ALMA MATER STUDIORUM - UNIVERSITÀ DI BOLOGNA

SCUOLA DI INGEGNERIA E ARCHITETTURA

DIPARTIMENTO DI INGEGNERIA INDUSTRIALE

CORSO DI LAUREA IN INGEGNERIA EDILE-ARCHITETTURA

TESI DI LAUREA

in
Acustica applicata ed Illuminotecnica

THREE CASE STUDIES OF ENHANCING SPEECH INTELLIGIBILITY IN UNIVERSITY LECTURE HALLS

CANDIDATO:
Domenico De Salvio

RELATORE
Chiar.mo Prof. Massimo Garai

CORRELATORI
Ing. Dario D'Orazio
Ing. Giulia Fratoni

Anno Accademico 2016/2017

Sessione III

"Deduco da una frase del Vangelo
che è meglio un imbianchino
di Le Corbusier"

FRANCO BATTIATO, Magic shop

Sommario

La qualità delle attività didattiche dipende in maniera preponderante dalla qualità della comunicazione fra insegnanti e studenti. Una buona comunicazione, in termini di facilità di elaborazione dei messaggi recepiti, permette di ottenere degli alti livelli di concentrazione da parte degli studenti ed un basso affaticamento fisico e mentale da parte degli insegnanti. È facile dedurre come la qualità della trasmissione della parola dipenda dalla sua intellegibilità. Questa è oggettivamente misurabile tramite complessi calcoli che tengono conto principalmente di due fattori: le caratteristiche acustiche degli spazi in cui l'attività si svolge ed i livelli di rumore di fondo che deteriorano la trasmissione dei segnali. Il presente lavoro affronta in maniera approfondita il tema dell'intellegibilità del parlato all'interno di tre aule universitarie e propone interventi di natura attiva e passiva al fine di migliorarla e conformarla agli standard richiesti dalle normative tecniche vigenti. Inoltre è stata realizzata un'analisi approfondita, sulla base di misure effettuate durante lo svolgimento delle lezioni nelle aule oggetto di studio, del rumore di fondo dovuto all'attività antropica. Ulteriori misure sono state effettuate, infine, per verificare gli effetti degli interventi di natura attiva e verificarne la conformità con i risultati previsti dalle simulazioni numeriche in fase di progetto.

Abstract

Teaching acoustic conditions strongly depend on communication quality between teachers and students. A good communication, mean as the ease with received messages can be processed, allows students and teachers to achieve, respectively, high levels of concentration and low physical and mental efforts. The quality of the transmission of words depends on their intelligibility. This is objectively measurable through complex calculations that take into account mainly two factors: the acoustic characteristics of the space in which the activity takes place and the background noise levels that are detrimental to the signal transmission. The present work is focused on the speech intelligibility of three university classrooms and the acoustic interventions proposed to achieve the standard requirements. In addition, a deep analysis of the background noise due to the anthropic activity was carried out, basing on a measurements campaign made during lessons in the lecture halls under study. Further measures were conducted to verify the effects of the interventions and their compliance with predicted results obtained from numerical simulations carried out during the design process.

Contents

List of Figures

List of Tables

Introduction

Communication is the most important aspect of the learning process. The learning mechanism is as much efficient as the speech intelligibility is improved. The quality of communication and thus the speech intelligibility can be measured as an objective parameter thus it's possible enhance it with specific acoustic interventions. The present work studies the acoustic behaviour of three university lecture halls in the faculty of Letters and Philosophy of the University of Bologna and proposes design interventions in order to achieve an improvement of speech intelligibility and acoustic comfort. A measurements campaign, according to ISO 3382-1 [1] requirements, allowed to analyse the lecture halls and to describe them with objective parameters. Subsequently, the design phase was developed defining the necessary active and passive interventions with the support of modelling and simulation software, specifically SketchUp [2], Autocad [3], 3ds Max Design [4], Odeon Room Acoustic [5] and Soundvision [6]. The design process was developed in order to achieve standard requirements provided by normative DIN 18041 [7], UNI 11532 [8] and BB 93 [9]. The acoustic characteristics of lecture halls, especially the reverberation time, are not the only factor which can deteriorate the speech intelligibility. This latter can be defined as the percentage of words correctly heard by listeners and it's strictly related to various acoustical quantities but the ambient noise has a fundamental role in it, too. The higher the speech level is respect to the background noise the greater the intelligibility speech, but the indoor noise in a lecture hall hasn't the single component of the ventilation system. Further investigations were made to analyse the background noise during lectures due to the student activity in each lecture hall. The signal-to-noise ratio for each lecture was examined with a peaks analysis in order to study in depth the speech intelligibility in ante-operam state. After the active treatments were completed, a furthers measurement campaign was carried out for the purpose of evaluate their effects. According to ISO 3382-1 [1] and IEC 60268-16 [10] objective parameters were measured and compared in order to highlight the different acoustical behaviour of the rooms and the improved speech intelligibility.

Chapter 1

Lecture halls acoustics

A general overview of the physic and acoustic characteristics of lecture halls is provided in this chapter. The theory on the basis of the present work is described after a first taxonomic classification of lecture halls according to normative. Acoustic descriptors for indoor spaces are illustrated referring to normative definitions. Each parameter is explained in depth through technical literature. The described room criteria concern the acoustic behaviour of rooms, speech intelligibility and the sound energy spatial distribution.

1.1 Lecture halls

A lecture hall is a room designed for teaching at university. While a high-school classroom has an occupancy of maximum thirty people, a lecture room can contain hundreds of people. Some lecture halls may be structured as an amphitheater both for a comfortable view and for acoustic reasons. Modern lectures require audio-visual equipment and thus a specific acoustic design is needed to make a lecture more efficient. A Public Address (PA) system isn't enough to increase the speech intelligibility if in the room there aren't the proper acoustic conditions. The DIN 18041 [7] distinguishes the small spaces (with a volume until 250 m^3) from the medium room (with a volume between 250 m^3 and 5000 m^3). The ISO 3382-2 [11] considers *"large"* a space with a volume greater than 300 m^3. The most important aspect concerning the acoustics of these spaces is the verbal communication. Its efficiency can be evaluates with objective parameters like Speech Intelligibility Index (STI) that takes into account considerations about the room's acoustic characteristics and background noise due to systems and student activity.

1.2 Acoustic requirements

The impulse response of a room is made of the direct sound and the successive reflections. Direct sound covers the minimum distance between source and receiver so it's not influenced by the environment. Instead all the reflections are affected by the complex interaction between sound and space geometry. It's usual to distinguish the first reflections (called *early reflections*) from the successive others (called *late reflections*). The early reflections arrive to the receiver within a certain range of milliseconds after the arrival of the direct sound and they can increase significantly the sound clarity. The threshold between early and late reflections is 50 ms for speech and 80 ms for music.

In a classroom the main activity is represented by verbal communication and thus achieving a suitable speech intelligibility is the most important acoustical issue. Speech intelligibility is strictly influenced by the acoustical response of the room which modifies considerably the speech signal arriving to receivers. The factors which affect the acoustic condition are the reverberation time, the signal-to-noise ratio (the level difference between the signal received and the background noise) and the occupancy state of the room. Speech intelligibility is described with parameters which highlight the behavior of sound energy in the room in function of time and source-to-receiver distance. Over the years standard requirements were developed on the base of many researches conducted to study in depth the problem. Many European countries have included regulatory requirements in their building codes. Some of these ones give a certain maximum value of a range of reverberation time in function of the volume and the room's use [12].

Recommendations and requirements took in consideration for this work are provided by normative, specifically the UNI 11532 [8], UNI 11367 [13], DIN 18041 [7], BB 93 [9].

1.2.1 Reverberation time T

The most significant parameter of energy decay is reverberation time T defined by ISO 3382-1 [1] as the time necessary for the sound pressure level to decrease by 60 dB after switching off the source. Smaller dynamic range can be used in technical difficulties to extrapolate a decay time of 60 dB. Thus, if T is derived from the time at which the decay curve first reaches 5 dB and 25 dB below the initial level, it is labelled T_{20}. If decay values of 5 dB to 35 dB below the initial level are used, it is labelled T_{30}.

In Sabine's formula T is directly proportional to volume V and inversely

Table 1.1: European countries normative and required reverberation time for classrooms [12].

Country	Standard/Guideline	Required T (s)
Denmark	BR 2010	$\leq$ 0.6
France	Arreté du 25 avril 2003	V<250 m^3 0.4 $\leq T \leq$ 0.8 V>250 m^3 0.6 $\leq T \leq$ 1.2
Germany	DIN 18041	T_{soll}=0.32log(V/m^3) - 0.17 (V=100 m^3 T_{soll}=0.47 V=250 m^3 T_{soll}=0.60)
Norway	NS 8175	$\leq$ 0.5 (Class C)
Spain	CTE DB-HR	V<350 m^3, T $\leq$0.5
UK	BB 93	Nursery & primary T $\leq$0.6 Secondary T $\leq$0.8

proportional to equivalent sound absorption area A in the room:

$$T = 0.161\frac{V}{A} \quad \text{(s)} \tag{1.1}$$

where:

- V is the volume of the room, in m^3;
- A is the total sound absorption area of the room, in m^2 Sabine.

The total sound absorption area of the room A can be expressed as:

$$A = \sum_i \alpha_i{\cdot}S_i \quad (\text{m}^2\,\text{Sabine}) \tag{1.2}$$

where:

- α_i the absorbing coefficient of the i-th surface;
- S_i is the i-th surface of the fixed elements in the room;

The hypothesis of this formula are:

- diffuse sound field conditions;
- a reverberant room with walls of a homogeneous geometrical acoustic nature;
- an omnidirectional source.

Despite its wide use the Sabine's formula doesn't have an accurate mathematical sense. In fact, in a zero reverberation time's room, namely anechoic chamber, the absorption coefficient α needs to be equal to infinity. Norris-Eyring's equation solves this problem expressing reverberation time in a different form, that is:

$$T = 0.161 \frac{V}{-S \log(1-\bar{\alpha})} \quad \text{(s)} \tag{1.3}$$

with:

$$\bar{\alpha} = \frac{1}{S} \sum_i S_i \cdot \alpha_i \tag{1.4}$$

where $\bar{\alpha}$ is the averaged absorption coefficient.

The differences between Sabine and Norris-Eyring equations (see equations 1.1 and 1.3) are in their assumptions. Sabine assumes that the sound wave in a room impacts the surfaces one after another while Norris-Eyring that all the surfaces are simultaneously hit by the initial sound and the successive waves are separated by mean free paths impacting the surfaces diminishing their energy with the average room absorption coefficient [14]. Despite the lack of math sense accuracy, the Sabine's equation is largely used for the evaluation of reverberation time in rooms with various usage. Following considerations are made on the base of Sabine's equation. Fixed and movable sound absorption surfaces in the room can be evaluated separately distinguishing S as the fixed and A_{obj} the movable objects, so the equation 1.1 can be expressed as:

$$T = 0.161 \frac{V}{\sum_i \alpha_i S_i + \sum_j A_{obj,j}} \quad \text{(s)} \tag{1.5}$$

- S_i is the i-th surface of the fixed elements in the room, in m^2;
- α_i the i-th absorbing coefficient of the surface;
- $A_{obj,j}$ the equivalent sound absorbing of movable elements in the room, in m^2 Sabine.

If the measures of reverberation time are made in furnished and unoccupied state a correction to evaluate the sound absorption due by people is needed. To simulate occupied state of the room (at 80% of the total occupancy) the sound absorption of people must be considered in the frequency range from 125 Hz to 4000 Hz. The T_{occ} is essential to assess the target range [7]:

$$T_{occ} = 0.161 \frac{T_{unocc}}{(1 + \frac{T_{unocc} \Delta A_{people}}{0.16V})} \quad \text{(s)} \tag{1.6}$$

where:

- T_{occ} is the reverberation time of the room in occupied state, in s;
- T_{unocc} is the reverberation time of the room in unoccupied state (measured values), in s;
- V is the volume of the room, in m^3;
- ΔA_{people} is the additional equivalent sound absorbing surface of people, in m^2 Sabine.

The additional equivalent sound absorbing surface of people ΔA_{people} depends on the occupancy density, the age and size of people, the type of clothing (winter or summer) and the acoustic characteristics of chairs. The ΔA_{people} values should be taken from technical literature or from normative. In a simplified way, the sound absorbing due to people can be evaluate with the table A.1 of DIN 18041 [7]:

$$\Delta A_{people} = N \cdot \Delta A_{1person} \quad (\mathrm{m}^2) \tag{1.7}$$

where

- N is the number of people (corresponding to 80% of occupancy);
- $\Delta A_{1person}$ is the additional equivalent sound absorbing area for each person following the Table A.1 of the DIN 18041 [7] in, m^2 Sabine.

Recommendations about the relationship of reverberation time T, volume V and use of room can be found in DIN 18041 [7] (see figure 1.1). For category A1 ("Music") the optimal reverberation time is expressed by the formula:

$$T_{60} = 0.45 \log V + 0.07 \quad (\mathrm{s}) \quad 30\,\mathrm{m}^3 \leq V < 1000\,\mathrm{m}^3 \tag{1.8}$$

where V is the volume of room in m^3. For category A2 ("Speech/Lecture" with one speaker) the optimal reverberation time is expressed by the formula:

$$T_{60} = 0.37 \log V - 0.14 \quad (\mathrm{s}) \qquad 50\,\mathrm{m}^3 \leq V < 5000\,\mathrm{m}^3 \tag{1.9}$$

where V is the volume of room m^3. For category A3 ("Teaching/Communication" with one speaker) the optimal reverberation time is expressed by the formula:

$$T_{60} = 0.32 \log V - 0.17 \quad (\mathrm{s}) \qquad 30\,\mathrm{m}^3 \leq V < 5000\,\mathrm{m}^3 \tag{1.10}$$

where V is the volume of room m^3. For category A4 (“Teaching/Communication” with more speakers) the optimal reverberation time is expressed by the formula:

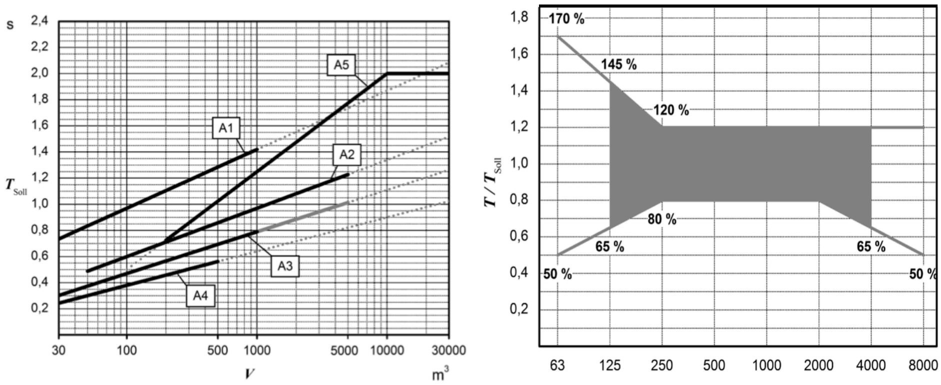
$$T_{60} = 0.26 \log V - 0.14 \quad \text{(s)} \qquad 30\,\mathrm{m}^3 \leq V < 500\,\mathrm{m}^3 \tag{1.11}$$

where V is the volume of room m^3. For category A5 (“Sport”) the optimal reverberation time is expressed by the formula:

$$T_{60} = 0.75 \log V - 1 \quad \text{(s)} \qquad 200\,\mathrm{m}^3 \leq V < 10000\,\mathrm{m}^3 \tag{1.12}$$

$$T_{60} = 2.0 \quad \text{(s)} \qquad V \geq 10000\,\mathrm{m}^3 \tag{1.13}$$

where V is the volume of room m^3.



(a) Recommended values

(b) Tolerance frequency range

Figure 1.1: Optimal reverberation time values (T_{soll} in the graph) as a function of the volume of room V and tolerance range of reverberation time compared to recommended values. The curves refer to the use of the room (A1: music, A2: speech/conference, A3: lecture/speech, A4: speech in communicative deficits, A5: sport) [7].

Research has long investigated the way to achieve the optimal reverberation time before defining parameters for the normative. Finding the optimal reverberation time in a room is the primary goal for achieve a good speech intelligibility in classrooms. Hodgson focus his works on the optimal reverberation time, hold to be the most important aspect concerning the speech intelligibility in lecture halls. Noise sources are multiple during a lecture and it’s useful to consider them in order to achieve a more realistic prediction model. Researches demonstrates contradictory results about the optimal reverberation time value. Experiments in various conditions show how this value should be zero while theoretical predictions give nonzero values as optimal value. Experimental method consists in testing speech intelligibility with a group of listener in reverberant and anechoic acoustical conditions while theoretical method uses speech intelligibility metrics, evaluated for various SNR (signal-to-noise ratio) and room characteristics. Comparing the researches made about the optimal reverberation time and analysing the method used and the supposed conditions of these works, Hodgson offers an explanation about the contradictory results. In some studies noise is not considered and in others the noise and signal sources are positioned at the same distance from the listener: these conditions lead to a value of zero for optimal reverberation time. Instead assuming a fixed noise sound level allows to realize the positive effect for the speech signal of the reverberation time. Physically it increases the early energy in the room and so the speech intelligibility. However, in reality, the noise sound level doesn't change so much. Reverberation time isn't the unique useful value to achieve a good speech intelligibility, an optimal value of it doesn't mean that the intelligibility issue is satisfied. It's needed evaluating the relationship with other parameters that consider the early energy quantity like U_{50} and C_{50}, explained in the sections 1.2.2 and 1.2.5 [15]. Hodgson highlights with STI_∞ parameter the effect of reverberation in the speech intelligibility. With typical speech and background noise levels reverberation should be detrimental so its optimum ideal value is zero. However reverberation increases the speech level so the optimum value has to be different from zero [16]. An optimal reverberation time could be found with the following equation depending on the volume of the room, increasing with its size [17]:

$$T_{opt} = 0.04V^{0.4} \quad \text{(s)} \tag{1.14}$$

Some simplifications are necessary to analyse the rooms behaviour. Reverberation time, in fact, has relative effects increasing or decreasing the intelligibility relatively to the signal and noise sources positions. If the source is far from the receiver and the noise source is near the receiver, the level of source can be greater than noise because of the amplification due to the reverberation of the room [18]. These relative effects are due to the non-diffuse nature of the sound field. The complexity of the use of a non-diffuse field is particularly in the difficult prediction of the material's absorption coefficient. Furthermore, the diffuse-field hypothesis leads to accurate results despite its simplification [19]. Diffuse-field theory permits to relate the reverberation time in occupied and unoccupied state. This is a fundamental reason to use this hypothesis since absorption due to people is determined with Sabine's formula (see equation 1.1). Starting from the reverberation time formula in occupied state:

$$T_{occ} = \frac{0.161V}{A_{unocc} + NA_p} \quad \text{(s)} \tag{1.15}$$

and unoccupied state:

$$T_{unocc} = 0.161 \frac{V}{A_{unocc}} \quad \text{(s)} \tag{1.16}$$

it's possible substitute in 1.15 the A_u in function of T_u obtaining:

$$T_{unocc} = \frac{1}{\frac{1}{T_{occ}} - \frac{A_p N}{0.161V}} \quad \text{(s)} \tag{1.17}$$

where:

- T_{occ} is the reverberation time in occupied state, in s;
- T_{unocc} is the reverberation time in unoccupied state, in s;
- A_{unocc} is the unoccupied room absorption surface, in m^2 Sabine;
- N is the number of occupants;
- V is the volume of the room, in m^3;
- A_p is the absorption per occupant, in m^2 Sabine.

According to the expression 1.17, rooms with different volumes and with the same T_{occ} and the same numbers of occupants per unit volume have the same T_{unocc} [15].

Bradley's works instead studies how change the behaviour of the room although the reverberation time remains the same. In fact, optimum value of reverberation time depends on absorption surfaces too. Adding absorbing surfaces isn't enough to reach the goal but it's important where these surfaces are positioned, as explained in section 2.6. An appropriate quantity of absorbing surfaces allows to have a good speech intelligibility. Less or more of this quantity can decrease substantially the goal of a good intelligibility. The best value of reverberation time corresponds to the maximum value of useful-to-detrimental ratio parameter U_{50} [20]. Reverberation time is the main parameter considered in an acoustical design. It is strictly related to the late reflected energy and contain the most important information about acoustical characteristics of the room. The speech intelligibility is degraded when the late sound energy covers the direct and the early sound energy. A sound absorption treatment should focus to reduce the late arriving energy to receivers. In fact, to increase the reverberation time leads to increase the late sound energy and thus decrease the early-to-late sound ratio. However, a reverberation time value too much low doesn't help the speaker to enhance greater sound levels. Thus it is necessary to find a precise reverberation time value which permits to obtain a good gain of early reflection and a little contribution of late ones. The most important aspect in speech intelligibility is the evaluation of early reflections benefit to listeners who can't take advantage of direct sound. Zero or a very short reverberation time is never desirable so the literature contradiction isn't insolvable . It's important to specify that reverberation time changes when speech and noise levels vary, this highlights the relationship with useful-to-detrimental ratio U_{50}. In order to achieve its maximum value, even though the variations of its parameter, it's useful to have a certain signal-to-noise ratio (SNR) [21].

Pelegrin Garcia's study shows a prediction model to evaluate the parameters that influence speaker's comfort. The decay time $DT_{40,ME}$ is the time necessary to the backwards integrated energy curve of an $OBRIR$ (oral-binaural room impulse response), which measures the impulse response with a microphone located at the position of ears of a dummy head using a speaker source inside its mouth (this is the mean of the subscript ME, mouth-to-ears), to decay 60 dB after the arrival of the direct sound calculated from the initial decay time of 40 dB and assuming a linear decay. This definition is important to obtain the range in which the optimal reverberation time fits. According to Pelegrin Garcia, the optimal reverberation time for vocal comfort (VC) is expressed by [12]:

$$T_{target,VC} = 0.032\sqrt[3]{V} + 0.38 \quad \text{(s)} \tag{1.18}$$

where V is the volume room in m^3. This prediction model expresses the room-averaged $DT_{40,ME}$ as a function of the reverberation time and the volume room.

1.2.2 Sound clarity index C_{t_e}

The clarity index C_{t_e} is defined as an early-to-late arriving sound energy ratio. The time intervals t_e can be assumed of 50 or 80 ms if the parameter concerns respectively speech or music. ISO 3382 [1] defines clarity index as:

$$C_{t_e} = 10 \log \frac{\displaystyle\int_0^{t_e} p^2(t)\ \mathrm{d}t}{\displaystyle\int_{t_e}^{\infty} p^2(t)\ \mathrm{d}t} \quad \text{(dB)} \tag{1.19}$$

where on the numerator there is the energy of the direct sound and the first reflections while on the denominator the energy of the late reflections. In particular:

- C_{t_e} is the early-to-late index, in dB;

- t_e is the early time limit (50 ms for speech and 80 ms for music), in s;

- $p(t)$ is the instantaneous sound pressure measured at the measurement point, in Pa.

