Volume 54, Number 1, 2013

ACTA TECHNICA NAPOCENSIS
Electronics and Telecommunications

ON THE CHOICE OF THE METHOD FOR OBTAINING THE ROOM IMPULSE RESPONSE

Adrian Călin FĂRCAŞ Marina Dana ŢOPA
Technical University of Cluj-Napoca
Str. G.Bariţiu nr.26-28, Cluj-Napoca, , tel.0264-401243, e-mail. calinf@radiocluj.ro

Abstract: The paper presents an overview on the direct or indirect methods used to obtain the acoustic room impulse responses. The analysis of the most relevant methods is based on experimental measurements and the goal is to achieve measurements for multiple position acoustic analysis in a live performance room. The objective acoustic parameters of an enclosed space may be obtained from the impulse response of the room.

Keywords: *Impulse Response, Maximum Length Sequence, Inverse Repeated Sequence, Time-Stretched Pulses, Logarithmic Sine Sweep*

I. INTRODUCTION

The objective acoustic characteristics of an enclosed environment are best described by the acoustic parameters. The *mother of all room acoustic parameters* is the reverberation time [1]. It is the time interval between the stop of a sound source and the 60 dB dim of the acoustic energy. Other relevant acoustic parameters are clarity C50, clarity C80 and definition D50 in accordance with ISO 3382 [2]. All these parameters can be evaluated if the room's impulse response is available.

The impulse response spectrum (*Figure 2*) can give information regarding the distribution of the signal's energy with respect to the frequency. For calculating the objective parameters it is compulsory that the impulse response's spectrum should exceed the noise level of the chamber in its rest state with 45 dB, within the respective frequency band [2].

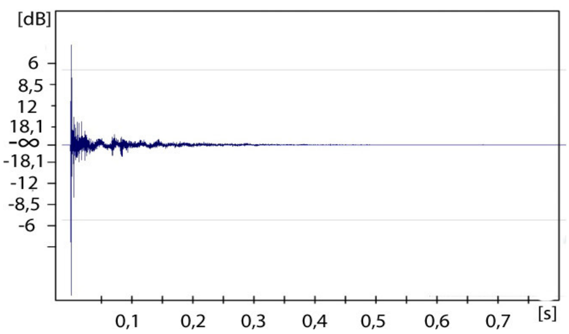

Figure 1. The response impulse in the time domain.

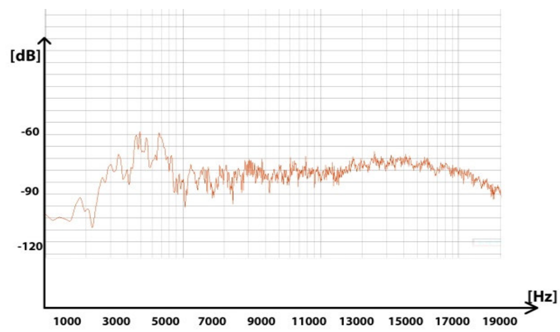

Figure 2. The impulse response's spectrum.

II. METHODS FOR OBTAINING THE IMPULSE RESPONSE

The methods for obtaining the impulse response can be separated in two categories: direct and indirect methods through convolution or deconvolution.

The direct methods of obtaining an impulse response use a sound source which produces an impulse type signal. The necessary equipment for this method comprises an impulse type signal source, a measurement microphone and a system for recording and analysis of the received acoustic signal, both with characteristics in accordance with ISO 3382 [2].The devices used for obtaining an impulse type signal are the balloons, firecrackers, pistols [3] and the most recent and popular device used in this field in Italy, *clappatore* [4]. Because of multiple disadvantages (pollution, fire, etc.) the balloons, the firecrackers and the pistols are avoided.

The indirect and most used methods are: MLS (Maximum Length Sequence), IRS (Inverse Repeated Sequence), Time-Stretched Pulses and SineSweep (linear or logarithmic) [5]. In the followings we considered only the indirect methods.

II.1. The MLS method

The MLS method was first proposed in 1979, by the German physician Manfred Robert Schroeder and was used with success for more than 20 years [5]. This method calculated for the first time the length of the sound dimming in an enclosed space, without using the direct impulse. It consists of producing a pseudo random

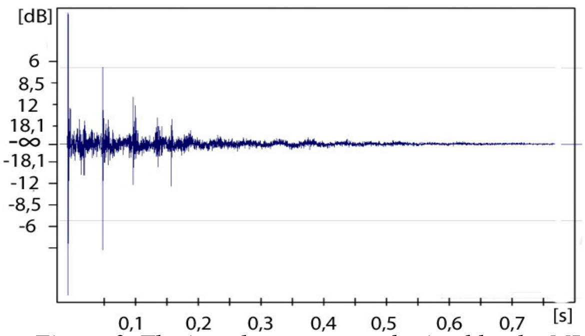

Figure 3. The impulse response obtained by the MLS method.

