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ACOUSTIC MEASUREMENTS IN OPERA HOUSES: COMPARISON BETWEEN DIFFERENT TECHNIQUES AND EQUIPMENT

P. FAUSTI

Department of Engineering, University of Ferrara I-44100 Ferrara, Italy

AND

A. FARINA

Department of Industrial Engineering, University of Parma, I-43100 Parma, Italy

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In room acoustics, many objective parameters to quantify subjective impressions have been introduced. These quantities can be measured by using a wide variety of powerful tools and equipment. The results can be influenced by the measurement techniques and instruments used. Furthermore, the results also depend on the measurement positions and on the condition of the hall (full, empty, etc.). The aim of this work is to define a tightly standardized measurement procedure for the collection of a complete objective description of an opera house's acoustics. In this paper some of the results obtained by the authors after measurements made in three different halls are presented. Comparisons were made both between different hardware and software tools (real-time analyzer, DAT, PC-board, source, microphones, post-processing software) and between different measurement methods (interrupted stationary noise, true-impulse, pseudo-random white noise with impulse–response doconvolution, sine sweep) as well as between different positions in the halls, with and without the presence of musicians and audience. The results have shown that the differences obtained when using different measurement techniques and equipment are not of significant importance. The only effective differences were found regarding the recording techniques, as the monaural measurements give appreciably different results from the average of left and right channel of binaural measurements. Slightly different results were also found between true impulsive sources (pistol shots, balloons) and omni-directional (dodecahedral) loudspeakers. Attention must be paid to the signal-to-noise ratio, as this can influence the correct calculation of some acoustical parameters. Some differences, not as great as expected, were found in the results with and without the musicians in the orchestra shell and with and without the audience in the hall. This is probably due to the high sound absorption that is typical in Italian opera houses even without an audience. However, important differences were found in the calculation of some acoustical parameters, particularly clarity $C80$, by changing positions in the hall.

1. INTRODUCTION

Measurements in room acoustics can be made by using a wide variety of powerful tools and equipment. The number of different combinations of tools, equipment, techniques and methods could be very large. The results are clearly influenced by these different settings, but it is not yet known how important these differences can be. The results also depend on the position of the listener and on the condition of the hall in which the measurements are made. The aim of this research is to find a procedure to qualify an opera house, which will always give comparable and reproducible results. The procedure must ensure that different researchers, with different measurement apparatus, will obtain the same results within a predefined admissible tolerance roughly corresponding to the subjective discrimination threshold for each objective quantity [1]. The choice of the preferred measurement methods, post-processing procedures and objective parameters to be retained will only be made after contrasting experimental results obtained by different research groups. The comparison takes into account the measurement techniques, the equipment, the measurement positions and the condition of the audience and the stage (e.g., empty, with the presence of orchestra equipment and/or musicians, with or without the presence of audience in the hall). In this paper, the results obtained by comparing the measurements in three different halls are reported. In hall 3, the Teatro Comunale in Ferrara, measurements were repeated twice: the first time with and without the musicians on the stage, and the second one with and without the audience in the hall. The Teatro Comunale in Ferrara is a typical Italian opera house but the measurements were made in both cases in the concert-hall configuration with an orchestra shell on the stage. Measurements were made in order to calculate the main objective parameters introduced to quantify subjective impressions in room acoustics. Many studies have been made in this field [2–5] and many objective parameters have been introduced both for the audience and for the musicians. In this research, the parameters used for the comparison are the following.

Clarity $C50$ *and* $C80$. This is the ratio, expressed in dB, between the "useful energy", which is received in the first 50 (80) ms of the impulse response, and the "detrimental energy", which is received after that. The "energy" is estimated by squaring the instantaneous values of the pressure impulse response although, particularly in the late reverberant tail, a correct energetic analysis of the sound field is generally much more complex [6].

Center time (TS). This is the first order momentum of the squared pressure impulse response, along the time axis, starting from the arrival of the direct wave. It is usually expressed in milliseconds.

Early decay time (EDT), reverberation time $T15$ and $T20$. These values of the reverberation time are estimated by the slope of the Schroeder-backward-integrated decay, respectively, in the dB ranges $[0, -10\}$ (EDT), $[-5, -20]$ (T15) and $[-5, -25]$ (T20).

Inter aural cross-correlation (IACC_E). This parameter comes from a binaural impulse response measurement, in which two impulse responses are measured through microphones located at the ear-canal entrances of a dummy head, aimed at the sound source. IACC_E is the maximum value of the normalized cross-correlation function computed for ± 1 ms (in the first 80 ms) of the two aural impulse responses.

Strength index (G). This parameter simply expresses the difference (in dB) between the sound pressure level (SPL) measured at the receiver position and the SPL produced by the same omni-directional source, in a free field, at a 10 m distance. In practice, it is obtained by the difference between the SPL and the SPL of the source, adding 31 dB.

These parameters are described in more detail in the ISO 3382 standard [7]; most of them can be calculated from the impulse response of the hall relative to the positions of the source and receiver. The impulse response is therefore the main characteristic needed for any comparison inside a hall.

2. MEASUREMENT TECHNIQUES

Measurements were made mainly in accordance with the ISO 3382 code, which describes the measurement techniques that could be used for the determination of the impulse response and the main characteristics that should be fulfilled by the equipment.

The measurement techniques used in this research are the following: technique based on the use of a real-time analyzer; technique based on the digital recording of the impulse response generated by impulsive sources (balloons or pistol shots) and its subsequent analysis; impulsive technique based on the deconvolution of a steady pseudo-random test signal (MLS); impulsive technique based on the deconvolution of an exponentially sweeping sine wave test signal.

