Issues affecting measurement and equalisation of loudspeakers

Authors:
Paul Glendenning BE, Dir. R&D Clarity EQ
Brett George BE (Hon)
Parastoo Sageghi BSc (Hon) MSc

In this paper we discuss measurement techniques suitable for loudspeaker analysis and the issues relating to loudspeaker equalization. Finally we discuss the need for a standard in speaker performance especially in studio monitoring.

It is often desirable to know the impulse response h(t) of linear systems such as rooms and loudspeakers. The impulse response defines the characteristics of the acoustic environment such that the response to any input can be found by convolving the input signal with the impulse response.

$$y(t) = \int_{-\infty}^{\infty} x(t).h(t-t).dt \qquad \textbf{(1)}$$

Where y(t) is the outputsignal and x(t) is the input signal
A method for determining the finite impulse response solution of a discrete time linear system [1] with a known input signal x(n), a measured output signal y(n), and an error function e(n) is described by the normal equations (2).

$$R_{xy} = R_{xx} \bullet h_v \qquad \textbf{(2)}$$

Where:
R_{xy} is the one sided cross-correlation matrix of the output signal y(n) with the input signal x(n) and
R_{xx} is one-sided auto-correlation matrix of the input signal and
h_v is the impulse response in vector form

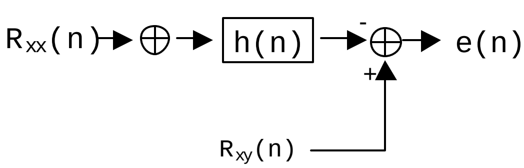

Figure 1 - Filtering configuration for normal equations.

The solution to equation (2) seeks to minimise the average energy of the error function when solving for h(n) (Figure 1). The complete solution of (2) is solved by inverting the matrix R_{xx}.

$$h_v = R_{xx}^{-1} \cdot R_{xy} \qquad \textbf{(3)}$$

Where R^{-1}_{xx} is the inverse of the auto-correlation matrix R_{xx}

A simple solution to equation (3) is given for the case of a periodic excitation of the system h(n). Typical x(n) is chosen to be a maximal length sequence [2] such that the auto-correlation of one period is an impulse. For a MLS signal equation (3) reduces to:

$$h_v \approx R_{xy} \qquad \textbf{(4)}$$

Much is discussed about MLS techniques such as their invariance to noise in the measurement signal y(n), and to the simplicity in calculating the cross-correlation using the Fast Hadamard Transform which requires only additions and some reordering to solve for h(n). The computational complexity of using the FHT is O(N.log(N)) where N is the dimension of the matrix R_{xx} (O means the mathematical order of complexity).

The advantages of using MLS for computational speed are these days irrelevant with the availability of high speed digital signal processors (DSP) which can do parallel operations of multiplications and additions in a single instruction. A direct solution to equation (3) with any input signal is possible by exploiting the symmetry of the auto-correlation matrix R_{xx}. Several algorithms exist that solve with complexity $O(N^2)$ and $O(N.\log^2 N)$[6], the most famous being the Levinson-Durbin recursion algorithm [3]. Using today's DSP, or even the power of a standard PC, the normal equations can be solved with minimal fuss.

An advantage often cited for MLS is noise rejection however this advantage exists for both techniques when the noise signal is uncorrelated. According to [4], the MLS method performs worse under pink noise conditions but in simulation we were unable to verify this. We have verified [4] that the direct solution performs best when input signal spectrum matches the noise spectrum and it is interesting to note that a logarithmic sweep, which has a 1/freq gain characteristic, performs better with pink noise. Figure 2 shows comparison of the direct and MLS technique under various noise conditions. Given that most real world systems will contain pink rather than white noise the logarithmic sweep is probably the better choice for measurement systems.

