

Conclusion

This paper presented the measurement method, hardware and software which is being employed in a large experimental survey of the acoustical properties of 20 Italian opera houses. After the project will be completed all the measured data will be made available on the WWW, including plans, photos and history of each theatre.

References

- [1] A. Farina, R. Ayalon – “Recording Concert Hall Acoustics for the Posterity” - 24th AES Conference on Multichannel Audio, Banff, Canada, 26-28 June 2003
- [2] Michael Barron, “Auditorium Acoustics and Architectural Design”, Spon Press, 1992
- [3] Leo L. Beranek, “Concert and Opera Halls: How They Sound”, Published for the Acoustical Society of America through the American Institute of Physics, 1996, 643 pp
- [4] Leo L. Beranek, “Concert Halls and Opera Houses Music, Acoustics, and Architecture”, Springer-Verlag, 2004
- [5] V.L. Jordan, “A group of objective acoustical criteria for concert halls”, *Applied Acoustics*, vol. **14** (1981)
- [6] Gade, A.C., "Acoustical Survey of Eleven European Concert Halls", Report No. 44, The. Acoustics Laboratory, Technical University of Denmark, 1989
- [7] J. S. Bradley, "An international comparison of room acoustic measurement systems." Institute for Research in Construction, IRC Internal Report IRC-IR-714, Jan. 1996
- [8] Rife, Douglas D. and John Vanderkooy, "Transfer-Function Measurement with Maximum-Length Sequences", *J. Audio Eng. Soc.*, vol. 37, 6/89, p. 419
- [9] Heyser R., “Acoustical measurements by time delay spectrometry”, *J. Audio Eng. Soc.* 15. 370-82 (1967)
- [10] M. Poletti – “Linearly swept frequency measurements, time-delay spectrometry, and the Wigner distribution” – *JAES* vol. 36, n. 6, 1988 June, pp. 457-468.
- [11] Hidaka, T. and Beranek, L. L. - “Objective and subjective evaluations of twenty-three opera houses in Europe, Japan and the Americas,” *J.A.S.A.*, 107, 368-383, (2000).
- [12] A. Farina – “Simultaneous measurement of impulse response and distortion with a swept-sine technique”, *110th AES Convention*, Paris 18-22 February 2000.
- [13] S. Müller, P. Massarani – “Transfer-Function Measurement with Sweeps”, *JAES* Vol. 49, Number 6 pp. 443 (2001).
- [14] G. Stan, J.J. Embrechts, D. Archambeau – “Comparison of Different Impulse Response Measurement Techniques”, *JAES* Vol. 50, No. 4, p. 249, 2002 April.
- [15] J. Vanderkooy, “Aspects of MLS measuring systems”, *JAES* vol. 42, n. 4, 1994 April, pp. 219-231.
- [16] W. T. Chu, “Time-variance effect on the application of the m-sequence correlation method for room acoustical measurements,” in Proc. 15th Int. Congr. on Acoustics (Trondheim, Norway, 1995), vol. 4, pp. 25-28.
- [17] A. Farina, F. Righini, “Software implementation of an MLS analyzer, with tools for convolution, auralization and inverse filtering”, Pre-prints of the 103rd AES Convention, New York, 26-29 September 1997.

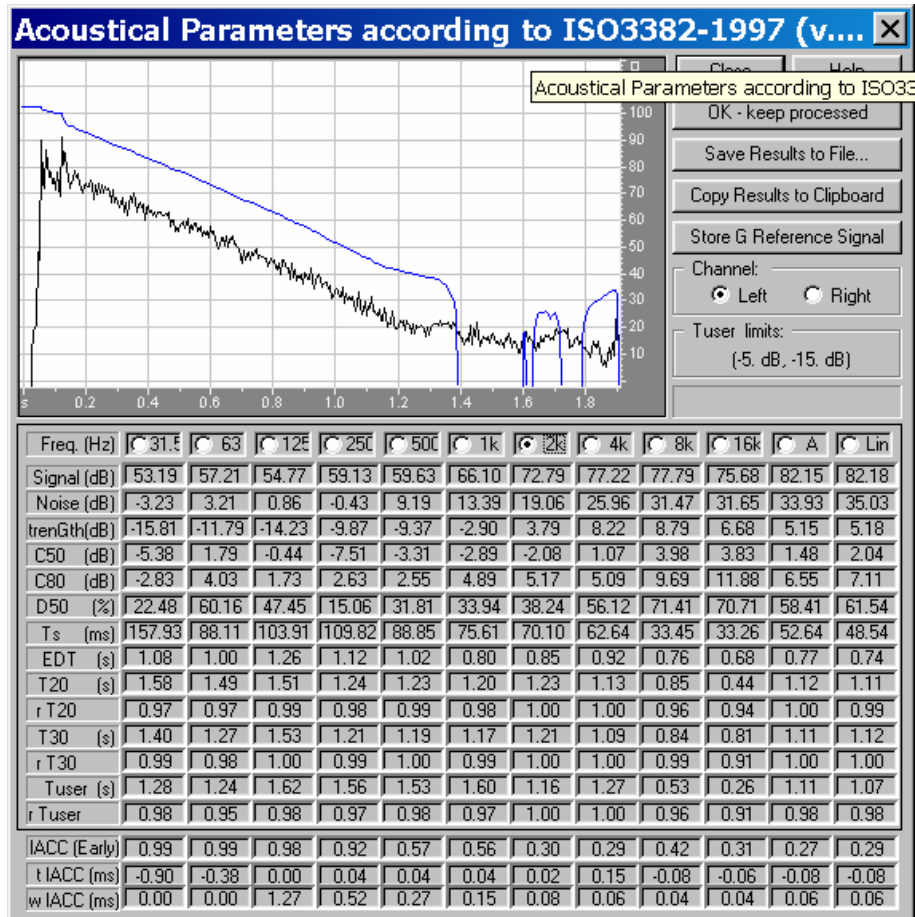


Figure 23 – “Acoustical parameters” plugin

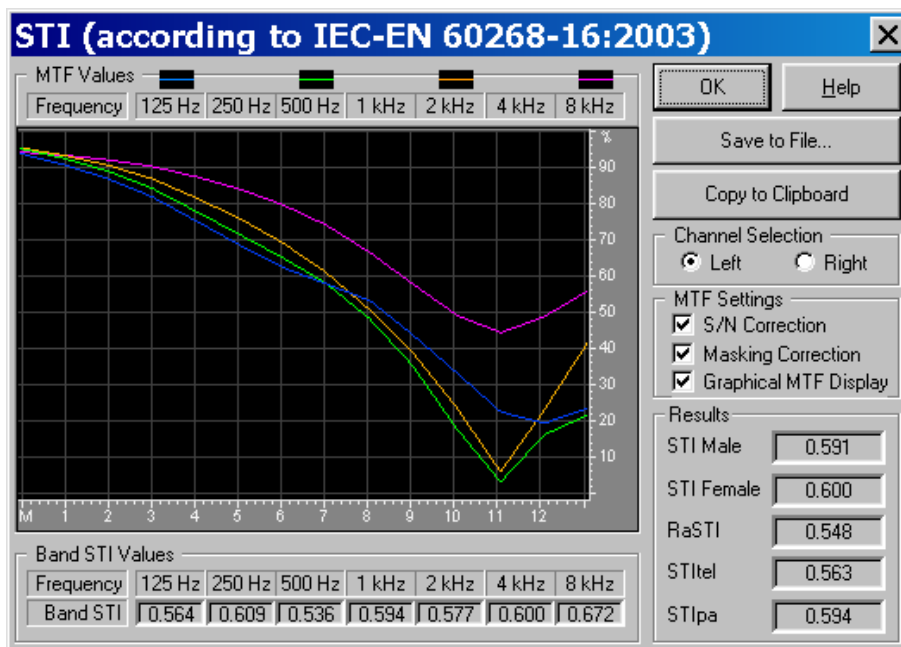


Figure 24 – “Speech Transmission Index” plugin



Figure 22: Multitrack session of a measurement

Each measurement takes approximately 15 minutes (25s x 36 repetitions); after the measurement is complete, Another 10 minutes are required for storing all the waveforms on the hard disk (in 32-bits format, for preserving all the available dynamic range); during this time, the source and/or the microphonic array are displaced into another position.

After the recordings are done, it is possible to deconvolve the impulse response: the Aurora “convolve with clipboard” plugin is employed, followed by “IR select” plugin, which extracts just the relevant portion of the measured impulse response.

Finally, other plugins being part of the Aurora package are employed for post-processing the measured impulse response, computing the acoustical parameters according to ISO 3382 and the values of STI (Speech Transmission Index) according to IEC-60268-16.

The following two pictures show these plugins at work.

The model employed for this research mounts a 1.8 GHz P-IV processor, and is equipped with 512 Mbytes Ram and an high speed (7200 RPM) hard disk. This allows for faultless operation when recording 8 channels and playing 2 channels at 96 kHz, 24 bits.

The sound card chosen for the task is an Aadvark Pro-Q10. Fig. 21 shows a picture of the equipment, which is installed inside a couple of fly-cases for easy transportation.



Figure 2: Liquid-cooled PC (FutureClient).