The technique based on the use of a real-time analyzer enables the user to measure directly a number of very important acoustical parameters such as the reverberation time, the sound level, the frequency response of the hall and the sound strength index, without the recording of the impulse response. A loudspeaker fed by a signal coming from the analyzer itself or (only for the reverberation time) an impulsive source can be used as sound sources.

All the other techniques are based on the computation of the impulse responses of the hall for each particular couple of source and receiver positions. From the impulse response, it is possible to calculate almost all of the most important acoustical parameters. The reverberation time is calculated through Schroeder's backward integration [8, 9]. By using two recording channels with a binaural microphone, it is possible, through the subsequent analysis, to calculate the value of the $IACC_E$.

The procedures based on the recording of the impulse response generated by impulsive sources (balloons or pistol shots) and its subsequent analysis could be carried out by using a small portable digital recorder (DAT) or directly a personal computer equipped with a sound board. Since the sound source is not stable and repeatable and does not have a normalized spectrum, it is not possible to obtain information neither about the absolute sound spectrum produced by a room, nor about the absolute value of the sound pressure level.

The impulsive technique called maximum length sequence (MLS) is based on the deconvolution of the response of the hall to a deterministic pseudo-random test signal. By using Hadamard's fast transformation [10] it is possible to obtain the correlation function between the test signal and room's response, which gives the impulse response directly in the time domain. As the MLS technique is based on a deconvolution of deterministic sequences, it is useful only for a time-invariant system. The signal-to-noise ratio can be improved by averaging many sequence repetitions. In this research we used two available MLS analysis systems. The system called MLSSA [11] uses a dedicated data acquisition board (A2D160), which also generates the deterministic pseudo-random signal with a maximum length of 65 536 points; the hardware generation ensures tight matching between generation of the signal and recording. This enables the system to calculate a decay of the sound field over 1·5 s with a sampling rate of 44·1 kHz. The acquisition board used gives the system a good signal-to-noise ratio and the dedicated software allows the direct calculation of all of the above-mentioned acoustical parameters. The system works only with one channel, although the acquisition board has two channels. The system called AURORA [12] uses a standard PC soundboard driven by software both for the generation and for the recording of the signal. The maximum length of the sequence is more than 2 million points and this permits the calculation of a very long decay; furthermore, the system can work with more than one channel both for generation and sampling, depending on the number of available channels on the soundboard employed. The signal-to-noise ratio also depends heavily on the quality of the soundboard used: although nowadays multi-channel sound boards equipped with 20- or even 24-bit converters are readily and cheaply available, in this case the Sound Blaster 16 soundboard already included in a notebook PC was employed, with obvious detrimental effects on the S/N ratio.

The new technique based on an exponentially sweeping sine wave test signal was used for the determination of the impulse response in hall 3 and was recently developed by one of the authors [13]. Although this technique is apparently similar to previously employed linear sweeping sine wave methods, such as TDS [14] and stretched pulse [15, 16], the exponential sweeping technique is quite new, and thus a more detailed explanation is needed here.

The CoolEditPro multi-channel wave editor program was employed as host program for the specialized plug-ins for generating the test signals and for deconvolving the impulse response. A first plug-in generates the test signal and also preloads in the Windows clipboard the proper inverse filter: this is simply the time reversal of the excitation signal, with an amplitude shaped according to the inverse of the spectral energy contained in it. This shaping is not necessary with a linearly swept sine, and it is the most innovative modification over the previous techniques. Figure 1 illustrates a very short excitation signal and its inverse filter.

Owing to the synchronous Rec/Play capabilities of CoolEditPro, the response of the system can be sampled simultaneously with the emission of the test signals: some repetitions are made, in order to ensure that the system has reached the steady state, and usually the response to the second or third repetition is analyzed.

To recover the system's impulse response, the inverse filter is simply convolved with the recorded system's response, owing to a second specialized plug-in. This method proved to be substantially superior to the maximum length sequence