There are other problems with the MLS method such as sensitivity to the non-linearity in the measurement system that mostly manifest itself as increase in the noise level and unwanted reflections in the impulse response. On the other hand there is more freedom in choosing a swept sine method that does not result in serious harmonic distortions. Also according to [5] if logarithmic swept sine is used, the spurious reflections can be pushed to the left of the linear response at precise known times, by means of inverse filtering, and thus enabling the measurement of the system's linear impulse response. The behaviour of these techniques under non-linear conditions are shown in Figure 3. The better performance of the sweep is useful when measuring real world components such as loudspeakers, which exhibit these characteristics

The main purpose of measuring the loudspeaker impulse response is to determine a suitable representation of the speaker in a way that accurately describes its performance. The measurement of the speaker: directly on-axis, at a fixed distance and with minimal interferences (outlined above, but not including the presence of echoes) is generally considered to be the best objective measure of its subjective performance [7]. It therefore follows that an equalisation method based on these measurements, which corrects the speaker as close as possible to an 'ideal' response, would provide the best subjective results.

It should be recognised that a single measurement can never accurately represent the response of a speaker in a non-ideal environment, because the radiation of the speaker into the room influences the perceived quality of the sound. It is these off-axis responses which also provide useful information for the description of the speaker, and they can also be considered as a factor in the equalization stage. Wilson [8] uses a technique which weights the measurements at various angles, providing an equalization method whose performance can vary depending on the importance of the on- or off-axis responses. If taken over large angles, such a method will compromise the on-axis measurement and is therefore more effectively used over a small angular range.

In determining the ideal response that the correction filter will aim at achieving, it is generally considered that a flat magnitude response is best [9], even though this may not exactly correspond to perceived loudness [10]. Such a speaker would reproduce the same signal level (in dB) for all frequencies. There are however limitations on achieving this ideal magnitude response, because some speakers will have significant attenuation of frequencies within the audio range. If the equalising filter attempted to provide an equal magnitude response at these frequencies, there is a risk of damage to the speaker or significant non-linear distortions. It is therefore important that the equalising algorithm does not significantly correct in these instances.

Equally important in determining the ideal response is to consider the phase response [15]. This effectively determines the time arrival of all frequencies, or groups of frequencies, and thus, together with the magnitude response, completely determines the response in the time-domain. It is interesting to note that the group delay (the negative derivate of the phase) determines the exact value of a frequencies time delay only when the magnitude is sufficiently flat [11][12]. This is further encouragement for using flat magnitude responses [13].

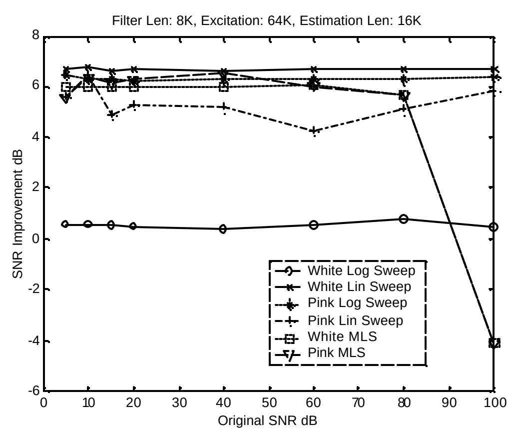

Figure 2 - Comparison of MLS to Direct solution with the addition of pink or white noise. Simulated using Matlab 6.0

Since speakers are often used in Home-theatre situations, where synchronisation with video signals is important, it is essential that the total delay through the system is kept to a minimum. This requirement limits the amount of equalisation that can be achieved on the phase. A compromise where the phase is corrected to a point that does not increase the overall perceived delay of the signal is necessary, and ensures that the phase is still improved. It is also important to consider that large group-delay variations between frequencies can cause considerable audio distortion [14], and this should also be kept to a minimum.

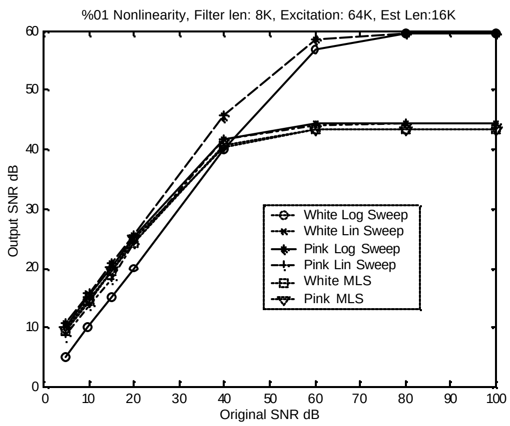

Figure 3 - Comparison of MLS to Direct solution under non-linear conditions. Simulated using Matlab 6.0

When correcting groups of speakers, uniformity of phase (between speakers) is important in improving imaging (the perceived direction of the sound), because the time arrival of each frequency will be the same for a listener sitting equidistant from all speakers. This provides a more realistic experience for the listener because each sound or instrument can be 'mixed' to fit precisely within this space (Figure 4).