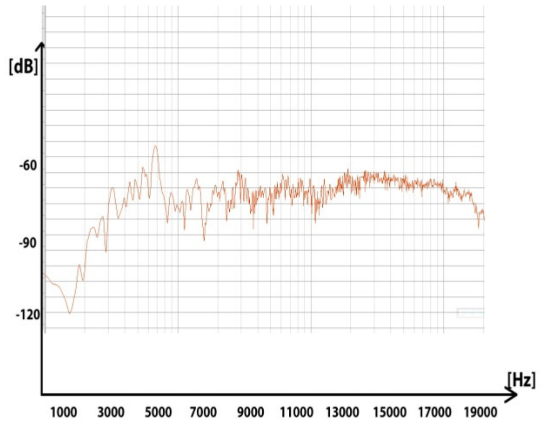

Figure 4. The impulse response's spectrum obtained by the MLS method.

signal with the same stochastic properties as the white noise in the measuring area, the receiving and recording of the sound in a given spot.

The correlation of the white noise with itself results in obtaining a unit impulse signal type [3], [6]. From the recorded signal the white noise is extracted by deconvolution, thus resulting the impulse response curve of the given area in the measuring point (*Figure 3*). The impulse response's spectrum is presented in *Figure 4*.

The MLS sequence of order *m*, has the maximum length for the impulse response $L = 2^m - 1$ samples. For a sampling frequency f_s, the measuring sequence is $t = \frac{L}{f_s}$ [3], [6]. For example, the AURORA plug-in, developed by Angelo Farina and attached to the Adobe Audition software, can generate a maximum 21th order MLS sequence, which means *L=2097151* samples, corresponding to a 47 seconds signal at a sampling frequency of 44,1 kHz [7].

The MLS method presents the advantage of making possible the correlation of the process with the Hadamard transform, resulting in a small number of mathematical operations in the frequency domain [3], [6], [8].

One can notice that (*Figure 3*), in the impulse response there are maxima and minima due to the inherent nonlinear errors of the play-record acoustic chain (distortion peaks unevenly distributed). The distortion peaks can generate anomalous values of the acoustic parameters. Using the playing, receiving, measuring and recording equipment in accordance to the specifications of ISO 3382, these errors are minimized [9]. However, this does not mean the elimination of the distorsion peaks which represents a disadvantage of the MLS method.

II.2. The IRS method

The IRS method was developed and proposed as a theoretical option for diminishing the distortion peaks that appear in the MLS method [5]. The IRS sequence $x[n]$ of *2L* length, is made of the MLS sequence with period $L(mls[n])$ doubled with its reverse:

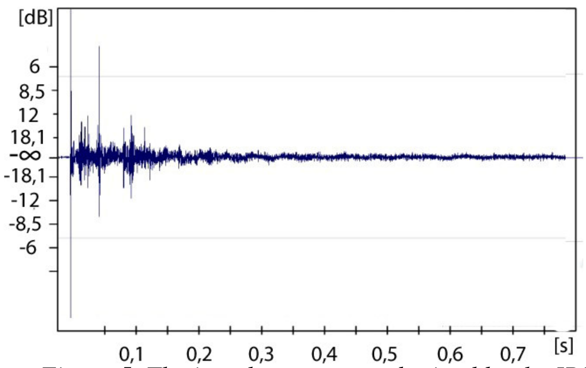

Figure 5. The impulse response obtained by the IRS method.

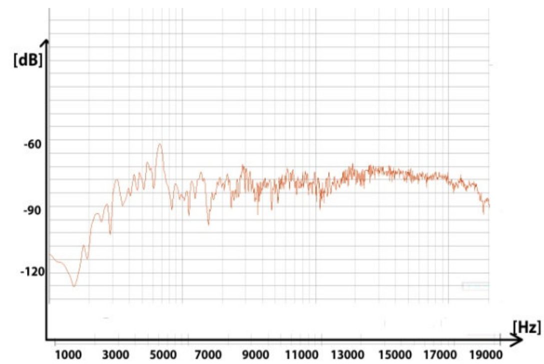

Figure 6. The impulse response's spectrum obtained by the IRS method.

$$x[n] = \begin{cases} mls[n] \ for \ n \ even \ with \ 0 \le n < 2L \\ -mls[n] \ for \ n \ odd \ with \ 0 < n < 2L \end{cases} \quad (2.1)$$

An advantage of this method is that the impulse response obtained by deconvolution presents more diminished distortion peaks than in the case of the MLS method (*Figure 5*). The IRS method is more robust to the changing in time of the measuring conditions than the MLS method.

In *Figure 6*, it can be seen that the impulse response's spectrum obtained by the IRS method is smoother than the one obtained by the MLS method (*Figure 5*). The disadvantage of this method is the longer time needed for the calculation of the deconvolution by using high order FFT and IFFT filters.

II.3. The TIME STRETCHED PULSES method

Short time after publishing the MLS method, Aoshima introduced the idea of TSP (Time-Stretched Pulses), which later on was taken and developed by Tatsuro Suzuki [5]. The development of the computerized techniques has allowed the digital processing of the high complexity signals.

The TSP method aims at the increasing of the SNR without increasing the system's nonlinearity, with the purpose of diminishing the peak distortions. They exist as a residue of the deconvolution filter.

The MLS, IRS and TSP methods are based on the thesis that the systems are time invariant. When the systems are varying, then the impulse response has nonlinear distorsions.

. II.4 The SINE SWEEP method

The Sine Sweep method was first proposed and developed by Angelo Farina. This method goes beyond the limitations of the previous methods and uses a sweep function with an exponential increase of frequency (*Figure 7*) [5].