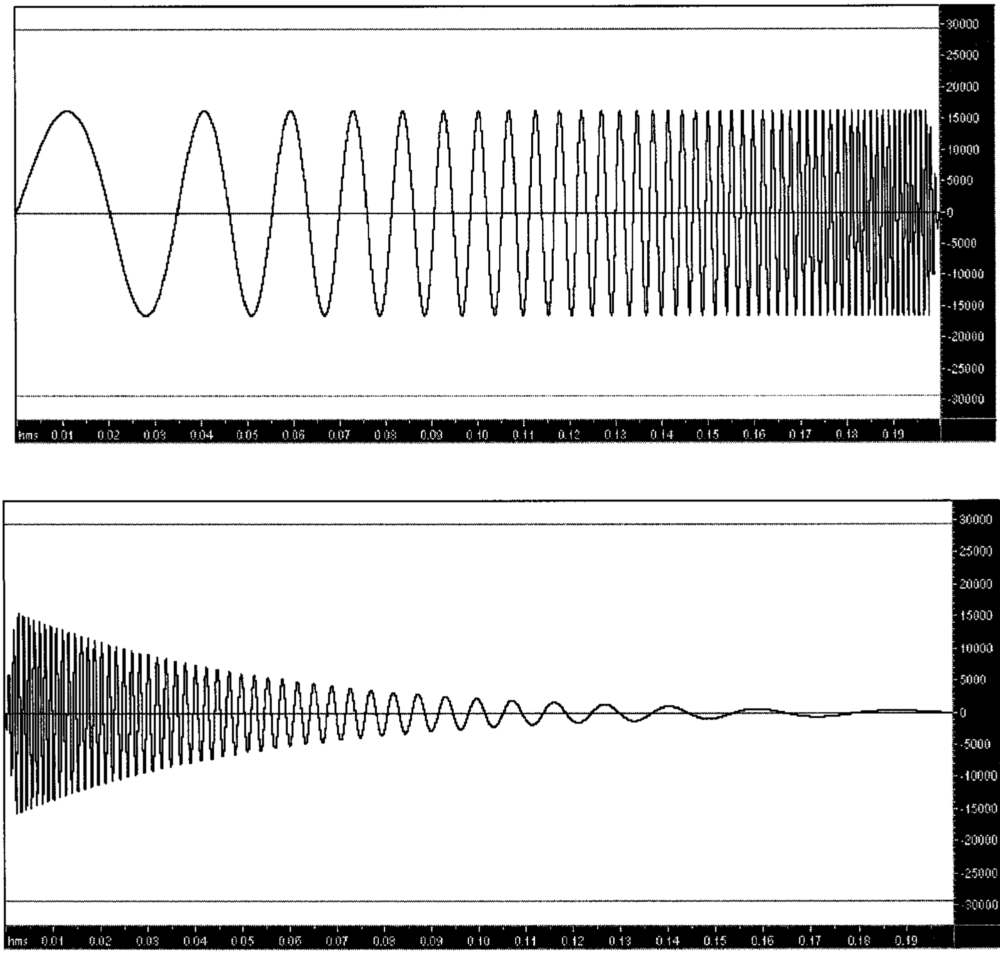

Figure 1. Test signal (above) and inverse filter (below).

(MLS) method previously employed [12]: when making use of the same excitation length, the S/N ratio is better, particularly at low frequency, thanks to the "pink" shape of the excitation spectrum, and the measurement is almost immune to non-linearity and time variance. Close matching between the clocks of the signal generation and sampling is no longer an issue (two different machines can be used without any problem). In addition, by properly setting the frequency limits for the sine sweep, it is possible to avoid damaging the transducers by applying too much signal outside their rated frequency response limits.

3. EXPERIMENTAL MEASUREMENTS

Measurements were made by employing the following instruments: two omni-directional dodecahedron loudspeakers, two binaural microphones, two

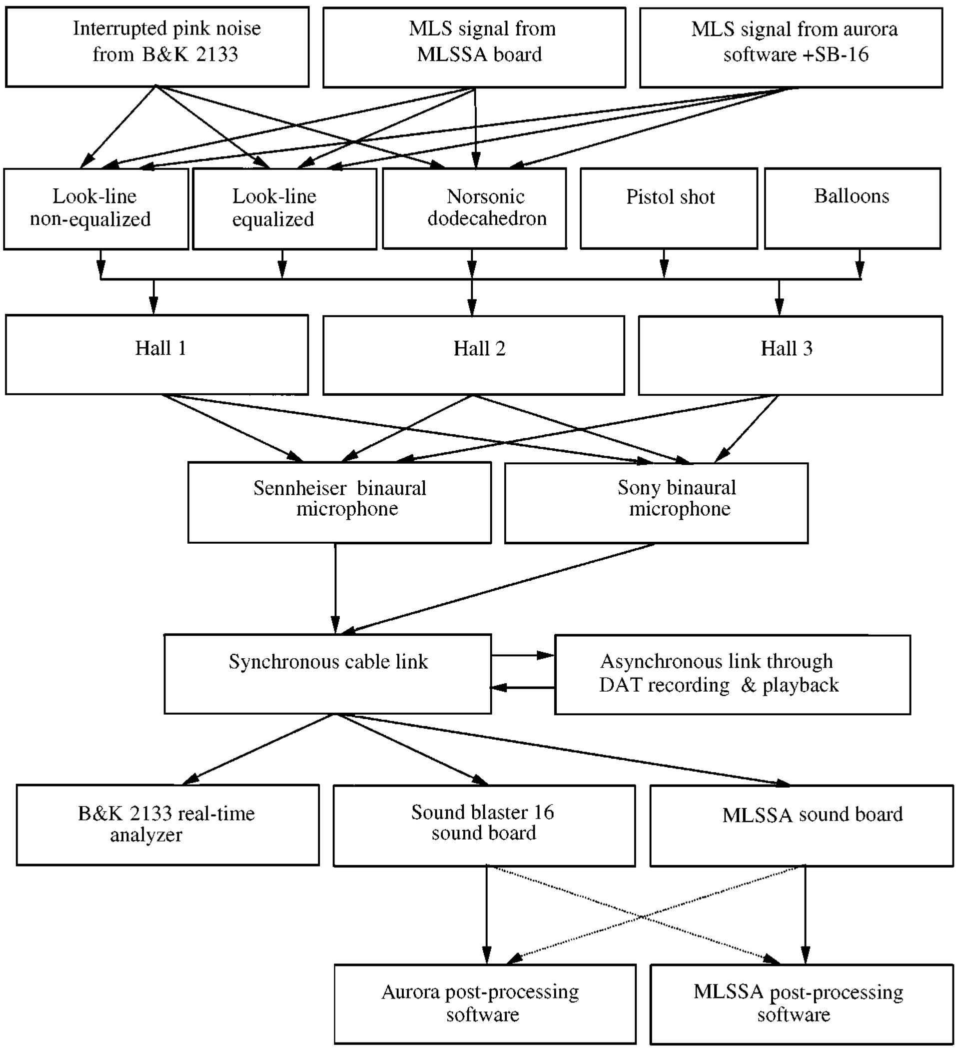

Figure 2. Block diagram of the instruments and measurement techniques employed.

computer-based MLS measurement systems, a real-time spectrum analyzer, a DAT recorder and two impulsive sound sources (pistol and balloons). Almost all combinations of these instruments were checked, although in the following only the most relevant comparisons are reported.

Also, different post-processing techniques of the same experimental results were attempted; the number of possible combinations was thus increased. Most of the comparisons were made in two halls.