Software for measurement and post processing

CoolEditPro 2.1 (now renamed Adobe Audition) was employed for the playback of the test signals and the simultaneous recording of the 8 microphonic channels. The test signal was looped 36 times, corresponding to the 36 steps of the rotating table along a complete rotation. The following picture shows a multi-track session, resulting from a measurement with the above-described approach.

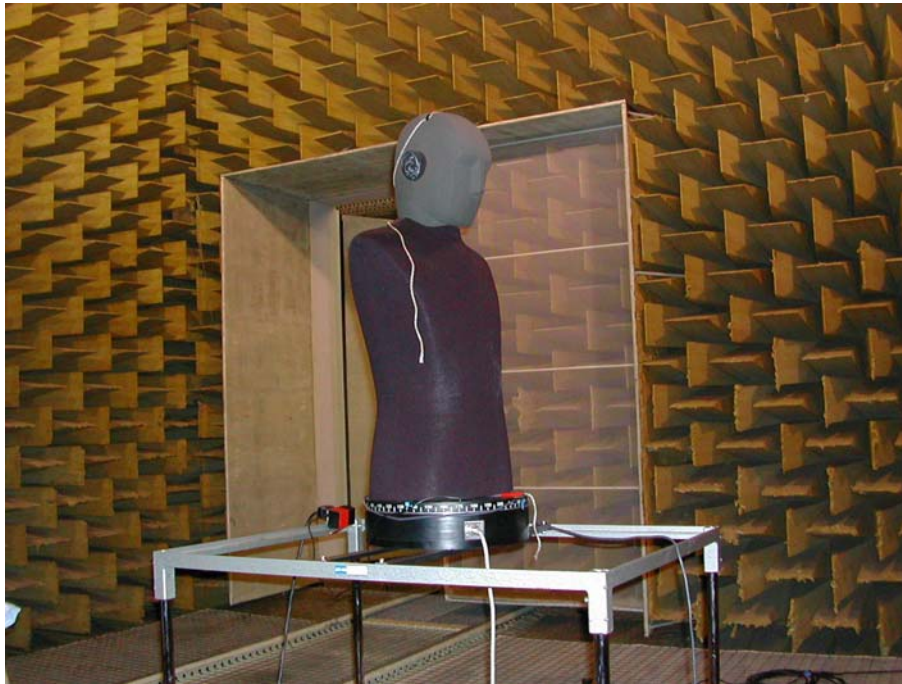


Fig. 19 – measurement of the polar patterns of the Neumann KU-100 dummy head

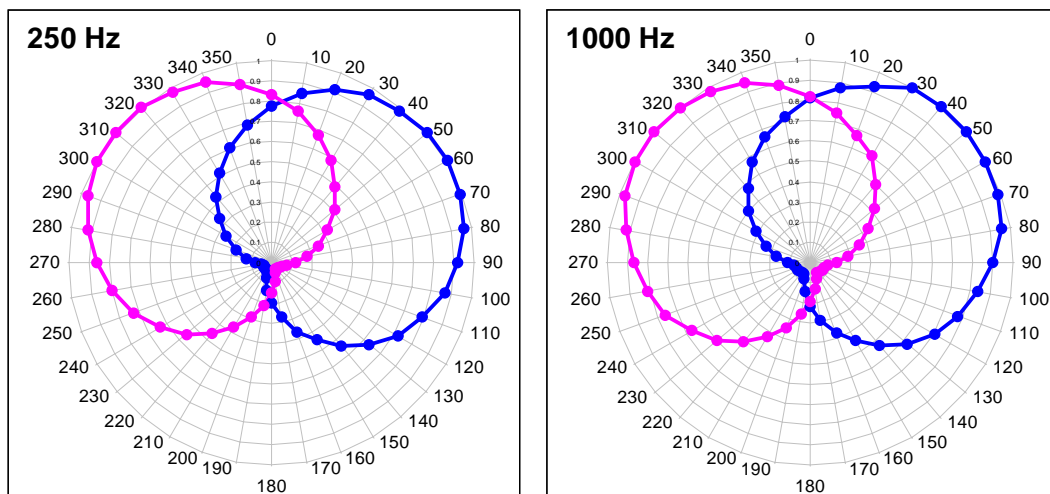


Fig. 20 – Polar patterns of the Neumann K-140 cardioid microphones at two frequencies

Computer and audio interface

The measurement method required the usage of a top-grade sound card, equipped with 8 analog inputs at 24 bits / 96 kHz, incorporating digitally controlled mic preamplifiers (for ensuring accurate control of the input gain, and relative and absolute calibration of the recordings). At the moment, these requirements can only be fulfilled by external rack-mounted units, connected to the computer by means of a PCI card.

This impeded the usage of any portable computer, and forced the choice of the only currently-available fanless PC, which stands out for its completely silent design: the Signum Data Futureclient.



Fig. 17 – The Soundfield ST-250 in the anechoic chamber

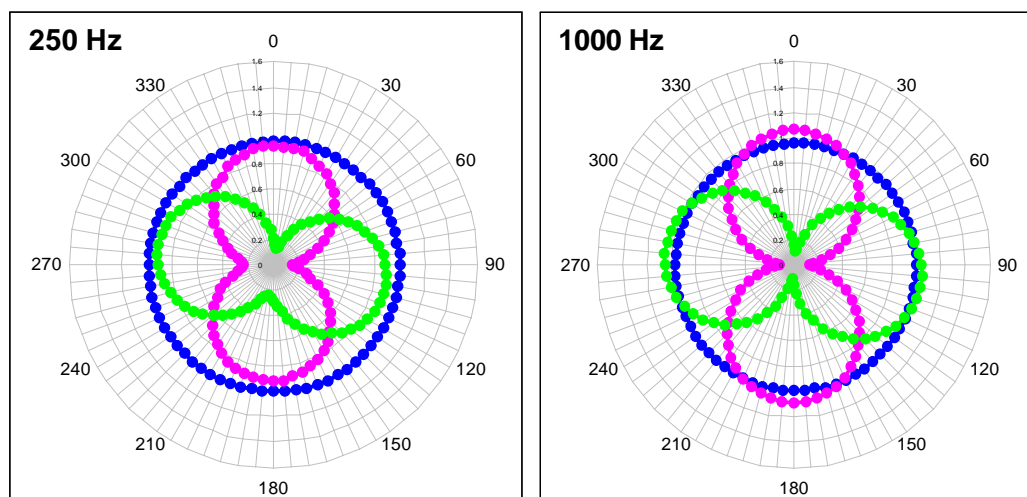


Fig. 18 – Polar patterns of the Soundfield ST-250 at two frequencies

Fig. 18 demonstrates the results obtained, showing how the Soundfield microphone closely matches the theoretical directivity patterns of a pressure-velocity microphone probe.

Similar tests were conducted over the Neumann cardioid microphones and over the Neumann binaural dummy head, as shown in figs. 19 and 20.

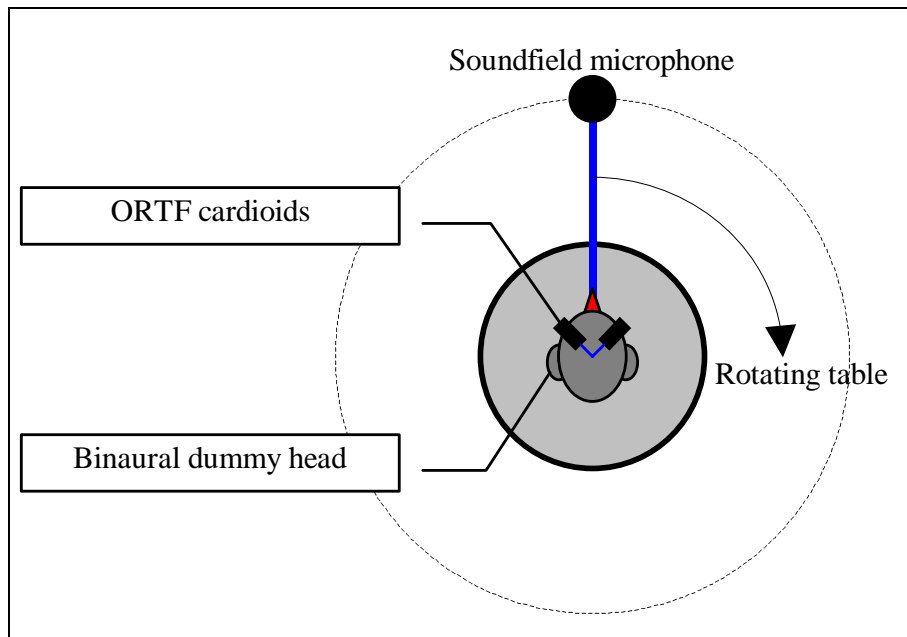


Figure 15 – microphones on the turntable



Figure 11: The microphones over the rotating table.

Of course, also the microphones have been accurately tested in anechoic conditions before being employed for the measurements. The following pictures show the test of the polar patterns of the Soundfield microphone.