Thus, based on the extreme used the formula can assume the form:

$$C_{50} = 10 \log \frac{\displaystyle\int_0^{50} p^2(t)\ \mathrm{d}t}{\displaystyle\int_{50}^{\infty} p^2(t)\ \mathrm{d}t} \quad \text{(dB)} \tag{1.20}$$

$$C_{80} = 10 \log \frac{\displaystyle\int_0^{80} p^2(t)\ \mathrm{d}t}{\displaystyle\int_{80}^{\infty} p^2(t)\ \mathrm{d}t} \quad \text{(dB)} \tag{1.21}$$

Requirement for the clarity index is provided by UNI 11367 [13] which suggests as suitable value in spaces designed for speech:

$$C_{50} \geq 0 \tag{1.22}$$

where C_{50} is intended as a mean value over all the source and receivers values in the octave band range of 500÷2000 Hz.

A predictive value of the sound clarity index can be expressed in function of reverberation time and the source-receiver distance by Barron's formula used in his revised theory [22] (see section 1.2.6):

$$C_B = 10 \log \Big[\frac{V e^{(0.0122r + 13.82t)/T}}{1024 T r^2} + e^{13.82t/T} - 1 \Big] \tag{1.23}$$

where:

- V is the room volume, in m^3;

- r is the source-receiver distance, in m;

- t is the ratio dividing time, in s;

- T is the reverberation time, in s.

The first term represent the direct-to-late sound energy ratio while the second the early-to-late ratio excluding the direct sound. UNI 11532 [8] uses Barron's formula as predictive model and also a simplified expression of it:

$$C_{50} \approx 10 \log \left(e^{0.691/T} - 1\right) \quad \text{(dB)} \tag{1.24}$$

If directivity factor Q is fundamental, it can be included in Barron's formula (see equation 1.23) which become, assuming t equal to 0.05 s:

$$C_B = 10 \log \Big[\frac{QVe^{(0.0122r+0.691)/T}}{1024Tr^2} + e^{0.691/T} - 1 \Big] \tag{1.25}$$

If inverted this formula can be used also to calculate the minimum Q to obtain a certain value of C_{50} [23].

1.2.3 Background noise

A fundamental aspect of speech intelligibility is background noise. The signal-to-noise ratio highlights how much the background noise is detrimental for intelligibility. In a lecture hall, during a lecture, the background noise is made by the HVAC devices, the student activity and, if the room is near the street, the external traffic. The DIN 18041 [7] suggests an optimal difference between signal level and background noise of almost 10 dB. Moreover it recommends that the background noise level ($L_{NA,Bau}$) doesn't exceed some values of the signal level based on the category seen in section 1.2.1:

- $L_{NA,Bau} \leq 30$ dB for rooms of category A1, for music performances;
- $L_{NA,Bau} \leq 35$ dB for rooms of category A2, for speech and lecture performances, A3, for teaching and communication performances with one speaker and A4, for teaching and communication performances with more speakers;
- $L_{NA,Bau} \leq 40$ dB for rooms of category A5, for sport performances.

These values can be augmented of 5 dB in case of PA system.

Many studies [20] [16] highlight that for an excellent speech intelligibility a signal-to-noise ratio of +15 dB is necessary. Bradley [24] relates also the intelligibility to the language used in speech: the negative effects are amplified if a person is listening in a second language. Hodgson [18], considering a non-diffuse sound field in the room, points out the relative effects of distance between signal source and noise source due to the reverberation of the room that can amplify the noise level if the receiver is far from the noise source and near to the signal source. Furthermore a prediction model for ventilation noise (VN) and student-activity noise (SA) has been derived [25]. This prediction model was built recording lectures, evaluating the long-term sound pressure level frequency distributions and fitting these with normal-distribution curves. The long-term sound pressure level frequency permits to separate the different levels due to the various sources type. The speech structure is a quasi-continuous signal with short and longer breaks due to the division of the single words or sentence and when the teacher, for example, writes on the blackboard. In the latter case, however, the long break allows the increase of the student noise so, for an accurate evaluation it's more interesting to study the short breaks between sentences. For what concerns the ventilation noise, its sound pressure level has to be recorded in the same day of the lecture but it isn't always possible. The calculation procedure was the following:

- the signals were squared;
- short-term, mean-square pressures were calculated with an interval time of a 200 ms;
- interval sound pressure levels were calculated;
- a statistical distribution of sound pressure levels were determined and plotted;
- the distribution were fitted by one, two, three or more normal-distribution curves to identify each source.

The total distribution of sound levels were generally non-symmetrical so the separation of sound components allows to make each distribution curve more symmetrical (see figure 1.2). Three peaks are notable:

- the lower sound pressure levels corresponding, considering the similar distribution in the unoccupied state measured values, to the ventilation system;
- the middle one, since doesn't decrease with the source-to-receiver distance, to the student-activity noise;
- the higher sound pressure levels which is consequently associated to the teacher's speech signal.

When the signal-to-noise ratio is too much low the distribution curve has just one peak so it's impossible to highlight the single sound components.

It's useful to note that the middle peak associated to the student-activity noise is an energetic sum of this with ventilation system noise. Thus this peak doesn't correspond to the level of that component which can be calculated subtracting the energy of the lowest sound pressure levels. Similarly it happens for speech signal levels obtained with the energetic subtraction between highest and middle levels. To specify the speaker's sex is necessary since the measures may differentiate about ±1.3 dB from the average level. The prediction model developed by Hodgson aims to give the levels of the

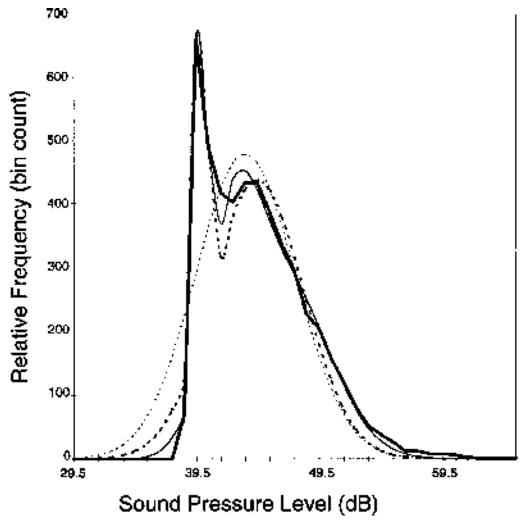


Figure 1.2: A type of total A-weighted sound pressure level recorded during a lecture [25]. The solid and thick, the dashed and the dotted lines are the normal-distribution fitting curves. It can be noted the non-symmetrical shape of the total distribution curve of sound pressure levels unlike fitting curves, more symmetrical.

various components of the sound so the following parameters are debated:

- Ventilation noise *VN*, that is the noise due to air conditioning system;

- Student-activity noise *SA*, that is the noise due to the presence of the students during lectures;

- Instructor speech signal *SL*, that is the received sound pressure level of the teacher;

- Instructor sound power *LW*, that is the instructor output sound power.

Ventilation noise (VN) in this prediction model depends on the type of equipment, the mechanical/acoustical power, the source-to-receiver distance and by the classroom's acoustical characteristics. It can be calculated as:

$$VN = 57.6 + 10.3 \log n + 0.68W - 21.3 \log A_0 \quad \text{(dB)} \tag{1.26}$$

where:

- n is the number of students;
- W is the room width, in m;
- A_0 is the total occupied-room absorption area, in m^2.

Student-activity noise (SA) can be expressed by the equation:

$$SA = 9.22 + 6.4 \log n + 0.71SL + 1.53I_{sex} - 8.0 \log A_0 \quad \text{(dB)} \tag{1.27}$$

where:

- n is the number of students;
- SL is the received speech level due to the instructor (see equation 1.29);
- I_{sex} is the speaker's sex and is equal to 0 for males and 1 for females;
- A_0 is the total occupied-room absorption area, in m^2.

An alternative expression of student-activity noise which doesn't depend on the instructor speech signal (SL) is given by:

$$SA = 83.0 + 10.0 \log n - 34.4 \log A_0 + 0.081A_0 \quad \text{(dB)} \tag{1.28}$$

where:

- n is the number of students;
- A_0 is the total occupied-room absorption area, in m^2.

Instructor speech signal (SL) is given by:

$$SL = 48.5 - 2.6I_{sex} + 0.58SA - 4.0 \log r + 0.013V - 11.7 \log A_0 \quad \text{(dB)} \tag{1.29}$$

where:

- I_{sex} is the speaker's sex and is equal to 0 for males and 1 for females;
- SA is the student-activity noise level, in dB;

- r is the source-to-receiver distance, in m;
- V is the room volume, in m^3;
- A_0 is the total occupied-room absorption area, in m^2.

Instructor sound power (LW) is evaluated with the equation:

$$LW = 54.8 - 2.6 I_{sex} + 0.5 SA + 0.016 V - 9.6 \log A_0 \quad \text{(dB)} \tag{1.30}$$

where:

- I_{sex} is the speaker's sex and is equal to 0 for males and 1 for females;
- SA is the student-activity noise level, in dB;
- V is the room volume, in m^3;
- A_0 is the total occupied-room absorption area, in m^2.

In a successive work, Hodgson specifies that noise sources are multiple and their contributions can be summed to obtain a total noise energy, considering that the furthest noise sources are negligible in prediction of optimal reverberation time. Only the noise sources that are closer ones to listener than the source have to be considered [15].

Speech intelligibility in a classroom is mainly determined by the signal-to-noise ratio that is almost constant because of *Lombard effect*. In fact, despite student-activity noise varies in time and so the total background noise, the signal-to-noise ratio remains constant since the speaker automatically adapt his output level. The *Lombard effect* doesn't affect only the speaker but the student-activity noise too. If the ventilation system varies its level the students increase consequently their level. Since the noise source level doesn't decrease after the addition of absorbing material, which affect most of all the late energy, in the room it's easy to guess that the noise level arrives meanly as direct sound and as early reflection. Thus, noise control should be focused on the source of the noise [21].

1.2.4 Speech Transmission Index STI

Speech is made up of fluctuations in the signal's intensity which correspond, depending on their speed, to the subdivision of sentences and individual words if they are slow or to individual phonemes within words if they are fast. These fluctuations are affected by a transmission channel with a Modulation Transfer Function (MTF). A good intelligibility can be obtain if the envelope intensity is preserved as much possible. The MTF highlights how much the signal is degraded by distortions like noise, reverberation and echo representing the decrease of modulation's depth in function of the modulation frequency. The Speech Transmission Index (STI) is an objective measure between 0 and 1 representing the quantity of speech understood from a listener to evaluate the sound quality. This parameter has been developed since the 1970s and today is defined by IEC 60268-16. It depends on the reverberation time, the sound pressure level and the background noise. The STI index is based on the concept of modulation of a carrier assuming that the human speech is simulated in this way. A complex signal consisting of 98 combination (14 modulation frequencies from 0.63 to 12.5 Hz x 7 frequency bands from 125 to 8000 Hz) is used with a speaker of the size of a human mouth, acting as a "speaker". The method associates the characteristics of the environment with the transfer function comparing the input (the modulated signal) and the output (the microphone signal at the location where the STI is to be determined). In this way the microphone signal is modified by the acoustic characteristics of the room (reverberation time and signal to noise ratio).

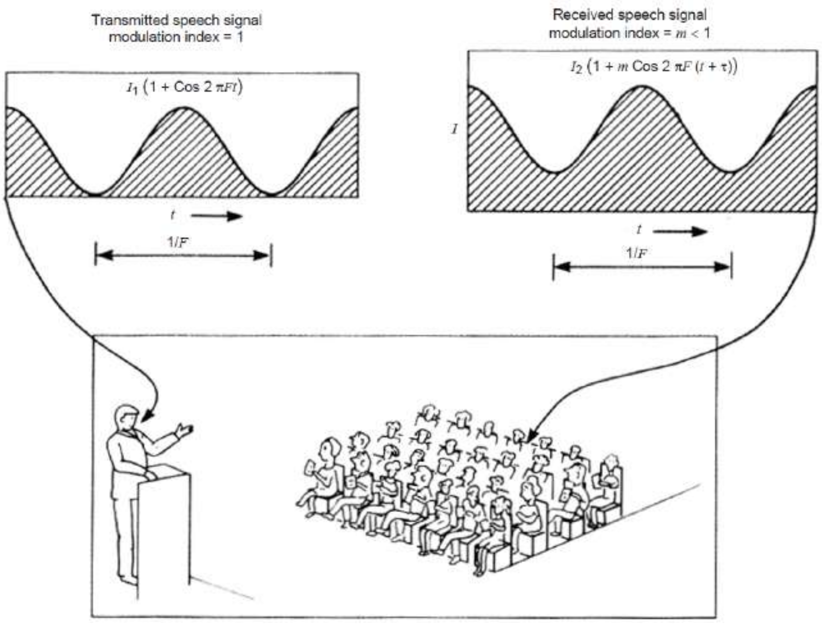


Figure 1.3: Concept of the effect of a transmission channel on the modulation of signal. The speech signal is represented on the left and the received signal reduced by the transmission channel is represented on the right [10].

STI can be measured in two ways:

- with the direct method which uses a suitably modulated signal;
- with the indirect method which is only applicable in the case of linear and invariant transmission systems, and it uses the impulse response and the signal-to-noise ratio.

The modulation transfer function is theoretically defined by the formula:

$$m(f) = \frac{1}{\sqrt{1 + \left(2\pi f \frac{T}{13.8}\right)^2}} \cdot \frac{1}{1 + 10^{0,1(-S/N)}} \tag{1.31}$$

where:

- f is the modulation frequency, in Hz;
- T is the reverberation time, in s;
- S/N is the signal-to-noise ratio, in dB.

An human ear has an inherent effect that masks higher frequencies when a louder low frequency sound makes them inaudible, because their relative levels exceed a certain threshold. So masking effect depends by the sound pressure level difference between two octave bands. The masking intensity $I_{am,k}$ for octave band k is given by:

$$I_{am,k} = I_{k-1} \cdot amf \quad (\mathrm{W^2/m^2}) \tag{1.32}$$

where:

- I_{k-1} is the intensity of the adjacent lower octave band, in $\mathrm{W^2/m^2}$;
- amf is the level dependent auditory masking factor which is a function depending by I_{k-1}.

The intensity I_{k-1} for an octave band $k-1$ is given by:

$$I_{k-1} = 10^{(L_{k-1}/10)} \quad (\mathrm{W^2/m^2}) \tag{1.33}$$

where L_{k-1} is the overall sound pressure level for the octave band $k-1$ in dB. The auditory masking factor amf for an octave band is given by:

$$amf = 10^{(amdB/10)} \tag{1.34}$$

where $amdB$ is the octave band level dependent auditory masking in dB which value is tabled in function of the sound pressure level of octave band $k-1$. The reception threshold intensity $I_{rt,k}$ for octave band k is given by:

$$I_{rt,k} = 10^{(ART_k/10)} \quad (\mathrm{W}^2/\mathrm{m}^2) \tag{1.35}$$

where ART_k is the absolute speech reception threshold for octave band in dB which is a tabled value and is defined by the absolute threshold of hearing and the minimal required dynamic range for the correct understanding of speech. Calculation of STI value is a complex procedure which requires some steps. Using the direct method the modulation transfer at each modulation frequency is evaluated deducing the modulation depth of the received signal mdr for each octave band output k:

$$mdr_{k,f_m} = 2\cdot \frac{\sqrt{[\sum I_k(t)\cdot \sin(2\pi f_m t)]^2 + [\sum I_k(t)\cdot \cos(2\pi f_m t)]^2}}{\sum I_k(t)} \tag{1.36}$$

where:

- f_m is the modulation frequency, in Hz;
- t is the time, in s;
- $I_k(t)$ is the intensity envelope in function of time for octave band k, in $\mathrm{W}^2/\mathrm{m}^2$;

Calculated the modulation indices of the received and transmitted signals, the modulation transfer ratio can be evaluated with the following formula:

$$m_{k,f_m} = mdr_{k,f_m}/mdt_{k,f_m} \tag{1.37}$$

where:

- mdr_{k,f_m} is the modulation depth of the received signal for octave band k and modulation frequency f_m;
- mdt_{k,f_m} is the modulation depth of the transmitted signal for octave band k and modulation frequency f_m.

The modulation transfer ratio obtained has to be corrected with the auditory masking effect with the formula:

$$m'_{k,f_m} = m_{k,f_m} \frac{I_k}{I_k + I_{am,k} + I_{rt,k}} \tag{1.38}$$

where:

- m_{k,f_m} is the derived modulation transfer ratio value for octave band k and modulation frequency f_m;

- I_k is the acoustic intensity level for octave band k, in $\mathrm{W}^2/\mathrm{m}^2$;

- $I_{am,k}$ is the acoustic intensity level for the level dependent auditory masking effect on octave band k, in $\mathrm{W}^2/\mathrm{m}^2$;

- $I_{rt,k}$ is the acoustic intensity level if the reception threshold for octave band k, in $\mathrm{W}^2/\mathrm{m}^2$.

The modulation transfer ratio m'_{k,f_m}, after its correction, has to be transformed into an effective signal-to-noise ratio with the formula:

$$SNR_{eff\,k,f_m} = 10\log\frac{m'_{k,f_m}}{1-m'_{k,f_m}} \quad \text{(dB)} \tag{1.39}$$

Its value is limited to the range of $-15 \div +15$ dB because the ratio could have an infinity value. $SNR_{eff\,k,f_m}$ serves to calculate the transmission index TI for each octave band and modulation frequency with the formula:

$$TI_{k,f_m} = \frac{SNR_{eff\,k,f_m} + 15}{30} \tag{1.40}$$

Founded the transmissions indexes they are averaged over modulation frequencies obtaining the modulation transfer index MTI_k per octave band with the formula:

$$MTI_k = \frac{1}{n}\sum_{m=1}^{n} TI_{k,f_m} \tag{1.41}$$

where:

- TI_{k,f_m} is the transmission index for each octave band k and modulation frequency f_m;

- m is the index of the modulation frequency;

- n is the number of modulation frequencies per octave band.

Obtained the MTI, it is possible to calculate the STI value with the formula:

$$STI = \sum_{k=1}^{7}\alpha_k{\cdot}MTI_k - \sum_{k=1}^{6}\beta_k{\cdot}\sqrt{MTI_k{\cdot}MTI_{k+1}} \tag{1.42}$$

where:

- α_k is the weight factor for octave band k;
- MTI_k is the modulation transfer index for octave band k;
- β_k is the redundancy factor between octave band k and octave band $k+1$.

Table 1.2: MTI octave band weight (α) and redundancy (β) factors in function of speaker's sex.

Octave band (Hz)		125	250	500	1000	2000	4000	8000
Males	α	0.085	0.127	0.230	0.233	0.309	0.224	0.173
	β	0.085	0.078	0.065	0.011	0.047	0.095	–
Females	α	–	0.117	0.223	0.216	0.328	0.250	0.194
	β	–	0.099	0.066	0.062	0.025	0.076	–

STI values are divided in a scale to evaluate a quality index that is reported in table 1.3. The STI method based on the 14 modulation frequencies for the

Table 1.3: Speech Transmission Index values corresponding to the intelligibility by IEC 60268-16 and correctly included syllables and words percentages.

STI value	Quality index	Percentage of syllables heard correctly (%)	Percentage of words heard correctly (%)
$0.00 \div 0.30$	bad	$0 \div 34$	$0 \div 67$
$0.30 \div 0.45$	poor	$34 \div 48$	$67 \div 78$
$0.45 \div 0.60$	fair	$48 \div 67$	$78 \div 87$
$0.60 \div 0.75$	good	$67 \div 90$	$87 \div 94$
$0.75 \div 1.00$	excellent	$90 \div 96$	$94 \div 96$

7 octave band which produces 98 test signals is called $FULLSTI$ method. Since its complex calculation requires many resources (with an average of 10 seconds per test signal, a $FULLSTI$ measurement requires approximately 15 minutes) simplified methods are used. They are:

- $STIPA$ that uses only 2 modulations in each of the seven octave bands. It's used to calculate STI values for public address systems (PA);

- $STITEL$ that uses only one test signal with seven modulation frequencies predefined, one per octave band. It's used to calculate STI values for telecommunication systems;

- $RASTI$ that use 9 modulation frequencies, 5 for the 2000 Hz octave band and 4 for the 500 Hz octave band. It's a condensed version of the $FULLSTI$ method but it's obsolete according to IEC 60268-16 [10].

Despite its definition, $STIPA$ method could be used also for natural speech measurements and not only for public address systems. $STIPA$ method is validated for male speech spectrum. There's a difference indeed between male and female speech spectrum: the male one is subjected to more distortions unlike the female one, that is considered more intelligible. Gender difference are expressed with different weighting and redundancy factors that influence the STI calculation as seen in equation 1.42.

Standard requirements for STI value can be found in UNI11367 [13] and in Building Bulletin 93 [9] that suggest a STI value ≥ 0.6 in spaces designed for speech, the Finnish standard suggest a value equal to 0.8 but nobody specifies the noise level to consider for STI calculation [26].

1.2.5 Useful-to-detrimental ratio U_{50}

Useful-to-detrimental ratio U_{50} is a further parameter for speech intelligibility. It's a more complex calculation of clarity index C_{te} because it adds to the early-to-late energy ratio the background noise contribution. Bradley defines this parameter as the ratio of useful fraction and the detrimental fraction of energy in the room [27]. The useful fraction energy which arrives in the t_e time's interval considered is given by:

$$Useful = \Big[\frac{E_e}{(E_e + E_l)}\Big] \cdot E_{SL} = \Big[\frac{C_{t_e}}{(Ct_e + 1)}\Big] \cdot E_{SL} \tag{1.43}$$

where:

- E_e is the early energy;

- E_l is the late energy;

- E_{SL} is the total energy of speech level;

- C_{te} is the clarity index calculated in the t_e time's interval.

The detrimental fraction, similarly, is calculated:

$$Detrimental = \left[\frac{1}{(C_{t_e}+1)}\right] \cdot E_{SL} + E_{BL} \tag{1.44}$$

where E_{BL} is the total energy of background level. Dividing the equations 1.43 and 1.44 it's possible to obtain an expression of useful-to-detrimental ratio for a certain time's interval t_e:

$$U_{t_e} = \frac{C_{t_e}}{[1 + (C_{t_e}+1) \cdot E_{BL}/E_{SL}]} \quad \text{(dB)} \tag{1.45}$$

Thus the calculation of this parameter requires previously the measure of clarity index C_{t_e} and it's evaluated in the same time limits. It's possible and useful to express the U_{50} in function of reverberation time in the context of classrooms, where the speech intelligibility is a fundamental aspect of design, but some considerations are previously needed. Starting from the traditional distribution of sound pressure level throughout the space:

$$L_p(r) = L_W + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4(1-\alpha)}{\alpha S}\right) \quad \text{(dB)} \tag{1.46}$$

where:

- L_W is the sound power level emitted from the source, in dB;
- r is the source-receiver distance, in m;
- α is the mean absorption coefficient;
- S is the total geometrical area, in m^2;
- Q is the directivity coefficient of a human speaker.

