ELEKTROTEHNIŠKI VESTNIK 78(3): 91-95, 2011
ENGLISH EDITION

MLS and Sine-Sweep technique comparison in room-acoustic measurements

Franco Policardi

Università di Bologna, viale Risorgimento 2, 40136 Bologna, Italia
franco.policardi@studio.unibo.it

Abstract. Modern room acoustic measurements are still in their evolution phase, as stated in ISO 3382. MLS test signals coexist with Sine-Sweep test signals because both show pros and cons. While MLS technique provides good rough data results, Sine-Sweep technique is more accurate butalso more time-consuming and computationally demanding. During 2009Trieste Politeama Rossetti room-acoustic measurement campaign, a comparison was made between the two techniques. After a theoretical discussion,the paper presents some testing results.(77)

Key words:room acoustics, ISO 3382, RT_{60}, impulse response, MLS, Sine-Sweep technique

1 INTRODUCTION

Every physical space is defined by specific acoustics and even in contemporary virtual-reality, acoustics plays a fundamental role in creating reality illusion. Reliable room-acoustic measurements have been developed between the end of XIXth and beginning of XXth centuries leading to a multitude of acoustic-parameter definitions through specific algorithms, first of all reverberation time decay for the first 60dB (RT_{60}). Reality still shows differences between calculated and measured parameters and ISO 3382 standard [1] gives some main guidelines in room acoustic measurements. Besides different measurement setups, this standard allows different room excitation signalsand a lot of freedom in different operator choiceswhichcan interfere with final results.

The fundamental hypothesis in analysingan acoustic or electroacoustic system behaviour is its linear time invariant (LTI) behaviour during measurement. This is necessary to determine its impulse response (IR) and to calculate from it its frequency response and other parameters. Room impulse response g(t) is the pressure-time response function at a specific receiver position as a result of the excitation by an impulsive source. This enables research ofa specific algorithm, defining the univoque relation between sound source and receiver through the unknown behaviour of the acoustic space. The time-energy function is therefore $E(t) = [g(t)]^2$.

By definition,IR is the system output signal obtained by the input signal,mathematically defined as Dirac δ, which is digitally PCM defined as a sequence of 0s containing a 1 at a full-scale strength. This sort of IR measurement (many times analogically approximated by pistol shots or balloon explosions) presents various problems as for example low S/N, possible non-linear phenomena due to too fast signal slew-rate and last but not least extremely low and high frequencies,manacing to damage electroacoustic transducers.

Early acousticians used hand-claps evaluation to understand room acoustic behaviour; but this was happening during a less technological era. It was a non-technological approachin a lot less measured room background noise.Different non-direct IR measurement techniques have then been developed, starting with organ pipes in 1895 by W. C. Sabine [2], thenthe artificially generated white noise and now pseudo-random maximum length sequence (MLS) and Sine-Sweep signals coexisting.

2 MEASUREMENT METHODS

2.1 Theory

A simple method to measure any system consists of applying a unitary impulse in input and to observe its output. The more the input signal is similar to the ideal one, the more accurate will be the system impulse response as shown in Figure 1.

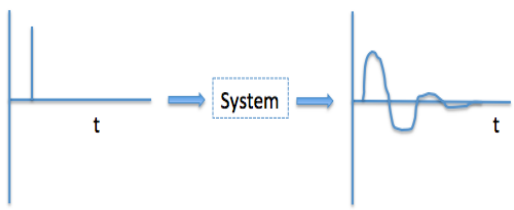

Figure 1: Dirac δ impulse, system under measurement and system IR

Received 25 May 2011
Accepted 10 June 2011

This method is accurate but,to give good results, input signal has to be as similar as possible to the ideal impulse. Unfortunately, in real conditions, Dirac δ is too short to contain enough energy and simple calculation shows that at normal PCM 44,1kHz sampling frequency, a single pulse lengthhas a 10^{-6}s magnitude order (about 22,68μs) and is absolutely too short to be correctly reproduced by any known power loudspeaker for at least 60dB above background noise. Even if this were possible, any small non-linearity might compromise good measurement results. A real situation is shown in Figure 2 [3], where at least transducerdistortions are surely added.