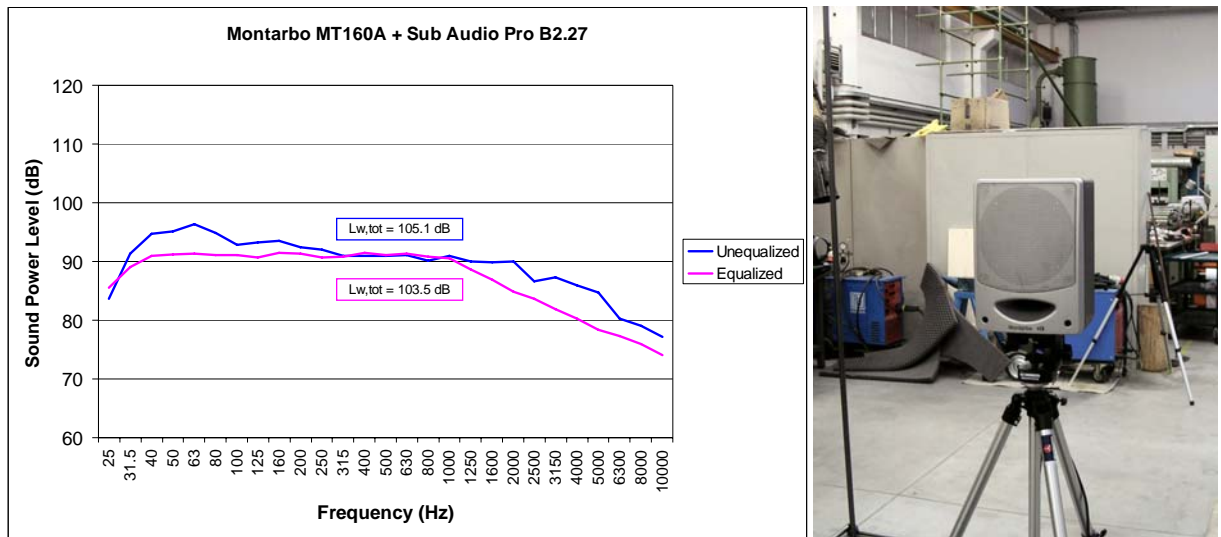


Fig. 13 – power response of Montarbo MT 160A

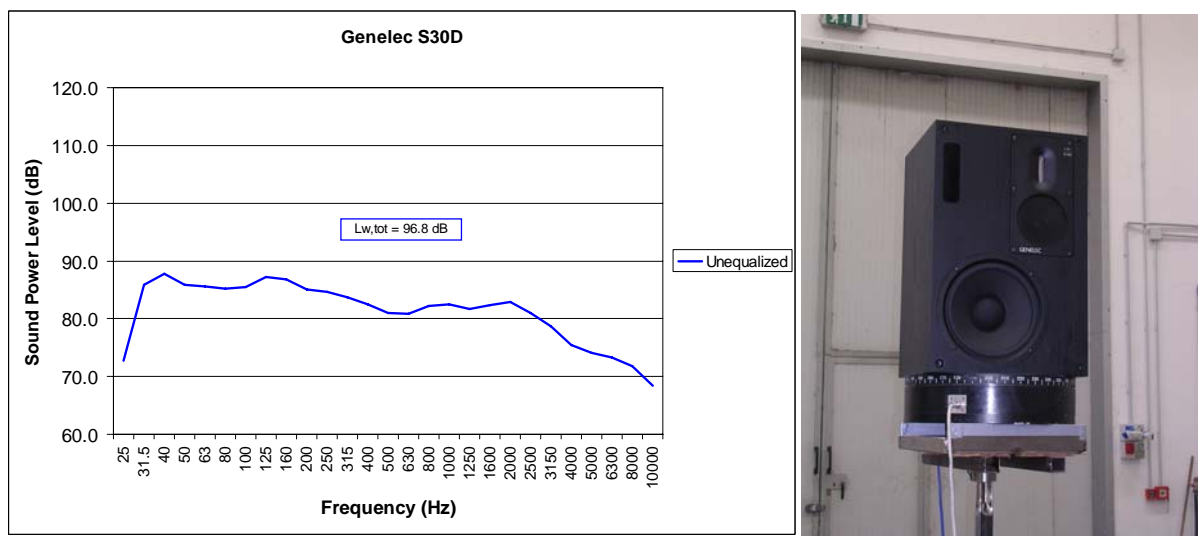


Fig. 14 – power response of Genelec S30D

Microphones

Three different microphonic probes were employed:

- a pair of high quality cardioids in ORTF configuration (Neumann K-140, spaced 180mm and diverging by 110°);
- a binaural dummy head (Neumann KU-100);
- a B-format 4-channels pressure-velocity probe (Soundfield ST-250).

All these microphones were installed over a rotating table, in such a way that the rotation center passed through the center of the dummy head, and through the point at the intersection of the axes of the two cardioids (which were mounted just above the dummy head). Alternatively the Soundfield microphone was displaced exactly 1m from the rotation axis, in front of the dummy head. The angular step was 10° (36 measurements for the complete rotation).

Figs. 15 and 16 show how the microphones were operated.

A further improvement can be obtained with a completely new design of the dodecahedron, recessing the loudspeaker drivers and shaping the corners of the faces in such a way to bend the radiated waveforms and merge them together smoothly.

Fig. 12 shows a prototype of this new type of dodecahedron, manufactured by Omnisonic, and actually under testing as a substitute of the LookLine models.

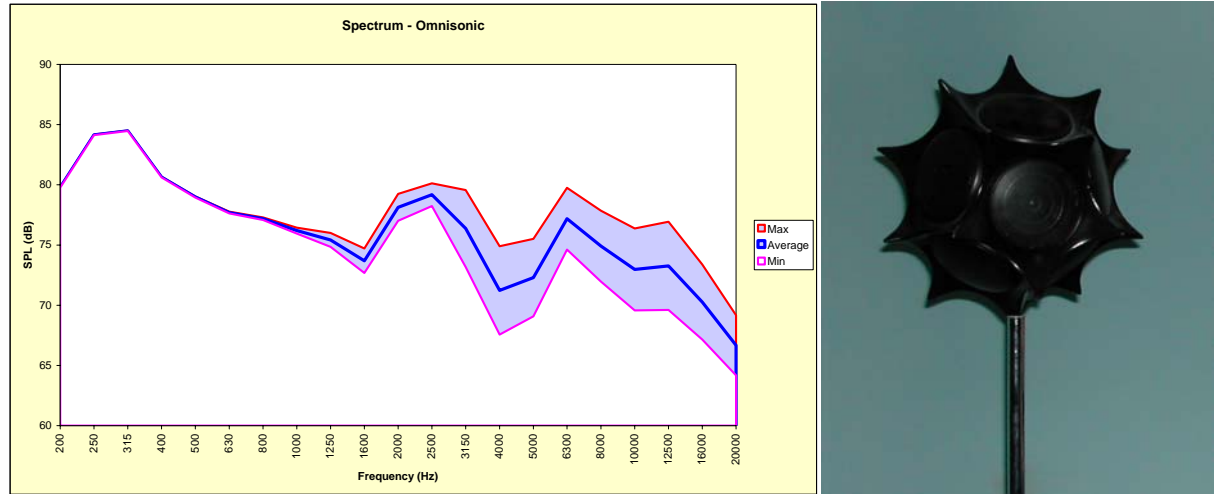


Fig. 12 – Spatial uniformity responses of the new Omnisonic loudspeaker

It can be shown that the new prototype breaks up at a much higher frequency (approximately 4 kHz) than the traditional dodecahedrons (which break up respectively at 1 kHz and 2 kHz), whilst providing a reasonably good frequency response end power.

However, it must be understood that the lack of omnidirectionality at high frequency is usually not a problem when the loudspeaker is placed inside the orchestra pit. In fact, the receiver is, in most cases, not receiving the direct sound, and the reflected sound coming out from the pit is already mixed-up by the multiple reflections, so re-orienting the loudspeaker in most cases does not have any relevant effect. This means that also dodecahedrons having performances equivalent to those of the LookLine models tested here are suitable for the task of measuring impulse responses from the orchestra pit.

Now we take in examination the sound sources to be employed over the stage: these source should not be omnidirectional, as they represent a model of the singer. Furthermore, they need to be coherent in time domain, because the responses measured through them have to be convolved with single voices, and no chorusing effects should occur.

In practice, the most immediate solution for this task is to choose a good-quality self-powered studio monitor. A variety of products is available in this field, ranging from very cheap solutions to professional monitors costing thousands of euros each.

We tested two extreme cases:

- a very cheap, but robust, Montarbo MT-160 A
- a high-end Genelec monitor (S30D)

If some music is played on these two loudspeaker, there is no race: the Genelec completely outperforms the Montarbo. But when the loudspeakers are employed for impulse response measurements, it turned out that, with proper equalization, the Montarbo can substantially equal the results obtainable with the Genelec monitor (which is so well-behaved that it does not require any equalization).

The following figures 13 and 14 show the power spectra of these two loudspeakers.

It must be emphasized that the “normal” dodecahedrons usually employed in building acoustics measurements, such as those manufactured by Bruel & Kjaer, Nosonic or Acoustic Engineering, are not suitable for these measurements, due to their limited frequency range and uneven response, which requires too much equalization for being flattened.

However, also the LookLine dodecahedrons suffer of not-perfectly-uniform emission at high frequency, causing the polar pattern to deviate significantly from the perfect omnidirectionality required (albeit remaining within the limits of the ISO 3382/1997 standard).