The terms within the brackets represent respectively the direct sound between source and receiver and the diffuse sound field. There is a certain distance from the source which makes equal the values of the two terms and it's called $reverberation\, radius$ and it's given by:

$$r_{rev} = \sqrt{\frac{\alpha S Q}{16\pi(1-\alpha)}} \quad \text{(m)} \tag{1.47}$$

Over the r_{rev} ends speech intelligibility depends by the sound energy reflected on the surfaces of the room. Taking into account Barron and Lee's revised theory [22] and the exponential decay of the energy the equation 1.46 becomes:

$$L_p = L_W + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4}{\alpha S}e^{-0.04r/T}\right) \quad \text{(dB)} \tag{1.48}$$

Using Eyring's formula the reverberation time can be expressed as follow:

$$T = \frac{-0.16V}{S\ln(1-\alpha)} = \frac{0.04\, mfp}{\ln(1-\alpha)} \quad \text{(s)} \tag{1.49}$$

where:

- V is the volume of the room, in m^3;
- mfp is the mean free path of the room, that is the average distance traveled by a particle between successive impacts, in m.

So the equation 1.48 becomes:

$$L_p = L_W + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4(1-\alpha)^{r/mfp}}{\alpha S}\right) \quad \text{(dB)} \tag{1.50}$$

The Barron and Lee's revised theory were developed for concert halls but with the Bradley and Sato correction it's possible to use it for the classrooms too [21]. Thus, adding a factor in the order of 2, the 1.50 becomes:

$$L_p = L_W + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4(1-\alpha)^{2\cdot r/mfp}}{\alpha S}\right) \quad \text{(dB)} \tag{1.51}$$

Since the sound power's decay for the Sabine's theory is predicted exponential, it's possible to express it as:

$$W(t) = W(t=0)e^{(-13.8t/T)} \tag{1.52}$$

Thus, integrating it into the time limits of early and late energy considered, the energy expression are obtained:

$$E_{early} = E_0[1 - e^{(-0.69/T)}] \tag{1.53}$$

$$E_{late} = E_0 e^{(-0.69/T)} \tag{1.54}$$

where E_0 is an arbitrary constant. Joining all the considerations made it's possible express the sound pressure levels of early and late fractions of energy as:

$$L_{p,early} = L_{W,speech} + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4(1-\alpha)^{2\cdot r/mfp}}{\alpha S}\cdot(1 - e^{(-0.69/RT)})\right) \quad \text{(dB)} \tag{1.55}$$

where:

- $L_{W,speech}$ is the output speech level, in dB;
- Q is the directivity factor;
- r is the source-receiver distance, in m;
- α is the absorption coefficient;
- S is the i-th surface of the fixed elements in the room, in m^2;
- mfp is the mean free path, in m;
- T is the reverberation time, in s.

$$L_{p,late} = L_{W,speech} + 10 \log \left(\frac{4(1-\alpha)^{fb \cdot r/mfp}}{\alpha S} e^{(-0.69/RT)} \right) \quad \text{(dB)} \tag{1.56}$$

where the $L_{W,speech}$ is the vocal output of the speaker. Nijs *et al.*, simplifying the calculation of noise and considering the reverberant contribution more important than the direct one, evaluate the noise level as:

$$L_{p,noise} = L_{W,noise} + 10 \log \frac{4}{\alpha S} \quad \text{(dB)} \tag{1.57}$$

Since its simplification this formula is inaccurate when the noise level isn't constant. Adding the noise contribution to the equation 1.56 it can be written as:

$$L_{p,late+noise} = L_{W,speech} + 10 \log \left(\frac{4(1-\alpha)^{fb \cdot r/mfp}}{\alpha S} e^{(-0.69/RT)} + \frac{4 \cdot 10^{-SN/10}}{\alpha S} \right) \tag{1.58}$$

where:

$$SN = L_{W,speech} - L_{W,noise} \quad \text{(dB)} \tag{1.59}$$

Thus, the useful-to-detrimental ratio U_{50} can be expressed as the difference between the equations 1.55 and 1.58:

$$U_{50} = L_{p,early} - L_{p,late+noise} \quad \text{(dB)} \tag{1.60}$$

In the context of a classroom the level of noise can vary during the time but the SN can be considered constant because of the $Lombard\ effect$ which brings the speaker to increase his speech level if the background noise grows.

The useful-to-detrimental ratio U_{50} is an evaluation's parameter of speech intelligibility like the speech transmission index STI but it permits the calculation of the early and late contributions. The relationship between this two parameter can be highlighted correlating the transmission index TI to the clarity index C_{50} with the equation [26]:

$$TI = 0.030C_{50} + 0.555 \tag{1.61}$$

or with:

$$C_{50} = 33.33TI - 18.5 \quad \text{(dB)} \tag{1.62}$$

By this relationship it is possible to obtain a matching scale between STI and U_{50} as in the following table.

Table 1.4: Relationship between STI and U_{50} parameters [26].

STI	Quality index	U_{50} (dB)
< 0.3	bad	< -8.5
$0.30 \div 0.45$	poor	$-8.5 \div -3.5$
$0.45 \div 0.60$	fair	$-3.5 \div 1$
$0.60 \div 0.75$	good	$1.5 \div 6.5$
> 0.75	excellent	$6.5 \div 11.5$

Bradley's researches give as an optimal value of U_{50} for a good speech intelligibility equal to 1 dB corresponding to a value of STI equal to 0.60 which is the standards requirements of many regulations. The useful-to-detrimental ratio can relate to speech intelligibility (SI) with the equation [20]:

$$SI = 98.24 + 0.861(U_{50}) - 0.0863(U_{50})^2 \quad (\%) \tag{1.63}$$

Considering Hodgson's research [15] it's interesting to treat the noise source in a certain position and generated by a point considering its relative effect on speech intelligibility and room's reverberation in order to express useful-to-detrimental ratio U_{50} as follows:

$$U_{50} = 10 \log \left(\frac{(r_{h_s}^2 / r_h^2) + 1 - e^{-k/20}}{e^{-k/20} + 10^{(L_{nf1} - L_{sf1})/10} \cdot \left(\frac{r_{h_s}^2}{r_n^2} + \frac{q_s}{q_n} \right)} \right) \quad \text{(dB)} \tag{1.64}$$

where:

- r_{h_s} is the reverberation radius of the source-receiver distance, in m;
- r_{h_s} is the source-receiver distance, in m;

- r_n is the noise-receiver distance, in m;
- q_s is the directivity index of speech source, in dB;
- q_n is the directivity index of noise source, in dB;
- L_{nf1} is the anechoic noise level at 1 m of distance from the source, in dB;
- L_{sf1} is the anechoic speech level at 1 m of distance from the source, in dB;
- k is a value derived from the Norris-Eyring's reverberation time's formula and equal to $\log(10^6)/T$.

Despite the assumption of the relative effects it's important to note that the equation 1.64 is based on diffuse-field theory.

Useful-to-detrimental ratio U_{50} is strictly related to reverberation time in fact find its maximum level permits to find the optimal reverberation time of the room [21].

Even though its importance in the evaluation of the vary contributions of direct, early and late energies and its tightly relationship to speech intelligibility and reverberation time, the useful-to-detrimental ratio U_{50} isn't considered in standard requirements.

1.2.6 Spatial distribution of sound energy

Sound strength G is a parameter which describes how much the room increases the sound level heard by listener and so how much it naturally amplifies the sound. ISO 3382-1 [1] defines G as the logarithmic ratio of the sound energy measured in a certain position in the room and the sound energy which arrives to listener at a distance of 10 m from the source in a free field. G can be calculated with the formula:

$$G = 10 \log \frac{\displaystyle\int_0^{\infty} p^2(t)\ \mathrm{d}t}{\displaystyle\int_0^{\infty} p_{10}^2(t)\ \mathrm{d}t} = L_p - L_{p,10} \quad (\mathrm{dB}) \tag{1.65}$$

where:

- $p(t)$ is the sound pressure measured in a certain position, in Pa;
- $p_{10}(t)$ is the sound pressure measured in a free field at a distance of 10 m from source, in Pa;

- p_0 is equal to 20 µPa;

- L_p is the sound pressure level of $p(t)$, in dB;

- $L_{p,10}$ is the sound pressure level of p_{10}, in dB.

The sound pressure level $L_{p,10}$ could be measured directly in anechoic room using a source-to-receiver distance of 10 m. If the anechoic room isn't big enough it's possible measure $L_{p,10}$ at a distance d ($\geq$ 3 m) from the source with the formula:

$$L_{p,10} = L_{p,d} + 20\log\left(d/10\right) \quad \text{(dB)} \tag{1.66}$$

where:

- d is the source-to-receiver distance that has to be of a minimum of 3 m;

- Lp, d is the sound pressure level measured at the distance d, in m.

Another method to obtain $L_{p,10}$ value is to measure the sound pressure level at a distance of 1 m. A distance like this permits to take only the direct sound without any reflections. After the measure $L_{p,10}$ is obtained with the formula:

$$L_{p,10} = Lp, 1 + 20\log\left(10/1\right) = L_{p,1} - 20 \quad \text{(dB)} \tag{1.67}$$

where:

- $L_{p,10}$ is the sound pressure level at a distance of 10 m from the source, in dB;

- $L_{p,1}$ is the sound pressure level at a distance of 1 m from the source, in dB.

When the power level of the source, that has to be omnidirectional, is known, the G can be calculated with the formula:

$$G = L_p - L_W + 31 \quad \text{(dB)} \tag{1.68}$$

Sound strength G is a fundamental parameter closely related to the subjective perception of loudness.

An important theory about sound decay in concert halls was introduced by Barron and Lee in 1986 [22] studying the behaviour of 17 halls. The main contribution brought consists in the evaluation of the total sound subdividing it in the three parts: the direct, early reflected and late reflected sound. The decay of each part was evaluated and compared with classical theory prediction. A fundamental aspect of the research was the study of the relationship between total decay sound and the source-receiver distance which isn't linear as the traditional theory predicts but is exponential. The results display that the reflection components decrease linearly but the direct slightly exponentially. Therefore the early part of the sound, which is the sum of the direct and the early reflected decays exponentially. These results are shown in the figure 1.4. The integrating energy in function of a certain

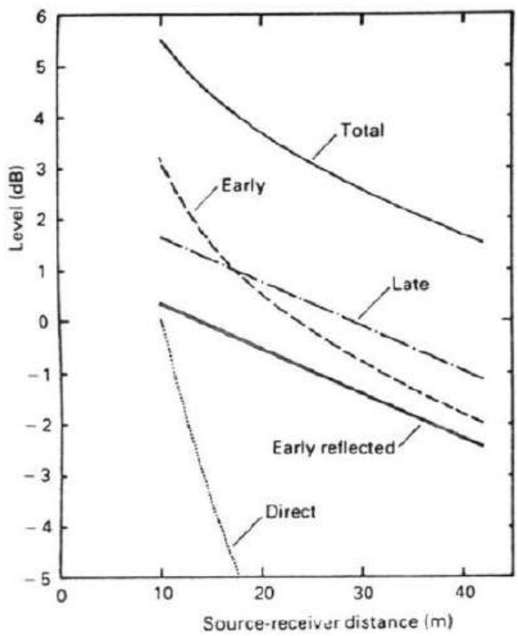


Figure 1.4: Tendency of total sound decay and its single contributions by Barron and Lee's revised theory. [22]

time interval, reverberation time and the volume of the room in the revised theory is given by:

$$i_t = (31200T/V)e^{-13.82t/T} \tag{1.69}$$

where:

- T is the reverberation time, in s;
- V is the volume, in m^3;
- t is the time interval, in s.

So the various components (indicated as d for direct, e_r for the early reflected and l for the late sound) of the sound decay, calculated in the time interval with threshold of 80 ms, are expressed by:

$$d = 100/r^2 \tag{1.70}$$
$$e_r = (31200T/V)e^{-0.04r/T}(1 - e^{-1.11/T}) \tag{1.71}$$
$$l = (31200T/V)e^{-0.04r/T}e^{-1.11/T} \tag{1.72}$$

where:

- r is the source-receiver distance, in m;
- T is the reverberation time measured in the 500 - 1000 - 2000 Hz octave bands, in s;
- V is the volume of the room, in m^3.

Subdividing the sound decay is important to evaluate how it's various the kind of the mean contribution that a certain position can receive compared to another one. The Barron and Lee's revised theory permits to have predicted results more accurate of a mean value of -2.5 dB. A relevant aspect of the Barron and Lee's work is the technique of comparison between various halls: it consists in the introduction of sound strength G parameter (see section 1.2.6). With this technique it's possible to compare the measured total energy value of halls and consequently compare their behavior. With the subdivided energy, G can be expressed in function of the source-receiver distance by:

$$G(r) = L(r) - L_{10} = 10 \log (d + e_r + l) \qquad \text{(dB)} \tag{1.73}$$
$$G(r) = 10 \log \left(\frac{100}{r^2} + 31200 \frac{T}{V} e^{-0.04r/T} \right) \qquad \text{(dB)} \tag{1.74}$$

And so the other energy parameter as clarity index C_{t_e} and early sound level E can calculate as:

$$C_{te} = 10 \log \left[\frac{(d + e_r)}{l} \right] \qquad \text{(dB)} \tag{1.75}$$
$$E - L_0 = 10 \log (d + e_r) \qquad \text{(dB)} \tag{1.76}$$

The Barron and Lee's theory is important not only for studying the concert halls behavior but it gives also a relevant contribution in the evaluation of the behaviour of other kind of rooms. Bradley and Sato [21] studied and adapted the revised theory in a classroom context. In classroom the acoustic behaviour concerns speech and not music so the time interval threshold is of 50 ms instead 80 ms. The equations 1.70, 1.71, 1.72 become:

$$d = 100/r^2 \tag{1.77}$$

$$e_r = (31200T/V)e^{-0.04r/T}(1 - e^{(-0.05)13.82/T}) \tag{1.78}$$

$$l = (31200T/V)e^{-0.04r/T}e^{(-0.05)13.82/T} \tag{1.79}$$

The total strength G is evaluated as seen in the equation 1.73 while the calculation of its each component is given by:

$$G_{e_r} = 10\log(e_r) \quad (\mathrm{dB}) \tag{1.80}$$

$$G_{50} = 10\log(d + e_r) \quad (\mathrm{dB}) \tag{1.81}$$

$$G_{late} = 10\log(l) \quad (\mathrm{dB}) \tag{1.82}$$

$$G_{total} = 10\log(d + e_r + l) \quad (\mathrm{dB}) \tag{1.83}$$

To evaluate the conformity with revised theory the reflection part of G were compared between measured and predicted values. The results show a difference between measured and predicted values of a factor of 2. So the correction to apply to the energy components, adding the factor of 2 to the reflection contributions and expressing the direct contribution in function of directivity, give new equations:

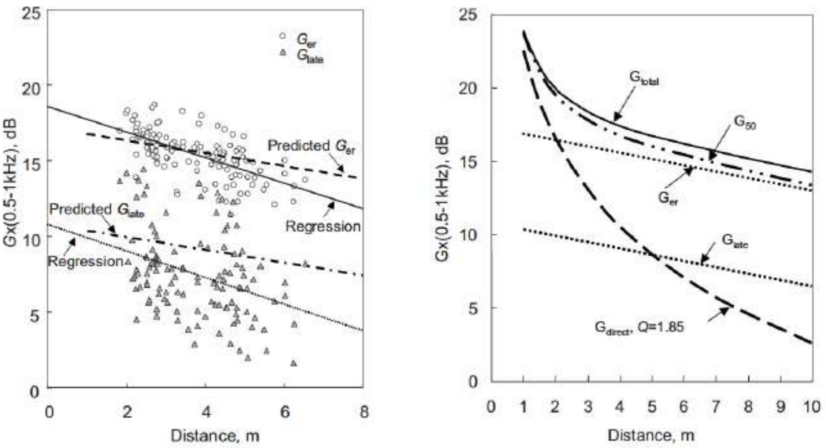
$$d = 100Q/r^2 \tag{1.84}$$

$$e'_r = (31200T/V)e^{2(-0.04r)/T}(1 - e^{(-0.05)13.82/T}) \tag{1.85}$$

$$l' = (31200T/V)e^{2(-0.04r)/T}e^{(-0.05)13.82/T} \tag{1.86}$$

Thus, in classrooms, the reflection energy decays twice rapidly as indicate by the revised theory. Results of the study are shown in the figure 1.5(a). The differences between measured and predicted values seem due to the object as desks and furniture in classrooms, according to the authors. The correction to revised theory permits to obtain a good accuracy in prediction of the behavior of classrooms. Bradley and Sato's study highlights how the relationship between reverberation time and early sound components is important. The G_{50}, which is the sound strength in the first 50 ms, measures the energy of the direct and the early reflections as a function of the source-to-receiver distance. When reverberation time is very low the room is poor of early reflections and this is disadvantageous especially for the farthest rows. An interesting parameter is the early reflections benefit ERB defined as:

$$ERB = G_{50} - G_{direct} \quad (\mathrm{dB}) \tag{1.87}$$



(a) (b)

Figure 1.5: Bradley and Sato's results. In figure a) G_{er} (open circles) and G_{late} (closed triangles) values compared with the revised theory prediction in function of the source-receiver distance. The solid line represents the regression line of measured values, the dashed line represents the predicted values. In figure b) Prediction of the modified energy sound components in function of source-to-receiver distance [21].

This parameter describes how much the early reflections increase the sound strength and thus the sound level of teacher's voice, especially for the farthest positions. The graph of the figure 1.6 shows the relationship between early reflections benefit and reverberation time in three different source-to-receiver distances. The importance of early reflections, considering the vocal effort of teachers, clarifies that a zero reverberation time should be avoided and that the optimal reverberation time isn't enough to evaluate a good speech intelligibility. Barron and Lee's theory differentiating the early and late reflections point out their different behavior. In fact the early reflections may be considered discrete while the late reflections statistical. Because of this, the late energy sound, described with the G_{late} parameter, doesn't correlate well with the source-to-receiver distance. Indeed the G_{late} varies, since of its statistical behavior, despite the position of measure but more with the reverberation time.

As seen in the graph in figure 1.7 a short reverberation time is useful to have a few of energy late and comparing it to the figure 1.6 is useful also to reach a good value of early-to-late energy ratio and thus a good clarity index. Evaluating only clarity index should seems that a zero reverberation

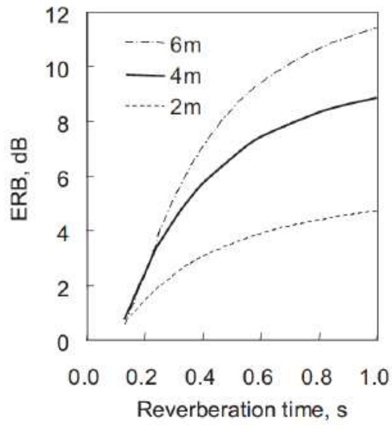


Figure 1.6: Early reflections benefit (ERB) as a function of reverberation time for source-to-receiver distance of 2, 4 and 6 m [21].

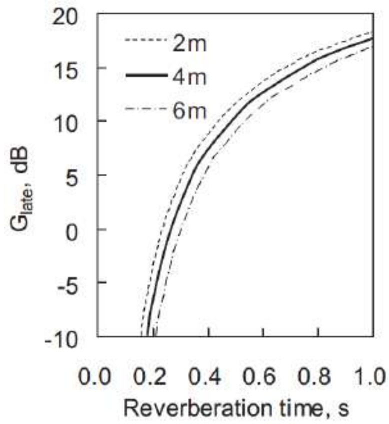


Figure 1.7: Relationship between G_{late} and reverberation time for source-to-receiver distance of 2, 4 and 6 m [21].

time permits to achieve its maximum value but this kind of consideration ignore the acoustic effects due to the noise levels.

Chapter 2

Method

The design procedure, from the measure methods, the simulation processes and the evaluation of the interventions according to standard requirements and technical literature guidelines is here described.

The case studies are presented in all that is necessary for the development of the interventions, like their geometrical features and their ante-operam conditions.

The measurement campaigns are illustrated describing the equipment and the technique used according to the normative. All the extracted parameters are shown for each room to evaluate the ante-operam acoustical behavior of the case studies.

The simulation process is explained in all its aspects as the modeling technique, the calibration method and the simulation algorithms. After that, the settings to reach the simulation parameters are described.

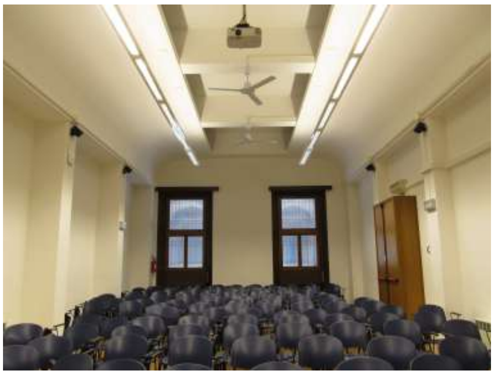
The design part is presented for each room dividing the active treatments, as the installations of a new public address system, and the passive ones, as all the interventions made for reducing the reverberation time with surfaces treatments. To illustrate the renovations brought by the projects renders and geometrical schemes are shown.

2.1 Case studies

The case studies are three lecture halls in the faculty of Letters and Philosophy of the University of Bologna. They have different forms but the same purpose that is university lectures (see figure 2.1). Despite they were designed for this use an acoustic discomfort is complained by teachers and students because of the excessive reverberation. The design goal is an increase of the speech intelligibility with accurate acoustic interventions. Aula

(a) Aula III

(b) Aula V



(c) Aula VI

Figure 2.1: Ante-operam layout of the lecture rooms under study (July 2017).

III and Aula V are two large and historical rooms with an amphitheater geometry. The walls are plastered and reflective while the seats and the benches are made of wood (figures 2.1(a) and 2.1(b)). Aula III and Aula V have respectively a volume of 1000 m^3 and 900 m^3 and they can be occupied by 250 and 200 students. Aula VI is approximately a shoe-box room with a volume of 850 m^3 hosting up to 170 students. Because of its form it can be used for non-traditional teaching activity, as theater rehearsal, despite of Aula III and Aula V. Every surface of the room is hard and reflective; on the top, between the ceiling and the false ceiling there are coupled volumes. The seats are movable and made of plastic.