Although the limit to how much a speaker can be improved is inevitably set by the speaker design itself, designing digital cross-over filters is still another area where significant improvement can be made. Cross-overs filters separate the full bandwidth signal into smaller bands more suitable for each individual loudspeaker within a complete speaker unit [16]. Using higher-order digital filters improves on the original advantages of cross-over filters, by further reducing the amount of energy to each speaker and providing improved polar response [16], and reduced lobing error [17].

Since the CD brought the first generation of digital audio to consumers, digital production processes for music and film and TV soundtracks have improved dramatically using digital recording and production techniques. Despite the improvements brought about by digital technology monitoring remains essentially unchanged. Many studios use a combination of active loudspeakers and defacto standards such as the Yamaha NS10 portable near field monitors, but to date there is no recognized standard reference which is applicable to audio production.

Since 1997 Clarity EQ has been working on the development of digital measurement, analysis, and playback equalization algorithms that enable loudspeakers to perform as perfectly as possible in any given listening space - this is the *Clarity Calibrated* Technology. Clarity EQ is seeking to set a new standard for loudspeaker performance where each speaker has been calibrated to a specified reference. With calibrated speakers the phase and gain characteristic of different loudspeakers are guaranteed to be with ±0.5db and phase to be within ±2º.

To achieve this end Clarity EQ will release in January the first in a series of Professional Digital Correction products (PDC), which provide all the hardware and software to implement equalisation filters and analyse speaker and room acoustics.

The PDC has specialised software for creating the equalisation function and implements a proprietary optimisation algorithm. The optimisation seeks to minimise the latency of the final equalisation response while meeting certain criteria which can be adjusted by the user. The PDC user can adjust parameters such as min cut and boost in certain frequency bands, allowable latency, as well as apply different optimisation techniques. The details of the optimisation techniques cannot be discussed in this paper as they are the subject of a patent application.

The PDC also provides the ability to equalise the tweeter, mid-range, and woofer loudspeakers separately and will design suitable digital crossovers. Unlike conventional crossovers these digital ones have an extremely narrow transition band which the authors believe enhances the perceived off-axis performance and dispersion at the crossover frequency.

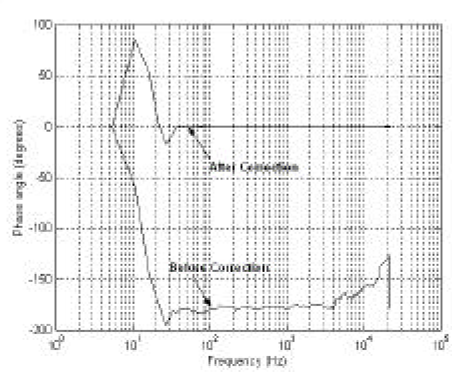

Figure 4 - Phase difference between a pair of speakers before and after correction.

Figure 5 shows the results of the ClarityEQ process on improving a standard pair of studio monitors.

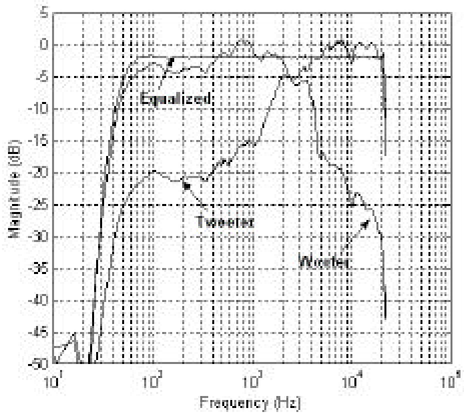

Figure 5 – Equalisation with digital cross-overs on a standard pair of studio monitors.

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