THE MLS ANALYSIS TECHNIQUE AND CLIO

CONTENTS

Rev 1.0 - 29 Dec 1997

Introduction

The MLS analysis aims to characterize the behaviour of the system under test both in time and frequency domains; the measurement itself achieves a precise time measurement of the basic parameter of a system, it's Impulse Response (IR), thus allowing several post-processing to be made; among these we recall:

- Frequency and phase response (Frequency)
- Acoustical anechoic analysis (Time and frequency)
- Acoustical room response (Time and frequency)
- Reverberation time RT60 (Time)
- Energy time curve (ETC) (Time)
- Cumulative spectral decay (Waterfall) (Time and frequency)

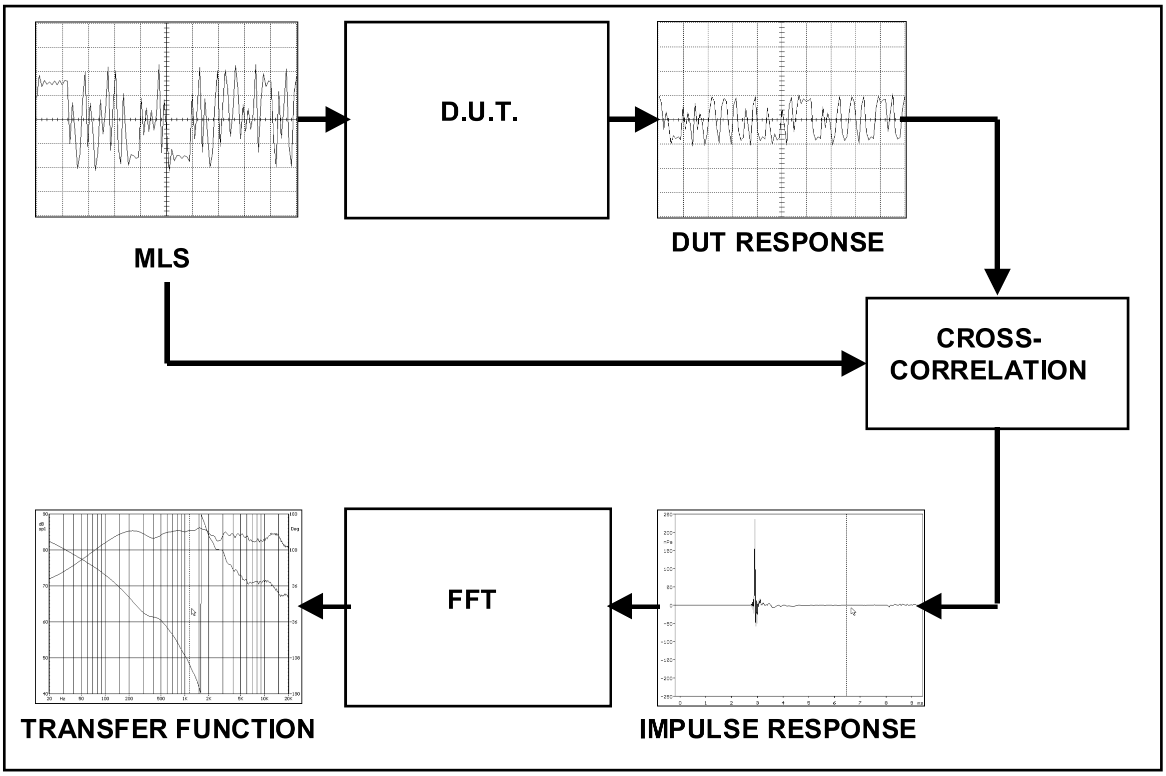


FIGURE 1 - The MLS analysis technique

Figure 1 shows the flow diagram of a MLS measurement. The analysis technique is based on the mathematical theory behind particular signals known as Maximum Length Sequences (MLS); these signals are digitally

synthesized binary sequences which, among other properties, share the following: when fed to the system input, their cross-correlation with the system output gives exactly the system impulse response; the cross-correlation can be carried out, in the time domain, with an efficient digital algorithm that minimizes the user's waiting time.
Regarding the phase response it has to be underlined that it is sufficient only a single measurement channel, due to the complete temporal control of the generated signal. In fact, the analyzer takes the generated signal instant by instant and compares it to the acquired one, so it is capable of performing real phase measurements. It is also possible to null the signal group delay caused, for example, by the propagation time from the loudspeaker to the microphone or by an equaliser network, therefore obtaining better quality presentations of phase measurements.
Pay attention to the fact that only an analyzer, like CLIO, that performs the digital MLS algorithm described in this article can achieve the performances offered by the MLS technique, and can therefore be called an MLS analyzer; there are some instruments that use MLS sequences only as stimuli and then directly perform FFT: the result is very poor.
Two fundamental quantities like the impulse and phase responses, that several analyzers only calculate, are instead measured ones; this let the user have the best picture possible of the system.