The lecture halls studied, since they are used only for lectures with one speaker, can be considered in A2 category of DIN 18041 [7] ("Speech/Lecture" with one speaker). The dimensional and volumetric characteristics of the rooms are described item by item in table 2.1.

Table 2.1: Lecture halls general data. The heights of Aula III and Aula V are mean values because of floor's inclination (figures 2.9(a) and 2.9(b)).

Elements	Symbol	Aula III	Aula V	Aula VI
Length [m]	l	16.7	12.7	17.0
Width [m]	w	13.3	13.3	7.9
Height [m]	h	5.0	5.0	6.2
Volume [m^3]	V	1000	900	850
Occupancy	N	250	200	170
Audience area [m^2]	S_A	100	100	81
Ratio Volume/Audience area [m]	V/S_A	10	9	11

2.2 Acoustic measurements

In July 2017, in order to qualify the lecture halls according to ISO 3382 [11] and the IEC 60268-16 [10] standards, a campaign of measurements was conducted to obtain the room criteria and the intelligibility criteria for speech. Acoustic measurements were performed in unoccupied state and with the presence of furniture. Monoaural technique and ESS (Exponential Sine Sweep) signal were used for the purpose. The equipment was made up of:

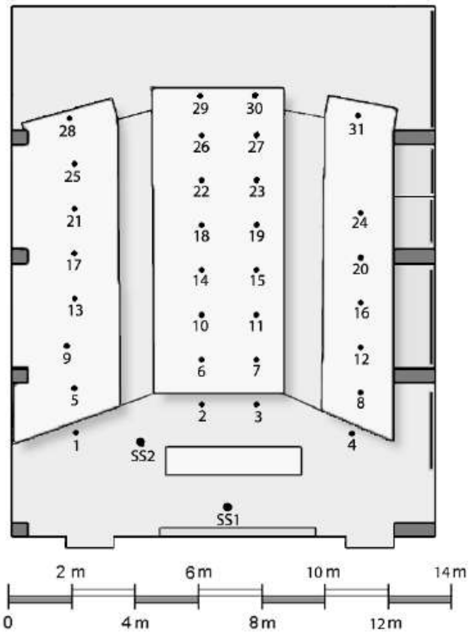
- a laptop that launched the ESS signal with length of 512k and sampled at 48 kHz;
- a signal converter (Motu UltraLite AVB);
- an amplifier to increase the signal power (Crown 2500 W);
- a dodecahedron with custom loudspeakers used as an omnidirectional source;
- one monoaural half inch free-field microphones (NTI audio MA220) as receiver.

The source was calibrated in reverberation room according to ISO 3741 [28] specifications. In figure 2.3 the placement of sources and receivers is shown for each room. Measured values are reported in table 2.2 averaged over all source-receiver positions. These values confirm the inadequate acoustic condition of the rooms. Graphs describing the tendency of the acoustic parameters in function of the frequency and source-receiver distance are shown below (see figures from 2.4 to 2.7).

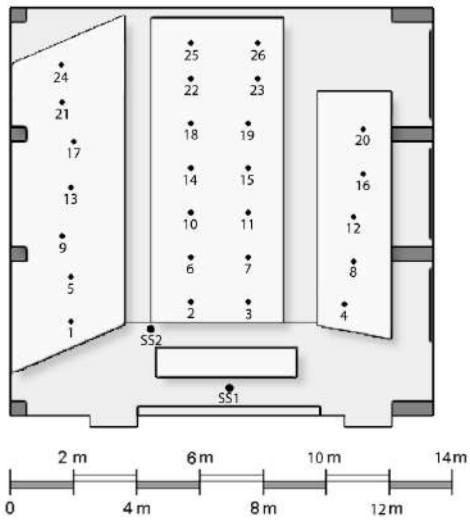
Figure 2.2: Equipment setup during ante-operam measures.

Table 2.2: Ante-operam state: mean measured values of room criteria for speech intelligibility. The subscripts “3” and “M” indicate in which octave-band the average was calculated ($500 \div 2000$ Hz and $500 \div 1000$ Hz).

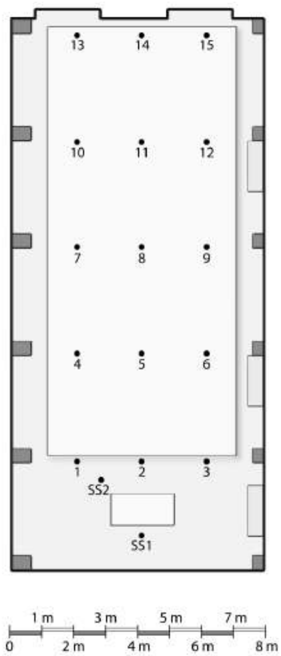
	Mean measured values		
	Aula III	Aula V	Aula VI
$C_{50,3}$ (dB)	-2.8	-2.4	-4.3
STI	0.49	0.47	0.44
$T_{M,unocc}$ (s)	1.70	1.72	2.54



(a) Aula III: sources and receivers placement

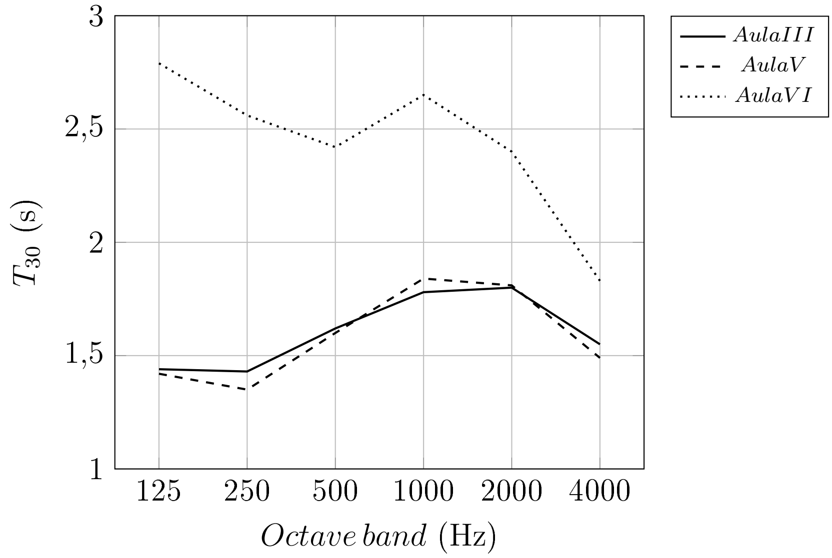


(b) Aula V: sources and receivers placement

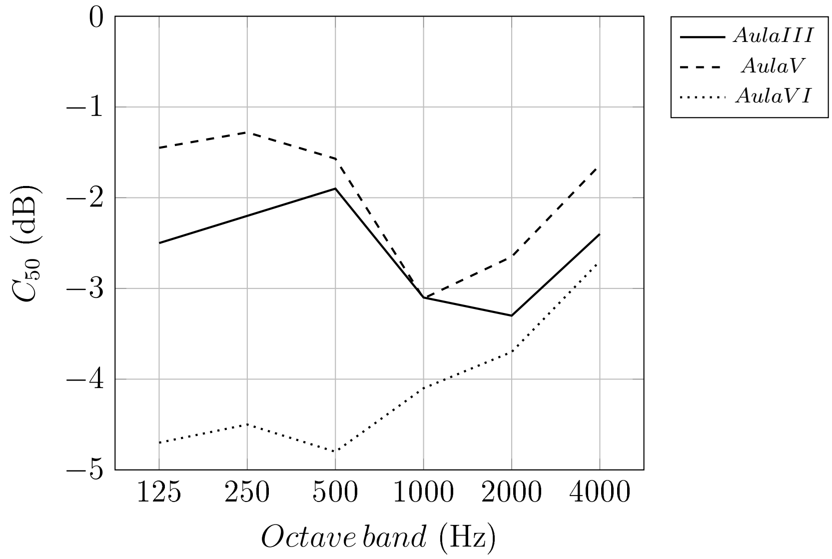


(c) Aula VI: sources and receivers placement

Figure 2.3: Lecture halls' plans: sound sources (SS1 e SS2) and receivers positions used for the measurement campaign of July.

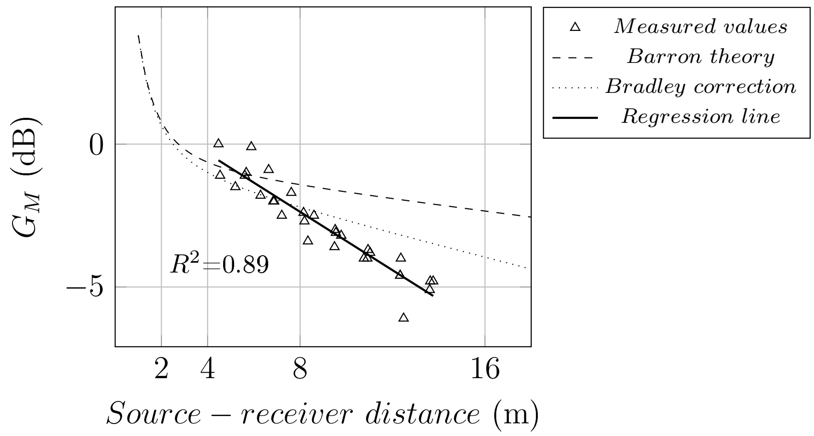


(a) Ante-operam measured T_{30} values.

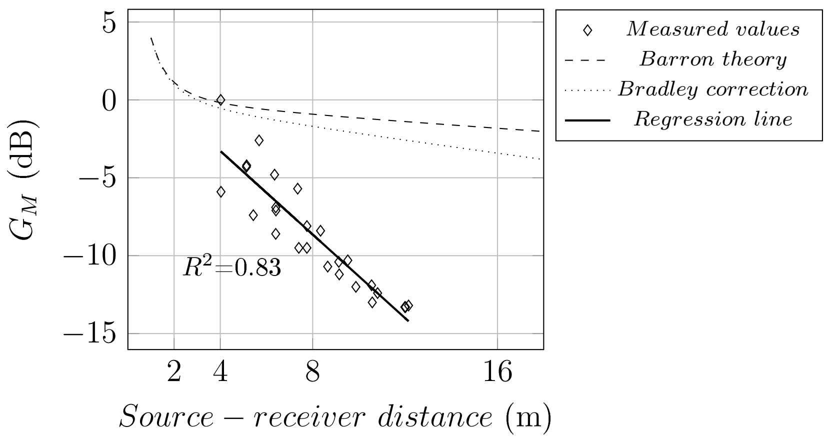


(b) Ante-operam measured C_{50} values.

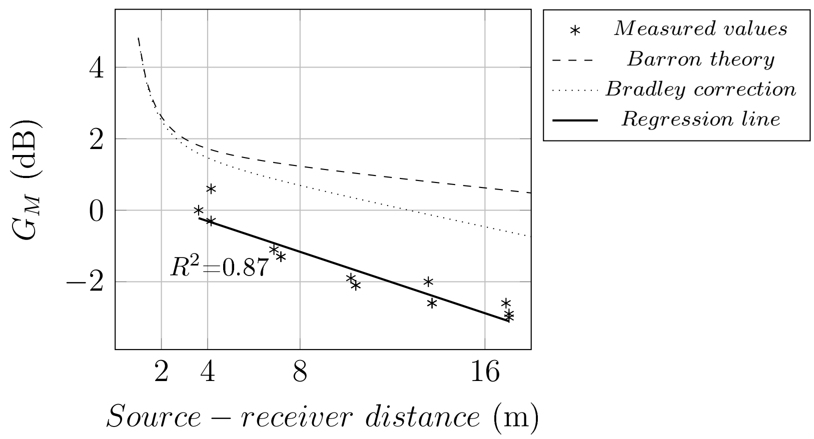
Figure 2.4: Ante-operam measured T_{30} and C_{50} values in function of the octave bands for each room: Aula III is plotted with a solid line, Aula V with a dashed line and Aula VI with a dotted line. Values were measured in unoccupied state with an omnidirectional source and are provided averaged over all source-receiver positions (see figure 2.3).



(a) Aula III: measured G values in function of source-receiver distance.

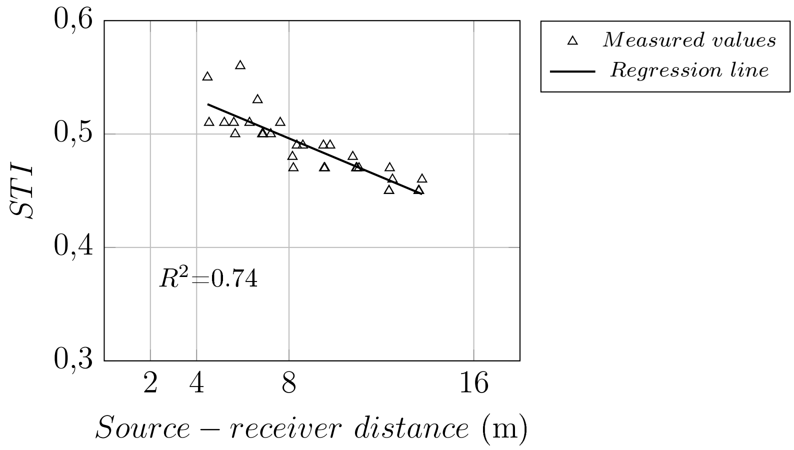


(b) Aula V: measured G values in function of source-receiver distance.

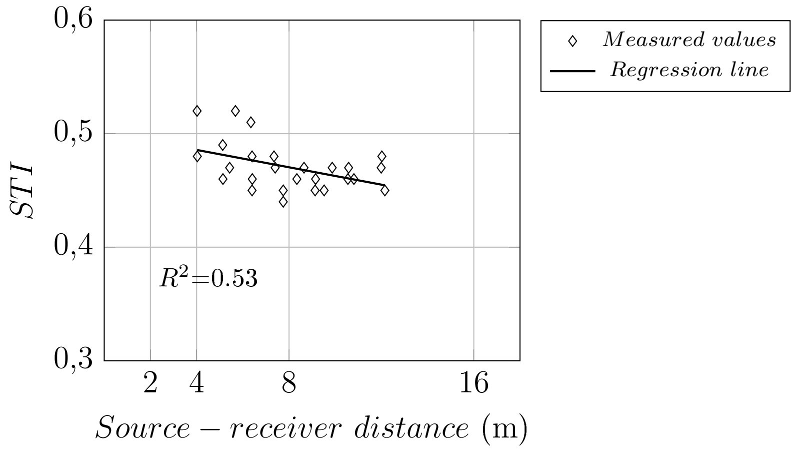


(c) Aula VI: measured G values in function of source-receiver distance.

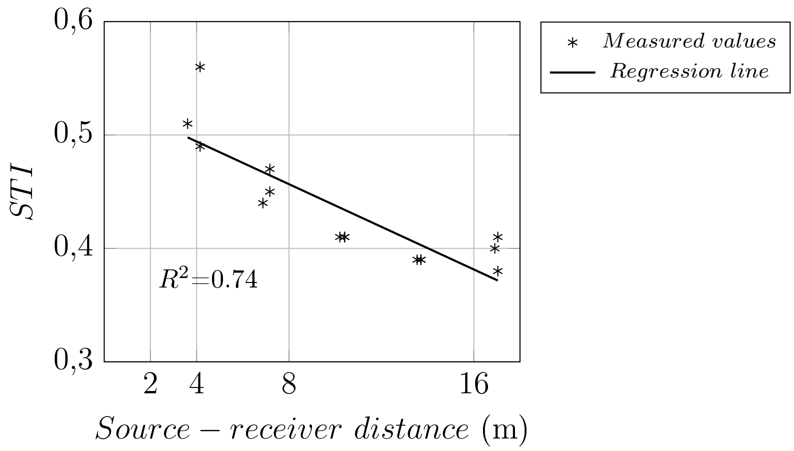
Figure 2.5: Ante-operam measured values of G_M in function of source-receiver distance. Values were measured in the campaign of July in unoccupied state with an omnidirectional source. A comparison with Barron and Lee's revised theory (dashed line) and Bradley and Sato's correction (dotted line) is visible (see section 1.2.6).



(a) Aula III: measured STI values in function of source-receiver distance.

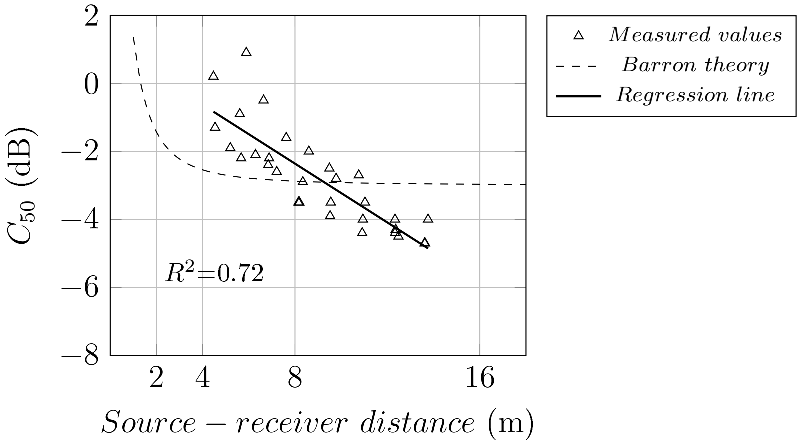


(b) Aula V: measured STI values in function of source-receiver distance.

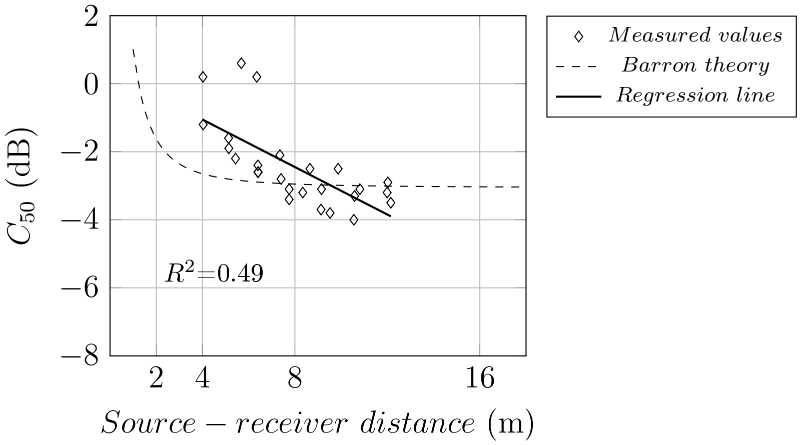


(c) Aula VI: measured STI values in function of source-receiver distance.

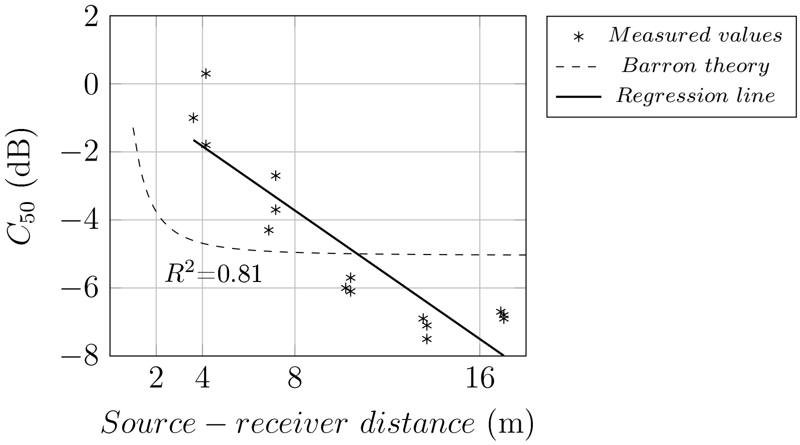
Figure 2.6: Ante-operam measured values of STI in function of source-receiver distance. Values were measured in the campaign of July in unoccupied state with an omnidirectional source.



(a) Aula III: measured C_{50} values in function of source-receiver distance.



(b) Aula V: measured C_{50} values in function of source-receiver distance.



(c) Aula VI: measured C_{50} values in function of source-receiver distance.

Figure 2.7: Ante-operam measured values of C_{50} in function of source-receiver distance. Values were measured in the campaign of July in unoccupied state with an omnidirectional source.

2.3 Background noise measurements

The background noise due to the HVAC system was measured using a class 1 sound meter level. On 18th and 19th of July 2017 the background noise was detected turning, according to technical standards, on the projector and the air condition system at medium power, for approximately two hours of measures. The noise level due to systems is necessary to define STI in numerical simulation and in prediction models. Equivalent sound pressure levels of background noise are reported in figure 2.8 in dB averaged on all receivers position.

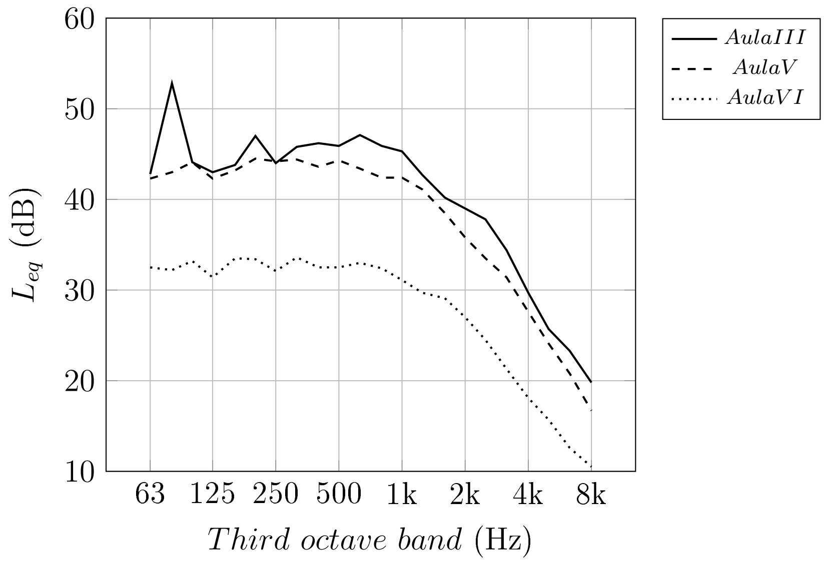
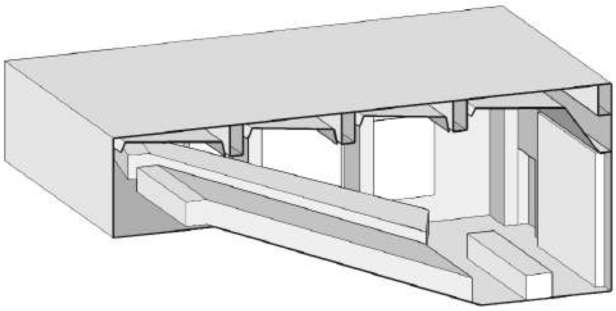


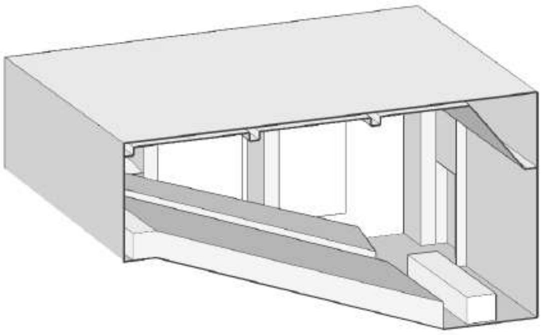
Figure 2.8: Measured background noise values during the campaign of July for each room. Aula III is plotted with a solid line, Aula V with a dashed line and Aula VI with a dotted line. During the measures the ventilation system was turned on.

