Previous scientific approaches to loudspeaker and room equalisation

For several years now electronic equalisation has been a topic occupying scientists, and while the breakthroughs in room acoustics (and to some extent also psychoacoustics) were made decades ago, recent developments in signal processing techniques and digital hardware now constitute the basis for realtime implementation of equalisation systems. The main areas treated in scientific papers the past fifteen years relating to equalisation are listed below.

- Correcting for bad room acoustics
- Improving sound quality in cars
- Developing robust inverse filtering techniques
- Developing adaptive equalisation systems
- Modelling the acoustic properties of rooms
- Realising simple and robust realtime equalisation systems
- Creating virtual acoustics, e.g. by phantom sources
Below, the content of ten essential papers on room equalisation are presented as the original abstracts in publishing order followed by discussions.

**Matti Karjalainen et al.:**

*Comparison of Loudspeaker Equalization Methods Based on DSP Techniques, J. Audio Eng. Soc., 47(1/2), 1999 January/February*

Methods of loudspeaker response equalization using digital filters are compared. In addition to generally known methods and techniques a recently introduced new principle, based on warped filters, is described. Different equalization methods are compared from the points of view of equalization error both objectively and subjectively, computational efficiency, as well as robustness and precision requirements of each method. The study is limited to the linear (magnitude and phase) equalization of loudspeaker free-field responses.

**R. Walker:**

*Equalisation of Room Acoustics and Adaptive Systems in the Equalisation of Small Room Acoustics, proc. of the Audio Eng. Soc. 15th International Conference, Copenhagen 1998*

The basic properties of acoustics in small rooms and the effects leading to uneven frequency responses are described. The principles underlying possible equalisation schemes are outlined. This paper concentrates on the underlying causes of response irregularities rather than discussing proprietary solutions. It is shown that additional low-frequency loudspeakers can help to optimize the low-frequency responses without detriment to the higher frequency image localization. By using several low-frequency sources, some response control can be achieved over a limited region. At high frequencies, the objective responses of rooms do not match the subjective impressions, and equalisation based on measure parameters is likely to be unsuccessful. Equalisation schemes based on relatively wide-band or short-term response parameters are likely to match the auditory perception mechanisms better and, thus, to be more successful. Automatic or active control of equalisation is essential for complex schemes. However, it adds little to the fundamental principles and the inherent limitations of either low- or high-frequency equalisation.

**John N. Mourjopoulos:**


Signal-processing methods such as digital equalization can in theory achieve a reduction in acoustic reverberation. In practice, however, the realization of these methods is only partially successful for a number of objective and subjective (perceptual) reasons. Two of these problems, the dependence of the equalizer performance on the source and receiver positions and the requirement for extremely lengthy filters, are addressed. It is proposed that all-pole modelling of room responses can relax the equalizer filter length requirement, and the use of vector quantization can optimally classify such responses, obtained at different source and receiver positions. Such classification can be
used as a spatial equalization library, achieving reduction in reverberation over a wide range of positions within an enclosure, as was confirmed by a number of tests.

Stephen J. Elliott et al.:  

Adaptive digital filters have been used in an experimental investigation of low-frequency equalization of a single-channel sound reproduction system in a car. The problems are discussed of adapting the digital filter so that a smooth transition is achieved between the equalized low-frequency response (below 400 Hz) and the unequalized high-frequency response. Equalization of the response at only one point in the car is found to cause degradation in the response at others. Multiple-point equalization, in which the response at four positions is best equalized in a least-squares sense, was found to give only modest overall improvements in this case. The best strategy for a single filter appears to be weighted multiple-point equalization, in which the error at the most important listening position in the car is more heavily weighted in the adaptation algorithm. This gave worthwhile improvements in the response at the selected location, without significant degradations at other points. A very similar effect can also be achieved with the single-point equalization systems either by using a leak in the adaptive algorithm or by using an adaptive filter with a smaller number of coefficients.

