}) \tag{4}$$

Thus, the resulting equalizer FIR linear-phase filter impulse response, $h(n)$, is given in (5).

$$h(n) = w(n)h_d(n) \tag{5}$$

2 Derivation of Artificial Reverberation Filter

The artificial reverberation (AR) filter has the following impulse response as shown by [2]. Specifically, the impulse response is given in (6).

$$h_r(n) = h_{fir}(n) + h_{iir}(n - n_d) + \delta(n) \tag{6}$$

The impulse response of the FIR filter $h_{fir}(n)$ has the following form.

$$h_{fir}(n) = \begin{cases} 10^{\frac{-60n}{20Fs(RT+0.001)}}, & \text{if } n = kfloor(\frac{Fs}{RD}); k = 0, 1, ..., k_{max} \\ 0, & \text{otherwise.} \end{cases} \tag{7}$$

The limit for k_{max} is determined by n_d, which is dependent on the IIR time delay ($T_{iir} = 0.160$ seconds) and the desired reverberation time, RT.

$$k_{max} = min(n_d, floor(RTFs)), n_d = floor(T_{iir}Fs) \tag{8}$$

For the IIR impulse response, all that is required is a scaled, first-order IIR filter with the following z-transform.

$$H(z) = \frac{g_1}{1 + a_1 z^{-1}} \tag{9}$$

The resulting impulse response is given in (10).

$$h_{iir}(n) = \begin{cases} g_1(a_1)^n, & n >= 0 \\ 0, & \text{otherwise.} \end{cases} \tag{10}$$

Lastly, the IIR coefficients, g_1 and a_1, are determined by (11), and (12) respectively.

$$g_1 = 10^{\frac{G_1}{20}}, G_1 = \frac{-60T_{iir}}{RT + .001} \tag{11}$$

$$a_1 = \begin{cases} 10^{A_1}, A_1 = \frac{-3-\frac{G_1}{20}}{max(0.001, RT - T_{iir})Fs}, & \text{if } a_1 <= 1 - 2^{-30} \\ 1 - 2^{-30}, & \text{otherwise.} \end{cases} \tag{12}$$

the bound on a_1 is to keep the IIR filter stable. Since we were required to vary RT from 0 to 5.0 seconds, the $.001$ term, was used to avoid a divide by 0 error in this implementation.

References

[1] John G. Proakis, Dimitris G. Manolakis, *Digital Signal Processing Principles, Algorithms, and Applications*. Prentice-Hall, Inc., 1996. (pp. 630–633)

[2] Robert L. Stevenson, EE598D – Project #1