MLS Sequences

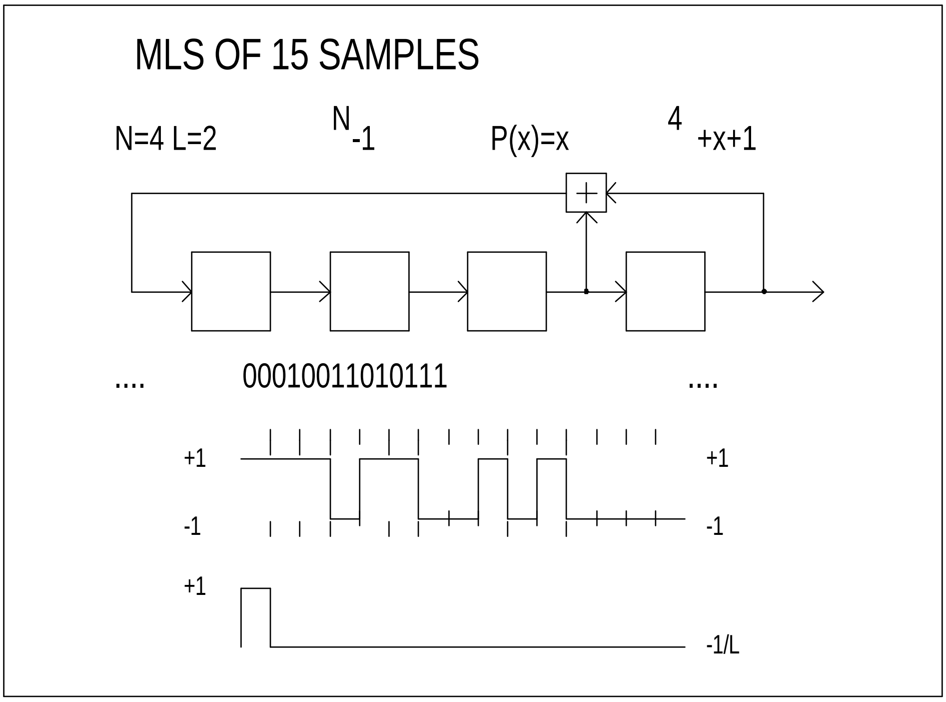
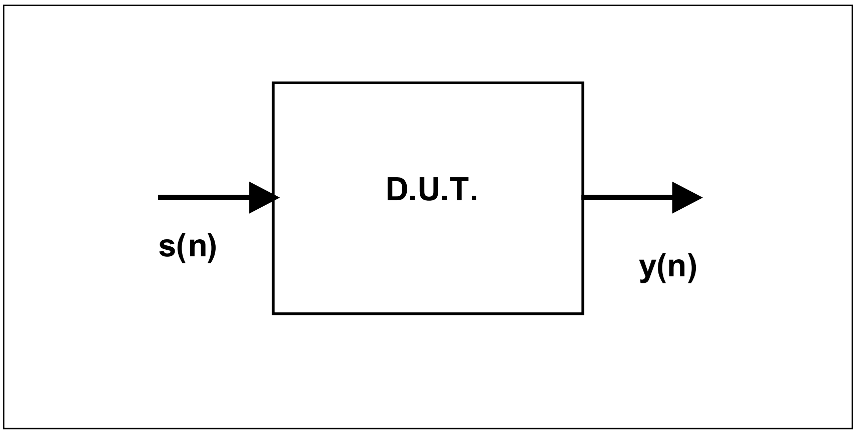


FIGURE 2

Figure 2 shows an example of Maximum Length Sequence; this is the simple case of a sequence of only 15 samples and we say that its order N is 4 and its length L is 15; in the figure we can see also the so called "primitive polinomial" P(x) and the relative "feedback shift register" that are needed to define and generate the sequence; in practice you obtain a bipolar signal assigning a high level to zero states and a low level to one states using integrated shift registers and D/A converters. Shown are also the sequence of binary values (zeros and ones), the associated sequence of electrical states (+1 and -1) and the value of the auto-correlation (cross-correlation of the sequence with itself).
From an electrical point of view the MLS signal has other interesting features; its spectral content closely resembles a white noise so, when you hear it, you feel like a noise burst is generated (in fact it is played for no longer than fractions of a second); but, on the other hand, it is deterministic (well known at any instant) and periodic with a certain temporal length (also referred as its order): this is why it is called a "pseudo-random noise". Being a noise signal it has a low crest factor, transfers a lot of energy to the system (compared with the classical impulse excitation) and achieves a very good signal-to-noise ratio.