The only possible solution about this problem is related to the shape and size of the dodecahedron. Of course, the smaller is the dodecahedron, the higher is the frequency where the response “breaks up”, as shown in fig. 11, which shows the comparison between the smaller D-200 and the larger D-300.

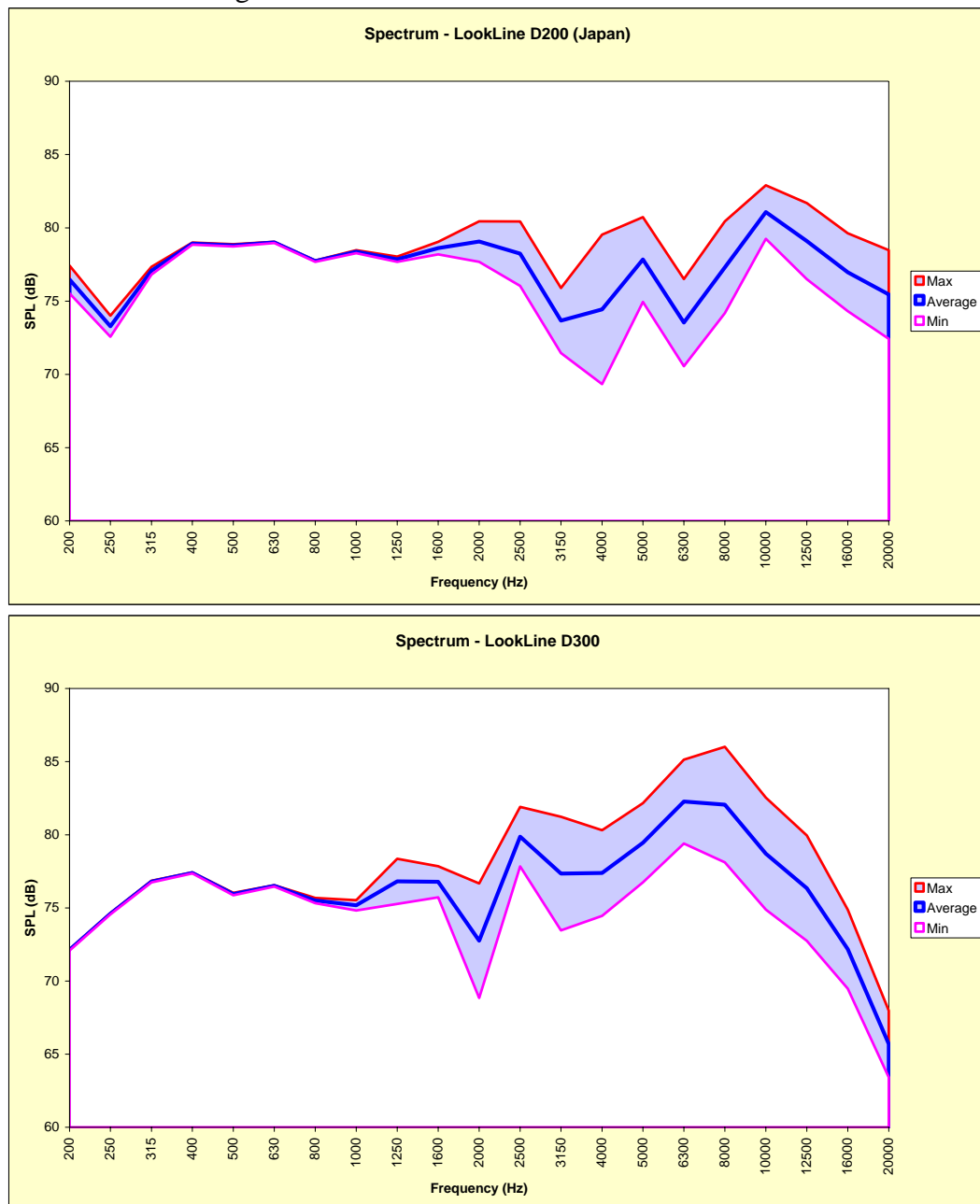


Fig. 12 – Spatial uniformity responses of the LookLine dodechaedrons

The second problem can only be solved by supplementing the dodecahedron with a separate low-frequency unit (a subwoofer). This additional unit is usually placed on the floor, just beneath the dodecahedron. If the distance between the two units is shorter than a quarter of the wavelength, at the cross-over frequency, in practice the radiated sound is still equivalent to that of a point source. As the cross-over frequency is usually below 100 Hz, being the wavelength 3.4m, a distance of 1m between the two units is usually OK.

The following figures 9 and 10 show the acoustic power level spectra of two currently-employed dodecahedron loudspeakers plus subwoofer, and photograph of each unit. The radiated sound power was measured with a sound intensity system, according to ISO 9614 part 2.

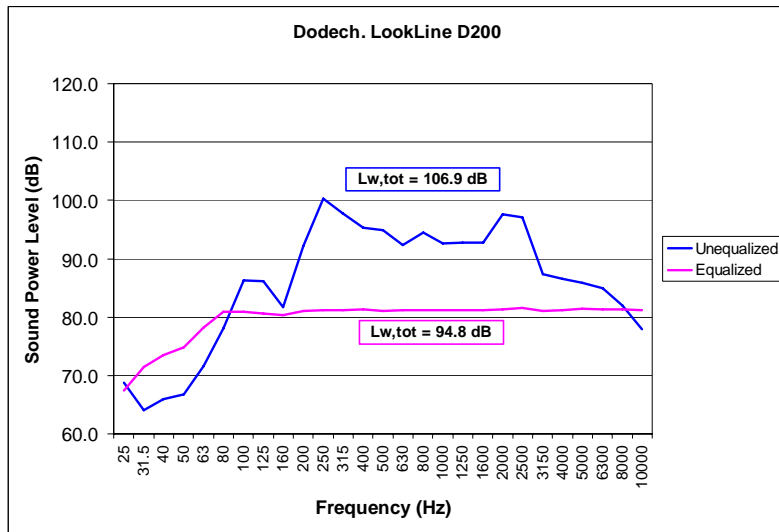


Fig. 9 – LookLine D-200 dodechaedron

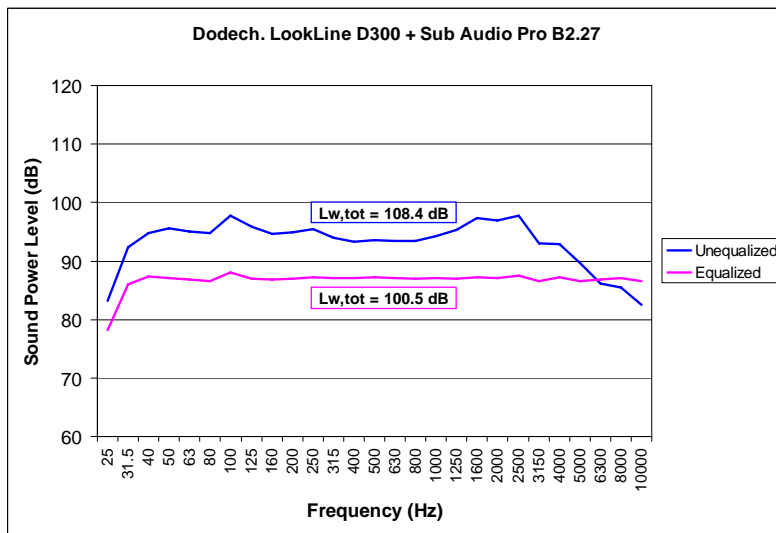


Fig. 10 – LookLine D-300 dodechaedron + Audio Pro B2.27 subwoofer

It can be shown how the larger model has both more acoustical power (100.5 dB instead of 94.8 dB, after equalization) and a better extension towards low frequencies, thanks to the professional AudioPro subwoofer which starts below 30 Hz.

It is easy to see how the exponential sine sweep method produces better S/N ratio, and the disappearance of those nasty peaks which contaminate the late part of the MLS responses, actually caused by the slew rate limitation of the power amplifier and loudspeaker employed for the measurements, which produce severe harmonic distortion.

The practical implementation of this measurement method is based on a standard Windows PC, equipped with a top-quality sound card (capable of recording multiple channels at 96 kHz, 24 bit sampling). The software employed is Adobe Audition, thanks to a series of specific additional plug-ins, which were written specifically for automating the tasks of generation of the test signal, computation of inverse filters and convolution of very long sequences [17]; these plug-ins were made available to all the research units participating to the project, and to the general community of acousticians, in the form of a shareware package (Aurora).

Sound sources

The measurement method employs loudspeakers as sound sources. In practice, three different kinds of loudspeakers can be employed:

- dodecahedrons
- studio monitors
- artificial mouths

Each of these three kind of loudspeakers has some advantage and disadvantages, as shown in the following table.

Type	Dodechaedron	Monitor	Artificial mouth
Acoustic power	high	medium	Very low
Directionality corresponding to theoretical curves	reasonably good	fair	good
Frequency response extension	poor	excellent	poor
Frequency response flatness	fair	excellent	poor
Time coherence	very bad	Good-excellent	Excellent (1-way)

In practice, dodecahedrons are mainly employed inside the orchestra pit, for emulating the radiation of a whole section of the orchestra. For this goal, the lack of time coherence is usually not a problem. The main problems with dodecahedrons are three, and two of them can be efficiently addressed:

- uneven frequency response
- no response at low frequencies
- bad omnidirectionality at high frequency

Flattening the in-band frequency response is easy, as the test signal is computer-generated. A proper equalizing filtering can be applied to the sine sweep signal employing the advanced filtering capabilities of Adobe Audition, or pre-convolving the test signal with a proper inverse filter computed with the Aurora plug-ins.