More specifically, the following comparisons were made: measurement of the reverberation time with the standard interrupted-noise method and with the backward-integration of the impulse response; MLS measurement of the impulse response with the two available systems and with synchronous/asynchronous correlation; measurement of the impulse response with impulsive sources (pistol shots, explosion of balloons); employment of two different dodecahedron loudspeakers, one of which has an optional electronic equalization circuit; employment of two different binaural microphones (on the same dummy head).

In Figure 2, a block diagram with the instruments and measurement techniques employed is shown. An attempt was made to maintain all the other variables unchanged when checking the effect of each of the above combinations. All the instruments employed are claimed to comply with the ISO 3382 standard.

In one of the halls measurements were repeated and the following comparisons were made: measurement of the impulse response on the stage and in the stalls with and without the presence of orchestra and choir inside the orchestra shell; this comparison was made without the audience in the hall; measurement of the impulse response in different positions of the hall with and without the presence of the audience in the hall; this comparison was made without the orchestra and choir but with their equipment inside the orchestra shell.

The measurements were repeated with various source and receiver positions, but great care was taken to ensure that these positions remained absolutely unchanged between the different sets of measurement. Furthermore, for each comparison a highly significant acoustical parameter was chosen, although the whole set of parameters was computed for each measurement set-up.

4. ANALYSIS OF THE RESULTS

Figure 3 shows the comparison between the reverberation times measured in hall 1 with the real-time analyzer (interrupted-noise method) and with the backward integrated impulse responses: these were obtained both with the MLS technique and with pistol shots. We found that the major differences are not between stationary and impulsive techniques but between stationary and impulsive sources.

Figure 4 shows the comparison between the signal-to-noise ratios obtained with the two MLS systems and two impulsive sources (balloons and pistol shots). The measurements based on the MLSSA board seems to have a better signal-to-noise ratio than those obtained using a standard Sound Blaster 16 PC board.

The comparison between the MLS direct measurements (synchronous correlation) and the measurements made after the DAT recording of the noise (asynchronous correlation) gave the same results. The MLS system based on the Aurora software has the advantage of permitting direct binaural measurements. As the results of the two MLS systems can be processed with both software tools, it was checked that the computation algorithms are perfectly interchangeable.

The effect of employing loudspeakers with very different frequency response was studied in hall 2, employing two different dodecahedrons (Norsonic and Look Line); the latter has two switchable frequency responses (unequalized and equalized). Despite the large discrepancy between the loudspeakers, the differences between the two results concerning clarity are less than 0·8 dB in the frequency range of interest (see Figure 5). The substantial coincidence of the results with sources having such a great difference in frequency response means that the

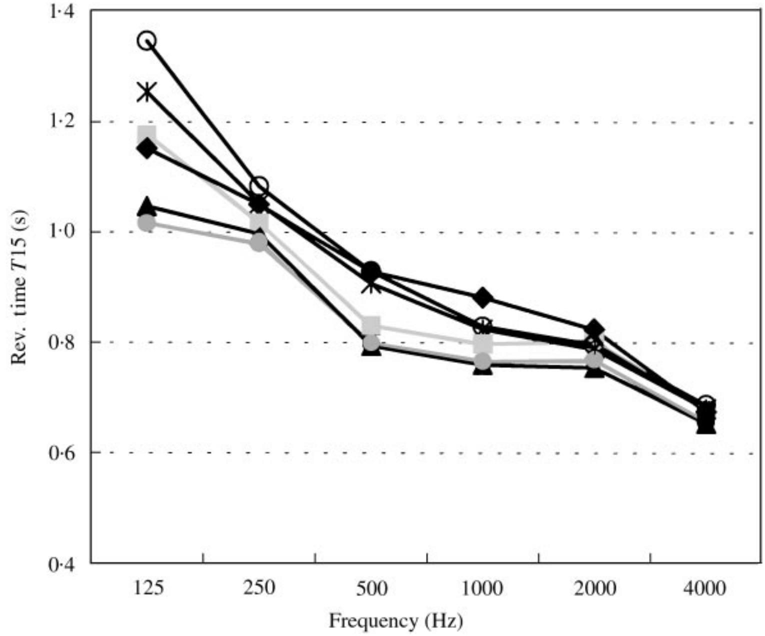

Figure 3. Comparison between the reverberation times measured with the interrupted-noise method and with the backward integrated impulse responses (hall 1): —■—, Interr.noise + B&K 2133; —◆—, Pistol shot + B&K 2133; —▲—, MLS + SB16 + Aurora software; —●—, MLS + SB16 + MLSSA software; —○—, Shots + SB16 + Aurora software; —✳—, Shots + SB16 + MLSSA software.

time-domain acoustical parameters are quite robust. But, when listening to the measured impulse responses, both directly or after convolution with anechoic signals, the effect is dramatically different: this means that the commonly accepted set of acoustical parameters does not properly include the characterization of the frequency response of the system. A new set of frequency-domain acoustical parameters is needed.

Using two different binaural microphones (on the same dummy head) gave comparable results. One of the microphones, equipped with very small capsules, gave a lower $C80$ (up to 0·5 dB at low frequencies) (see Figure 6).

Figure 7 shows the comparison between the values of clarity $C80$ obtained with the two measurement techniques used in hall 3: the MLS technique based on the deconvolution of the deterministic pseudo-random signal and the SWEEP technique based on an exponentially sweeping sine wave test signal. The results obtained with the two techniques do not indicate significant differences for all the calculated parameters and for all the positions. In some cases, the results were exactly the same. In other cases, differences less than 0·5 dB were found, particularly in the presence of the audience, where the MLS technique gave a non-optimal signal-to-noise ratio, probably due to the imperfect time invariance of the system (the people were not perfectly still).

