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**Subject:** **Recommendations for optimal audio system characterization technique**

**References:** "Recommendations for new product line." 9 March 2003
http://www.elysianaudio.org/class/tc496/tc496_rpt3_JBG.pdf

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## Foreword

The new Adaptive Audio Room Equalizer ($A^2$REq) project requires a measurement system in order to characterize the time and frequency response of an audio system under test. It was my task to investigate possible measurement systems and identify the best one for use with $A^2$REq. It is the aim of this report to present my findings and propose an optimal measurement solution.

## Summary

Elysian Audio has recently embarked on project to build an adaptive equalizer to complete its product line of hi-fi audiophile equipment. The purpose of the equalizer is to make adjustments to the audio signal going through the audio system that correct for environmental, technical and user factors. Frequency response variations due to early reflections of sound waves account for the majority of errors in sound reproduction. These errors cause significant sonic degradation.

To make the equalizer easy to use and separate it from competing products, it is to be self-adaptive. This means that it will automatically determine the correct filter coefficients. In order to calculate the error correcting filter coefficients, and measurement system is needed that uses a signal source and a microphone to accurately determine the response of the room and audio system. This measurement system must have reasonable immunity to background noise, account for acoustic reflections, and work over the entire audio passband.

Possible measurement systems were identified through an exhaustive literature search of both academic sources and commercial works. Journals from IEEE and the Audio Engineering Society were investigated. Two different measurement methods stood out and have been used in the past to characterize audio systems and environmental response: stepped-sine wave frequency response and MLS impulse response.

MLS stands for maximal length sequence because of the way it is generated. It is similar to noise, but is completely deterministic, meaning its value is known at all times. The MLS impulse response technique is a common and known method for characterizing audio systems.

The professional audio software packages we use at Elysian Audio all use MLS for measurement. A large amount of material exists on MLS methods for audio systems and can be found in many journals. Compared to other methods, such as swept-sine, the MLS technique is many times faster and is an order of magnitude less complex to implement. Furthermore, the MLS method provides additional data, which could be used to augment the functionality of $A^2$REq so it could compete with other professional audio software packages.

It is therefore my recommendation that an MLS impulse response method be used to characterize the audio system and environment for the $A^2$REq project. MLS impulse response offers more useful results and lower implementation costs in terms of both time, effort, resources, and hardware.

## Discussion

There are two domains in which to characterize an audio system, one of which offers a better solution. To understand the implications of using an MLS impulse response method for characterization, it is important have a general idea of why the impulse response is important, and why it is almost impossible to do directly. This directly leads into what MLS is and why it is desired. Decoding the response from the MLS signal is discussed and finally further applications of MLS data is mentioned.

### *Characterization Domains*

There are two domains, time and frequency. Any linear system can be completely characterized by measuring one domain. Furthermore, once the measurement of one domain is know, the other can be computed directly. Since the initial goal was to correct the frequency response of the system, it makes sense to measure in the frequency domain. The stepped-sine method does this.

Stepped-sine frequency response measurement works by injecting a single frequency sine wave into the audio system. Simultaneously, it records the sound waves with a microphone and filters for that frequency and measures the level. The frequency of the sine wave is adjusted through the desired range at a number of frequencies. It requires a complex signal generation and measurement system that takes a long time to make one measurement to reduce noise interference to an acceptable level. Furthermore, such a measurement is only practical if the number of discrete frequencies measured is limited. Reducing the number of frequencies measured reduces the overall accuracy of the entire system. There is also the problem that the human ear hears frequencies logarithmically, and almost all swept-sine methods are meant to work on a linear scale. Because only one single frequency is measured at a time, and there is a long integration, only the stead-state of the room can be determined. By measuring steady state, no transient information can be ascertained and therefore, early room reflections cannot be corrected for.