The cross-correlation



And now a closer look at the cross-correlation. In figure 3 and in the associated equations we may describe mathematically what happens in an MLS measurement. We supply to our DUT an MLS signal s(n) (equation 1) and measure its response y(t). The value of an MLS auto-correlation is given in equation 2 and we may

$$s(n) = MLS \qquad (1)$$

$$\Phi_{ss}(n) = 1 \quad (n = 0) \qquad (2a)$$

$$\Phi_{ss}(n) = -\frac{1}{L} \quad (0 < n < L) \qquad (2b)$$

$$\Phi_{ss}(n) \approx \delta(n) \qquad (3)$$

$$y(n) = s(n) \otimes h(n) \qquad (4)$$

$$\begin{aligned} \Phi_{sy}(n) &= s(n)\Phi\, y(n) = \\ &= s(n)\Phi\,[s(n) \otimes h(n)] = \\ &= [s(n)\Phi\, s(n)] \otimes h(n) = \\ &= \Phi_{ss}(n) \otimes h(n) \end{aligned} \qquad (5)$$

$$h(n) \approx \Phi_{sy}(n) \qquad (6)$$

FIGURE 3

well state that it is almost equal to a perfect impulse (eq. 3; and this approximation is more and more valid as the length of the sequence increases with practical lengths starting from 4095 points). From theory we know that the output of our system is equal to the input signal convoluted with the system impulse response (eq. 4). We may now see the expression of the cross-correlation of the input MLS sequence with the measured output signal; with very simple steps described in equation 5 (and substituting the value of the auto-correlation as in eq.3) we obtain the desired result: the system impulse response equals the cross-correlation of the MLS input with the measured output.

Example

We will now describe the classical application of the MLS analysis for recovering the anechoic frequency response of a loudspeaker, i.e. the frequency response as if the loudspeaker were positioned in an anechoic