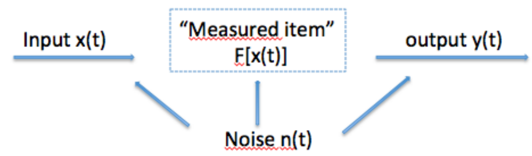

Figure 2: Linear system measurement with noise

2.2 *MLS method*

MLS signal is useful to measure any input-output system asmost equipment and room acoustics are. At the beginning (1965)MLS signal was generated by primitive polynomials and subsequently by digital shift registers. MLS signal is white noise comparable and white noise is non-periodical and random, thus its measurement requires long measurement time averaging (minutes) to be sure to correctly estimate its spectrum. In practice, this kind of signal consists of a randomly distributed sequence of same amplitude, same positive and negative impulses, so that the sequence is symmetrical around 0. This sort of signal contains at least 1 x 10^4 pulses per sequence, thus having higher energy content than the theoretical Dirac δ signal, which contains just one pulse per sequence [4]. MLS numerical sequence length is 2^n - 1, where n is the number of digital shift registers; in acoustics 16 digital shift registers are normally used implying 65.535 samples for a typical MLS room-acoustics measurement signal.

A very important feature ofMLS signal use is that its self-correlation calculation generates a perfect Dirac δ, as its self-correlation generates the single pulse shown in Figure 3.

Advantages of the pseudo-random MLS signal are 1) its complete frequency spectrum, 2) given its determined sequence, it is easy to mathematically find its time reversal MLS^{-1}, 3) because MLS is a binary sequence, its convolution product is simplified as multiplications become sums, 4) because of signal and time-domain convolution computational simplicity, it only requires Fast Hadamard Transform (FHT) and last but not least 5) given its flat sound spectrum, it is easy to calculate the IR signal frequency response.

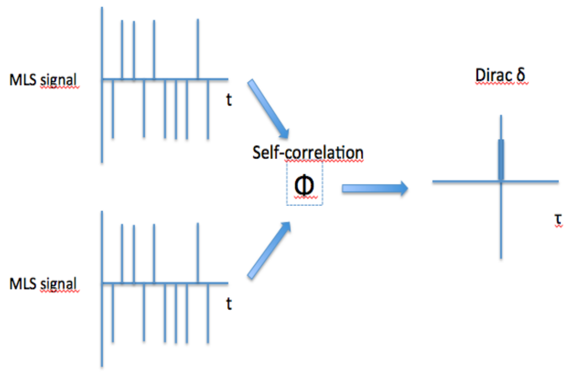

Figure 3: MLS self-correlation produces a Dirac δ

Any difference from Dirac δ shows acousticnon-linearbehaviour. From this difference,the measured item frequency response and many other significant parameters can be easily obtained through FFT. That is why this measurement method has been so popular between 1975 and 2000 (known as TEF or MLSSA), when computational power was still limited. In the particular case of room acoustics, it has to be noted that the MLS signal duration must be longer than room reverberation time RT_{60} to allowsound field to reach a steady-state in the room and thus ensure correct measurement results.

Thorough and extensive studies devoted to analysing this specific measurement technique have revealed its intrinsic limits as different types of distortion can occur in IRs because the entire system is not LTI, determining uncertainty in different results. Instead of anhomogenous Schroeder function decay, different peaks occur in measurement,mainly due to non-LTI electroacoustic system behaviour.

2.3 *Linear Sine-Sweep method*

Little by little other methods have entered the acoustic measurement field:first the linear Sine-Sweep method (LSS) better known as a time-delay spectrometry or TDS using TEF analyser and afterwards exponential sine sweeps (ESSs). Introduced by R. C. Heyser in 1967 [6],the LLS method uses a linear time-growing frequency to »sweep« the measured item, in this remembering W. C. Sabine's five different-frequency organ-pipes early experiments in room acoustics [2]. TDS specifically consists of a broad-spectrum technique using a signal stimulus mathematically expressed as $\exp[i\ \theta\ (t)]$; delay-tracking filters process the response of the measured item extracting useful informations.