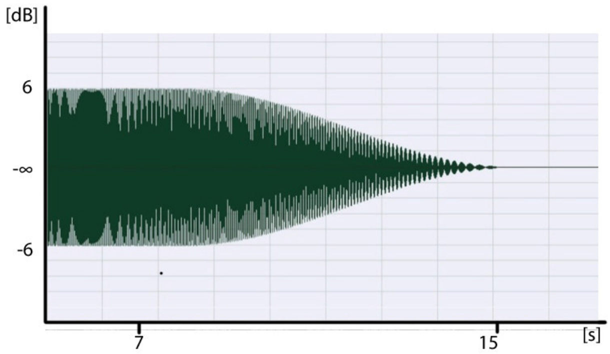

Figure 7. The Logarithmic Sine Sweep signal

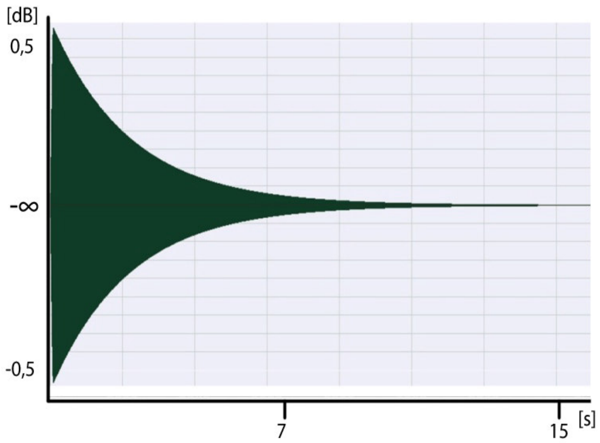

Figure 8. The Logarithmic Sine Sweep signal with reversed filter

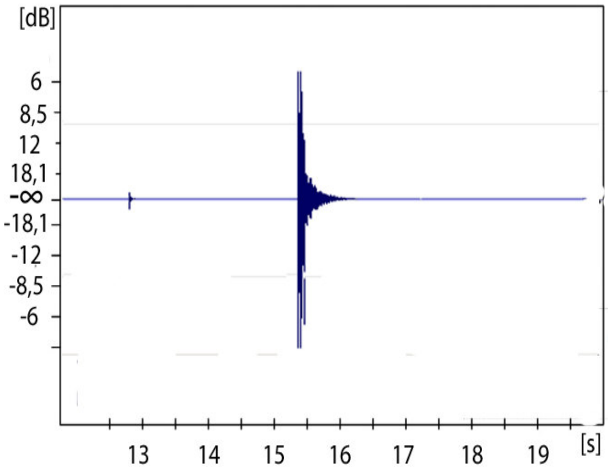

Figure 9. The impulse response obtained by the LSS method

The LSS (Logarithmic Sine Sweep) signal consists of:

$$x(n) = sin\left[\frac{T\omega_1}{\ln\frac{\omega_2}{\omega_1}}\left(e^{\frac{1}{T}\ln\frac{\omega_2}{\omega_1}} - 1\right)\right] \quad (2.2)$$

where and ω_1 and ω_2 are the initial and final frequencies of the signal, T is the length of the signal [9] [10] [5].

The logarithmic Sine Sweep signal emitted by the source is recorded and the recorded signal is subject to the convolution with the logarithmical sine sweep signal with its reversed filter (*Figure 8*). The result is two signals: the impulse response for the linear system and the impulse response for the nonlinear distortions. The advantage of

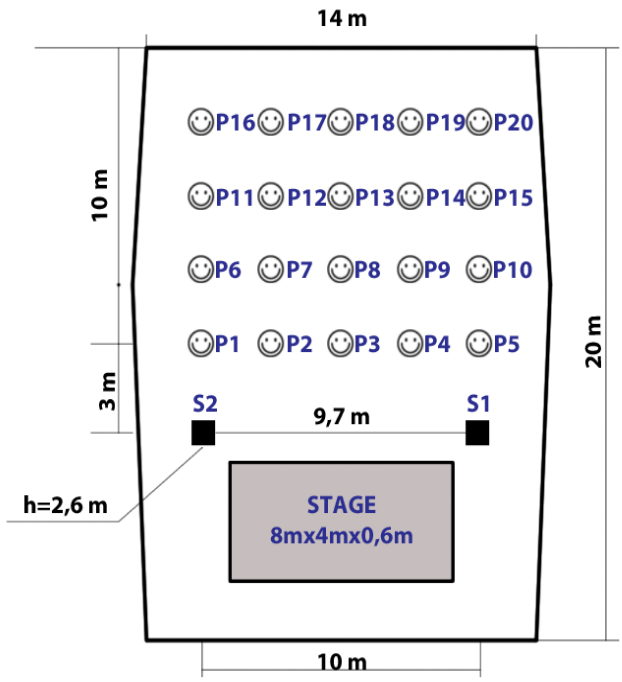

Figure 10. The positions of the measuement points and of the sound sources in the performance room.

this method is that the distortion peaks will appear before the impulse response and can be measured (*Figure 9*) [9] [5].