This equalization reduces the overall radiated power, as its effect is to reduce the gains where the response of the loudspeaker is “fat”, but of course it is not possible to boost too much the signal at those frequency where the response of the loudspeaker is inherently “weak”.

At this point, applying a suitable time window it is possible to extract just the portion required, containing only the linear response and discarding the distortion products.

The advantage of the new technique above the traditional MLS method can be shown easily, repeating the measurement in the same conditions and with the very same equipment. Fig. 8 shows this comparison in the case of a measurement made in an highly reverberant space (a church).

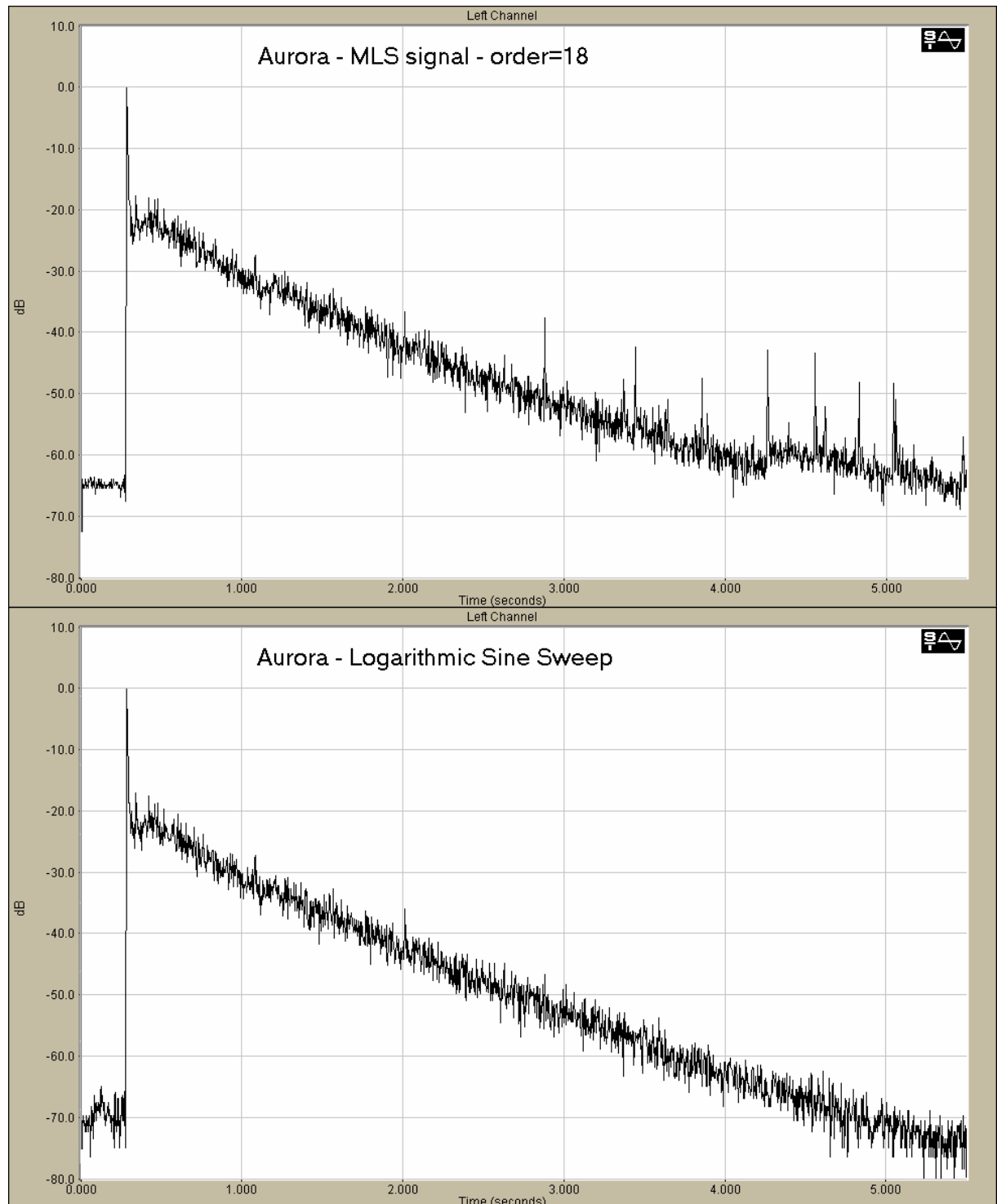


Fig. 8 – comparison between MLS and sine sweep measurements

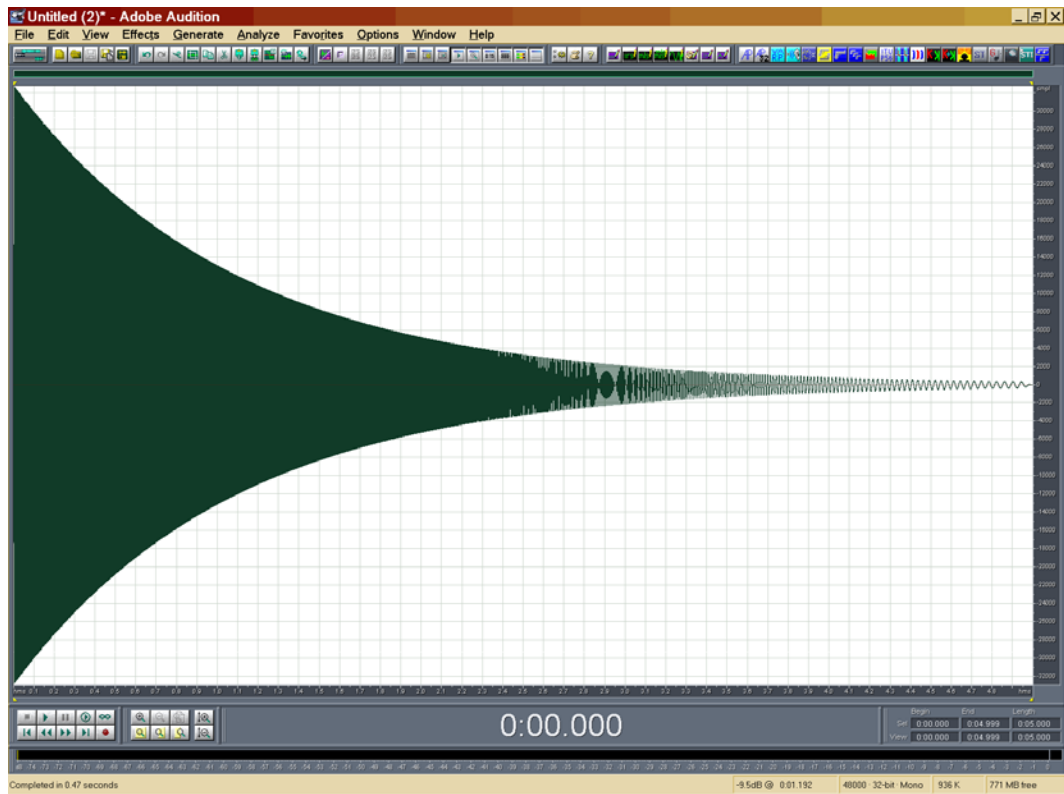


Fig. 6 – waveform representation of the inverse filter $f(t)$

Fig. 7 shows a typical result after the convolution with the inverse filter has been applied.