Sine-Sweep techniques show important computational advantages as the test signal inverse filter $x(t)^{-1}$ is just test signal x(t) time reversal. Unfortunately, MLS FHT computational simplicity is lost but the actual computational power allows for a real-time and precise solution through specific »select-save FFT« fast convolution. The main advantage of Sine-Sweep techniques is that it becomespossible to discriminate distortions from the signal even if the whole measuring chain is not LTI or in any way non linear as for example electroacoustic transducers, as shown in Figure 4.

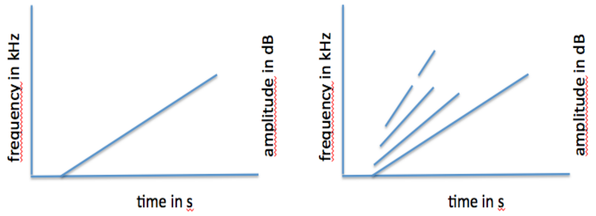

Figure 4: LSS and measurement with distortion products(upper not parallel)

Convolving y(t) output signal with$x(t)^{-1}$ test-signal inverse filter, it becomespossibleto separate high-order distortion products to obtain vertical right desired IR containing the F[x(t)] measured item behaviour, as shown in Figure 5.

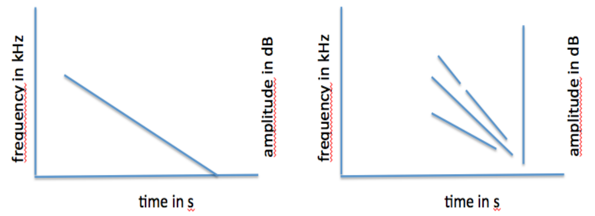

Figure 5: LSS $x(t)^{-1}$ inverse filter and IR in time-frequency domain

LSS techniquemain contribution is its possibility to discriminate between system IR and electroacoustic non-linearities. LSS technique hardly permits to well separate different harmonic-distortion products and unfortunately still suffers from time aliasing if output signal is not significantly longer than RT_{60}, in the last remembering MLS technique.LSS method is actually seldom in use in room acoustics because of the growing interest in ESSs.

2.4 ExponentialSine-Sweep method

An interesting development from LSS is ESS familyin which frequency-growth law can be arbitrarily chosen. ESS measurement technique is a frequency-determined sine sweep exponentially growing with time,which in the specific case of logarithmic lawis much more similar tohuman ear and music logarithmic behaviours. In this case, y(t) shows a parallel time-frequency logarithmic-plane slope of x(t) as shown in Figure 6.

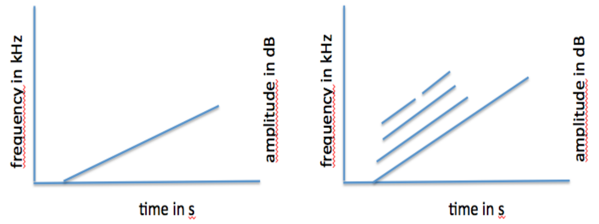

Figure 6: ESS and measurement with distortion products (upper parallel)

In the specific case of ESS, test-signal deconvolution inverse filter $x(t)^{-1}$ has to be slightly modified to avoid uncorrect frequency filtering,and to compensate ESS test signal which is energetically not frequency-flat but decreases by 3dB per octave, in this similar to pink noise. The solution is very simple consisting in a 6dB per octave positive-amplitude filtering directly during computational generation oftest signal deconvolution inverse filter $x(t)^{-1}$ [8].

As for LSS, ESS output $y(t) \otimes x(t)^{-1}$ inverse filter = IR and different harmonic non-linear contributions forestall IR. IR is on the right and other order harmonic distortions are now perfectly distinguishable on its left, in this being better than LSS, as shown in Figure 7. Some electroacoustic experiments showed up to 50 harmonic products, allowing very deep measured-item understanding.

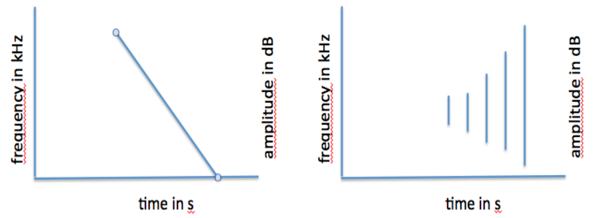

Figure 7: ESS $x(t)^{-1}$inverse filter and IR in time-frequency domain

3 EXPERIMENTAL RESULTS

During Trieste Politeama Rossetti 2009 measurement campaign, different measurement systems were used to investigate their result comparability.