The comparison between the values of the acoustical parameters calculated from the monaural measurement and the binaural measurement has shown differences of

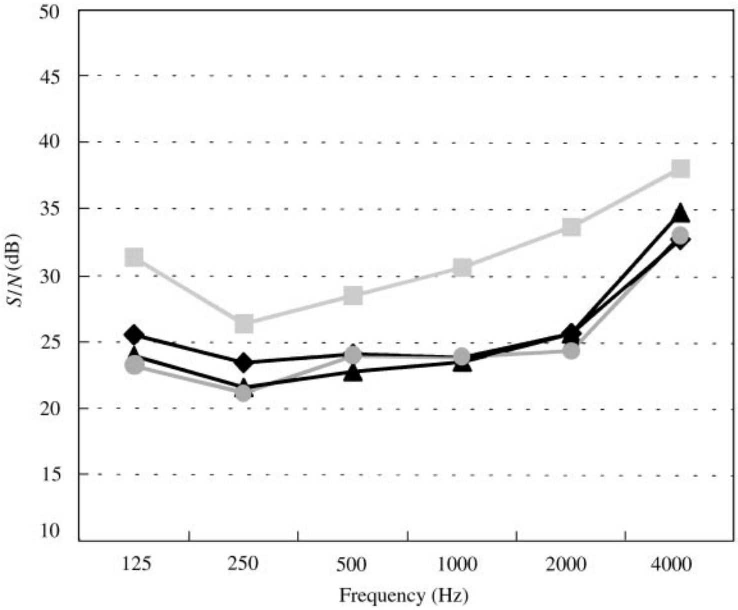

Figure 4. Comparison between the signal-to-noise ratios obtained with different measurement techniques (hall I): —■—, MLS + MLSSA board + MLSSA software; —◆—, MLS + SB16 + MLSSA software; —▲—, Balloons + SB16 + MLSSA software; —●—, Pistol shot + SB16 + MLSSA software.

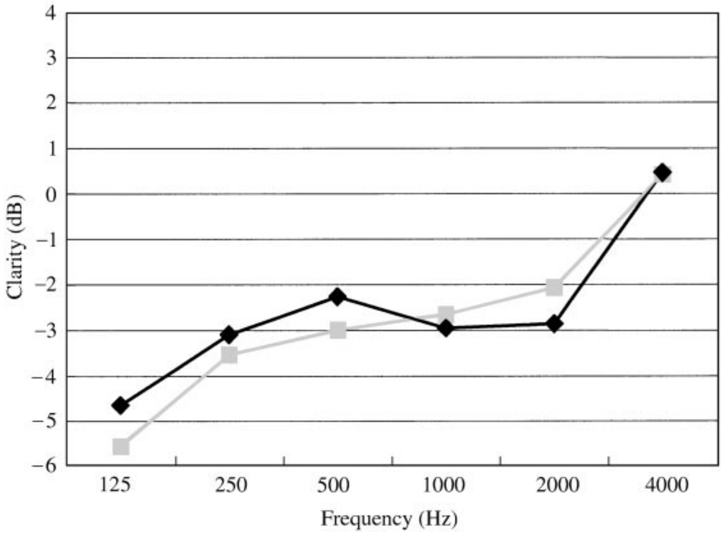

Figure 5. Comparison between the values of the clarity *C*80 obtained with two different dodecahedron sound sources one of which with an electronic equalization (hall 2): —■—, Norsonic source non-equalized; —◆—, look line source equalized.

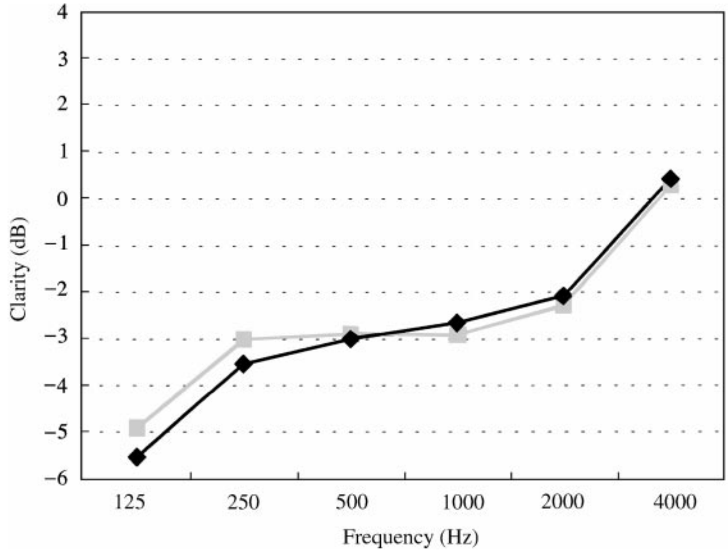

Figure 6. Comparison between the values of the Clarity $C80$ obtained with two different binaural microphones (hall 2): —■—, Sennheiser microphones; —◆—, Sony microphones.

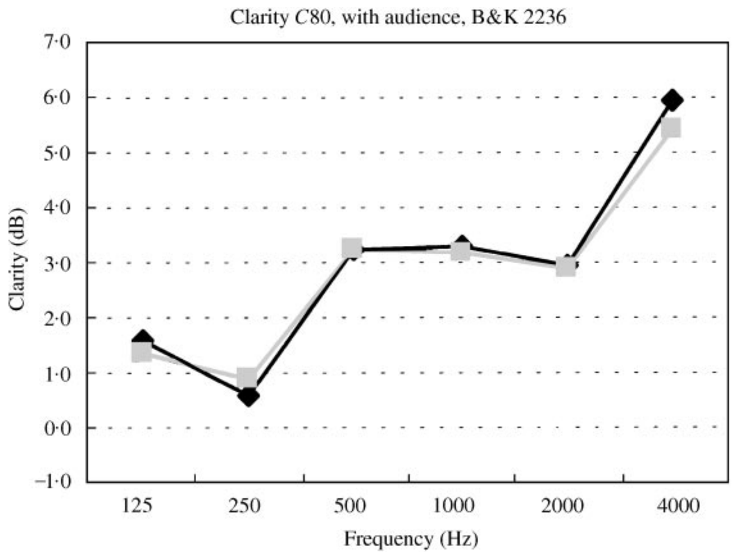

Figure 7. Comparison between the values of the Clarity $C80$ obtained with the two measurement techniques (hall 3): deterministic pseudo-random-signal (MLS) and exponentially sweeping sine wave signal (SWEEP): —◆—, MLS; —■—, SWEEP.