2.4 Numerical models

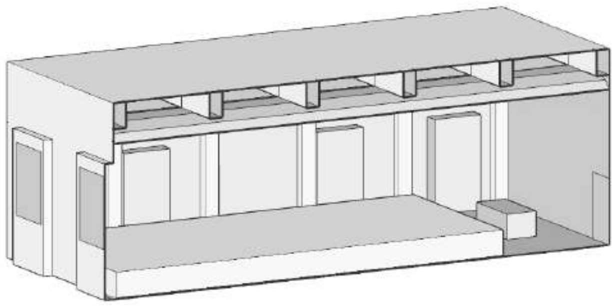
Numerical models were made and calibrate with measurements results to evaluate the criticalities in every room and identify the appropriate interventions. These were created with SketchUp software for model surfaces and geometries and then imported in Odeon Room Acoustics, a GA (geometrical acoustic) software simulations. Odeon uses an hybrid algorithm which takes advantage of two methods: the image source method and the ray tracing.



(a) Aula III: cross section of SketchUp model used for Odeon simulation



(b) Aula V: cross section of SketchUp model used for Odeon simulation



(c) Aula VI: cross section of SketchUp model used for Odeon simulation

Figure 2.9: Sketchup models of rooms current state realized with SketchUp software according to Odeon's guidelines for acoustic simulation modeling seen in section 2.4.1.

The first is based on the principle that considers the point in which the reflection happens as a new source while the ray tracing uses the tracing of the particles emitted by a source point loosing energy after every reflection according to the material properties of the surface. The hybrid algorithm uses the positive concepts of both simplifying some steps in their calculations to obtain a less computation.

2.4.1 Modeling process

The models were realized with SketchUp software to export them with SU2Odeon plugin to avoid holes and overlapped surfaces and according to the Odeon's manual guidelines that are [30]:

- ignore all the irregularity of surfaces smaller than 30 cm, which corresponds to the 1000 Hz wavelength, to obtain a minor computation without losing accuracy of results;
- curved surfaces are approximated into a certain number of planar surfaces;
- the subdivision in layers is based on the function and the material of each object in order to assign the absorbing and scattering coefficients to a defined group;
- avoid to model each step between the rows but approximate the audience area to a box with an height of 0.8 meters above the floor and a scattering coefficient of 0.7.

2.5 Calibration

Calibration is an iterative procedure which consists in modeling the current state of rooms and assigning to their surfaces absorption and scattering coefficients in order to obtain an acoustical response of the model within the JND (just noticeable difference) of the measured parameters. The criteria used for the calibration are T_{30} and C_{50}. Modeling was made with the less possible layers to simplify the calibration procedure. Layers were thought to distinguish objects by material, function and position in the room. Sound absorption and scattering coefficients were assigned following values provided by technical standards and scientific literature. The organization of layers allows to divide the hard and reflective surfaces (walls, ceiling, glasses) from the sound absorbing and scattering elements (desks and seats). Specifically, each type of benches and chairs was associated with a range of absorption coefficients, depending on the shape and material, as shown in table 2.3.

Table 2.3: Scattering (s) and absorption (α) coefficients used for numerical simulation, provided by scientific literature and technical standards. Scattering coefficients referee to the average frequency of 707 Hz while the absorption coefficients are shown for each octave-band frequency, according to Odeon algorithms.

Materials	s	α					
		125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Plaster/Floor	0.05	0.02	0.02	0.03	0.03	0.04	0.06
Bottom wall (Aula V)	0.10	0.42	0.21	0.10	0.08	0.06	0.06
Seats (Aula III, Aula V)*	0.70	0.40	0.42	0.30	0.27	0.12	0.15
Seats (Aula VI)**	0.70	0.10	0.12	0.10	0.08	0.06	0.04

*Benches and wooden fixed chairs (figure 2.12(b))
**Plastic furniture chairs (figure 2.12(a))

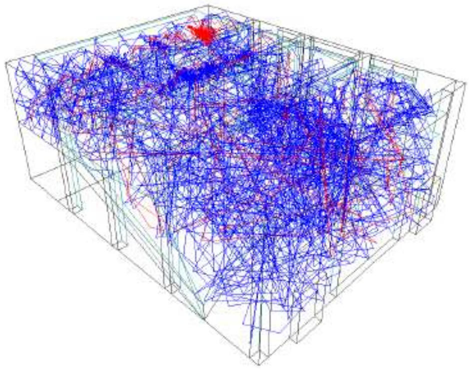
(a) Types of sitting in Aula III and Aula V

(b) Type of chair in Aula VI

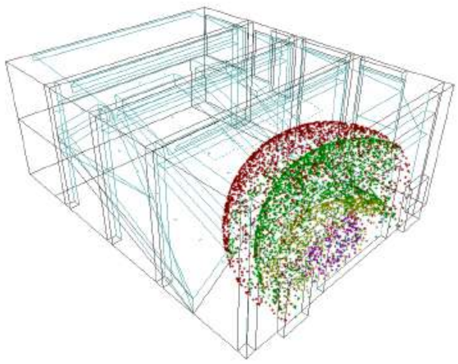
Figure 2.10: Details of chairs in rooms. In Aula III and Aula V the seats are made of wood and are fixed on the floor while in Aula VI there are plastic chairs movable.

Sources and receivers were placed into the models with the same layout of the measurement setup. Using *Source-receiver list* tool the omnidirectional sources and receivers were setted respectively at 1.5 m and 1.2 m above the floor. A first check of the models with Odeon's simulation tools was made to ensure the correctness of geometry before proceeding with the calibration. Odeon permits to use the following instruments:

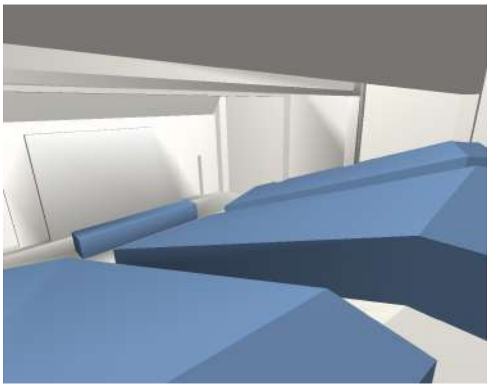
- *3D Invesigate Rays* which simulates the ray tracing, if models have holes the rays go out of them (see figure 2.11(a));



(a) View of *3D Invesigate Rays* tool.



(b) View of *3D Billiard* tool.



(c) View of *3DOpenGL* tool.

Figure 2.11: View of the Odeon tools to check the models and simulate its acoustical behaviour.

- *3D Billiard* which simulates the wave front with some balls and, as the previous tool, if holes are present in the models the balls go out (see figure 2.11(b));

- *3DOpenGL* which is a visual tool that permits to navigate into the model and to discover the possible missing surfaces (see figure 2.11(c)).

Concerning the calculation settings in *Room setup* of the acoustic simulation software, 2500 ms (Aula III, Aula V) and 3000 ms (Aula VI) impulse responses were used, a transition order of 2 for all classrooms, a number of rays of 16000 (*Precision*).

2.6 Design of acoustic treatments

Improvements were developed considering the different typology, characteristics and specific use of each lecture hall. The intervention proposal consists of two macro-categories, the first one is the passive acoustics treatment and the second one is the introduction of a proper public address (PA) system.

Passive acoustic treatments were designed in order to achieve the requirements provided by the national standard UNI 11532 [8], which refers to DIN 18041 [7] method. Therefore, optimal reverberation time was found with the DIN 18041 [7] formulas in occupied state, as seen in section 1.2.1. Revers formula of reverberation time permits to calculate the needed equivalent absorption area A and thus, the quantity and the properties of sound absorbing panels to be introduced in each lecture hall. Nevertheless, adding the adequate A isn't enough to obtain a good speech intelligibility because the placement of these surfaces play a key role in enhancing the sound clarity and the sound energy distribution throughout the space. Understanding where positioning the sound absorbing panels becomes a fundamental aspect to achieve the goal since it could heavy change C_{50}, with variations until 4 dB, and speech levels received, with variations until 3 dB values [20]. DIN 18041 [7], BB 93 [9] and scientific literature describe how to obtain a good placement. It's important to leave the ceiling free from absorption because it's helpful to enhance the early reflections. By the way its edges can be covered with some absorbing material along the perimeter. The rear wall, instead, represents the ideal zone to cover with passive acoustic treatments because it is the place the most of late reflections come from. In addition, the rear wall should be treated to prevent echo effects especially if the distance between the rear wall and the speaker is greater than 9 meters.

For interventions design the useful-to-detrimental ratio U_{50} was not evaluated because the relationship between the intelligibility and the signal-to-noise ratio is considered in the STI yet.

Passive treatments were designed and optimized combining accurately sound absorbing and diffusing devices.

The absorbent panels chosen for the intervention project is made of wood and carved to take advantage of absorbent properties of Helmholtz resonators and vibrant membranes. These panels are made up of more layers, the first is a set of strips spaced by an empty space which permits to brake the wave front. Behind this, a layer made of a succession of holes allows to absorb sound energy based on the principle of Helmholtz resonators. The last layer is spaced from the previous one to create an air cavity and it is made with a porous absorbing material. This kind of panels have an high absorbing

Table 2.4: Ante-operam state: comparison between mean measured values of room criteria for speech intelligibility and standard requirements (DIN 18041 [7], UNI 11532 [8], BB93 [9]). The subscripts "3" and "M" indicate in which octave-band the average was calculated (500 $\div$ 2000 Hz and 500 $\div$ 1000 Hz). The reverberation time in occupied state ($T_{M,occ}$) was calculated with DIN 18041 formulas 1.6.

	Mean measured values			Standard requirements
	Aula III	Aula V	Aula VI	
$C_{50,3}$ (dB)	-2.8	-2.4	-4.3	≥ 0
STI	0.49	0.47	0.44	≥ 0.60
$T_{M,occ}$ (s)	0.90	1.00	1.23	$0.66 \div 1.07$

power in the middle and high frequencies as shown in table 2.5.

The slat panels, unlike the previous ones, have a greater absorption power at low frequencies (see table 2.5) and use the properties of Helmholtz resonators. They are made with a porous absorbing material panel covered with wooden stripes less or more thick to reduce the absorption of a certain percentage chosen by the designer. The panels are assembled with a certain distance from the wall to obtain an air cavity. The wooden stripes give reflecting and diffusing properties to the panels (see figure 2.12).

Table 2.5: Scattering (s) and absorption (α) coefficients of devices introduced by design proposals. Scattering coefficients referee to the average frequency of 707 Hz while the absorption coefficients are shown for each octave-band frequency, according to Odeon algorithms.

Materials	s	α					
		125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Sound absorbing panels (Aula III, Aula V)	0.10	0.03	0.33	0.73	0.89	0.85	0.77
Slats absorbers (Aula VI)	0.50	0.35	0.45	0.73	0.89	0.85	0.77

In the Aula III, sound absorbing panels, as described before (see table 2.5), are designed to be installed on the rear wall, on the beams' surfaces and the pillars' sides facing the speaker (see figure 2.13). The absorbing ma-

(a) Sound absorbing panels designed for Aula III and Aula V.

(b) Slat absorbing panels designed for Aula VI.

Figure 2.12: Passive acoustic treatments of the design proposal.

terial is installed along the pillar starting from 2 meters above the floor, to reduce material degradation due to the activities of the students.

In Aula V the acoustic correction project is quite similar to the one developed for Aula III because this room is very similar to the previous one in form, size and material. Therefore, also in this case, a sound absorbing material (table 2.5) is installed on the rear wall, the beams' surfaces and the pillars' side facing the speaker with the same logic followed for Aula III (see figures 2.14).

Passive treatment in Aula VI involves the insertion of slats absorbers at the top of the side walls (see table 2.5), between one pillar and another. These are both sound absorbing and diffusing devices designed to decrease the reverberation time, to spread the reflected sound energy throughout the space and to avoid eco-flutter phenomena between one pillar and another (see figure 2.15).

2.7 Design of PA system

The PA (Public Address) system is a critical element for speech intelligibility within a lecture hall. According to the numerical simulations, with passive acoustic correction interventions it's possible to achieve STI values below the recommended 0.60. These values can be considered as precautionary in order to obtain an adequate intelligibility for two reasons:

- the voice's directivity provides higher parameter results than the omnidirectional source used during the project development. The use of this kind of sound source in the design and testing is necessary to be con-

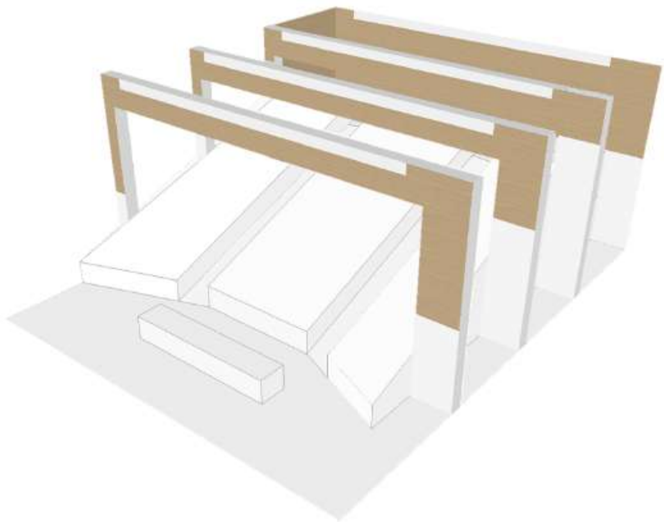


Figure 2.13: Passive acoustic treatments in Aula III: placement of the sound absorbing panels.

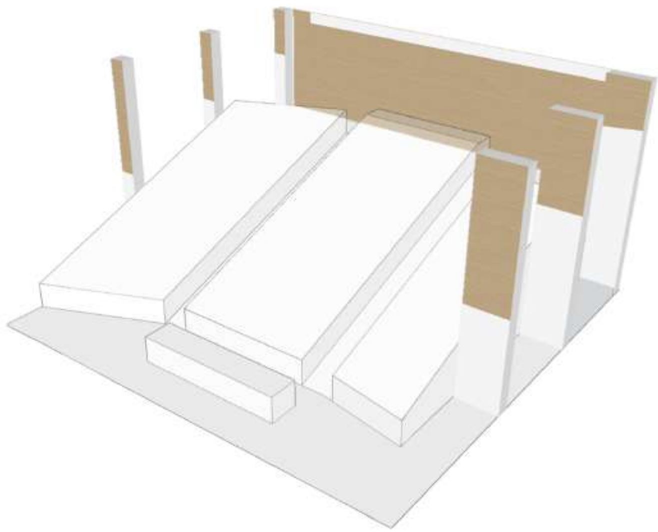


Figure 2.14: Passive acoustic treatments in Aula V: placement of the sound absorbing panels.

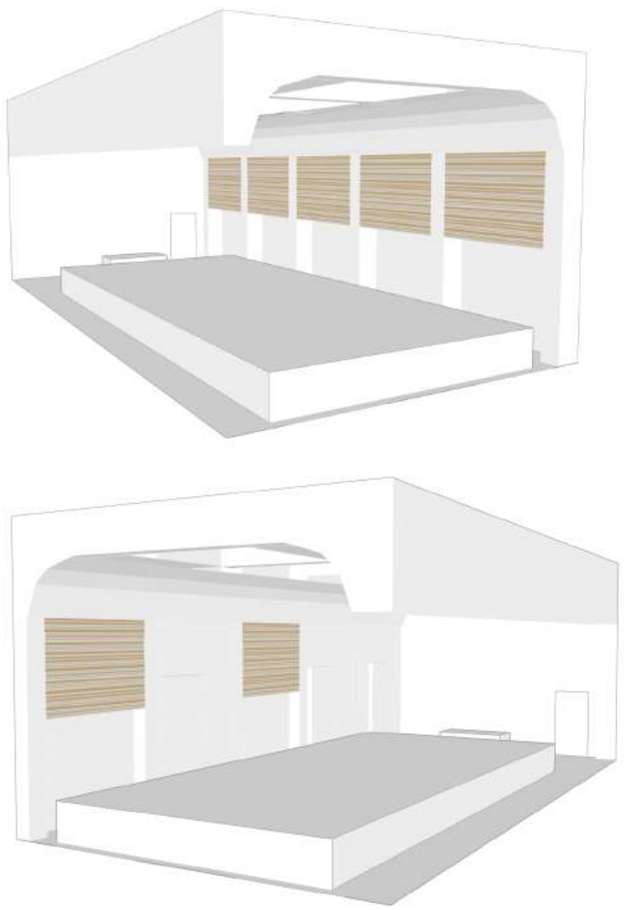


Figure 2.15: Passive acoustic treatments in Aula VI: placement of the slat absorbing panels.

sistent among ante-operam qualification, numerical model calibration and post-operam measures;

- the presence of a properly designed and installed amplification system drastically increases intelligibility.

Furthermore, it should be considered that:

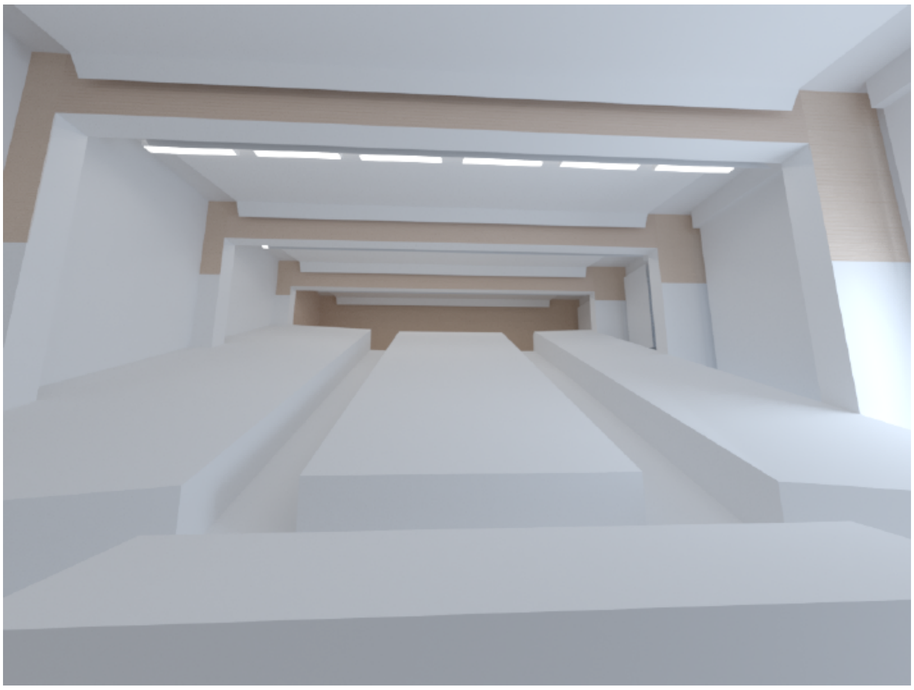
- the use of "normal" loudspeakers (90 degree vertical opening, horizontal 40 to 60 degrees at medium high frequencies) does not seem to be the optimal solution to increase speech intelligibility;

- the use of a mini line array (vertical aperture 25 degrees, horizontal 130-140 degrees on range $250 \div 2000$ Hz) is the most powerful solution, guiding the direct sound towards the students with an accurate placement.

The mini line array placement was optimized with Soundvision [6] simulation software, which simulates the coverage from the only direct field generated by the array (figures 2.19(a), 2.19(c) and 2.19(d)). A line array permits to spread a cylindrical wave front and obtain a low decay along the length of the room. A powerful public address system allows to focus the speech signal to the students area and to achieve an optimal signal-to-noise ratio. The sound level values shown in the figures 2.19 are purely indicative and should be considered in a relative way. The PA system should be calibrated to have an adequate coverage of background noise (considering the contribution of the system and the contribution of the students themselves).

Compatibly with the furnishings and the geometry of the rooms, an appropriate placement was studied to have an adequate coverage of the listening area, with a good homogeneity among all the student positions. The detected differences (contained within 3 dB between maximum and minimum points) are further reduced by the contribution of the reverberant sound field. The seats closer to the speaker will also benefit from the helpful contribution of the speaker's voice.

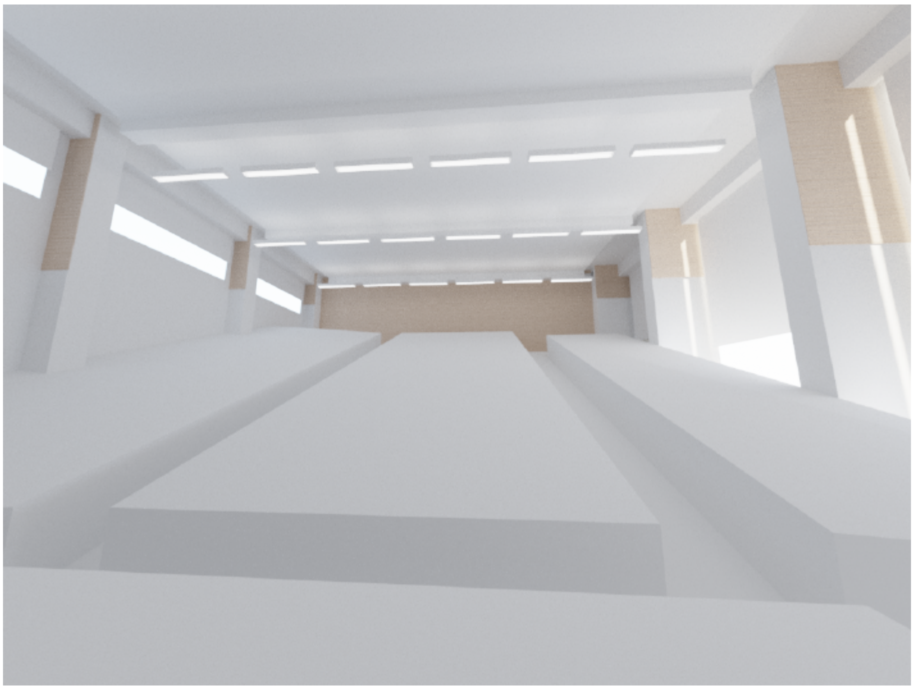
Following, from picture 2.16 to 2.18, photorealistc renders show the aesthetics of all interventions, active and passive ones. For Aula III and Aula V two renders are shown, one from the position of the speaker and one from the bottom of the audience area. From the first is possible to see the complete passive treatments with the positioning of the acoustical absorbers and from the views on the bottom the active treatments with the positioning of the public address system. For Aula VI two render images permit to see treatments, the active one with the slats absorbers on the sides of the walls and the passive one with the mini line array above the blackboard.