The computer-generated Dirac δ (Section 2.1), MLS (Section 2.2), and ESS (Section 2.4), signals drovea 01dB acoustic-measurement power amplifier linked to a 01dB Dodecaedronacoustic-measurement loudspeaker system. Receivers were two Neumann KU100 dummy-head microphone systems. Even if planned, LSS-TDS (Section 2.3),and TEF equipment was unfortunately not available during the measurement campaign.

Starting from the beginning as stated in 2.1, a progressively more energetic Dirac δ was sent tothe power-amplifier and loudspeaker system, but rather than power amplifier temperature slight growth because of the joule effect, almost no appreciable impulse came from loudspeaker system. As expected, apart from some clicks from loudspeaker system tweeters, no appreciable Dirac δ resulted, because mid-frequency and even more low-frequency power loudspeaker moving system inertia was too big. Just to remember, a normal loudspeaker moving system is made of diaphragm, dust cap, voice coil, suspension system (spider + surround)voice coil connexion cables and voice coil front + rear moved air mass [9].

MLS and ESS methods were then compared using the same electroacoustic measuring chain in the same Trieste Politeama Rossetti theatre at the same time. Experimentswere made at various receiver positions and most interesting comparison results are here shown for the same microphone in the same position.

As expected, MLS technique showed to be sensitive toacoustic noise,uncorrect low-frequencies omnidirectional measurement loudspeaker system reproduction, infrasound, amplifier insufficient high-frequencies slew-rate, joule effect in the electroacoustic chain, harmonic and/or intermodulation products and many more as show different peaks in Figure 8. RT_{60} decay plot can be correctly calculated for 20dB only and so for RT_{20} only, not being sure if at lower energy levels or on the other hand in the remaining part of the RT (which is a time expression) the room behaviour is extensible.

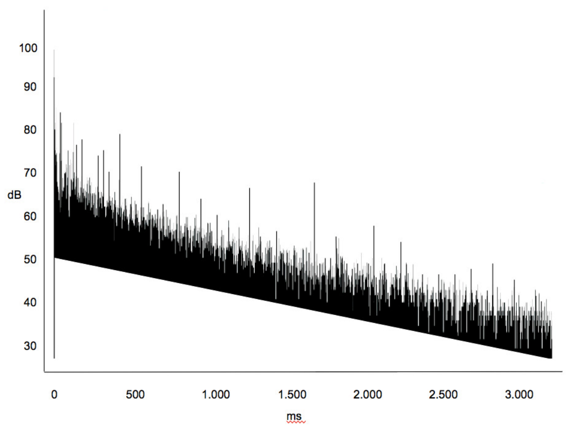

Figure 8: MLS measurement with distortion products

Even if still in use, MLS signal implies an incorrect or at least aleatory results risk. In these conditions, it normally becomes necessary to average many measurements(losing in result accuracy).Here is then discovered one of the main causes preventing further correct calculations during room acoustics measurement or restoration.

As expected, ESS techniqueallowedfor acoustic-system linear IR extraction with no electroacoustic distortions and quantifiedthe whole system non-linear response weighting every harmonic order IR. Linear IR measurement had better S/N and was completely exempt from system non LTI, as shown in Figure 9 and because averaging was not performed. A comparison between Figures 8 and 9 clearly shows a greatly better S/N, which can be correctly calculated for 60dB (now at lower energy levels, too) and RT_{60} is now surely detected in late room-acoustic decay also. Its slope is now evident and measurement results can be assumed much more correct, compared to MLS method.