III. EXPERIMENTAL MEASUREMENT OF THE IMPULSE RESPONSE IN A PERFORMANCE ROOM

The impulse response has been obtained in the *Vasile Rebreanu* performance room from the Radio Romania Cluj studio. Twenty measuring points P1-P20 have been chosen on the area normally occupied by the audience and two points S1-S2 for the measuring signal emission in accordance to the usual positions of the loudspeakers used in that room (*Figure 10*) [2].

The measurements were made in a room considered fully furnished (150 upholstered chairs, a 32 square meters stage covered with carpets, a 2 square meters table and a 8 square meters place with measurements equipment).

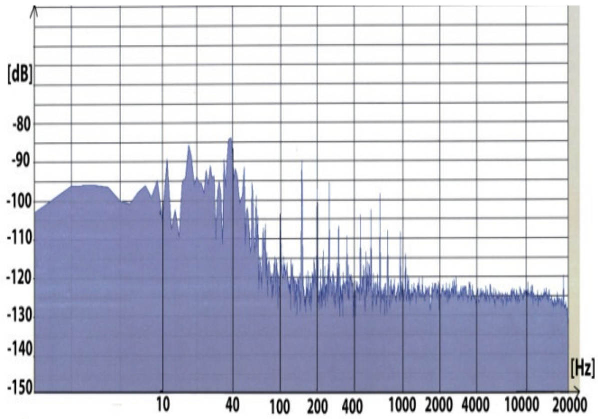

Figure 11. The frequency characteristic of the performance room in a resting state.

The MLS, IRS and LSS signals were successively emitted with the sound system from the performance room (JBL-VRX932LAP) [11]. We captured the sound with PCB130 type omnidirectional microphones [12], connected to an ASUS Lamborghini VX1 laptop [13] through a MOTU 896HD type external sound card [14]. The recorded signal was processed using the Adobe Audition software with the AURORA plug-in [7] and with SONY's Sound Forge 10 Pro [15].

Before starting the measurements the entire acoustic chain was calibrated. The noise level of the performance room was measured and the frequency characteristic was plotted (*Figure 11*). A low frequency noise of the amplifying system and of the illumination can be perceived.

Three measurement microphones were grouped in the P1 position and were calibrated by analogic adjustments of the MOTU 896HD external sound card. For calibration a 1 kHz to 1,55V_{ef} sine signal was used. The same manner was used for the positions P5 and P18.

III.1. Obtaining the impulse response by the MLS method

A pseudo random Multi MLS (MMLS) signal is generated; it has the same stochastically properties with the white noise with N=16 sequences, a 12 seconds length and 65 dB amplitude above the noise level (*Figure 12*).

The length of the sequence is of 2^{15}-1=32767 samples. The first sample is cast away because at this stage the system has not reached the equilibrium.

The frequency response of the JBL-VRX932LAP amplifying system is verified by plotting the frequency characteristic of the amplified MLS signal in the performance room and recorded in the P1 and P3 positions (*Figure 13 and Figure 14*).

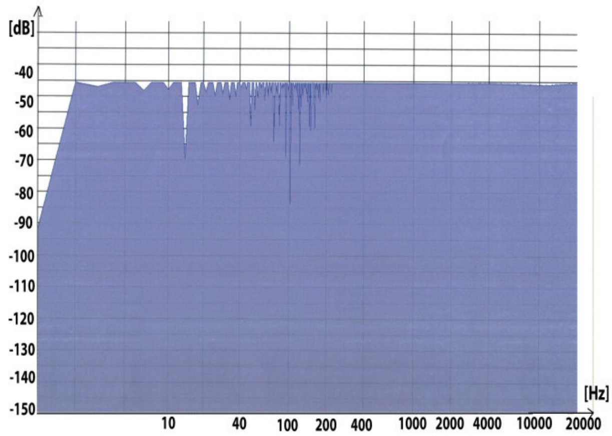

Figure 12. The MMLS spectrum.

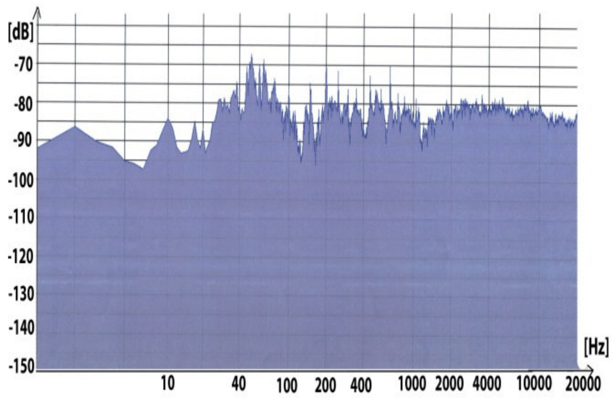

Figure 13. The spectrum of the MMLS signal recorded in the P1 position.

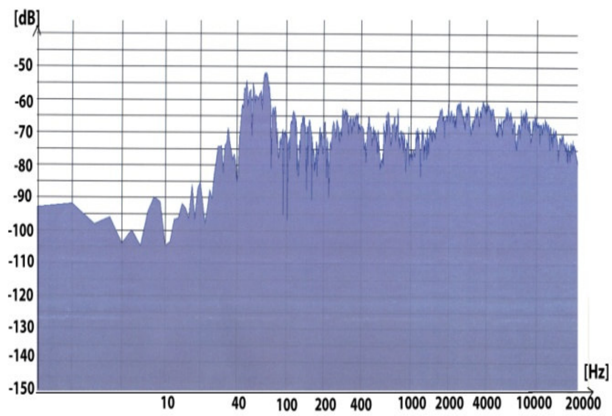

Figure 14. The spectrum of theMMLS signal recorded in the P3 position.