### *Impulse Response*

The time domain equivalent of frequency response is impulse response. The impulse response of a given system is the time varying output of the system given a dirac impulse is the input. A dirac impulse is a theoretical pulse waveform with infinite amplitude and zero width. If a perfect audio system with a speaker were placed in the middle of an open space, a microphone would pick up the single impulse. Now if a wall was placed somewhere in this space as well, the microphone would first pick up the impulse from the speaker and then the impulse reflected from the wall. This impulse response could then be used to make a filter, which would greatly attenuate the reflected impulse from the wall. The impulse response is also affected by the frequency response of a system, so for instance a speaker that attenuates high frequencies would make the impulse oscillate and have reduced amplitude.

The problem with the dirac impulse is that it is an impossible function. Furthermore, because it just a single pulse, there is minimal energy in the signal. The microphone would pick up a large amount of spurious background noise. The system would be prone to great error, even in an anechoic chamber designed to be absolutely quiet, which is far from the desired operating environment for $A^2$REq.

*Maximal-Length-Sequence* (MLS)

In order to overcome the limitations of the dirac impulse but still make an impulse response measurement, a technique involving Maximal-Length-Sequences (MLS) is used. An MLS is a somewhat pseudo-random periodic sequence that is either 1 or –1. It is easily generated using a simple shift register generator (SSRG) with minimal processor overhead. Just a ten-position simple shift register generator can create a MLS with over one thousand values before repeating. This makes the MLS have the desirable properties of random noise yet remain sequential and deterministic, meaning the value of the sequence at any time is always known. The SSRG is a standard form that can be expanded to any number of values resulting in a wide range of possible MLS lengths.

The MLS sequence contains all possible frequencies, just like the true impulse, but time-spread over a much longer interval. This is what gives the MLS its noise advantage. When an MLS sequence is correlated against itself, the result is a single sharp impulse. Since the MLS is a time-spread impulse, techniques can be used to use a robust MLS measurement instead of a direct dirac impulse measurement.

To be used in a measurement system, an MLS is played through the audio system over the loudspeaker. On the receive end, only one period of the MLS needs to be recorded with the microphone. After being sent through the audio system and environment (and thereby exposed to the room reflections) the MLS sequence contains all the important time information. A number of methods exist for extracting the time information, but the fastest and most efficient by far is the Fast Hadamard Transform (FHT) or Fast Walsh-Hadamard Transform. The FHT is a known algorithm that is relatively easy to implement.

*Fast Hadamard Transform*

The FHT can be broken down into 2-point matrix multiplications that are identical to the 2-point butterfly computations used in the ubiquitous Fast Fourier Transform (FFT). However, when scaled up to multiple points, the FHT does not require the Twiddle Factors needed by the FFT algorithm and does all computations using addition. The FHT can be computed on data in-place, so the memory requirements are reduced by many times compared to other techniques. By eliminating the significant computation of exponentials, the FHT can operate on significantly lower cost and lower complexity hardware.

*Implications of using MLS Impulse Response*

Once the system impulse response has been computed from the recorded MLS signal, a number of possibilities exist for the data. The DFT of the impulse response could be taken to generate a frequency domain filter that would perform at least as well as a filter generated with a stepped-sine measurement method. Given the higher noise immunity of the MLS method, it would be better in normal conditions. Furthermore, since all frequencies were measured simultaneously, the measurement time and therefore annoyance to the user would be significantly less.

If the filter coefficients were generated directly from the impulse response, a further benefit could be realized. Because the impulse response is time-domain, it would correct time-delay mismatch between multiple drivers of one speaker. This would be impossible with steady-state stepped-sine analysis.

Lastly, the impulse response data could be used to generated waterfall type plots, or other acoustic diagrams of the audio system. Doing so would put us in direct competition with professional products such as MLSSA without adding significant development time or effort.

*Conclusion*

The MLS impulse response measurement technique is the optimal solution for audio system characterization in the $A^2$REq project. It offers the greatest benefits with the least overhead compared to all other known systems.