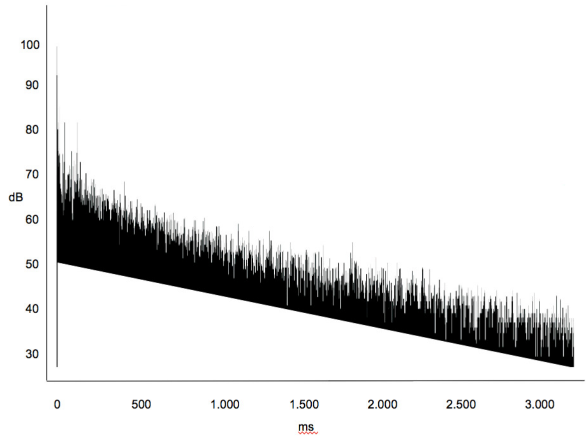

Figure 9: distortion products free ESS measurement

4 DISCUSSION AND CONCLUSION

As sound perception in humans is not homogenous [10], some acoustic objective evaluation parameters had to be developed to describe room acoustics as for example RT_{60}. Slow evolution in room acoustics seems nownot to be mainly due to recently solved computational requirements but to imperfect electroacoustic chain components and to measurement techniques. Electroacoustic-chain components drawbacks are related to component high costs and mainly to their physical limits, as linear power amplifiers do not exist as long as linear loudspeakers on 3D propagation planes [9]. Operators link the two main problems, as they are mainly not enough trained and measurements are consequently not comparable because ofISO 3382 uncertainties. Huge work in this particular applied physics sector is still needed, where ESS solved at least previously impossible discrimination between measurement chain distortion and room acoustic behaviour even in non LTI situations.

As theory first and experimental results afterwards demonstrate, ESS in room-acoustic measurements is a big step towards the comprehension of the room IRat a specific listener position and this is the main difference between MLS and ESS measurement methods. ESS shows the specific algorithm which defines the univoque relation between the sound source and the sound receiver towards the step by step less unknown behaviour of acoustic spaces.

ACKNOWLEDGEMENTS

The author acknowledges ing. Gianni Amadasi from 01dB Padova for the 01dB acoustic measurement power amplifier and 01dBDodecaedron loudspeaker system and ing. Stephan Peus from Neumann Gmbh Berlin for KU100 microphone systems lending during measurement sessions.

BIBLIOGRAFY

[1] ISO 3382 standard.

[2] Sabine Wallace Clement “Collected papers on acoustics”, Harvard University Press 1922 (post).

[3] Cuomo Mario “Analisi qualitativo-comparativa di sistemi di riverbero artificiali”, tesi magistrale, Adria 2009.

[4] Karjalainen Matti “Transfer-function measurements in audio”, (Eds.) *Äänenkäsittelyn seminaari 1996: Digitaaliaudion signaalinkäsittelymenetelmiä* (Seminar on audio signal processing: “Signal processing methods in digital audio”), Helsinki University of Technology , Laboratory of Acoustics and Audio Signal Processing, Report 41, Espoo, Finland 1996, pp.181-204.

[5] Schroeder Manfred Robert “Synthesis of low-peak-factor signals and binary sequences with low autocorrelation”, IEEE Trans. Information Theory, January 1970, pp.85-89.

[6] Heyser Richard Charles“Acoustical measurements by time delay spectrometry”, J. Audio Eng. Soc. 1967, 15 pp. 370-82.

[7] AA. VV. “Acustica musicale ed architettonica”, chapt. 19, ed. UTET, Torino 2009.

[8] Farina Angelo, Capra Andrea, Campanini Simone “La misura dela risposta all'impulso per la caratterizzazione di sistemi acustici e vibrazionali”, Seminario di Acustica “Strumenti e metodi di misura per l’acustica e le vibrazioni”, Università Politecnica delle Marche - Dipartimento di Energetica, Ancona 2008, pp. 75-90.

[9] Policardi Franz “Linearity and reliability in measurements”, European Acoustical Association - EAA Euroregio, Ljubljana 2010, Acta Acustica vol 96 pp. 96.

[10] Burnik Urban, Tasič Jurij Franc “Človeško zaznavanje in objektivno vrednotenje zakasnitev multimedijskih storitev”, Elektrotehniški vestnik, 2006, 73 (1), pp. 7-12.

Franco Policardiwas born in Trieste, Italy in 1963 where he graduated in 1983. He received his Master in Music technologies degree in Adria and Ph.D. degree in applied physics (Industrial engineering) from Bologna University in 2005 and 2010, respectively. From 1985 to 2002 he was incharge of top quality recording and live music concert amplification (Wiener Philharmoniker Orkester, Pavarotti and friends...) and has afterwards extended his early interests in room acoustics.Currently he works towards his second Ph. D. in Mechanical engineering at the L'Aquila University.