(a) Aula III: photorealistc render showing the passive treatments from the speaker's view.

(b) Aula III: photorealistic render showing a part of the passive treatments on the rear wall of the room and the active one with the placement of the public address system above the blackboard.

Figure 2.16: Aula III: photorealistic renders from the speaker's point of view (a) and from the bottom of the room (b).



(a) Aula V: photorealistc render showing the passive treatments from the speaker's view.

(b) Aula V: photorealistic render showing a part of the passive treatments on the rear wall of the room and the active one with the placement of the public address system above the blackboard.

Figure 2.17: Aula V: photorealistic renders from the speaker's point of view (a) and from the bottom of the room (b).

(a) Aula VI: photorealistc render showing the passive treatments near the speaker's view.

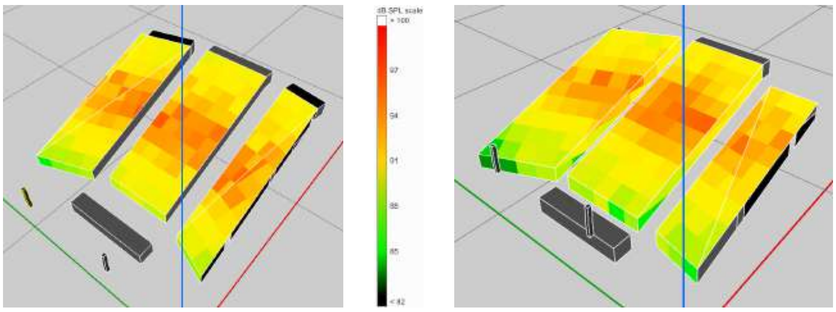
(b) Aula VI: photorealistic render showing passive treatments on the sides of the walls and the active one with the placement of the public address system above the blackboard.

Figure 2.18: Aula VI: photorealistic renders near the speaker's point of view (a) and from the bottom of the room (b).

2.8 Numerical simulations

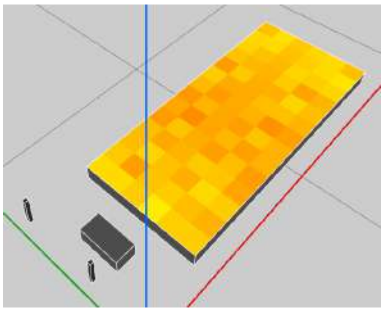
In addition to the prediction methods, an estimate of post-operam results was also obtained with numerical simulations (see tab. 3.1). The contribution of the passive acoustic treatments was simulated with Odeon software introducing the new elements, the absorbing panels and the slat absorbers into the numerical models (absorption and scattering coefficient are provided in table 2.3). STI values are simulated taking into account the measured background noise levels due to the ventilation system (see figure 2.8). Finally, with Soundvision software, the accurate position of line array speakers was optimized assessing the coverage provided on the students area (see figure 2.19).

In order to estimate the effects of the interventions throughout the space, simulated STI and C_{50} ante-operam and post-operam values are compared using *Grid* tool with squares of 1 m^2 each one at 1.2 m above the floor.



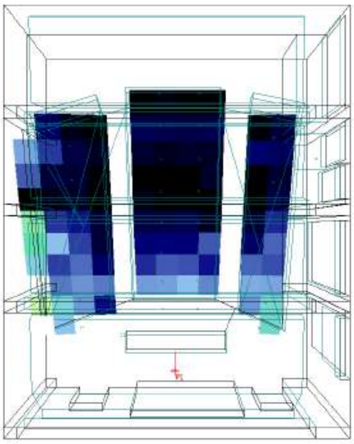
(a) Aula III: coverage area of the public address system

(c) Aula V: coverage area of the public address system

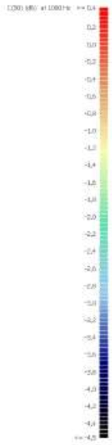


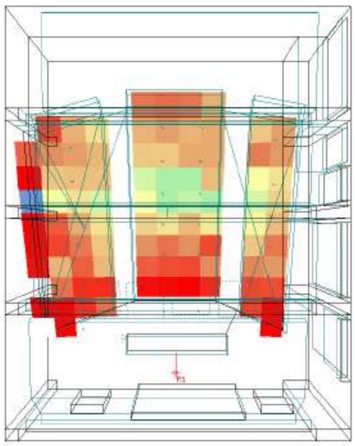
(d) Aula V: coverage area of the public address system

Figure 2.19: Numerical simulations performed with Soundvision software to evaluate the right placement of the public address system.



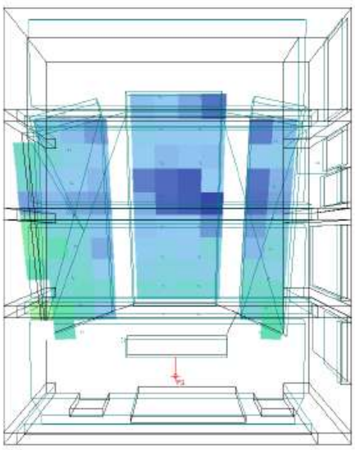
(a) Aula III: C_{50} ante-operam values



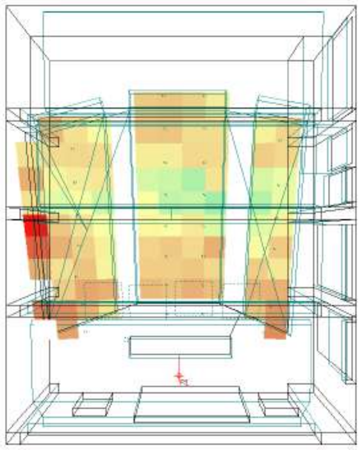


(c) Aula III: C_{50} post-operam values

Figure 2.20: Aula III: numerical simulations performed with Odeon software to evaluate the spatial distribution of sound clarity index C_{50} before (a) and after (b) the passive interventions.

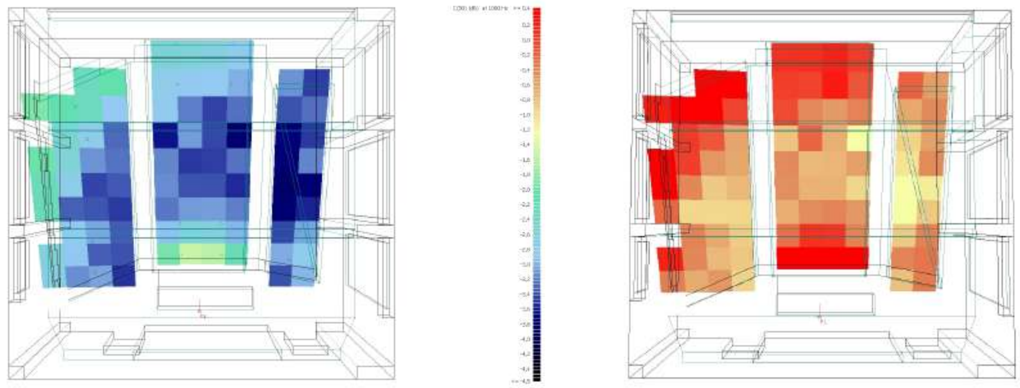


(a) Aula III: STI ante-operam values



(c) Aula III: STI post-operam values

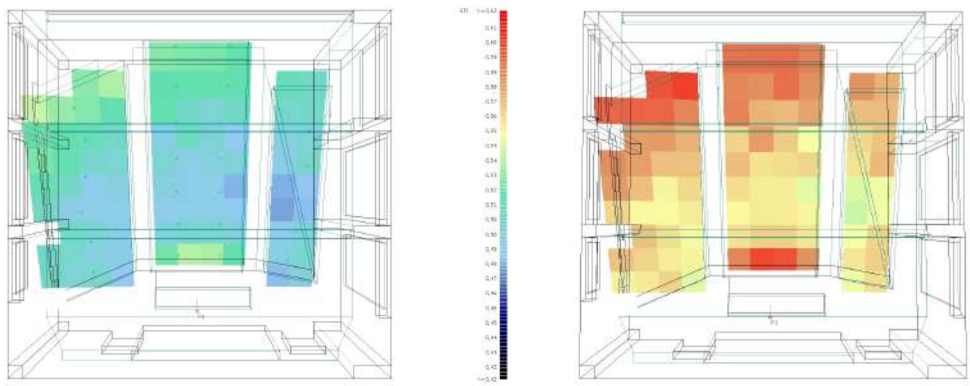
Figure 2.21: Aula III: numerical simulations performed with Odeon software to evaluate the spatial distribution of speech intelligibility index STI before (a) and after (b) the passive interventions.



(a) Aula V: C_{50} ante-operam values

(c) Aula V: C_{50} post-operam values

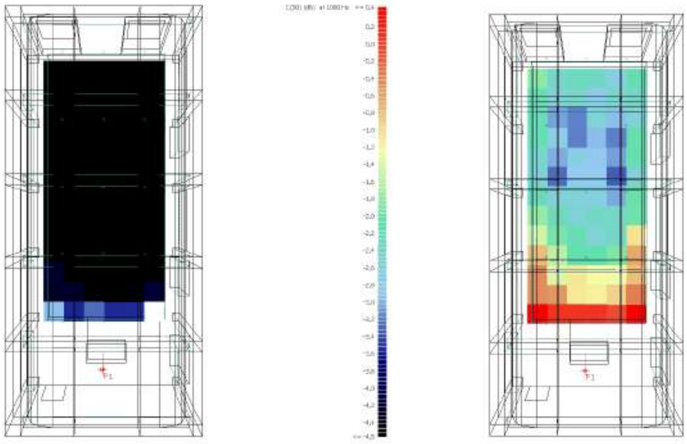
Figure 2.22: Aula V: numerical simulations performed with Odeon software to evaluate the spatial distribution of sound clarity index C_{50} before (a) and after (b) the passive interventions.



(a) Aula V: STI ante-operam values

(c) Aula V: STI post-operam values

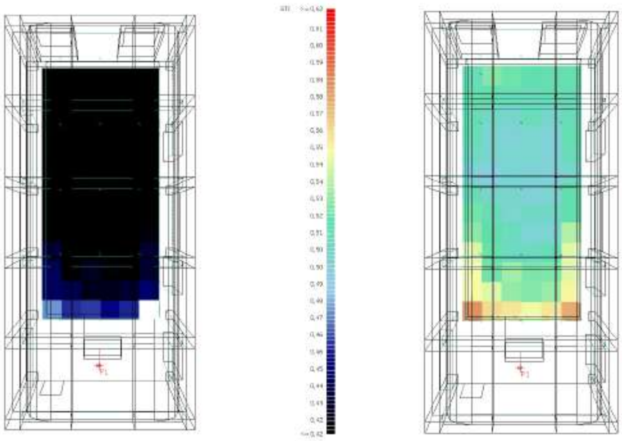
Figure 2.23: Aula V: numerical simulations performed with Odeon software to evaluate the spatial distribution of speech intelligibility index STI before (a) and after (b) the passive interventions.



(a) Aula VI: C_{50} ante-operam values

(c) Aula VI: C_{50} post-operam values

Figure 2.24: Aula VI: numerical simulations performed with Odeon software to evaluate the spatial distribution of sound clarity index C_{50} before (a) and after (b) the passive interventions.



(a) Aula VI: STI ante-operam values

(c) Aula VI: STI post-operam values

Figure 2.25: Aula VI: numerical simulations performed with Odeon software to evaluate the spatial distribution of speech intelligibility index STI before (a) and after (b) the passive interventions.

Chapter 3

Results

The results of measurements campaigns carried out are here reported. Values obtained from simulations are first presented in order to evaluate the efficiency of acoustical interventions provided. Subsequently the results obtained from the measurements campaign during lectures concerning the background noise due to the student activity and their cumulative distribution for each receiver are shown. Thus curve fittings obtained from the peaks analysis are reported with their corresponding student activity (SA) and speech level (SL) values. Finally the results gained by the measurements campaign carried out after the installation of the public address system are shown in order to assess the effects of the active interventions.

3.1 Comparison between ante-operam and post-operam values

Comparison between the ante-operam measured values and post-operam simulated values of the acoustic descriptors considered for the speech intelligibility is reported in table 3.1. Targets required by normative are also shown in order to highlights the inadequacy of ante-operam state.

3.2 Analysis of student activity

In order to investigate deeper the sources of the background noise for what concerns university lessons, according to Hodgson's research work [25] a measurements campaign was carried out to qualify the noise due to student activity during the lessons. The measurements were recorded in three days, one for each room, on the 27th, 30th and 31th of October 2017. The

Table 3.1: Acoustic design: comparison between the ante-operam measured and post-operam simulated values of the acoustic descriptors considered for the intelligibility of speech, simulated post-operam mean values and requirements standards (DIN 18041, UNI 11532, BB93). The subscripts "3" and "M" indicate in which octave-band the average was calculated ($500 \div 2000$ Hz and $500 \div 1000$ Hz). The reverberation time in occupied state $T_{M,occ}$ was estimated with DIN 18041 formulas (see section 1.2.1).

		Mean values			Target
	Parameter	Aula III	Aula V	Aula VI	
Ante–operam measured values	$C_{50,3}$ (dB)	-2.8	-2.4	-4.3	≥ 0
	STI	0.49	0.47	0.44	≥ 0.60
	$T_{M,occ}$ (s)	0.90	1.00	1.23	$0.66 \div 1.07$
Post–operam simulated values	$C_{50,3}$ (dB)	0	0	-2	≥ 0
	STI	0.57	0.57	0.53	≥ 0.6
	$T_{M,occ}$ (s)	0.71	0.79	0.84	$0.66 \div 1.07$

first room measured was Aula III, then Aula V and finally Aula VI. Sound pressure levels were recorded during the entire lessons activity collecting 8 hours of measures for Aula III, 10 hours for Aula V and 8 hours for Aula VI. Two 01*dBDuo* sound level meters were used for recording, placed in opposite points in the room. Calibration of microphones were made two times during each day, at the beginning and at the end of the measures. Furthermore temperature and relative humidity were detected with a sensor (see figures 3.2(a), 3.2(b) and 3.2(c)). These latter parameters were monitored because their variations, depending on the occupancy and on the opening of windows and doors, could change acoustic behaviour of the rooms creating a different air density and consequently changing the sound propagation speed. Sound level meters were signed univocally and called $R1$ and $R2$ to relate the extracted data with the receiver position. During the record the occupancy of the room and how people were clothed were noticed. The setup of the room was also noted which means the description of all the details that could influence the acoustic response of the room like the presence or not of a curtain, a door or a window left open. Using *dBtrait* software, after analyzing the temporal histories of each recording and separating the lessons time from the intervals time, the following parameters were extracted:

- global equivalent A-weighted sound pressure level $L_{eq,A}$, in dB;

- global maximum and minimum A-weighted sound pressure levels L_{MAX} and L_{min}, in dB;

- global statistical A-weighted sound pressure levels, defined as the level exceeded for a certain percentage of time (i.e., $L99$ is the sound pressure level exceeded for the 99% of recording time), $L99$, $L95$, $L85$, $L80$, $L75$ and $L70$, in dB;

- A-weighted sound pressure level in function of frequency in third-octave bands;

- statistical cumulative sound pressure level distribution in percentage to detect with multi-peaks analysis the student activity and speech received levels.

Multi-peaks analysis and curve fitting were made with *OriginLab* software in order to obtain a significant gaussian regression of the data. Measured asymmetrical curves were divided in two symmetrical normal-distribution curves with the maximum values coincident with the measured peaks (see figures 4.1 - 3.8). Subsequently a further curve fitting and peak analysis were made with the mean measured values of two receivers (see section reliability of student activity). After curve fitting, a prediction model according to Hodgson's formulas for student activity noise (see section 1.2.3) was calculated and averaged on the two positions used during the measures (see tables 4 - 6). Afterwards a comparison between predicted and mean measured values was made (see table 3.5 - 3.7). In tables 3.2, 3.3 and 3.4 the data extracted from the measures are shown.

Figure 3.1: Sound meter levels setup during measurements.

Table 3.2: Aula III: occupancy, global A-weighted equivalent, maximum, minimum, statistical sound pressure levels and recording time measured during lectures.

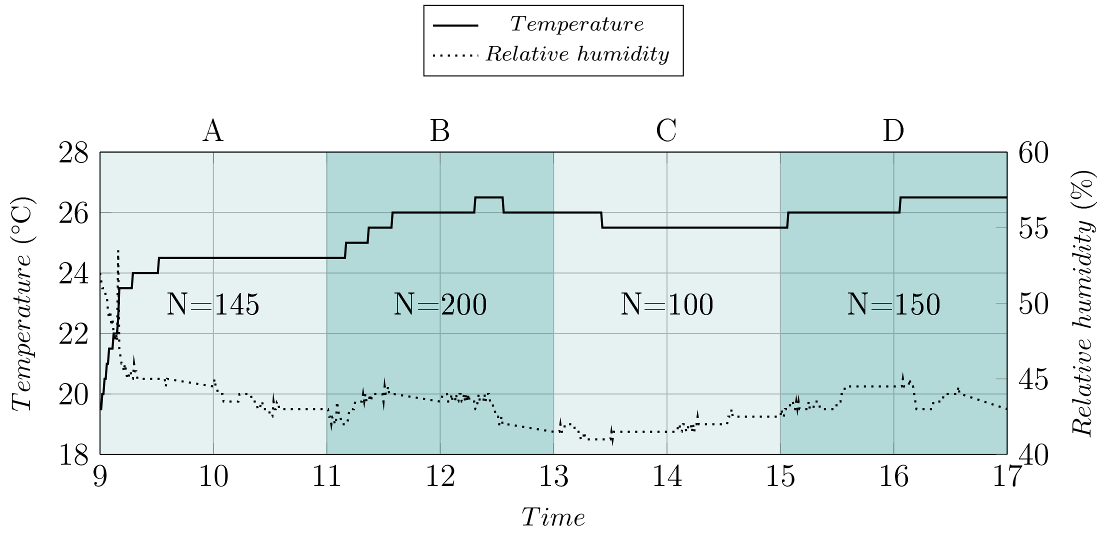
Time	Lesson	Occupancy (N people)	$L_{eq,A}$ (dB)	L_{min} (dB)	L_{MAX} (dB)	$L95$ (dB)	$L90$ (dB)	Recording time (h:m)
09.00-11.00	A	145	66.5	36.9	85.1	45.2	47.6	01:16
11.00-13.00	B	200	61.3	37.0	76.8	43.1	45.4	01:22
13.00-15.00	C	100	65.7	36.6	84.6	48.4	51.8	01:38
15.00-17.00	D	150	66.4	36.4	90.8	45.2	48.5	01:24

Table 3.3: Aula V: occupancy, global A-weighted equivalent, maximum, minimum, statistical sound pressure levels and recording time measured during lectures.

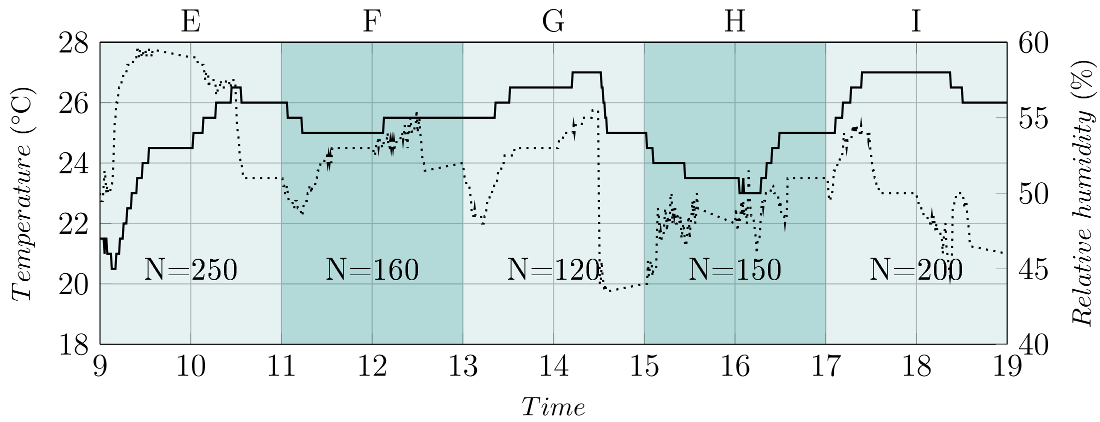
Time	Lesson	Occupancy (N people)	$L_{eq,A}$ (dB)	L_{min} (dB)	L_{MAX} (dB)	$L95$ (dB)	$L90$ (dB)	Recording time (h:m)
09.00-11.00	E	250	67.3	37.4	85.2	43.4	45.4	01:32
11.00-13.00	F	160	69.1	37.9	87.5	46.4	49.2	01:24
13.00-15.00	G	120	74.8	38.4	91.6	51.4	57.8	01:19
15.00-17.00	H	150	76.6	39.7	94.4	48.6	51.4	01:30
17.00-19.00	I	200	71.9	40.1	99.1	50.2	54.6	01:29

Table 3.4: Aula VI: occupancy, global A-weighted equivalent, maximum, minimum, statistical sound pressure levels and recording time measured during lectures.

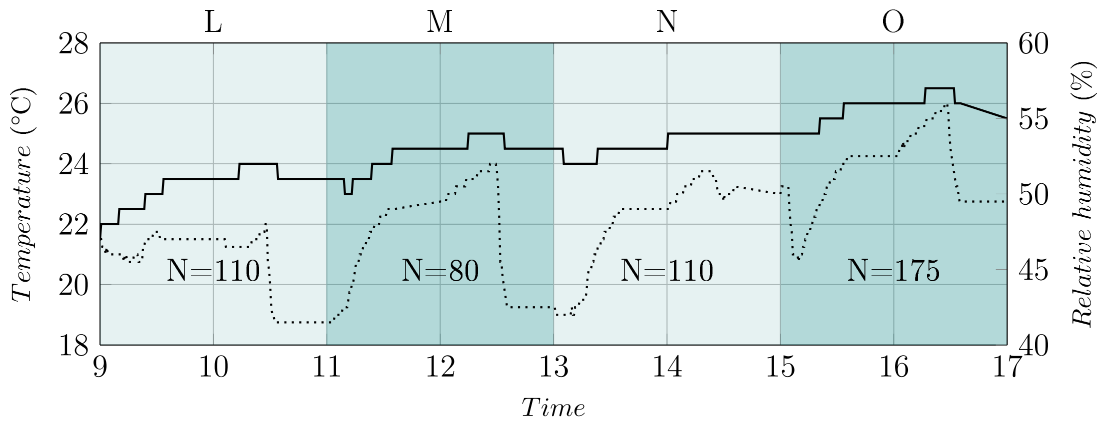
Time	Lesson	Occupancy (N people)	$L_{eq,A}$ (dB)	L_{min} (dB)	L_{MAX} (dB)	$L95$ (dB)	$L90$ (dB)	Recording time (h:m)
09.00-11.00	L	110	64.9	39.4	89.1	49.0	51.8	01:35
11.00-13.00	M	80	69.4	41.0	82.4	46.9	48.9	01:30
13.00-15.00	N	110	65.3	42.2	87.7	52.6	54.9	01:35
15.00-17.00	O	175	66.3	41.8	82.9	49.3	52.7	01:27



(a) Aula III: temperature and relative humidity measured values during the entire day of lessons.