up to 1 dB for clarity $C80$ and of up to 0·2 s for reverberation time $T15$, as reported in Figures 8 and 9. This result was the same for all the calculated acoustical parameters and both with and without the presence of the audience. This is very important because usually the average value of the left and right channels of a

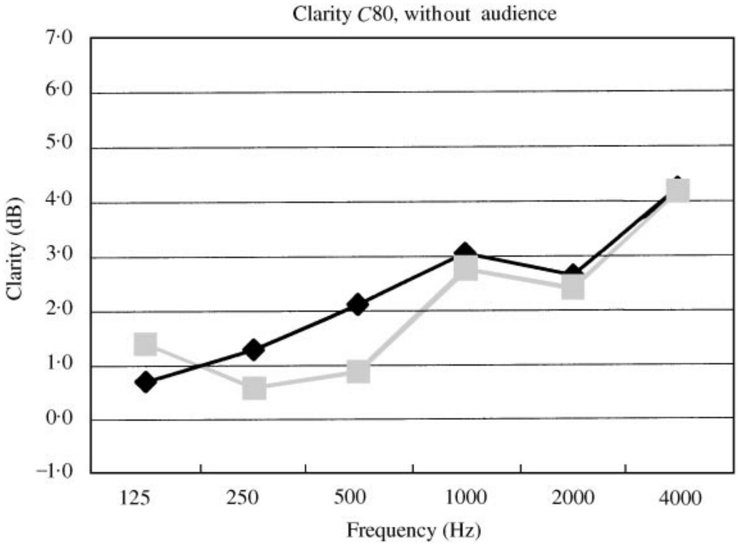

Figure 8. Comparison between the values of the Clarity C80 obtained from a monoaural measurement and an average of a left and right channels of a binaural measurement (hall 3): —◆—, B&K 2236-monaural; —■—, Sennheiser-binaural.

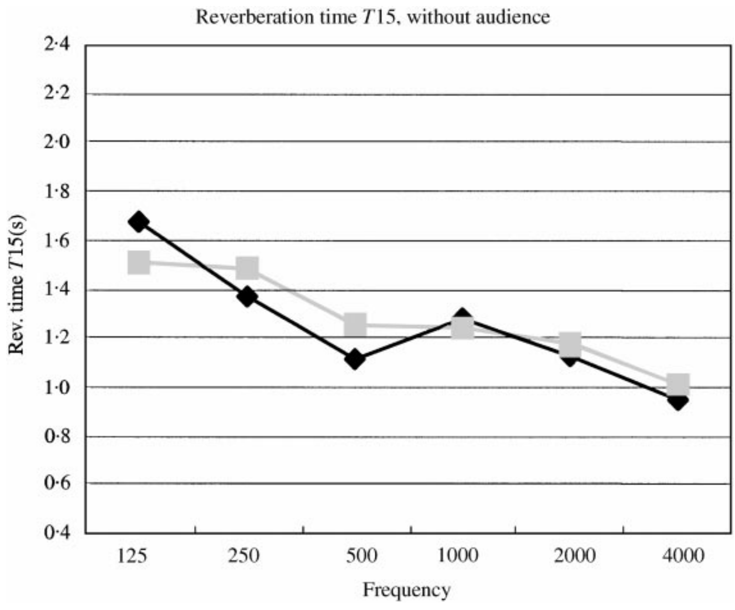

Figure 9. Comparison between the reverberation time (T15) obtained from a monoaural measurement and an average of a left and right channels of a binaural measurement (hall 3): —◆—, B&K 2236-monaural; —■—, Sennheiser-binaural.

binaural measurement is used to express many of the monaural acoustical parameters.

The measurements made with and without the musicians inside the orchestra shell gave differences in all the frequency ranges of interest of up to 1 dB for C80 and of up to 0·2 s for T15. In Figures 10 and 11, the results obtained with the source

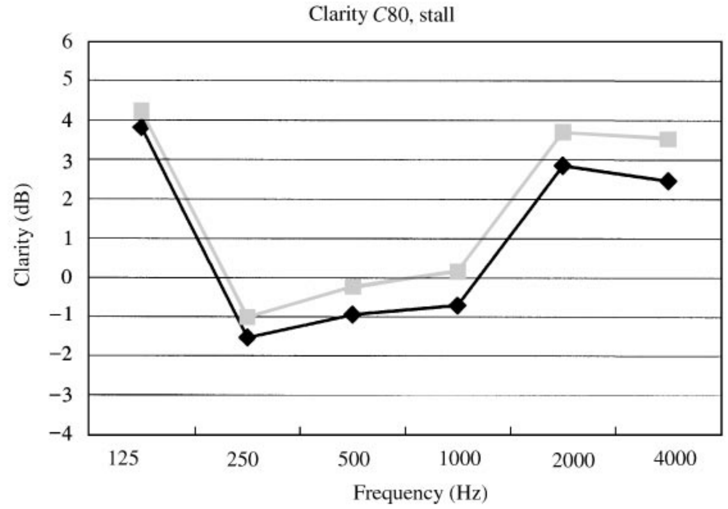

Figure 10. Comparison between the values of Clarity $C80$ obtained with and without the presence of the musicians inside the orchestra shell: the source was in the position of the first violin and the receiver was in the fourth row of the stall (hall 3): —■—, With orchestra and choir; —◆—, Without orchestra and choir.