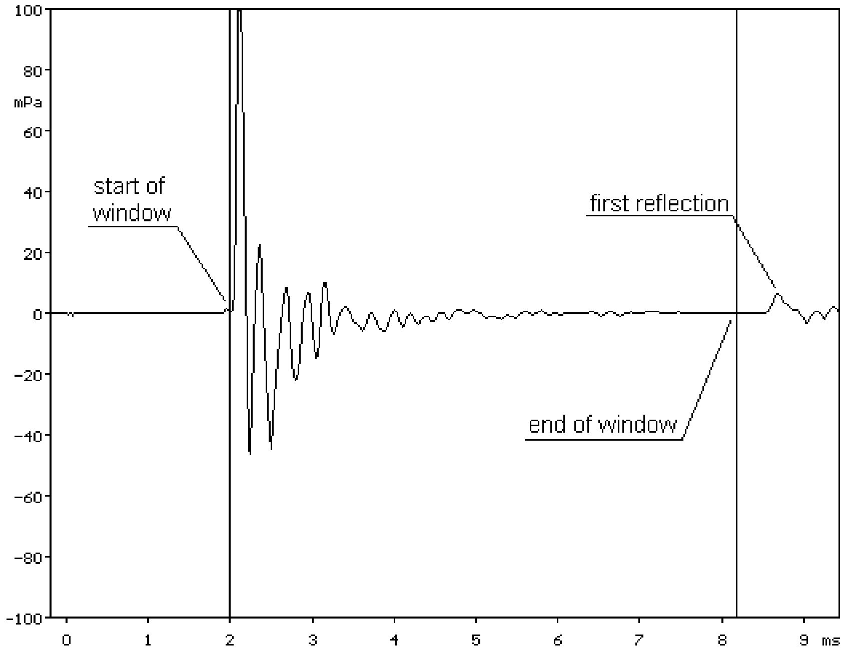


FIGURE 4

chamber but measured in a normal room.
Figure 4 shows an impulse response that we may obtain as the result of a MLS measurement; the important thing to underline is that, being this the product of our measurement, we can think of it as the start point from which we can obtain the desired final result after we have made some particularly important choices; the impulse response shown is of a loudspeaker measured in a normal room; as we can see the impulse is something like 2.1 ms after time zero which identifies the beginning of the MLS stimulus and with some easy calculations involving the speed of sound (around 344 m/s) we are able to measure exactly the distance of the measuring microphone from the acoustic center of the loudspeaker (in this case 72 cm); we can also see that after some time (9 ms) that the impulse is decayed to negligible values more energy arrives, it is due to the first reflection of sound in the room; it is this the time to choose if we are interested to the total frequency response of the loudspeaker plus the room or we want to obtain the anechoic behaviour of the former.
In the first case we have to transform the complete impulse response measured and we will obtain the

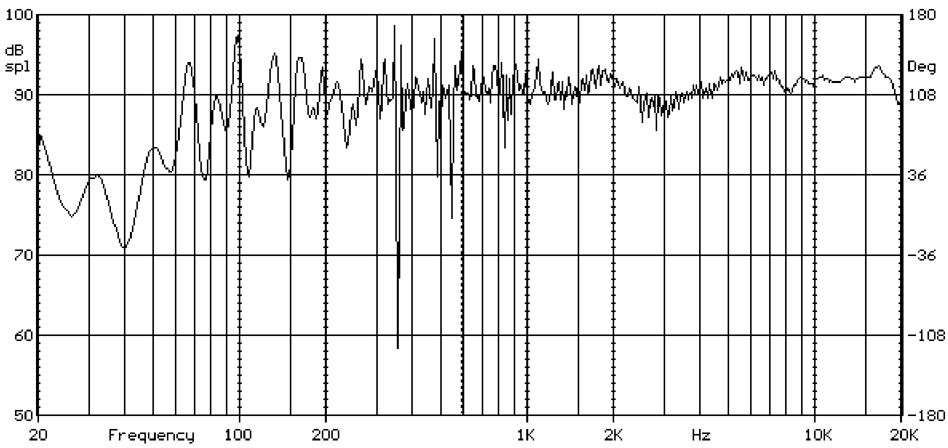


FIGURE 5

response as in figure 5.
In the second case we have to transform only the part of the impulse response that is due to the loudspeaker; in other words we have to select this part by means of a time window (also seen in figure 4); we will obtain the

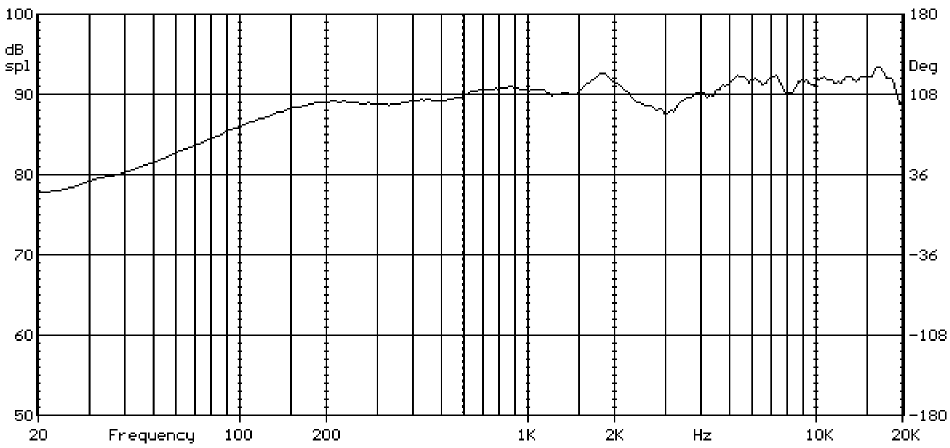


FIGURE 6

response as in figure 6.
These two figures clearly show the contribution due to the environment that adds to the proper frequency response of the loudspeaker; other measuring strategies may lead us to investigate different parts of the system impulse response; for example we could isolate one wall reflection and study the absorbing properties of the wall material in function of frequency.

Bibliography

[1] D.D. Rife and J. Vanderkooy, "Transfer Function Measurement with Maximum-Length Sequences," J. Audio Eng. Soc., Vol. 37, 1989 June.
[2] W.D.T. Davies, "Generation and properties of maximum length sequences," Control, 1966 June/July/August.