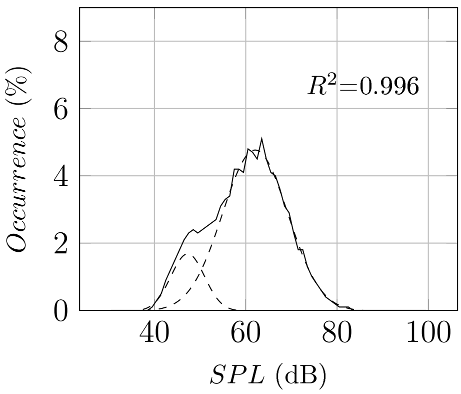


(b) Aula V: temperature and relative humidity measured values during the entire day of lessons.

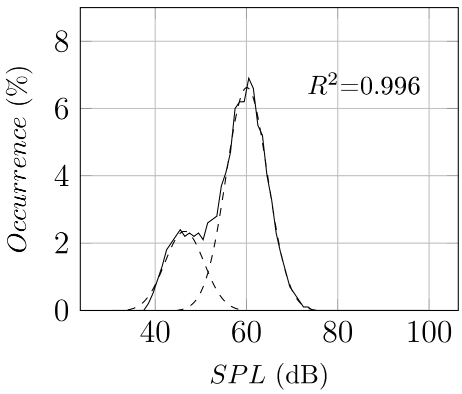


(c) Aula VI: temperature and relative humidity measured values during the entire day of lessons.

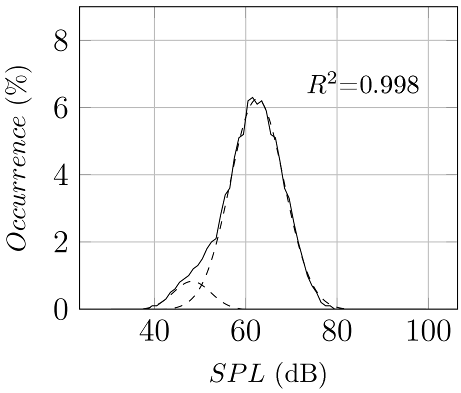
Figure 3.2: Measured values of temperature and relative humidity during lesson activity in the three lecture halls. Fluctuations of values show the periods while the doors were opened and how the parameters stabilize during the lectures. The “N” values indicate the occupancy state of the room during the measurements.



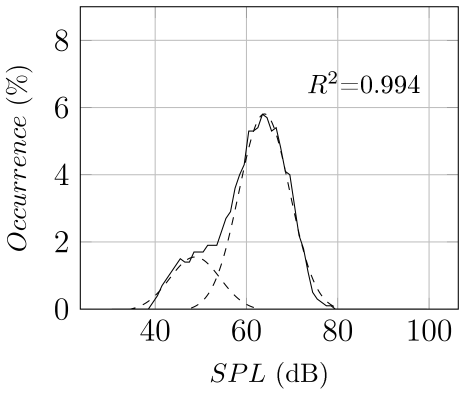
(a) Measured values during lesson A.
Student activity peak = 47.3 dB
Received speech level peak = 62.1 dB



(b) Measured values during lesson B.
Student activity peak = 46.4 dB
Received speech level peak = 60.1 dB

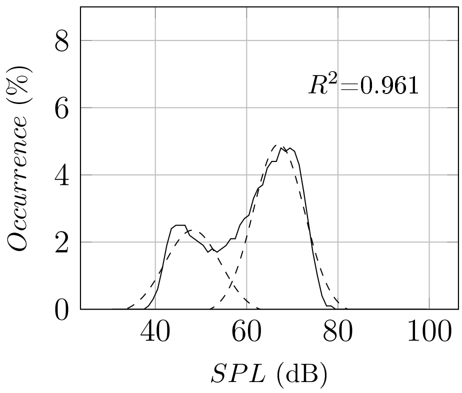


(c) Measured values during lesson C.
Student activity peak = 48.2 dB
Received speech level peak = 62.5 dB

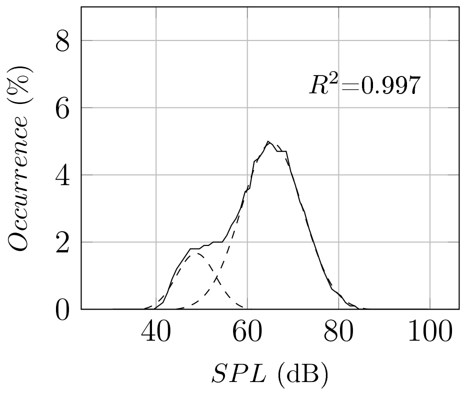


(d) Measured values during lesson D.
Student activity peak = 48.7 dB
Received speech level peak = 63.9 dB

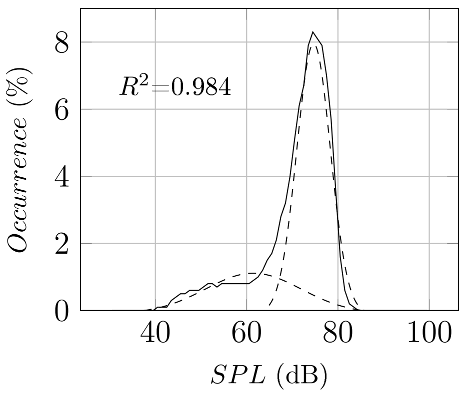
Figure 3.3: Aula III, sound level meter $R1$: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



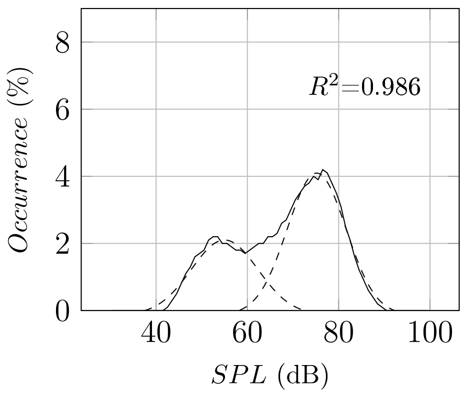
(a) Measured values during lesson E.
Student activity peak = 48.2 dB
Received speech level peak = 67.1 dB



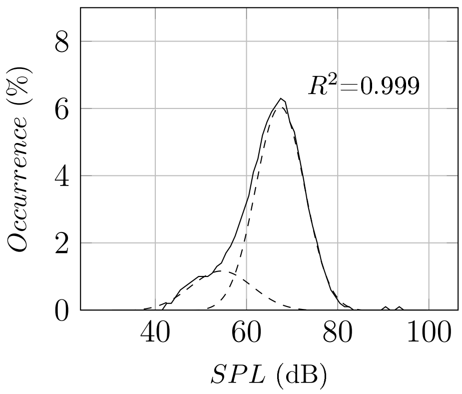
(b) Measured values during lesson F.
Student activity peak = 48.7 dB
Received speech level peak = 65.4 dB



(c) Measured values during lesson G.
Student activity peak = 61.5 dB
Received speech level peak = 74.7 dB

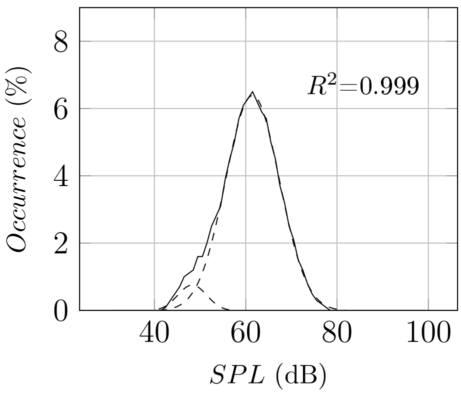


(d) Measured values during lesson H.
Student activity peak = 55.0 dB
Received speech level peak = 75.2 dB

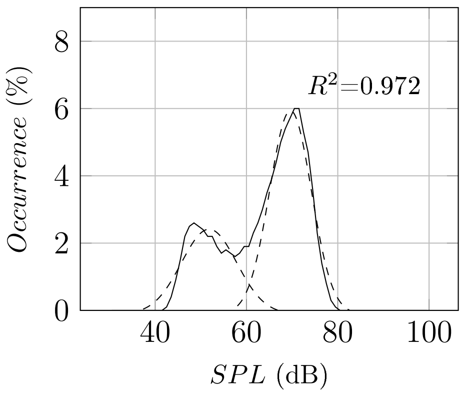


(e) Measured values during lesson I.
Student activity peak = 54.5 dB
Received speech level peak = 67.4 dB

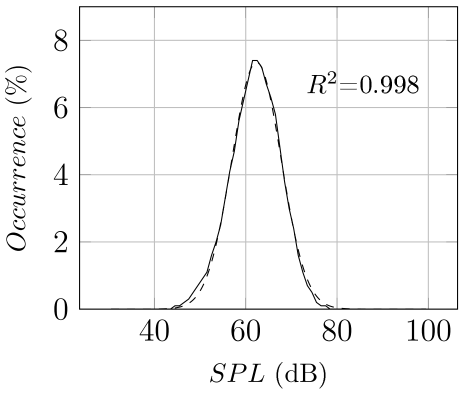
Figure 3.4: Aula V, sound level meter $R1$: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



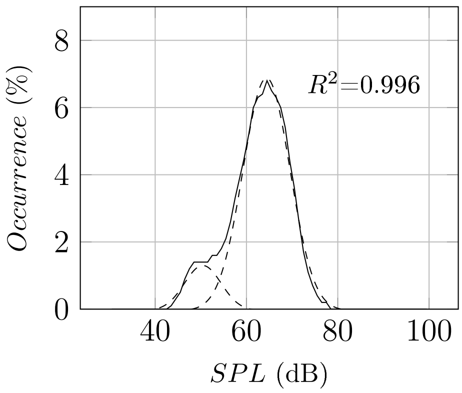
(a) Measured values during lesson L.
Student activity peak = 48.3 dB
Received speech level peak = 61.5 dB



(b) Measured values during lesson M.
Student activity peak = 51.8 dB
Received speech level peak = 69.6 dB

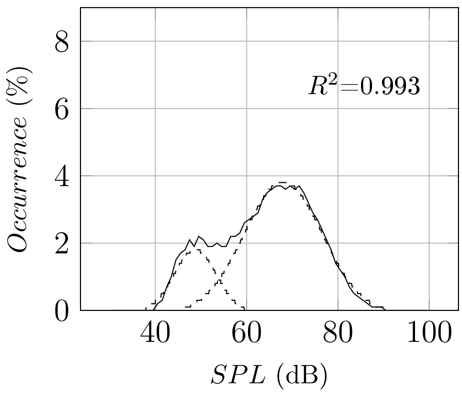


(c) Measured values during lesson N.
Student activity peak = not detected
Received speech level peak = 62.4 dB

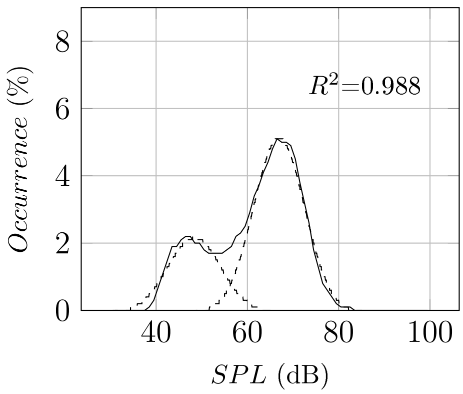


(d) Measured values during lesson O.
Student activity peak = 50.4 dB
Received speech level peak = 64.4 dB

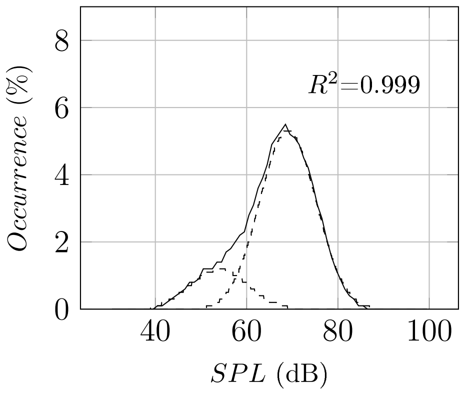
Figure 3.5: Aula VI, sound level meter $R1$: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



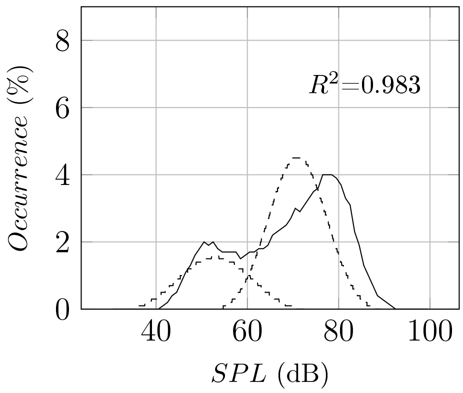
(a) Measured values during lesson one.
Student activity peak = 48.7 dB
Received speech level peak = 68.0 dB



(b) Measured values during lesson two.
Student activity peak = 48.4 dB
Received speech level peak = 66.9 dB

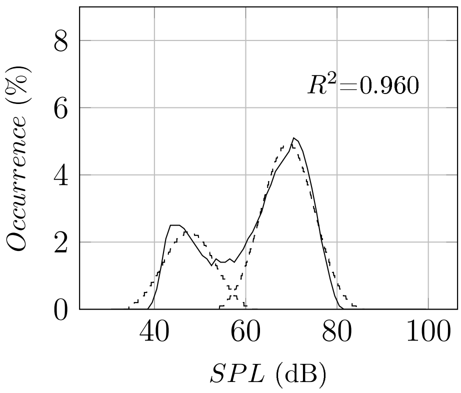


(c) Measured values during lesson three.
Student activity peak = 53.9 dB
Received speech level peak = 69.0 dB

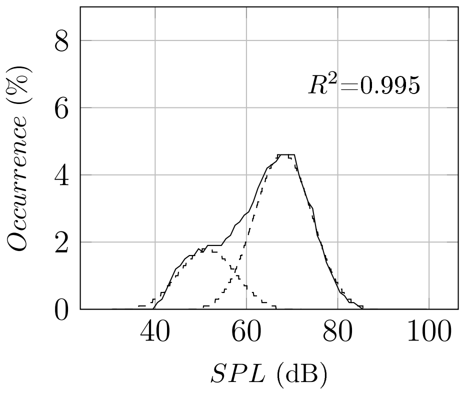


(d) Measured values during lesson four.
Student activity peak = 52.9 dB
Received speech level peak = 70.9 dB

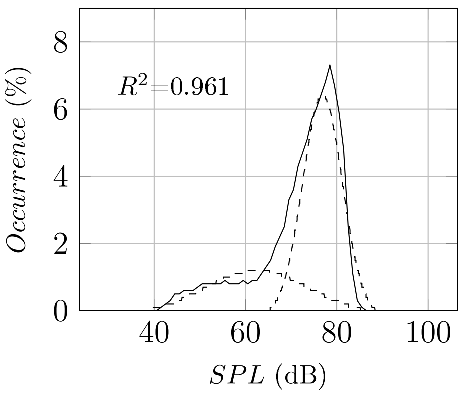
Figure 3.6: Aula III, sound level meter $R2$: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



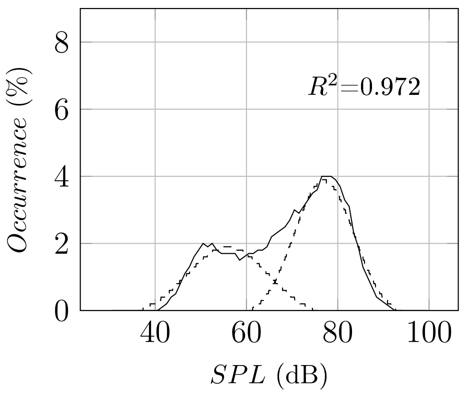
(a) Measured values during lesson one.
Student activity peak = 47.6 dB
Received speech level peak = 69.2 dB



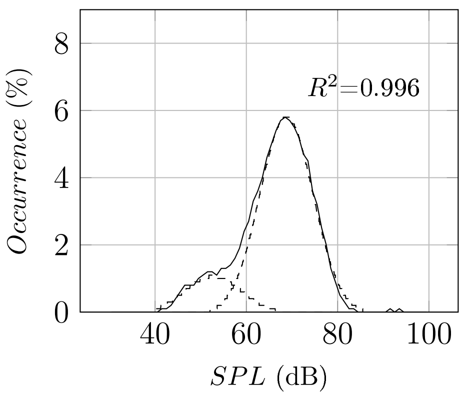
(b) Measured values during lesson two.
Student activity peak = 51.4 dB
Received speech level peak = 68.0 dB



(c) Measured values during lesson three.
Student activity peak = 62.5 dB
Received speech level peak = 76.9 dB

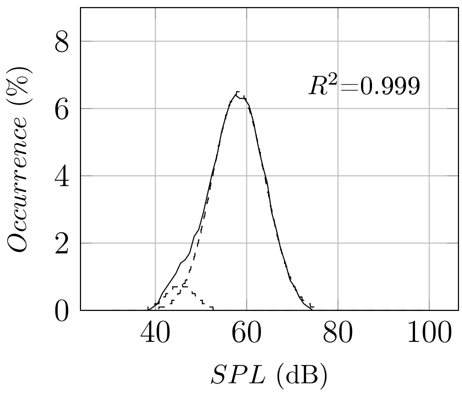


(d) Measured values during lesson four.
Student activity peak = 55.9 dB
Received speech level peak = 77.0 dB

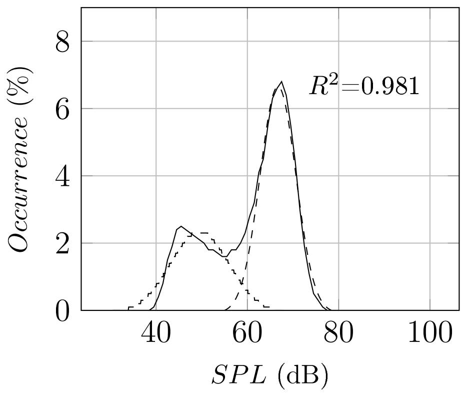


(e) Measured values during lesson five.
Student activity peak = 52.7 dB
Received speech level peak = 68.8 dB

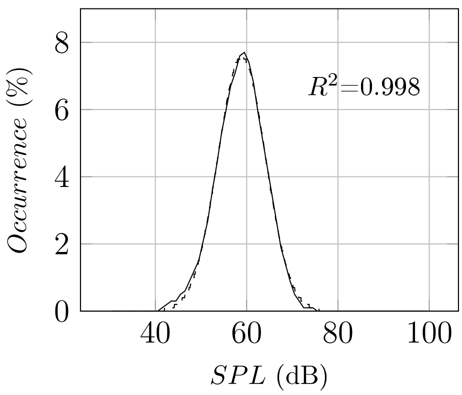
Figure 3.7: Aula V, sound level meter $R2$: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



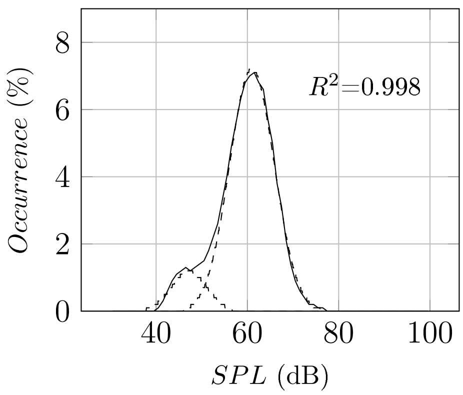
(a) Measured values during lesson one.
Student activity peak = 45.5 dB
Received speech level peak = 58.3 dB



(b) Measured values during lesson two.
Student activity peak = 49.5 dB
Received speech level peak = 66.7 dB



(c) Measured values during lesson three.
Student activity peak = not detected
Received speech level peak = 58.9 dB



(d) Measured values during lesson four.
Student activity peak = 47.2 dB
Received speech level peak = 61.0 dB

Figure 3.8: Aula VI, sound level meter $R2$: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.

Table 3.5: Aula III: measured A-weighted of equivalent, student activity and received speech sound pressure levels and corresponding signal-to-noise ratio values and averaged on the two positions during lessons after curve fitting.

Lesson time	$L_{eq,A}$ (dB)	Student activity (dB)	Speech level (dB)	Signal-to-noise ratio (dB)
09.00-11.00	69.9	48.0	65.1	17.1
11.00-13.00	64.8	47.4	63.5	16.1
13.00-15.00	68.9	51.1	65.8	14.7
15.00-17.00	69.7	50.8	67.4	16.6

Table 3.6: Aula V: measured A-weighted of equivalent, student activity and received speech sound pressure levels and corresponding signal-to-noise ratio values and averaged on the two positions during lessons after curve fitting.

Lesson time	$L_{eq,A}$ (dB)	Student activity (dB)	Speech level (dB)	Signal-to-noise ratio (dB)
09.00-11.00	72.3	47.9	68.2	18.9
11.00-13.00	71.9	50.1	66.7	16.7
13.00-15.00	78.6	62.0	75.8	13.2
15.00-17.00	79.2	55.5	76.1	20.2
17.00-19.00	74.6	53.6	68.1	12.9

Table 3.7: Aula VI: measured A-weighted of equivalent, student activity and received speech sound pressure levels and corresponding signal-to-noise ratio values and averaged on the two positions during lessons after curve fitting. During the lesson of 13.00-15.00 it's not possible to detect a student activity from the peak analysis.

Lesson time	$L_{eq,A}$ (dB)	Student activity (dB)	Speech level (dB)	Signal-to-noise ratio (dB)
09.00-11.00	63.3	46.9	59.9	13.0
11.00-13.00	67.8	50.4	68.2	17.8
13.00-15.00	63.6	–	60.7	–
15.00-17.00	65.1	48.8	62.7	13.9

3.3 Performance of public address

At the time of writing not the entire project was completed but only the active acoustic intervention with the installation of the public address system. A campaign of measurements was carried out on the 21th, 22th and 23th December 2017 to test the correct installation of the public address system and its effect on the acoustic behavior of the rooms. In each room a pair of loudspeakers produced by L-Acoustic, specifically the model called Syva was installed. This is a colinear source system designed for medium throw applications with a directivity very wide on the horizontal plane and narrow on the vertical plane. The high sound pressure level emitted by the speakers permits to cover homogeneously the audience area.