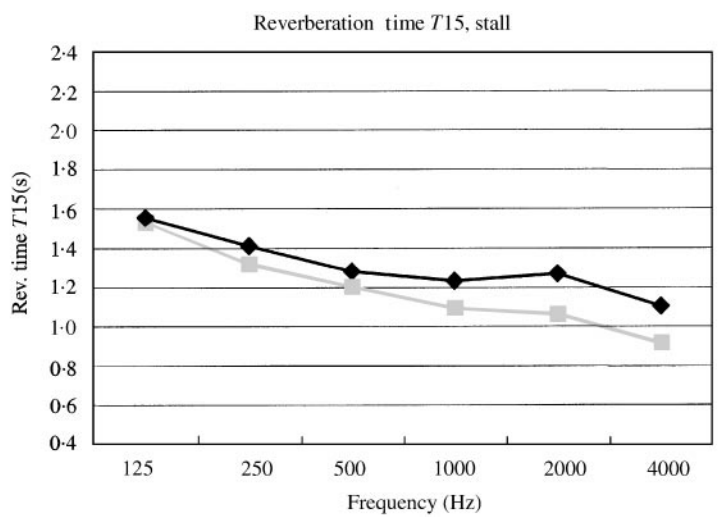

Figure 11. Comparison between the values of the reverberation time obtained with and without the presence of the musicians inside the orchestra shell: the source was in the position of the first violin and the receiver was in the fourth row of the stall (hall 3): —■—, With orchestra and choir; —◆—, Without orchestra and choir.

in the position of the first violin and the receiver in the fourth row of the stall are reported. The differences are evident both for the clarity and for the reverberation time.

The differences obtained with and without the audience in the hall were evident but not as large as expected considering the presence of 700 people (maximum

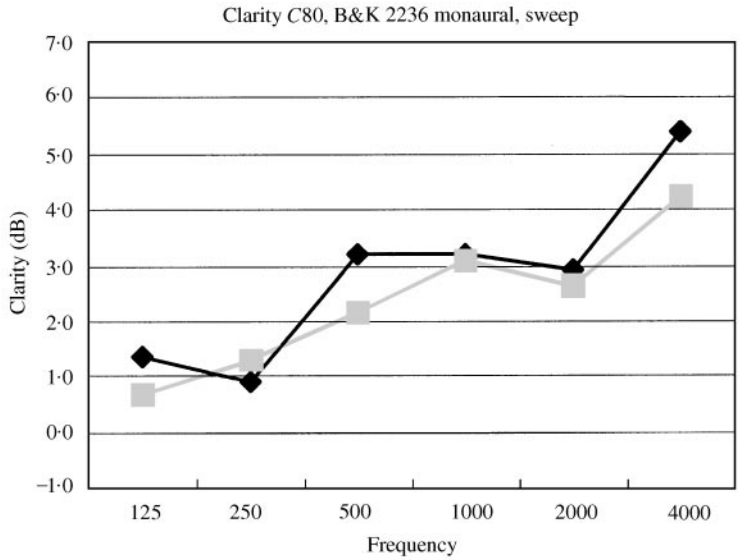

Figure 12. Comparison between the values of the Clarity C80 with and without the audience in the hall (hall 3): —◆—, With audience; —■— , Without audience.

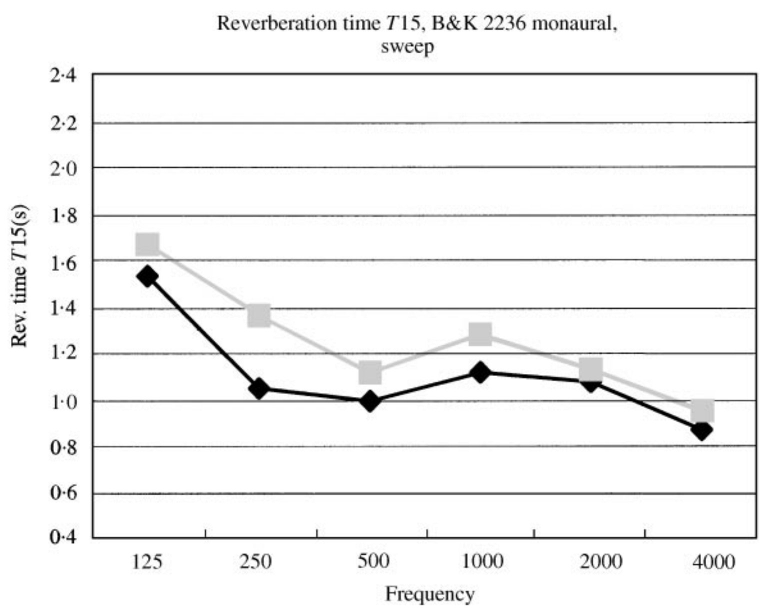

Figure 13. Comparison between the values of reverberation time T15 with and without the audience in the hall (hall 3): —◆—, With audience; —■— , Without audience.

capacity 800 people). In Figures 12 and 13, a case in which the differences were more evident is reported, with maximum differences of 1·1 dB for C80 and 0·3 s for T15.