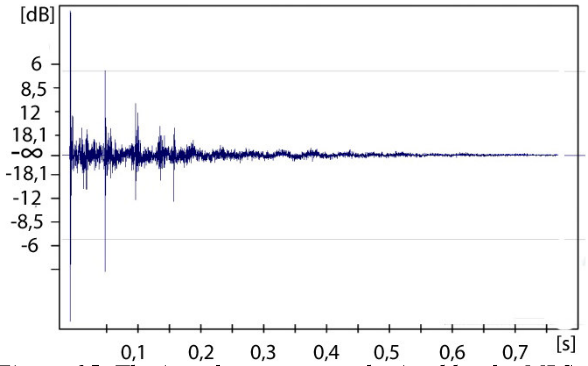

Figure 15. The impulse response obtained by the MLS method in position P10.

In the technical specifications of the JBL-VRX932LAP amplifying system its frequency response has the bandwidth between 57 Hz and 20 kHz. This aspect can be clearly noticed in *Figures 13* and *14*. The amplifying system does not respond to frequencies lower than 57 Hz and so we will ignore in the acoustic parameters analysis the values calculated for the first octave of 31,5 Hz.

The recorded signals in all of the twenty measuring points are subject of a deconvolution with the initial sequence and the impulse responses of the performance room for each point are obtained.

The impulse response in any measurement position will look similarly to the one from position P10 (*Figure 15*). It can be noticed that the impulse is followed by several distorsion peaks caused by the nonlinearity due mainly to the amplifying system.

By using the AURORA plug-in attached to Adobe Audition 3 objective acoustic parameters are extracted in each of the twenty measuring points.

III.2. Obtaining the impulse response by the IRS method

The experiments were conducted similarly to the MLS method: the IRS was emitted through the amplifying system of the performance room and the sound received by the measuring microphones in the preset audition positions is recorded. The recorded signal is subject to a deconvolution with the initial IRS signal thus obtaining the impulse response for each measuring point (*Figure 16)*. Next the objective acoustic parameters are calculated.

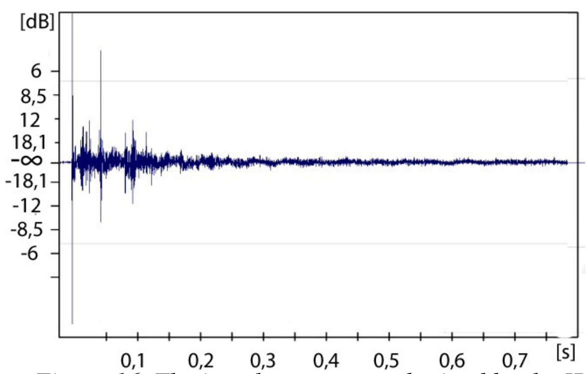

Figure 16. The impulse response obtained by the IRS method in position P10.

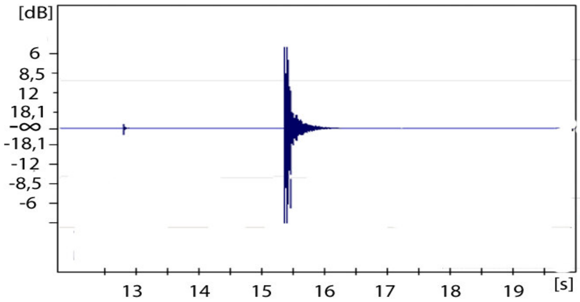

Figure 17. The impulse response obtained by the LSS method in position P10.

III.3. Obtaining the impulse response by the LSS method

The following signals are generated: a Logarithmic Sine Sweep (LSS) between the 22 Hz - 22 kHz frequency, with 65 dB for 15 seconds followed by 0 dB signal for 10 seconds (*Figure 7*) and the signal through the reversed filter for the convolution of the recordings in the measuring points (*Figure 8*).

The first signal is emitted through the JBL-VRX932LAP sound system. In a measurement point this sound is being received, recorded using the measurement microphones and the recording system. The recorded signal is then subject to the convolution by the second signal obtained with the reversed filter, thus obtaining the impulse response in the measuring point. The same procedure will be used for each of the 20 measuring points and obtained the impulse response in each of them.

The impulse response of the linear system is obtained preceded by false responses of the nonlinear systems (*Figure 17*).

After eliminating the impulse responses of the nonlinear systems, the acoustic parameters are calculated.

IV. EVALUATON THE ACOUSTIC ROOM PARAMETERS

In each method the following objective acoustic parameters were extracted: clarity C50 and C80, definition D50 and reverberation time T20 and T30 in according with ISO 3382 [2]. These parameters are the most significant for the acoustic description of a room.

The results were gathered for each point, each parameter and each measuring method.

