Trevor de Clercq
February 26th, 1999

Digital Audio Processing II
E85.2601, Prof. K. Peacock

Article Summary:

"Comparison of Loudspeaker Equalization Methods Based on DSP Techniques"
by Matti Karjalainen, Esa Piirila, Antti Jarvinen, and Jyri Huopaniemi
(<u>Journal of the Audio Engineering Society</u>, Jan/Feb 1999, pgs. 14-31)

In this article, the authors explored different problems and methods associated with the equalization of loudspeakers using digital signal processing. Digital filtering is used to make loudspeakers' performance closer to a flat frequency response with respect to decibels. A new DSP principle in particular, that of warped filters, is analyzed specifically as it relates to loudspeaker equalization. The study compares different filtering methods on the grounds of equalization error, both from an objective viewpoint (through graphs resulting from the use of a spectrum analyzer) and a subjective viewpoint (through listening tests). Only the linear EQ response of speakers is examined, without consideration for room acoustics or other signal flow elements.

The new filtering method that is introduced with respect to loudspeaker EQ is that of the warped filter. Warped filters are basically new models of the standard FIR (finite impulse response) and IIR (infinite impulse response) designs. Without going into mathematical detail, warped filters (both WFIR and WIIR) differ from their non-warped counterparts by affecting frequency response in a manner that more closely resembles auditory perception (such as logarithmic) as opposed to the purely linear response of FIR and IIR filters. The main benefit of warped filters is that they require a lower order (degree of complexity) to compare with the response of traditional filters; lower order and thus less complexity equals the ability to digitally process such information more quickly and efficiently.

The results of the experiments proved indeed that warped filters offer better equalization response with a lower order of complexity. An FIR filter of order 1000, for example was needed to completely flatten the on-axis response of a typical home speaker, as compared to a WFIR filter of only order 300 and an WIIR filter of only order 80. Morevover, with respect to bass and mid-range, the warped filters more quickly began to level out the frequency response as compared to the traditional FIR filter (IIR filters were not considered in these experiments due to their need for high computing power and overall inferior frequency response). When examining off-axis reponses, the high-frequency detail that was incurred by the FIR filter dropped off significantly; in this case, therefore, the clarity of the bass and mid-range response of the WIIR filter was to its advantage.

The listening tests showed a different side of the story. Listeners reported that FIR filters with an order of 80 were sufficient to be unable to appreciate the Just Noticeable Difference (JSD) between higher orders. WFIR filters scored at around 75 and WIIRs at 35. The substantial closeness of filter orders in these listening tests argue for the FIR filter since it inherently requires less computing power and an order-to-order basis. The authors suggest, however, that the listening tests might be flawed, offering their listening test's design (based on spectral distance measure) or the design of the filters themselves as a possible blame for the unexpected results.

From an informed skeptic's point of view, I can say two things: 1) that listening tests are the definitive conclusion to audio research and that therefore these studies need to be conducted with a wider variety of participants (only seven were used) and/or the data needs to be reanalyzed to match the listening results; and 2) that hopefully these experiments are only the beginning for the study of DSP based loudspeaker equalization; the authors themselves admit to not having tackled the true test of analying loudspeaker response with respect to the acoustic environment of a room or any other part of the signal path.