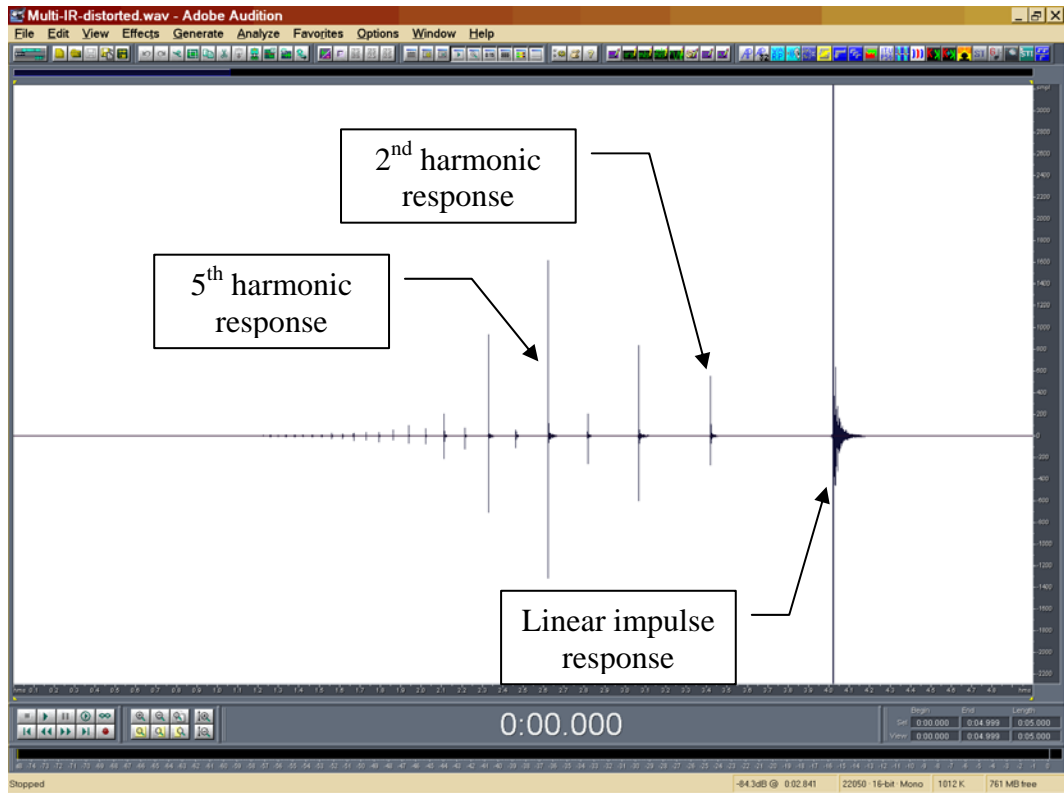


Fig. 7 – output signal $y(t)$ convolved with the inverse filter $f(t)$

Now the output signal $y(t)$ has been recorded, and it is time to post-process it, for extracting the linear system's impulse response $h(t)$.

What is done, is to convolve the output signal with a proper filtering impulse response $f(t)$, defined mathematically in such a way that:

$$h(t) = y(t) \otimes f(t)$$

The tricks here are two:

- implement the convolution aperiodically, for avoiding that the resulting impulse response folds back from the end to the beginning of the time frame (which would cause the harmonic distortion products to contaminate the linear response)
- employ the Time Reversal Mirror approach for creating the inverse filter $f(t)$

In practice, $f(t)$ is simply the time-reversal of the test signal $x(t)$. This makes the inverse filter very long, and consequently the above convolution operation is very "heavy" in terms of number of computations and memory accesses required (on modern processors, memory accesses are the slow operation, up to 100 times slower than multiplications).

However, the authors developed a fast and efficient convolution technique, which allows for computing the above convolution in a time which is significantly shorter than the length of the signals.

However, it must be taken into account the fact that the test signal has not a white (flat) spectrum: due to the fact that the instantaneous frequency sweeps slowly at low frequencies, and much faster at high frequencies, the resulting spectrum is pink (falling down by -3 dB/octave in a Fourier spectrum). Of course, the inverse filter must compensate for this: a proper amplitude modulation is consequently applied to the reversed sweep signal, so that its amplitude is now increasing by +3 dB/octave, as shown in fig. 5.

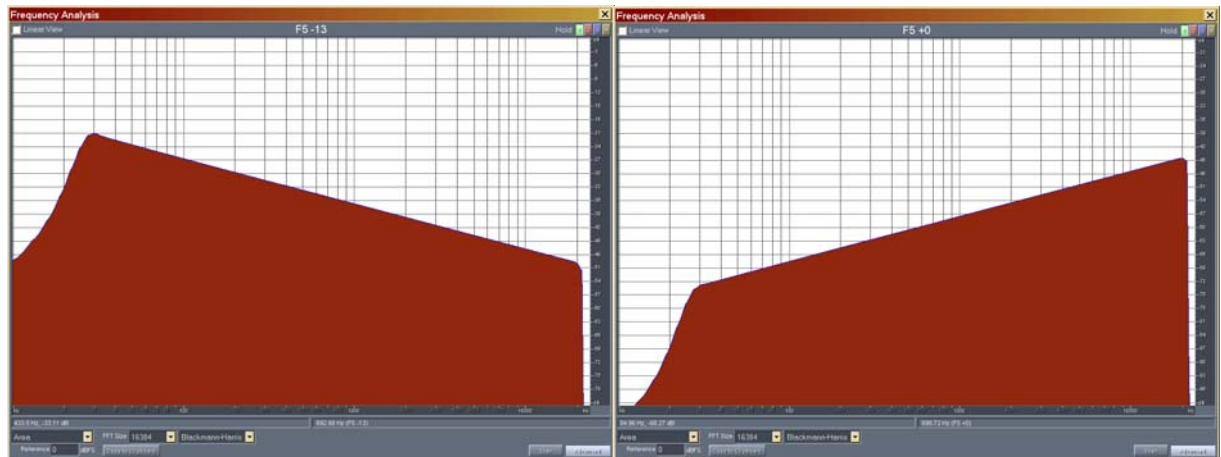


Fig. 5 – Fourier spectrum of the test signal (left) and of the inverse filter (right)

Fig. 6 shows the waveform of the inverse filter $f(t)$, which makes evident the amplitude modulation employed for obtaining the above equalization.

When the output signal $y(t)$ is convolved with the inverse filter $f(t)$, the linear response packs up to an almost perfect impulse response, with a delay equal to the length of the test signal. But also the harmonic distortion responses do pack at precise time delay, occurring earlier than the linear response. The aperiodic deconvolution technique avoids that these anticipatory response folds back inside the time window, contaminating the late part of the impulse response.

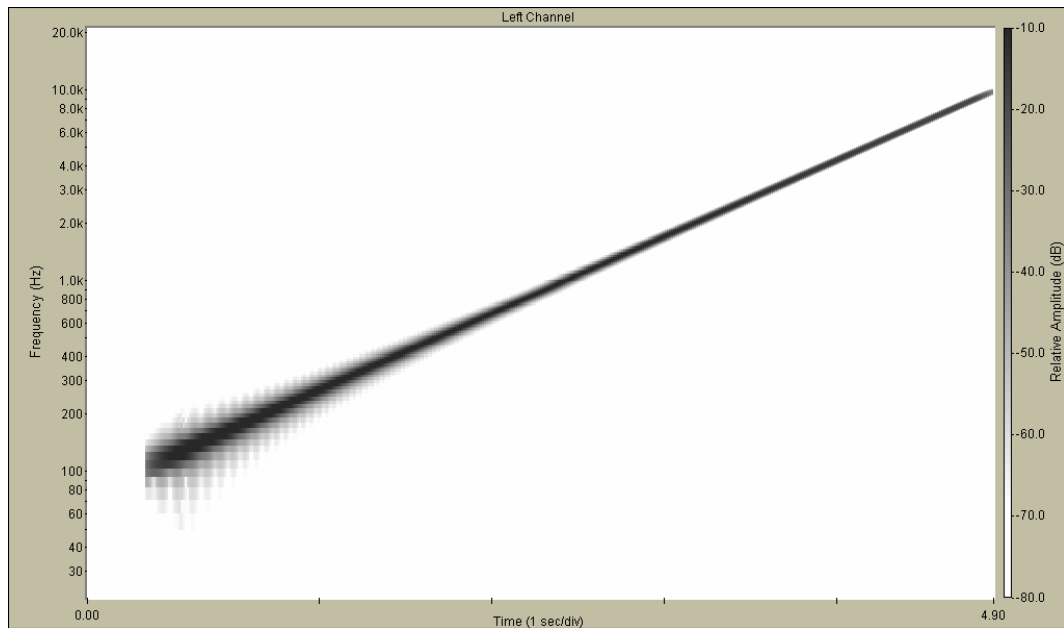


Fig. 4 – sonograph of the test signal $x(t)$

It must be noticed that the vertical axis in the sonograph has logarithmic scale, which makes the trace of the sine sweep to appear as a straight line. This is the reason for which this test signal is also called logarithmic sweep.

When this signal, which has constant amplitude and is followed by some seconds of silence, is played through the loudspeaker, and the room response is recorded through the microphone, the resulting signal exhibit the effects of the reverberation of the room (which “spreads” horizontally the sweep signal), of the noise (appearing mainly at low frequencies) and of the not-linear distortion (which creates harmonic signals at frequencies double, triple, etc. of the one of the test signal).

These “distorted” harmonic components appear also as straight lines, above the “main line” which corresponds with the linear response of the system. Fig. 5 shows the signal recorded through the microphone.