P. G. Craven and M. A. Gerzon:  
*Practical Adaptive Room and Loudspeaker Equaliser for Hi-Fi Use, a preprint of the Audio Eng. Soc. 92nd Conv., Vienna 1992*

This paper describes a soon to be available system for stereo hi-fi use for equalising room and loudspeaker impulse responses across a listening area, using a chirp measurement and decimated digital equaliser implemented using a single DSP chip per channel. Filter impulse response lengths up to one second are achievable at low frequencies. Psychoacoustic requirements are discussed for subjectively satisfactory results and naive strategies, such as mean-square optimisation or minimum-phase equalisation are found to be inadequate.

Ronald Genereux:  

Loudspeaker designers have long recognized the influence of the acoustic environment on the perceived quality of an audio system. The availability of powerful digital signal processor (DSP) integrated circuits has created interest in the application of adaptive digital filtering techniques to the equalization of loudspeakers in rooms. A review of some recently proposed implementations is followed by a discussion of the issues which must be considered in order to address the problem successfully. Results from an experimental system are presented.
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S. J. Elliott and P. A. Nelson:

*Multiple-Point Equalization in a Room Using Adaptive Digital Filters*, *J. Audio Eng. Soc.*, 37(11), 1989 November

A method is presented for designing an equalization filter for a sound-reproduction system by adjusting the filter coefficients to minimize the sum of squares of the errors between the equalized responses at multiple points in the room and delayed versions of the original electrical signal. Such an equalization filter can give a more uniform frequency response over a greater volume of the enclosure than a filter designed to equalize at one point only. The results of computer simulations are presented for equalization in a “room” with dimensions and acoustic damping typical of a car interior, using various algorithms to adapt automatically the coefficients of a digital equalization filter.

C. Bean and P. Craven:


This paper concerns digital signal processing techniques used to equalize loudspeakers and correction of room effects well away from the loudspeaker itself. Measured results for the linear and minimum phase corrections are presented. Various algorithms for calculating digital equalisation filters are discussed, including techniques to limit the length of the digital filter. Some of the problems inherent in the correction of room acoustics are highlighted and some of the perceived effects after correction are outlined.

M. Miyoshi and Y. Kaneda:


A novel method is proposed for realizing exact inverse filtering of acoustic impulse responses in a room. This method is based on the principle called multiple-input/output inverse theorem (MINT). Because a room impulse response generally has non-minimum phases, it has been impossible to realize exact inverse filtering of room acoustics using previously reported methods. However, the exact inverse of room acoustics can be realized using the proposed method. With this method, the inverse is constructed from multiple FIR filters (transversal filters) by adding some extra acoustic signal-transmission channels produced by multiple loudspeakers or microphones. The coefficients of these FIR filters can be computed by the well-known rules of matrix algebra. Inverse filtering in a sound field is investigated experimentally. It is shown that the proposed method is greatly superior to previous methods that use only one acoustic signal-transmission channel. The results in this paper prove the possibility of sound reproduction and sound receiving without any distortion caused by reflected sounds in a room.
When a conversation takes place inside a room, the acoustic speech signal is distorted by wall reflections. The room’s effect on this signal can be characterized by a room impulse response. If the impulse response happens to be minimum phase, it can easily be inverted. Synthetic room impulse responses were generated using a point image method to solve for wall reflections. A Nyquist plot was used to determine whether a given impulse response was minimum phase. Certain synthetic room impulse responses were found to be minimum phase when the initial delay was removed. For these cases a minimum phase inverse filter was successfully used to remove the effect of a room impulse response on a speech signal.

2 Papers on room modelling and subjective evaluation

The following five papers address the room modelling issue in order to establish a basis for equalisation which is more feasible than the blind point-to-point inverse filtering scenario and subjective tests done on equalised room transfer functions.