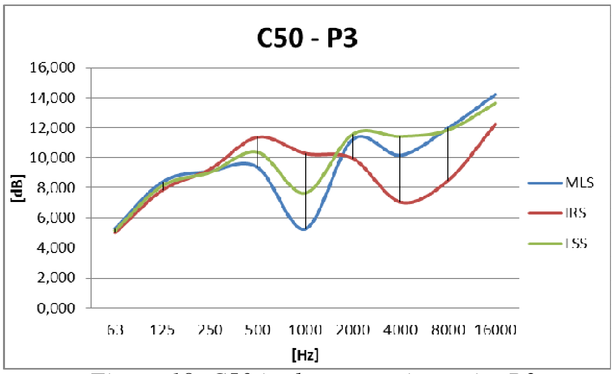

Figure 18. C50 in the measuring point P3.

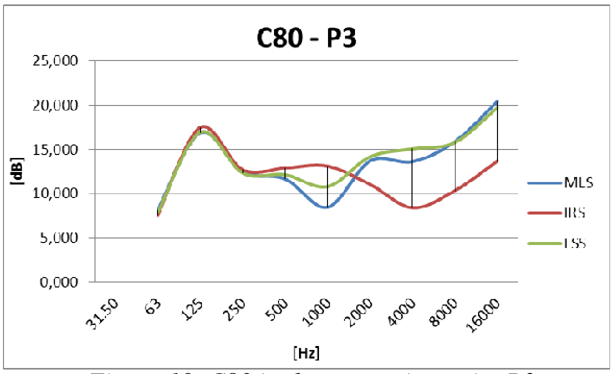

Figure 19. C80 in the measuring point P3.

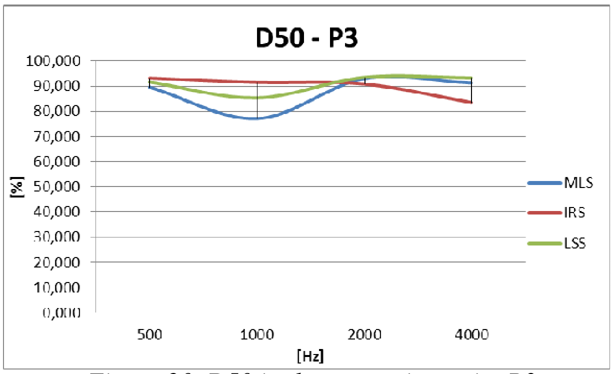

Figure 20. D50 in the measuring point P3.

The points P3, P8, P13 and P18 correspond to the good audition seats. To illustrate the results we selected the plots of the parameters in the point P3 (*Figures 18* to *22*). It can be noticed that the IRS method gives aberrant values far away from the values obtained by the other two methods.

In terms of reverberation times T20 and T30, in all twenty measuring points, the values obtained by using MLS and LSS, are almost identical and far away from the IRS values (*Figure 21, Figure 22*).

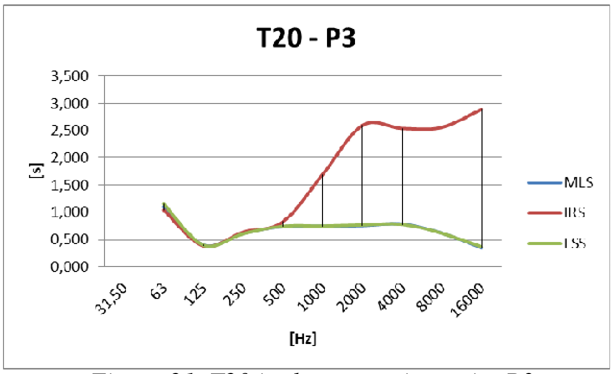

Figure 21. T20 in the measuring point P3.

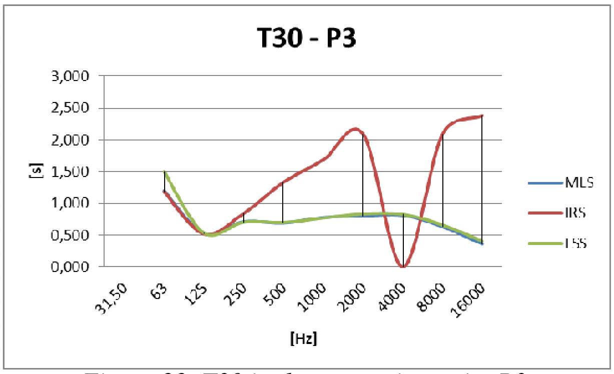

Figure 22. T30 in the measuring point P3.

In the followings we will compare only the MLS and LSS methods and leave the IRS as inefficient. (*Figures 23* to *25*).

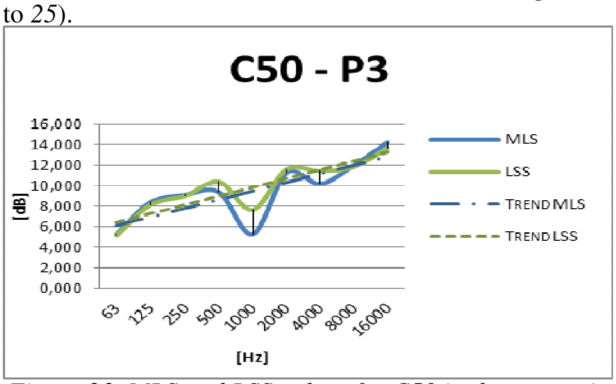

Figure 23. MLS and LSS values for C50 in the measuring point P3.