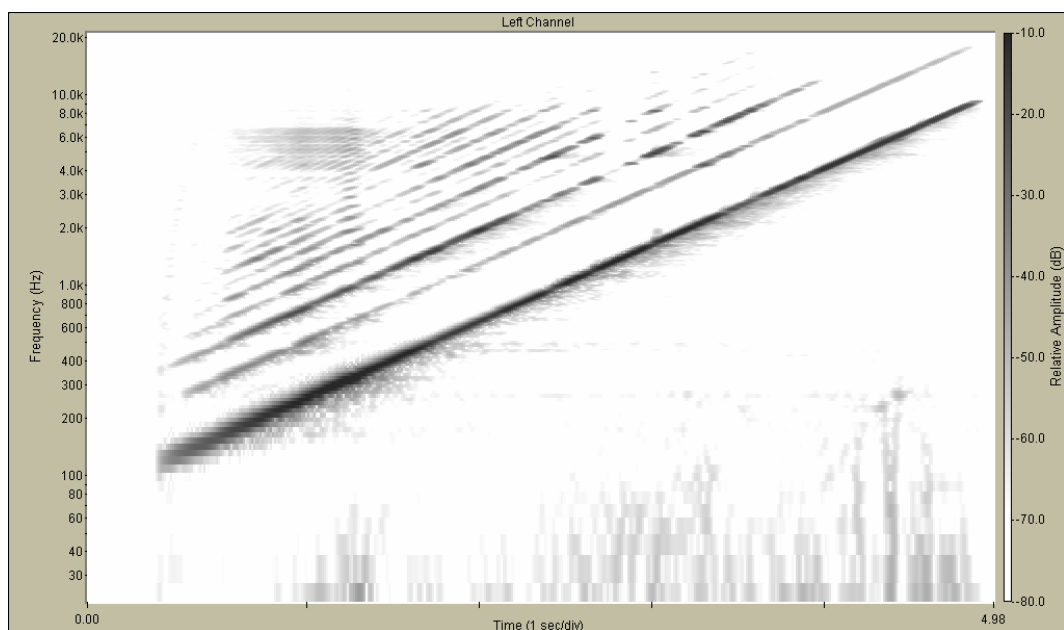


Fig. 4 – sonograph of the response signal $y(t)$

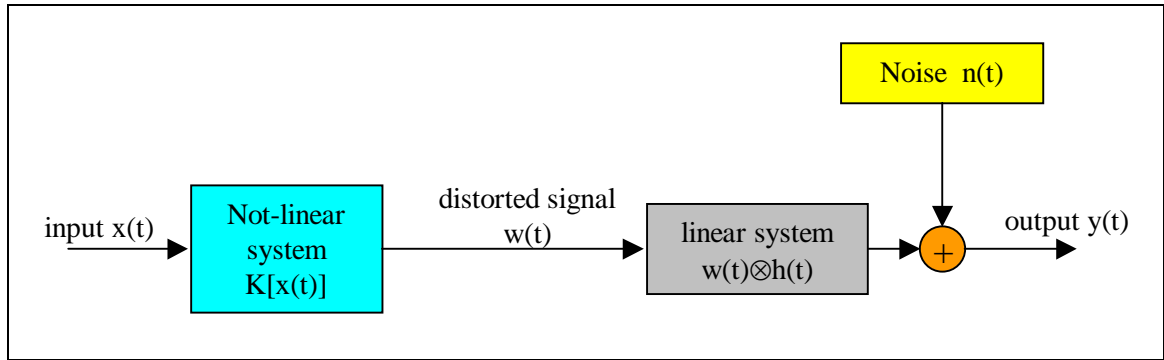


Fig. 3 - A more complex system, in which a not-linear, memoryless device drives a subsequent linear, non-time-invariant reverberating system

The quantity which we are interested to measure is the impulse response of the linear system $h(t)$, removing the artifacts caused by noise, not-linear behavior of the loudspeaker and time-variance.

As already recalled in the introduction, a good noise rejection is provided by employing the MLS test signal. The deconvolution done with this technique, however, is usually severely affected by even minor nonlinearities [15]. Furthermore, as usually the MLS method is employed together with synchronous averaging (for further improving the S/N ratio), also some small time-variance can cause significant underestimation of the late reverberant energy at high frequency [16].

The method chosen, based on an exponential sweep test signal with aperiodic deconvolution, provides a good answer to three above problems: the noise rejection is better than with an MLS signal of the same length, not-linear effects are perfectly separated from the linear response, and the usage of a single, long sweep (with no synchronous averaging) avoids any trouble in case the system has some time variance.

The mathematical definition of the test signal is as follows:

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

This is a sweep which starts at angular frequency ω_1 , ends at angular frequency ω_2 , taking T seconds. A sonograph display of the test signal is shown in fig. 4.

higher-order Ambisonics formats can be derived) and a large-size circular array of pressure-velocity microphones (suitable for driving Wave Field Synthesis reproduction arrays).

In the following chapters, each of these microphonic systems will be described and technically evaluated, by means of a series of preliminary laboratory investigations aimed to detecting the operational limits and deviations for the theoretical response of the transducers available.

Measurement of the impulse response

The system employed for making impulse response measurements is conceptually described in fig. 1. A computer generates a special test signal, which passes through an audio power amplifier and is emitted through a loudspeaker placed inside the theatre. The signal reverberates inside the room, and is captured by a microphone. After proper preamplification, this microphonic signal is digitalized by the same computer which was generating the test signal.

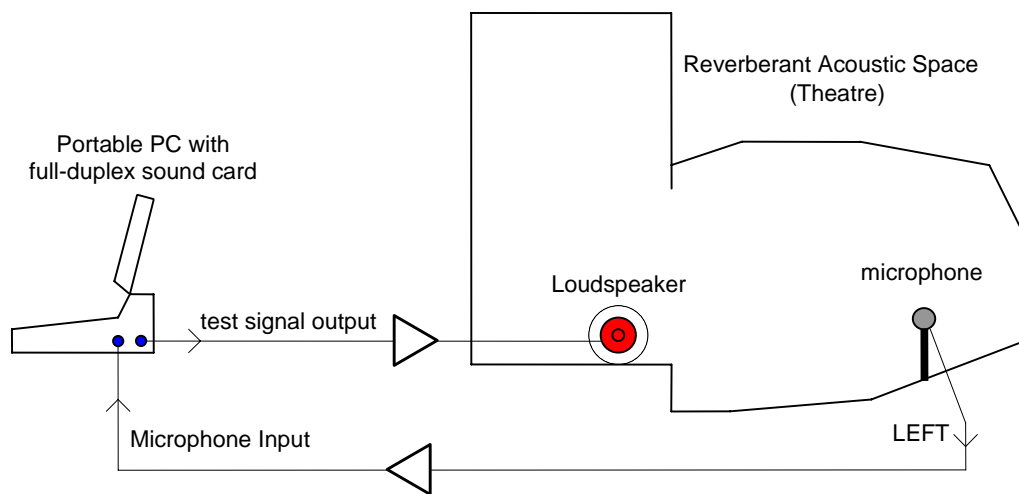


Fig. 1 – schematic diagram of the measurement system

A first approximation to the above system is a “black box”, conceptually described as a Linear, Time Invariant System, with added some noise to the output, as shown in fig. 2.

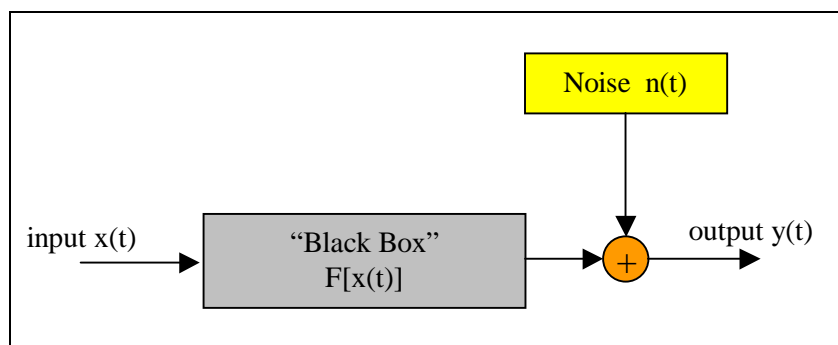


Fig. 2 – A basic input/output system

In reality, the loudspeaker is often subjected to not-linear phenomena, and the subsequent propagation inside the theatre is not perfectly time-invariant. So a more refined schematization of the system is the one shown in fig. 3.

of Rife [8] and the linear sine sweep (this method was named in various ways, depending on the technique employed for extracting the impulse response from the recording of the room response): known implementations of linear sine sweep methods were TDS (Time Delay Spectrometry), as initially suggested by Heyser [9], linear frequency sweep (a more rigorous deconvolution method, as the original TDS approach only works for very slow sweeps) as explained by Poletti [10] and the more recent FFT-based time-stretched pulse, employed by many Japanese researchers (i.e. Hidaka [11]).

In 2000, one of the authors proposed a modified sweep technique, employing an exponential sweep and an aperiodic deconvolution process [12]. This method proved to be significantly superior to the previous ones, as reported in the independent verifications conducted by Muller [13] and Stan [14], as it removes the two main constraints that the system has to be linear and time invariant, and provides significantly better S/N ratio for a given measurement time.

The details of this aperiodic exponential sweep (AES) method are expanded in the following chapter.

But a measurement procedure is not only based on the signal processing approach: other important parts are the transducers (loudspeakers and microphones) and the choice of their placement.

Traditionally, both source and receiver were meant to be point-form and omnidirectional. This makes the impulse response independent from swapping the source/receiver positions, and is an easy approach for numerical simulations. However, in reality, we have to deal with sources which are not point-form, not omnidirectional: think to musical instruments or to the human singing voice.

Similarly, also the receivers are not omnidirectional: the human hearing system is equipped of substantial directional cues, and the presence of the human body (in particular of the head) modifies significantly the sound field around it. Of consequence, a realistic evaluation of the listening conditions requires to analyze the direction-of-arrival of the wavefronts and their interaction with the receiver.

An analysis of previous work show that the sound radiated from the orchestra in the orchestra pit is almost always assimilated to the “omnidirectional” radiation produced by a dodecahedron-like loudspeaker. Instead, various researchers analyzed the effect of the directivity of singers employing directive loudspeakers, although usually in a very empirical way, as done by

As shown in the following chapters, instead, here a more detailed approach was taken, starting from the currently available data about the directivity of singers, and supplementing them with a series of specific measurements conducted in an anechoic chamber. As a consequence of this analysis, a special “artificial singer” torso was built and tested, and will be tested during measurements on the stage of the Italian opera houses.