Y. Haneda et al.: 

Multiple-Point Equalization of Room Transfer Functions by Using Common Acoustical Poles, 
IEEE Trans. on Speech and Audio Proc., Vol. 5, No. 4, 1997 July

A multiple-point equalization filter using the common acoustical poles of a room transfer function is proposed. The common acoustical poles correspond to the resonance frequencies, which are independent of source and receiver positions. They are estimated as common autoregressive (AR) coefficients from multiple room transfer functions. The equalization is achieved with a finite impulse response (FIR) filter, which has the inverse characteristics of the common acoustical pole function. Although the proposed filter cannot recover the frequency response dips of the multiple room transfer functions, it can suppress their common peaks due to resonance; it is also less sensitive to changes in receiver position. Evaluation of the proposed equalization filter using measured room transfer functions shows that it can reduce the deviations in the frequency characteristics of multiple-point room transfer functions better than a conventional multiple-point inverse filter. Experiments show that the proposed filter enables 1-5 dB additional amplifier gain in a public address system without acoustic feedback at multiple receiver positions. Furthermore, the proposed filter reduces the reflected sound in room impulse responses without the pre-echo that occurs with a multiple-point inverse filter. A multiple-point equalization filter using common acoustical poles can thus equalize multiple room transfer functions by suppressing their common peaks.
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The feasibility of using all-pole or all-zero model approximation of room transfer functions was examined especially in respect to the degree that such approximations are suitable for removing room reverberation from signals. Two aspects of the above models were assessed: Their success in reducing room transfer function order and their insensitivity to measurements taken for different source and receiver position inside a room. Both these aspects are crucial to the success of practical dereverberation methods. The tests were carried out on simulated and measured data and it was found that all-pole models are more suitable than all-zero models when both of the above two aspects must be satisfied by the approximation model. It was also shown that, for a given room, an optimum all-pole model exists which approximates the room transfer functions.

P. L. Schuck et al.: **Perception of Perceived Sound in Rooms: Some Results of the Athena Project**, *Audio Eng. Soc. 12th Int. Conference*

Experiments comparing the use of multiple versus paired comparisons for loudspeaker evaluation, the use of digital band splitting and equalization, and assessing the scope of the loudspeaker/room interaction effect on loudspeaker preference ratings, are dis-closed. Finally, results of an experiment which utilizes loudspeaker/room equalization to lower the variability of preference scores, as well as increase to them, are presented.

Søren Bech: **The Influence of the Room and of Loudspeaker Position on the Timbre of Reproduced Sound in Domestic Rooms**, *Audio Eng. Soc. 12th Int. Conference*

A round robin test has been conducted to examine the interaction between a loudspeaker and the room with respect to fidelity of timbre of reproduced sound. Three rooms, three loudspeaker positions, four loudspeakers, four programs, and six subjects were used for the experiment. The statistical analyses show that the main factors which have a significant influence on the assessment of fidelity of timbre are room, loudspeaker position, loudspeaker, and program. Several of the interactions between the main factors are significant, however the most important is that between loudspeaker and positions. The results show that the room will influence a) the overall fidelity of timbre of reproduced sound in all positions; b) the perceived differences between loudspeakers in similar positions; and c) the perceived differences between loudspeakers in different positions in the same room. The loudspeaker position will also have a significant effect on the level of fidelity of timbre. The degree of influence of the room and of the positions will depend on the directivity characteristics of the loudspeaker.
3 Discussions on the material

Basically, the topics addressed in these fifteen papers can be condensed into the issues below:

Physical limitations

Why will correction/equalisation never work all over the room, and what artifacts are observed when designing optimal (in some sense) equalising to one part of the room and in fact experience the reproduced sound in another. No magic solutions are given. Essentially, the more optimal equalising in confined parts (or even points) of the room, the more prices are paid in other parts in the sense of deteriorating reproduced sound compared to the one without a correction system employed. For low frequencies, however, it seems like more loud-speakers spread around the room can ease up the problem. Also, appropriate modelling of a room response, e.g. by an all-pole function, may prove beneficial. Some report that equalisation based on multiple receivers (microphones) improves overall performance - there is no general agreement though.

Mathematical limitations

Equalisation is in most of the papers related to inverting the frequency spectrum of a measured impulse response. If the response is not representing a minimum-phase system (and it rarely is in ordinary listening rooms) the inversion is not possible without introducing artefacts of some kind. Desired and relatively easy correction must be established then within the set of minimum-phase transfer functions.