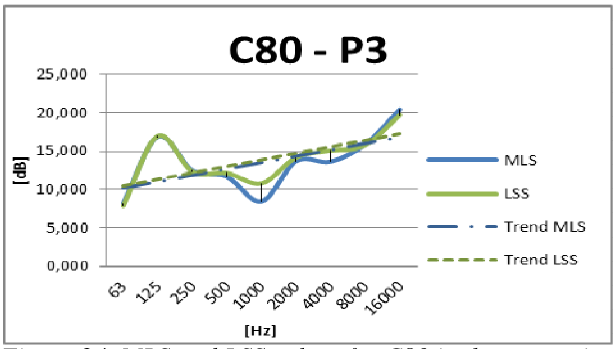

Figure 24. MLS and LSS values for C80 in the measuring point P3.

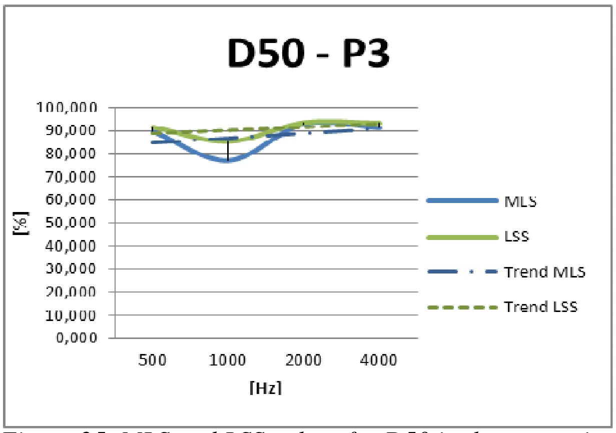

Figure 25. MLS and LSS values for D50 in the measuring point P13.

We have analyzed each pair of MLS and LSS plots from the *Figures 23* to *25* with respect to two aspects:

- The speed of the parameter's variation between two octaves;
- The closeness of the plots to the global trend of the parameters.

We compared the MLS and LSS plots in each of the measuring points and selected the plot with a slower variation and closer to the global trend. The selected plot has a better stability for the given geometry and noise level of room.

The results regarding the method's selection are drawn in *Figures 26 to 28*.

The measuring points where MLS was better than LSS are coloured in blue and when opposite in green. When none of the methods prevailed, the points are coloured in white.

In Figure 26, the clarity C50 clarity is computed better with LSS method in 14 measuring points and with MLS in 6 points.

In Figure 27, for C80 there are 10 "LSS" points, 9 "MLS" points and 1 point where none of the two methods was clearly prevalent. In Figure 28, for D50 there are 10 "LSS" measuring points, 8 "MLS" points and 2 "LSS equal MLS" points in which none of the two methods were clearly prevalent.

From the analysis of the maps from Figures 26-28 it is easy to see a slight advantage in using the LSS method for obtaining the objective acoustical parameters in the *Vasile Rebreanu* performance room from Radio România Cluj and hence for obtaining the impulse response of that specific hall. Using the Adobe Audition application with the AURORA plug-in, the LSS method was quicker, although the logarithmic sine sweep signal was more unpleasant in usage.

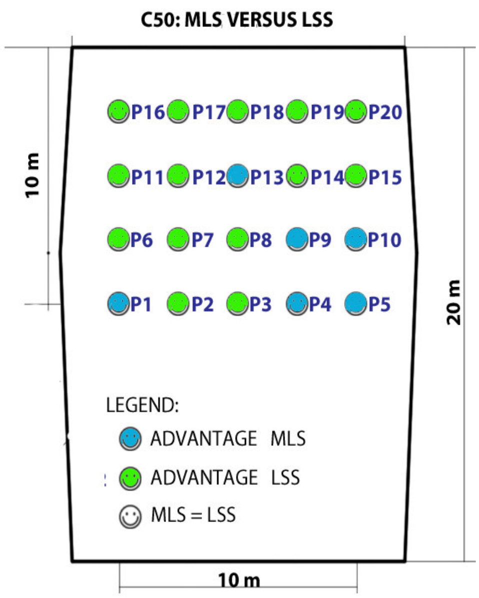

Figure 26. Selection of MLS or LSS with respect to C50 values in the measuring points.

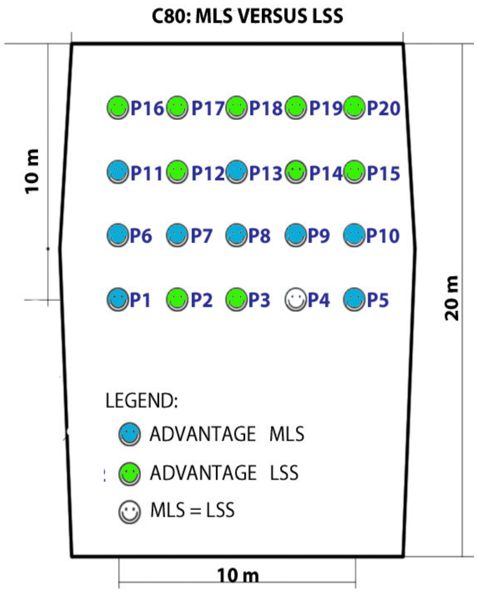

Figure 27. Selection of MLS or LSS with respect to C80 values in the measuring points.