Finally, regarding the microphones employed for recording the room’s response, an hybrid approach was taken: on one side, a “binaural dummy head” is employed, which captures two signals by means of microphones placed at the entrance of the ear channel of a human-shaped dummy head. This provides signals suitable for listening by headphones, or for the calculation of objective “head related” parameters, such as the IACC (Inter Aural Cross Correlation).

But, as nowadays high quality multi-loudspeaker reproduction systems do exist, it was decided to measure also the impulse responses captured by an array of directional microphones, so that signals suitable for the above-mentioned multi-loudspeaker rigs are captured. These include the Ambisonics B-format 4-channels microphone (commercially known as the Soundfield microphone), a small-size circular array of cardioids (from which

different kinds of sound source must maintain a proper relative balance, but they must also be “processed” by the room differently. In fact, the singed voice should be heard with significant clarity, so that the verbal content of the song can be understood (which is required for enjoying the action). On the other hand, the music should instead be more soft and diffuse, and possibly “surround” the listener.

These two requirements are partially opposite, so a compromise must be reached.

In Italian opera houses, this usually points to a quite short reverberation time, around 1.1-1.4 s at medium frequencies, and without the typical increase towards low frequencies which instead is common in central and northern Europe concert halls.

This is obtained with a lot of drapery, velvet and upholstery for controlling medium and high frequencies, and to the wide usage of wood panels and floor, which absorb significantly the low frequencies.

Italians are used to these very dry rooms, and usually accept them also for symphonic music, where instead a reverberation time close to 2s would be considered more appropriate in almost any other country. This difference in musical taste and customization explains why the Italian theatres are usually judged too dry by international ensembles and conductors, and why some world-acclaimed opera singers refuse to sing in some selected Italian opera houses, such as Teatro Regio in Parma, which are considered very dangerous for their career (being very dry, around 1.1s empty, and populated by very sharp-eared music lovers, who do not refrain by destroying the reputation of a singer for any very minor error).

When an Italian opera house is employed for other purposes, significant modifications are applied: the floor of the orchestra pit is raised (at the level of the stalls, for increasing the seat capacity, or at the level of the stage, for accommodating large scenic requirements or a full orchestra plus choir). Also the arrangement of the stage is often strongly modified, by building over it a partially open, or even a completely closed orchestra shell, with the goal of obtaining a better acoustical coupling between the stage and the main room.

As we are mainly interested in the usage of the theatre for the opera, we will not take into account specific issues related to these modified configurations of the theatre.

The main issues which instead are relevant here are the following:

- Projection of the sound of singers towards the audience
- Projection of the sound of the orchestra towards the audience
- Bi-directional communication between the orchestra pit and the stage
- Acoustical conditions for musicians inside the orchestra pit
- Acoustical conditions for singers over the stage
- Acoustical conditions for the conductor

Various acoustical parameters were introduced for assessing objectively these points. We are not discussing here the effectivity of these parameters, nor their correspondence with subjective perceptions: these will be the goal of the two-year research undertaken, and thus we shall evaluate these topics only at the end of the scheduled work.

The main goal here is how to capture raw acoustical measurement data, from which all these objective parameters can be used, and which are suitable for performing listening tests and collection of questionnaires by means of auralization techniques.

Regarding the measurement techniques which can be applied to opera house, the most remarkable contributions were given by Jordan [5], by Gade [6] and by Bradley [7]. In all cases, the basis of the measurement is the recording of the so-called “impulse response”. This was possible initially only employing a true pulsive sound source (balloons, blank shots of a starter pistol, etc.), but later electroacoustic techniques were developed, allowing for the usage of special not-pulsive test signals, suitable for being reproduced through loudspeakers.

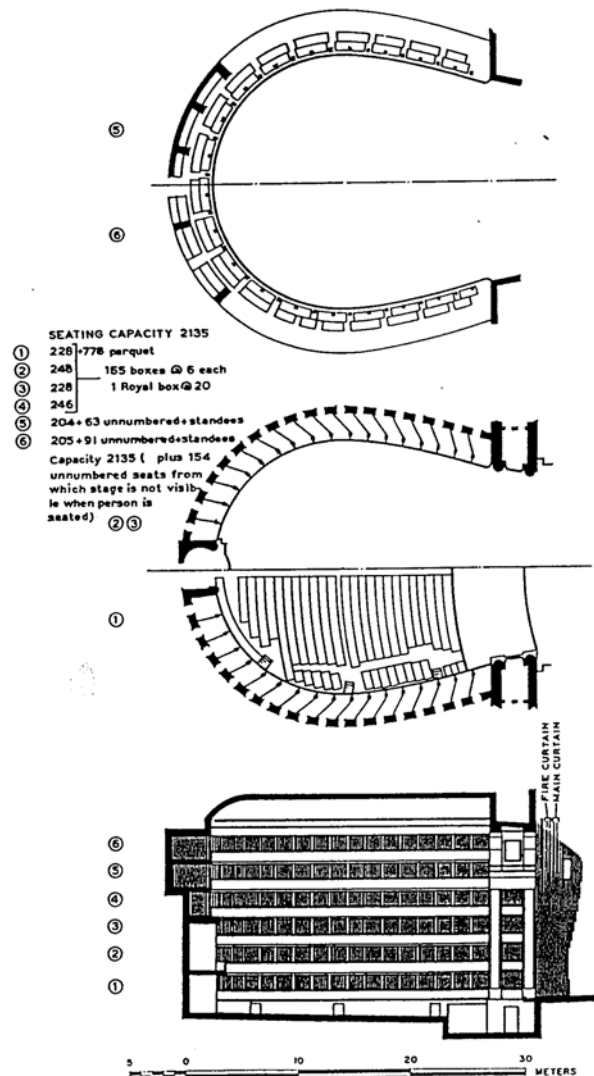
The most widely employed test signals for impulse response measurements were MLS (Maximum Length Sequence), which was made viable thanks to the practical implementation

Nowadays Italy is rich of almost 100 opera houses, with seating capacity ranging between less than 100 (some court theatres such as in Gonzaga or Busseto) to approximately 3000 (San Carlo in Naples). Most of them are also used for other kind of performances, such as cabaret, ballet, drama, and symphonic concerts.

However, we are mainly interested here in the usage of the theatre for operas. Under these conditions, it is assumed that the musicians are located in the orchestra pit, below and in front of the stage. Of course, on the stage some scenery is present (usually placed behind the curtains), and the singers act in front of the scenery, sometimes “coming out” onto the forestage. The choir, instead, always stays inside the scenic tower, usually behind the protagonists.

At the beginning of the “melodrama period”, the main stalls of the theatre were not equipped with seats: the people staying there were standing, and often they did dance during some part of the performance.

The following picture show the typical plan and section of an Italian opera house (La Scala in Milan).



The most comprehensive work on the acoustical behavior of Italian-style opera houses was done by Barron [2], following the fundamental approach of Beranek [3,4].

The main characteristics of an opera house are to provide simultaneously a good “blend” of the sound coming from the orchestra with the voice of the singer. In this blend, the two

An experimental comparative study of 20 Italian opera houses: measurement techniques

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Abstract

This is the preliminary exposition of a huge research project about the acoustical properties of Italian opera houses, which was started at beginning of 2004 and will last 2 years. The project is co-financed by the Italian Ministry for University and Research, and is aimed to take “acoustical photographs” of existing top-grade opera houses.

For “acoustical photography” we mean a set of measured impulse responses, which enable to “listen” at the measured room by means of advanced auralization methods.

Once these data sets have been measured, they can be employed in two different ways: objective analysis and listening tests.

In fact, it is possible to compute dozens of acoustical objective parameters, describing the temporal texture, the spatial effect, the frequency-domain coloring, and the stage/pit balance of each opera house. On the other hand, by means of the auralization technique, it becomes easy to conduct listening experiments with human subjects.

This paper focuses principally on the development and specification of the measurement technique, which is the topic assigned to the research unit of Parma, to which the authors belong.

It describes the hardware equipment, the software, the electro-acoustic transducers (microphones and loudspeakers), the measurement positions, the system for automatic displacement of the microphones and the conditions of the room during the measurements [1].

Experimental results are reported about a couple of opera houses which were employed for testing the measurement procedure, and showing the benefits of the new method against the previously employed ones.

Introduction and previous work

The project presented here is related to high-quality measurement of the acoustical properties of rooms. More specifically, the goal is focused on opera houses, a very specific kind of theatres, emerged during 17th century as the typical performing space for a new kind of art, developed mainly in Italy during the renaissance: the “melodrama”.

At the time, this was a very popular form of entertainment, both for people of low social classes and for lords and their courts. It had very important effects on the culture and social relationship of the times, which evolved and remained very hot during the whole 19th century. This evolution was accompanied, on one side, by a corresponding modification of the artistic contents of the dramas, which saw the greater importance of the musical content above the “action”. Of course, this also drove modification on the performing space (the theatre), which reached the definitive form, shape and structure at the end of 18th century.