Psychoacoustic issues

More of the papers express their concern to what is in fact audible by the human hearing and transform that knowledge into relaxed criteria for the equaliser/correction system. There seems to be a common agreement on not to attempt too detailed equalisation in both time- and frequency domains as this is likely to cause undesired artifacts even in the sweet spot.

Signal processing issues

In relation to equalisation systems, the late eighties and early nineties were dominated by the fact that it suddenly became possible to implement real-time digital correction electronic hardware. Mostly, however, the reports remain at the discussion level. In fact in this context only the paper by Craven and Gerzon reveals a practically functional system. Maximum equalisation filter length is also discussed as a necessary concern due to limits in available hardware. Although more papers have been presented on the practical issues, few, if any, report on entire stand-alone correction systems, i.e. a system which by itself can do the acoustic environment measurements (impulse response acquisitions), calculate the equalisation filters/algorithms, and finally perform the real-time handling of the electrical signals.

Room modelling issues

Modelling the room acoustics seems to be a good way of eliminating the effects in a measured room transfer function that usually mess up equalisation attempts. By the use of discrete-time models based on polynomial ratios...
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ARMA models), the perhaps most annoying acoustical phenomena are captured (the modal resonances), and the beautiful thing is that they immediately relate to digital correction filter design.

4 Some commercial room correction systems

In 1991 B&W launched a room correction system called the B&W Digital Control Unit. Essentially it was based on the seventies Teledyne experimental system and to a large extent Colin Bean and Peter Craven’s earlier theoretical work (see above). A technical presentation can be found in Hi-Fi News & Record Review 1991 December.

Marantz AX-1000, a combined room correction system and room simulation system, was accessible in 1992. An impressive piece of equipment for that time representing a power of 13.3 mips.

TacT Audio introduced a room correction system in 1997 called RCS 2.2. The equipment only works together with a PC for calculating the equalisation filters, impulse response acquisition and real-time processing are both done independently. Splitting the measured impulses into three frequency bands, the RCS 2.2 uses different strategies for designing the correction filters. In the lowest frequency region, the correction resolution is 0.6 Hz, up to 1,500 Hz the resolution is 5 Hz, and above it is 300 Hz. Processing is powered by four Motorola DSP56002 processors. The RCS 2.2 was received with some excitement in the professional world, and although really expensive it has been reviewed (with good marks) in the home theatre / Hi-Fi literature. Many prejudices pointing towards digital Hi-Fi equipment was put to death, and the reviews reported on noticeable improvements in the subjective quality of reproduced sound. Later, TacT has launched simpler and cheaper versions producing almost the same subjective quality improvement.

Roister Digital Audio has come up with a quite new set of digital room equalisers (1999), the Acoustics Compensators, having a processing capacity between 160 and 480 MIPS for the top-of-the-range model. That allows running two-channel FIR filters at 48 kHz with 768 up to 2336 filter coefficients. Input impulse responses must be supplied using external pieces of measuring equipment.

Other recent pieces of room correction equipment are listed below. Also Philips launched a room equaliser in the early nineties (and a not too expensive set of active speakers in fact) - neither of those obtained much commercial success.

- SigTech AEC-1000/2000. Can handle 2,200 FIR filter taps in 48 kHz sampling and works adaptively, i.e. the correction filters are changed on-the-fly,
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• **Snell Acoustics RCS 1000.** Room correction system with 1.5 Hz resolution,
• **Behringer Ultracurve Pro 8024.** A 1/3 octave equaliser that produces the band corrections automatically based on measurements.

*Why not more systems?* Not many systems of the kind have emerged yet, and the world still waits for a simple, cheap, and completely automatic room correction system. Perhaps that is why the concept of room correction has not yet gained widespread popularity. Another explanation is that just ten years ago, digital Hi-Fi equipment was still approached with much scepticism (at least among purists). Perhaps still not completely vanished, the “fear-of-digital” has greatly reduced here at the beginning of the new century. Thus, a breakthrough for the digital room correction may be just around the corner.