ASPECTS OF INVERSE FILTERING FOR LOUDSPEAKERS

Scott G. Norcross
Advanced Audio Systems, Communications Research Centre, 3701 Carling Ave., P.O. Box 11490 Station H, Ottawa, Ontario, K2H 8S2 e-mail: scott.norcross@crc.ca

1. INTRODUCTION

Inverse filtering has been proposed for numerous applications in audio and telecommunications such as loudspeaker equalization and room deconvolution. Its attraction is that it can potentially "undo" a system and correct both the magnitude and phase responses [1][2]. Two methods are compared in this paper: a time-domain least-squares approach and a frequency-domain deconvolution method. Formal subjective tests conducted in previous studies [5][6][7] have shown that the inverse filter can actually degrade the audio quality by producing artifacts or distortions. These artifacts, which include pre-echo and timbre changes, can in some cases result in subjectively poorer performance than if no inverse filter was used. The severity of the degradation depends on both the inversion method and the system that is being corrected. This paper will review the results of the formal subjective tests evaluating the two methods as well as some strategies, such as regularization and smoothing, to remove or control the severity of audible artifacts.

2. INVERSE FILTERING

The inverse filtering process is based around the concept of linear filtering. Assuming that a flat frequency response is desired, we have the following equation to solve

$$\delta(n) = c(n) \otimes h(n) \qquad (1)$$

where $\delta(n)$ is the Kronecker delta function or unit impulse function and *c(n)* is the impulse response (IR) and *h(n)* is the inverse filter.

A least-squares time-domain approach can be expressed in matrix form, as in [2] and a solution for the inverse filter *h(n)* is given by,

$$h(n) = (\mathbf{C}^T\mathbf{C})^{-1} \bullet \mathbf{C}^T \mathbf{a}_m \qquad (2)$$

where $\mathbf{C}$ is the convolution matrix of *c(n)* and $\mathbf{a}_m$ is a modeling delay vector.

Although fast algorithms exist for calculating the time-domain solution given by Equation (2), a fast frequency-domain deconvolution method can be used to speed up this computation. This approach is based on the fact that a time-domain convolution becomes a multiplication in the frequency domain and can be written as

$$H(k) = \frac{D(k)}{C(k)} \qquad (3)$$

where *H(k)* and *C(k)* are the discrete Fourier transform (DFT) of *h(n)* and *c(n)* respectively, and *D(k)* is the DFT of the desired signal, usually a delayed pulse.

A potential problem with inverse filtering is when the denominator in equation (3) is zero or very small. This will result in *H(k)* having an excessive boost. Kirkeby et al. [2] and Craven et al.[3] have used regularization to limit this effect. Kirkeby's implementation can be expressed as,

$$H(k) = \frac{D(k)C^*(k)}{C(k)C^*(k) + \beta B(k)B^*(k)} \qquad (4)$$

where *B(k)* is the regularization term and β is a scaling factor controlling the amount of regularization.

Another approach to overcome the problem of small values of *C(k)* is the use of smoothing. Hatziantonious and Mourjopoulos [4], have suggested complex smoothing given as,

$$C_{cs}(k) = \sum_{i=0}^{N-1} C((k-i) \bmod N) \bullet W_{sm}(m,i) \qquad (5)$$

where W_{sm} is the smoothing window.

3. MEASUREMENTS AND SUBJECTIVE TESTS

The IRs from two different loudspeakers were measured on- and off-axis (45°) in an anechoic environment. The IRs were sampled at 44.1kHz and were 1024 samples in length. Test files were created from a mono recording of a castanets passage. The test strategy was to filter the audio source signal with one of the measured IRs and then process these filtered audio signals with an inverse filter to correct for the loudspeaker's response. Ideally, if the inverse filter were perfect, the results of this process should yield an audio file identical to the original audio source file.

To evaluate the subjective performance of the different inverse filtering methods (correction methods), double-blind subjective tests were conducted using the MUSHRA method (Recommendation ITU-R 1534) test over headphones.

4. RESULTS

Figure 1 shows mean subjective grades for the five loudspeaker conditions versus correction method. Six frequency-domain inverse filters of increasing length (2k to 64k) are shown along with a 2k time-domain inverse filter. Also included is the case where no correction (labeled 'Filter') was made to the filtered audio signal.