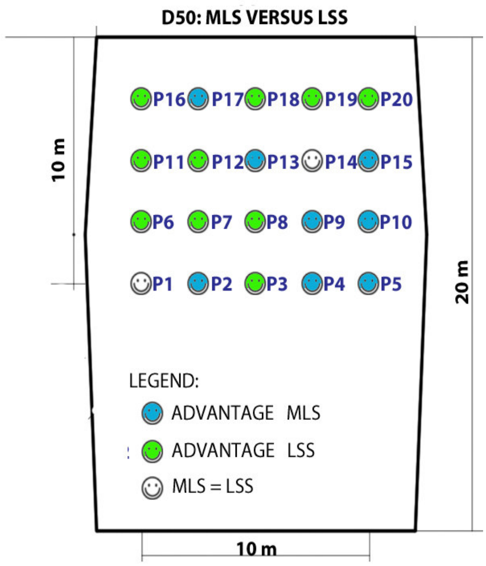

Figure 28. Selection of MLS or LSS with respect to D50 values in the measuring points.

V. CONCLUSIONS

The acoustic analysis of a room implies the measurement of the impulse room. The paper is dedicated to the choice of the best method to measure the room's impulse response among the most relevant methods: Maximum Length Sequence (MLS), Inverse Repeated Sequence (IRS) and Logarithmic Sine Sweep (LSS).

The measurements were carried out in the *Vasile Rebreanu* performance room from Radio România Cluj in 20 measuring points in the area normally occupied by the audience. From the resulted 20 impulse responses we extracted the objective acoustic parameters in accordance with the ISO 3382 regulations. Comparing the parameters we noticed aberrant values obtained by the IRS method, at frequencies higher than 1 kHz for clarity and definition and at frequencies higher than 250 Hz for the reverberation time.

We focused on the analysis of the MLS and LSS methods regarding the clarity and definition values, the reverberation time being identical for the two methods. By comparing the parameters plots with respect to the frequency, taking into consideration the speed variation and the global trend, we consider that in the conditions of the given geometry and noise level, the most stable and suitable method for obtaining the impulse response in multiple positions of audience is the LSS method.

On the other hand, the Vasile Rebreanu performance room from Radio România Cluj is a 3370 cubic meters chamber with the walls and ceiling covered in wooden planks, and the floor covered with epoxidic clay.

Analysing the values of the C50 and C80 clarity parameters, we conclude that this is a deaf hall, unsuitable for orchestral music concerts with audience and less suitable for vocal performances. The human voice intelligibility, illustrated by the D50 definition parameter (the D50 values are high above the 50% level on the entire audition zone for the 5^{th}, 6^{th}, 7^{th} and 8^{th} octave bands).

In terms of reverberation time, the hall is suitable for concerts of rock music and theatre performances. Nonetheless the room can be successfully used as for cinema or lecture room, especially in the positions P11-P20.

REFERENCES

[1] Magne Skalevik, *Reverberation time – the mother of all room acoustic parameters,* BNAM 2010, Bergen Norway 10-12 may 2010

[2] International Standard ISO 3382 – 1, *Acoustics — Measurement of room acoustic parameters - "Part 1: Performance spaces*" – 2009

[3] Franco Policardi, *MLS and Sine-Sweep measurements* Università di Bologna, Italia ELEKTROTEHNIŠKI VESTNIK 78(3): 91-95, 2011 ENGLISH EDITION

[4] Armani Alberto (SPECTRA SRL – ITALIA) – *Clappatore Eccitazione con sorgente impulsiva* – http://www.spectra.it/download/PDF -PRESENTAZIONI-FLIP/Clappatore/index.html , septembrie 2009

[5] Stan Guy-Bart, Jean-Jacques Embrechts, Archambeau Dominique *Comparison of different impulse response measurement techniques*, Sound and Image Department, University of Liege, BELGIUM, December 2002

[6] Jens Hee, *Impulse response measurements using MLS*, http://jenshee.dk, august 2003

[7] http://pcfarina.eng.unipr.it/Aurora_XP/index.htm

[8] Ivo Mateljan, Kosta Ugrinovic, *The comparison of room impulse response measuring system*, Faculty of Electrical Engineering, University of Split, Croatia

[9] Angelo Farina, *Advancements in impulse response measurements by sine sweeps*, Convention Paper Presented at the 122nd AES Convention 2007 May 5–8 Vienna, Austria

[10] Angelo Farina, *Simultaneous Measurement of Impulse Response and Distortion with a Swept-Sine Technique*, Convention Paper Presented at the 108nd AES Convention February 2000

[11]http://www.jblpro.com/catalog/general/Product.aspx?PId=33&MId=2

[12]http://www.pcb.com/TestMeasurement/Acoustics/Prepolarized_Array.aspx

[13]http://hothardware.com/Reviews/ASUS-Lamborghini-VX1-Notebook/

[14]http://cdn-data.motu.com/manuals/firewire-usb-audio/896HD_Manual_Mac.pdf

[15] http://www.sonycreativesoftware.com/soundforge