In Figures 14 and 15 the comparison between the values of clarity C80 and that between the values of reverberation time, obtained in three different positions of

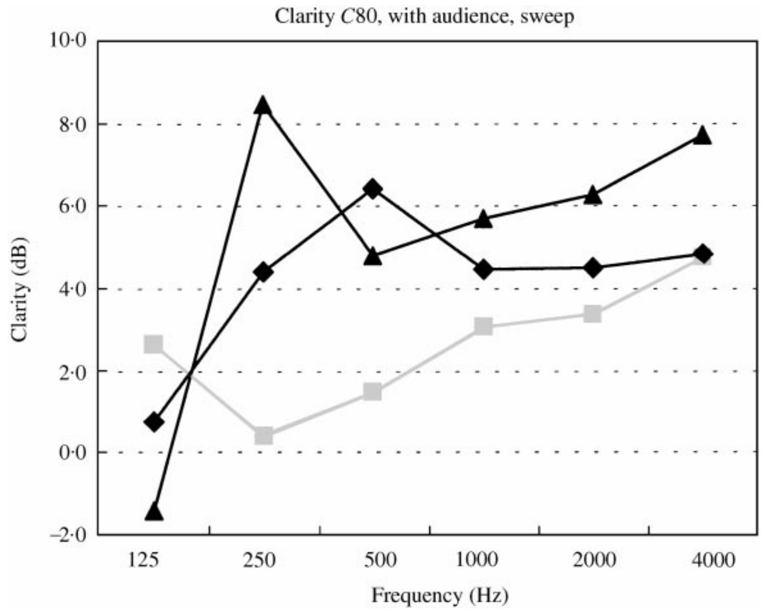

Figure 14. Comparison between the values of the Clarity $C80$ obtained in three different positions of the hall 3: —■—, Stall; —◆—, Second balcony; —▲—, Gallery.

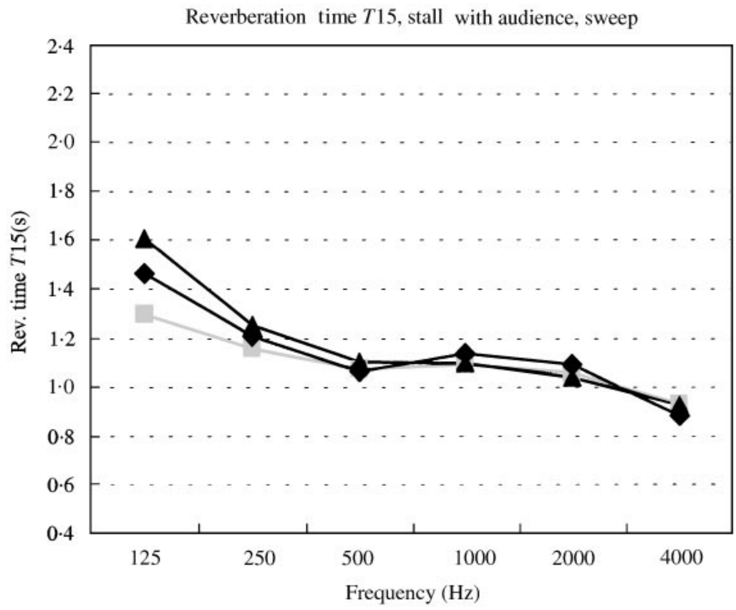

Figure 15. Comparison between the values of the reverberation time $T15$ obtained in three different positions of the hall 3: —■—, Stall; —◆—, Second balcony; —▲—, Gallery.

hall 3, are reported. As shown, significant differences were found for clarity (up to 8 dB at 250 Hz) and small differences were found for reverberation time (up to 0·3 s at 125 Hz). Significant differences, although not as evident as for clarity, were found for many other acoustical parameters such as center time, definition and early decay time. This result shows that, for clarity, the influence of the position is greater than that of the audience or any of the other factors that can influence the results. This means that clarity is not useful parameter for the comparison between different theatres or different settings. The reverberation time is instead very stable with respect to the position, as it should be according to its definition.

5. CONCLUSIONS

The aim of the research was to compare the results, in terms of acoustical parameters, obtained using different measurement techniques and equipment. There are three aspects to the conclusions: acoustical parameters, equipment and measurement techniques.

In the calculation of the reverberation time, small but significant differences between different excitation techniques were found. On the other hand, large differences, particularly at low frequencies, were found for clarity $C80$ and for early decay time. Clarity and early decay time seem to be better correlated than clarity and reverberation time.

The signal-to-noise ratio is limited mainly by the soundboard used in the measurements.

The differences obtained in the calculation of the acoustical parameters using loudspeakers with different frequency response are of less importance than the differences obtained between loudspeaker and impulsive sources. The differences are almost negligible when using different binaural microphones (on the same dummy head) and no appreciable alteration is induced by the recording/playback over the DAT recorder. The measured time-domain parameters were not influenced by the equalization of the sound source, as they remained substantially unchanged. Obviously, there is a great subjective difference when listening to the impulse responses, both directly or by convolution with anechoic signals. It appeared that none of the measured parameter took account of such large subjective differences related to the frequency content of the impulse responses. This means that a new acoustical parameter is required for evaluating the spectral flatness (for example, frequency-dependent strength).

The two software implementations of the MLS method (MLSSA and Aurora) are substantially equivalent. Aurora has the advantage of processing simultaneously both channels of a binaural impulse response, and the MLS maximum order is 21 instead of 16, but in these experiments it was penalized by the use on a poor-quality soundboard.

The interrupted stationary noise method agrees well with MLS measurements made with the same loudspeaker and less well with integrated impulses coming from pistol shorts or balloons. The new exponentially swept sine test signal produced slightly better results than MLS in terms of S/N ratio, although all the computed parameters are almost the same. Its many advantages (immunity from clock mismatch, time variance and non-linear distortion) are certainly worth the longer post-processing time required for deconvolving the impulse response, considering also the continuously increasing speed of personal computers

Effective differences were found regarding the recording techniques, as the monaural measurements give appreciably different results from the average of left and right channel of binaural measurements.

Significant differences, but not as great as expected, were found in the results with and without the musicians in the orchestra shell and with and without the audience in the hall. This is probably due to the high sound absorption that is typical in Italian Opera Houses even without the audience. However, important differences were found in the calculation of some acoustical parameters, particularly for clarity, by changing positions in the hall.

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