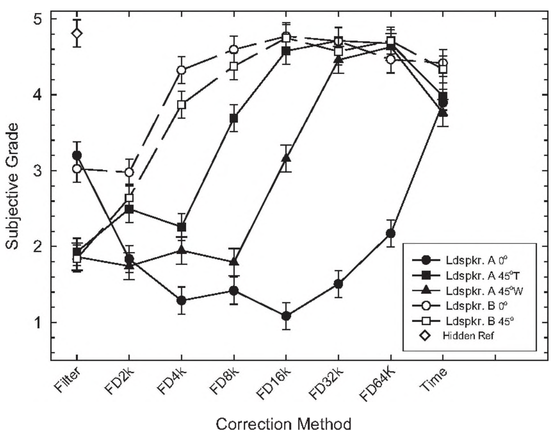

Figure 1. Test results for the five IRs showing the mean subjective grade versus correction method. 'FD' refers to the frequency-domain method and *X*k refers to the length of the inverse filter. Also included are the uncorrected condition ('Filter') and the time-domain inverse (2k in length) labeled with 'Time'.

It can be seen that the time-domain inverse filters scored higher than the frequency-domain filters of similar lengths. The time-domain filters also scorred very similarly for all five loudspeakers whereas with the frequency-domain method there was a larger spread between them. Most of the frequency-domain filters subjective performance increased when the length was increased, except for one. This might indicate the presence of time-aliasing with the frequency-domain deconvolution method.

Three forms of regularization, one frequency-independent and two frequency-dependent, were used to try to improve the subjective performance of the frequency-domain method. One frequency-dependent regularization method consisted of regularizing the extreme low and high frequencies with a value of 1 (0 elsewhere), and the other method was based on the 1/3 octave spectrum of the impulse response. All the regularization methods did improve the performance over the case with no regularization, but the amount required to achieve the performance increase was dependent of the IR.

Third-octave complex smoothing of the IR prior to calculating the inverse was also explored and showed some improvement in the subjective performance. The smoothing did not tend to broaden the corrected IRs as was the case when regularization was added.

5. SUMMARY

Two methods for calculating the inverse filter of a loudspeaker's IR were examined. Formal subjective tests have shown that the inverse filter can produce audible artifacts. The frequency-domain method is not as robust as the time-domain method. Regularization and complex smoothing can help improve the subjective performance of the inverse filter. The amount of regularization required is dependent on the IR that is being inverted, thus requiring hand-tuning to optimize the subjective performance. Complex smoothing of the impulse response with a third-octave smoother can also be used to improve the subjective performance of the inverse filter. Complex smoothing did not broaden the main pulse of the corrected IR as the regularization tended to do.

6. REFERENCES

[1] Mourjopoulos, John, N. (1994). "Digital Equalization of Room Acoustics", *J. Audio Eng.*, vol. 42, pp. 884-900.

[2] Kirkeby, Ole and Nelson, Philip A., "Digital Filter Design for Inverseion Problems in Sound Reproduction", *J. Audio Eng.*, vol. 447, pp. 583-595.

[3] Craven, Peter G. and Gerzon, Michael A., "Practical Adaptive Room and Loudspeaker Equaliser for Hi-Fi Use", AES UK DSP Conference (Sept 1992).

[4] Hatziantoniou, Pantagiotis D., Mourjopoulos, John N.,"Generalized Fractional-Octave Smoothing of Audio and Acoustic Responses", *J. Audio Eng. Soc.*, vol. 48, pp. 259-280 (April 2000).

[5] S. G. Norcross, G. A. Soulodre and M. C. Lavoie, "Evaluation of Inverse Filtering Techniques for Room/Speaker Equalization", 113th AES Convention, LA, 2002, Preprint 5662

[6] S. G. Norcross, G. A. Soulodre and M. C. Lavoie, "Subjective Effects of Regularization on Inverse Filtering", 114th AES Convention, Amsterdam, 2003, Preprint

[7] S. G. Norcross, G. A. Soulodre and M. C. Lavoie, "Further Investigations of Inverse Filtering", 115th AES Convention, New York, 2003, Preprint

[8] Louis D. Fielder, "Analysis of Traditional and Reverberation-Reducing Methods of Room Equalization", *J. Audio Eng. Soc.*, vol. 51, pp.3-26 (January/February 2003).