Measurements were conducted in unoccupied state using the public address system installed as signal source. Two kinds of STIPA measures were made, with the direct and indirect method as seen in section 1.2.4. The equipment used was similar to that seen in section 2.2, specifically:

- a laptop that launched the ESS (exponential sine sweep) signal with length of 512k and sampled at 48 kHz;
- a signal converter (Motu UltraLite AVB);
- one monoaural half inch free-field microphones (NTI audio MA220) as receiver.

And for the STIPA measures with the direct method [10]:

- a source to launch the modulated noise signal (NTI Minirator MR-PRO);
- a sound level meter with post-processing (NTI XL2 audio and acoustic analyzer).

The indirect method was used for all the receivers positions as seen in figure 2.3 while the direct method was used only for some of them as seen in figures (see figure 3.12). A comparison between the measured values before and after the installation of the new public address system is shown in the following tables 10, 11 and 12.

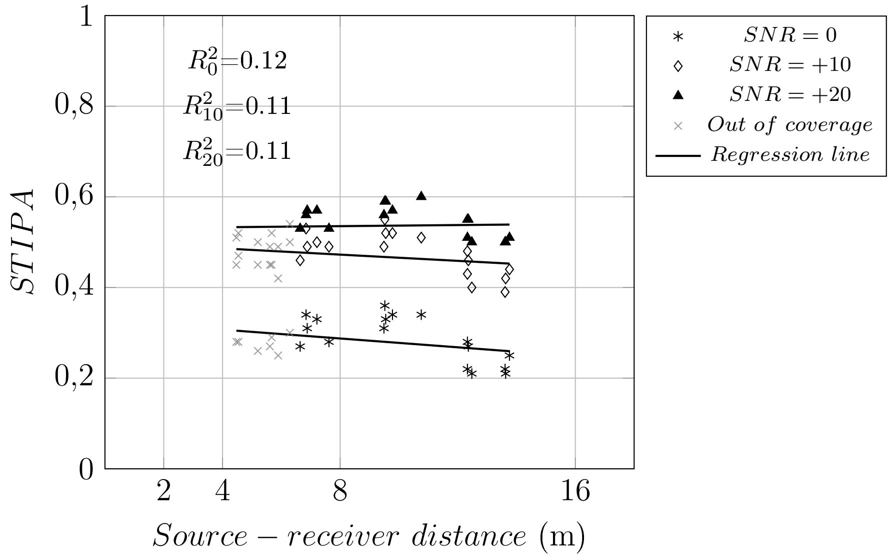


Figure 3.9: Aula III: measured values of STIPA, with direct method, in function of source-receiver distance with fixed signal-to-noise SNR values. Public address sound pressure level was considered at 0 when equal to the HVAC (heating, ventilation and air conditioning) noise. Thus the public address signal was increased first of +10 dB and then of +20 dB. The values in gray show how the nearest rows don't benefit from the public address system since STIPA values float in function of the width of the room as expected from simulations. The area covered by the system make the values higher and quite homogeneous. The small volume beyond the last truss doesn't take advantage from the public address source. The subscripts of the coefficient of determination means the corresponding SNR.

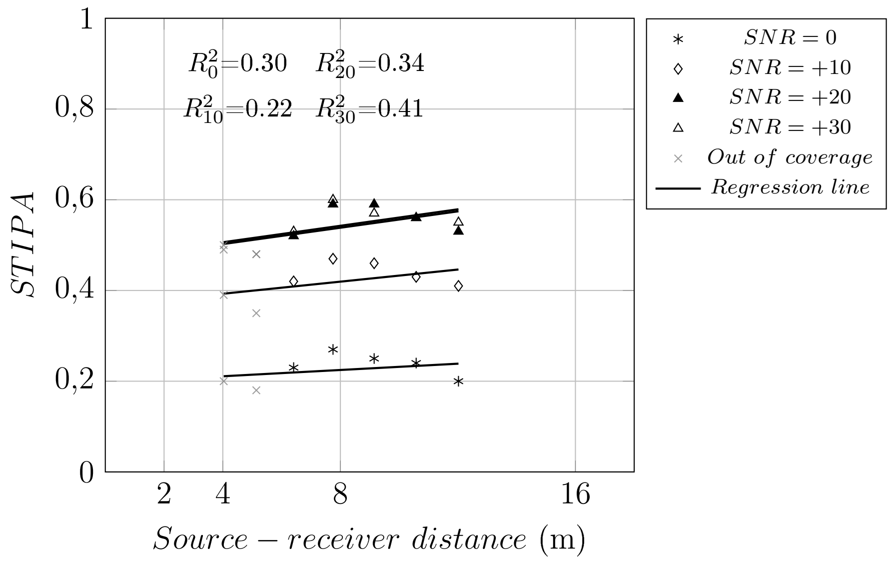


Figure 3.10: Aula V: measured values of STIPA, with direct method, in function of source-receiver distance with fixed signal-to-noise SNR values. Public address sound pressure level was considered at 0 when equal to the HVAC (heating, ventilation and air conditioning) noise. Thus the public address signal was increased first of +10 dB, of +20 dB and then of +30 dB. The values in gray show how the first rows don't benefit from the public address system since measured STIPA values are lower then the farther rows as expected from simulations. Nevertheless unlike Aula III the values float less. With the distance STIPA values are more homogeneous. The lines of +20 db and +30 dB highlight how over an SNR of +20 dB it's almost impossible increase the speech intelligibility.The subscripts of the coefficient of determination means the corresponding SNR.

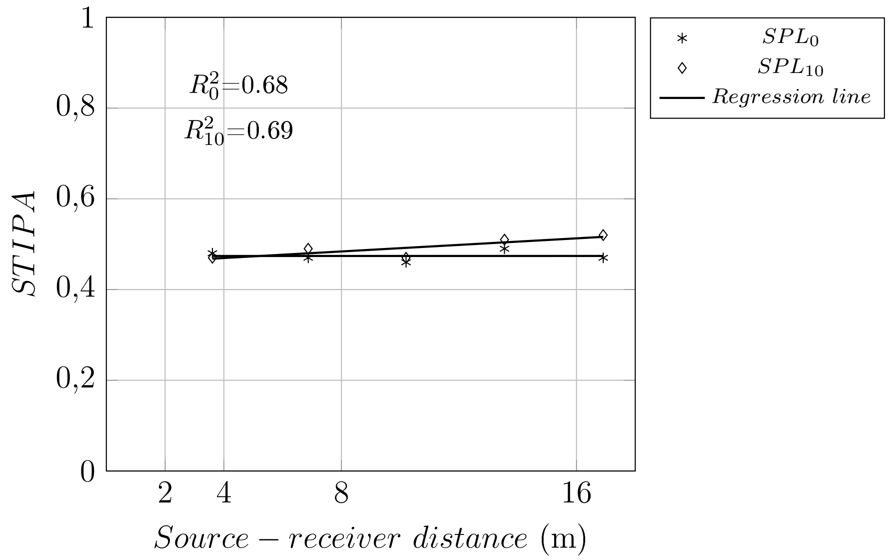
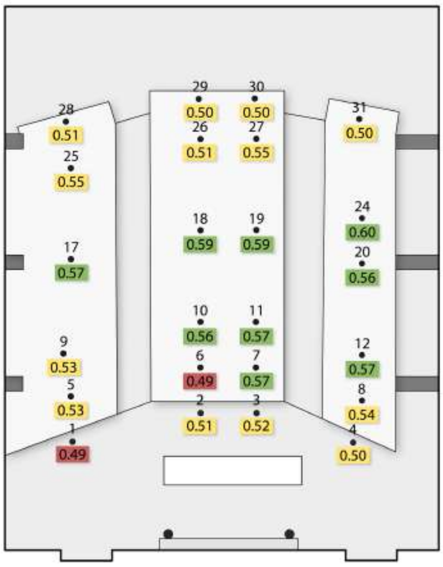
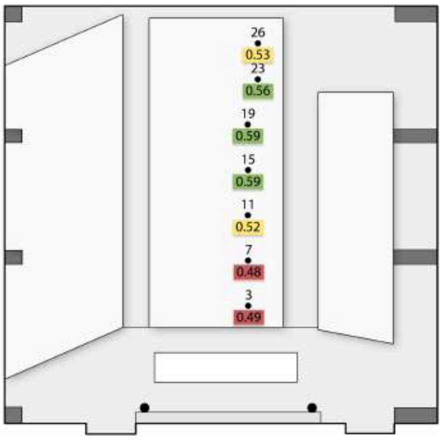


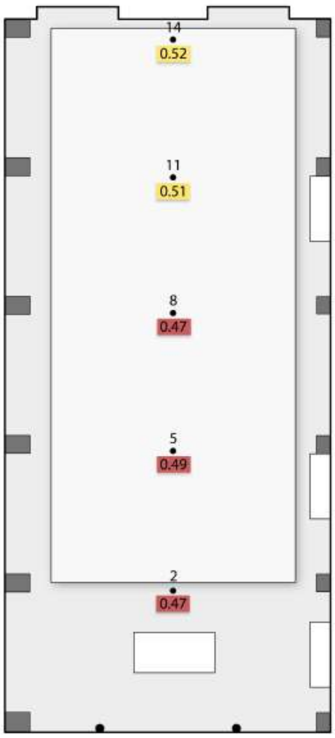
Figure 3.11: Aula VI: measured values of STIPA, with direct method, in function of source-receiver distance. Since the ventilation system was switched off two measurements were made without a fixed signal-to-noise ratio SNR. A typical value of sound pressure level of received speech signal, called SPL_0, was used and then increased of +10 dB, called SPL_{10}. The graph lines show how the public address signal spreads homogeneously in whole room in both cases. The subscripts of the coefficient of determination means the corresponding sound level pressure used.



(a) Aula III: measured STIPA values with direct method.

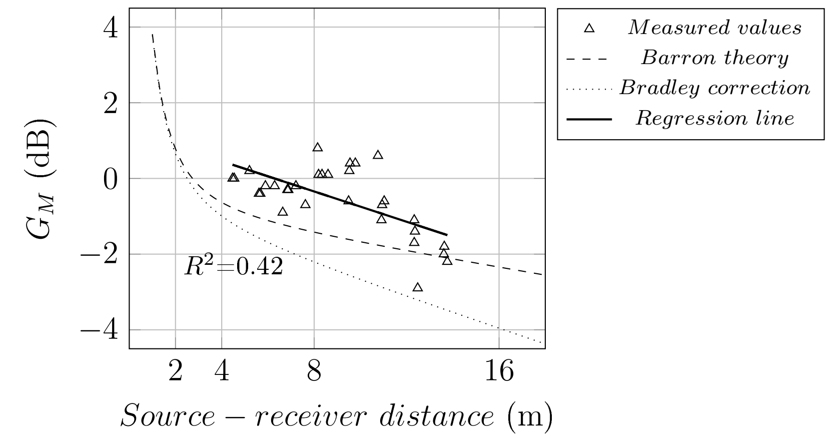


(b) Aula V: measured STIPA values with direct method.

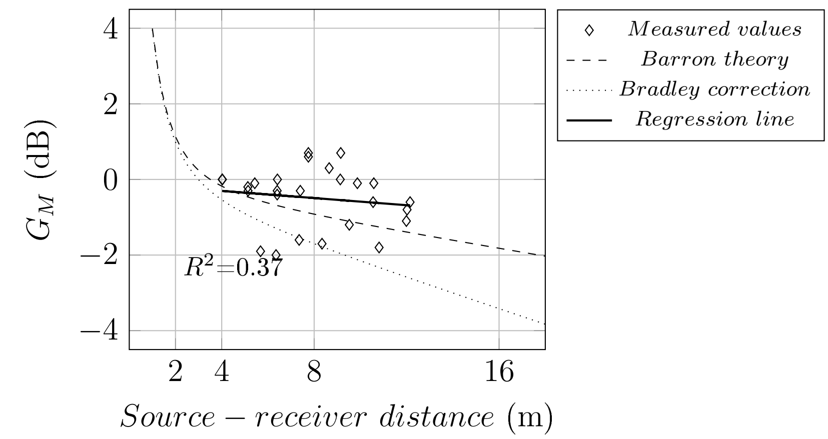


(c) Aula VI: measured STIPA values with direct method.

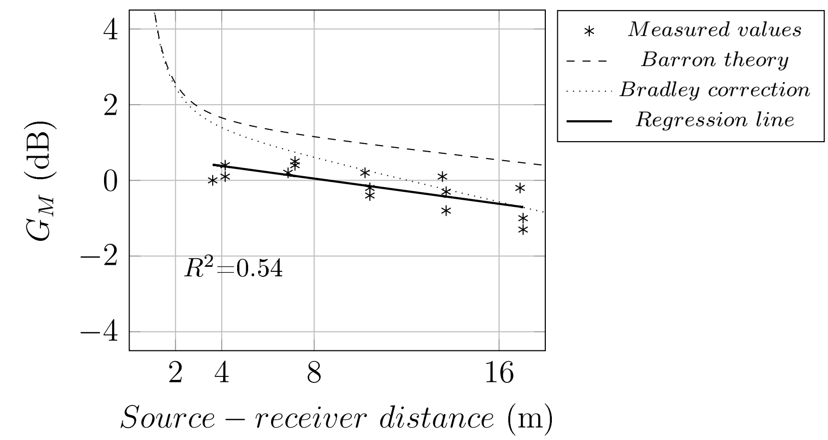
Figure 3.12: Measured STIPA values with the direct method at corresponding receiver positions. Values lower than 0.5 are indicated in red, that included between 0.50 and 0.55 in yellow and that higher than 0.55 in green. The colors subdivision shows values classified in "fair" category of IEC 60268-16 [10] in base of their distance from the 0.6 value, corresponding to the minimum of "good" category.



(a) Aula III: measured G values in function of source-receiver distance.



(b) Aula V: measured G values in function of source-receiver distance.



(c) Aula VI: measured G values in function of source-receiver distance.

Figure 3.13: Measured values of sound strength G using the public address system as source in function of source-receiver distance. In each graph the coefficient of determination to evaluate the goodness of fit is indicated. A comparison with Bradley and Lee's revised theory (dashed line) and Bradley and Sato's correction (dotted line) is visible (see section 1.2.6).

Chapter 4

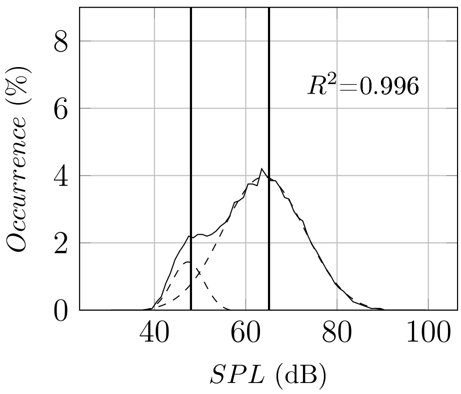
Discussions

The discussion of results are here spread out. Background noise due to the student activity is presented with averaged values between the two receivers used. Particular cases are individually analysed and discussed. A comparison between the predicted values from the Hodgson's model and measured values is shown. Subsequently discussions concerning the effects of the sound energy spatial distribution due to the installation of the public address system are presented. A comparison between ante-operam and post-operam values and theory predictions by Barron and Lee's revised theory and its correction provided by Bradley and Sato is shown for sound strength G parameter. Additionally, a comparison between ante-operam and post-operam STI values are presented in order to highlight the effects of new loudspeakers.

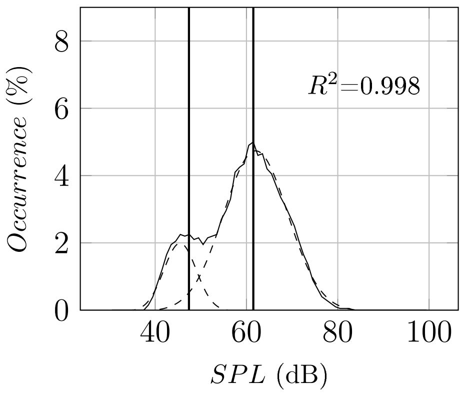
4.1 Reliability of student activity

The results deserve some considerations about the acoustic conditions during the lessons, their effects and the prediction model made by Hodgson seen in the section 1.2.3.

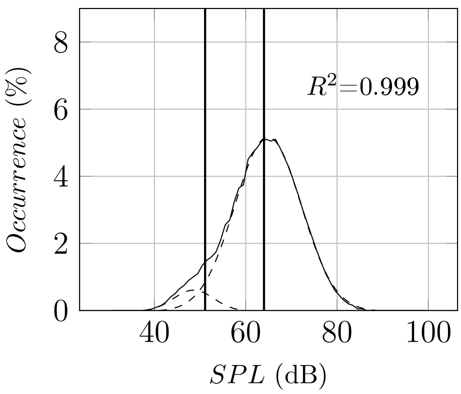
The measured student activities during lectures in each room have in most cases a similar value of background noise due to ventilation measured in the campaign of July seen in figure 2.8. This is an expected data according to Hodgson's measurements [25]. In fact, as specified in his work, the student activity level noticed has to be considered as a sum of itself and ventilation noise so to evaluate the single contribution a difference between this levels has to be done. If, during the measurement campaign done to evaluate the student activity noise, the ventilation system had been turned on higher values about +3 dB student activity were expected. Values in Aula VI make exception since ventilation noise measured values are lower than the other



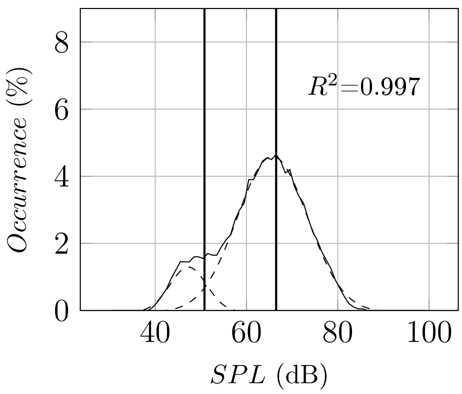
(a) Measured values during lesson A.
Student activity peak = 48.0 dB
Received speech level peak = 65.1 dB



(b) Measured values during lesson B.
Student activity peak = 47.4 dB
Received speech level peak = 63.5 dB

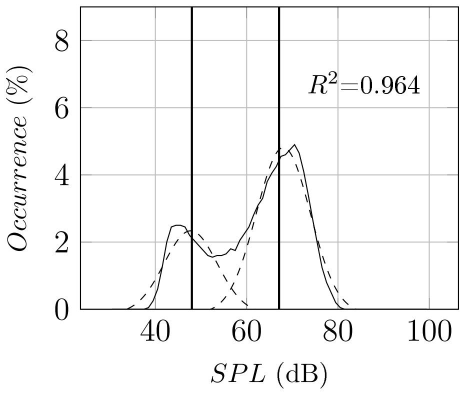


(c) Measured values during lesson C.
Student activity peak = 51.1 dB
Received speech level peak = 65.8 dB

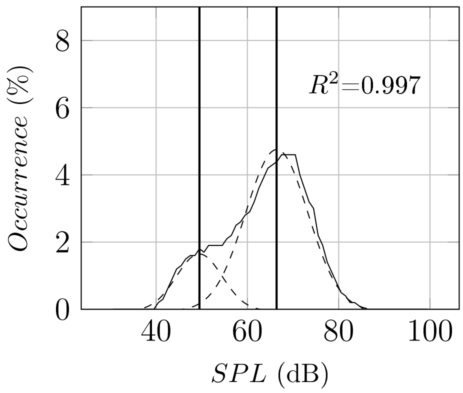


(d) Measured values during lesson D.
Student activity peak = 50.8 dB
Received speech level peak = 67.4 dB

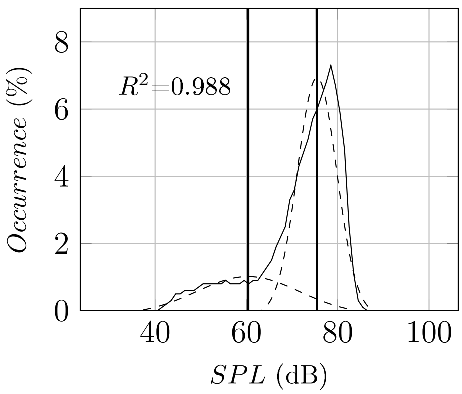
Figure 4.1: Aula III: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



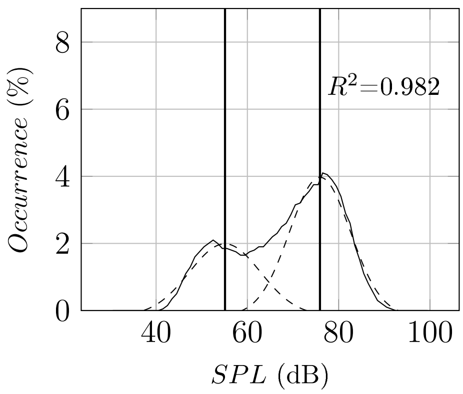
(a) Measured values during lesson E.
Student activity peak = 47.9 dB
Received speech level peak = 68.2 dB



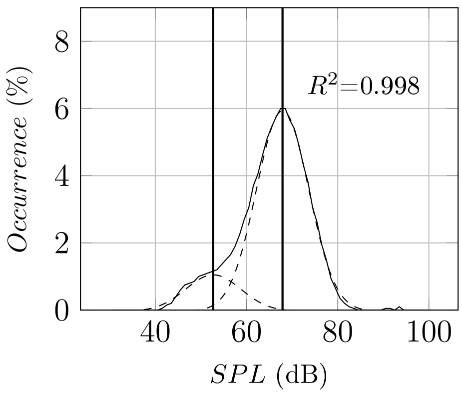
(b) Measured values during lesson F.
Student activity peak = 49.5 dB
Received speech level peak = 68.2 dB



(c) Measured values during lesson G.
Student activity peak = 62.0 dB
Received speech level peak = 75.8 dB

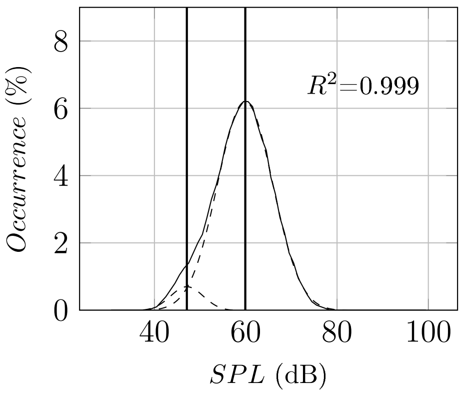


(d) Measured values during lesson H.
Student activity peak = 55.5 dB
Received speech level peak = 76.1 dB

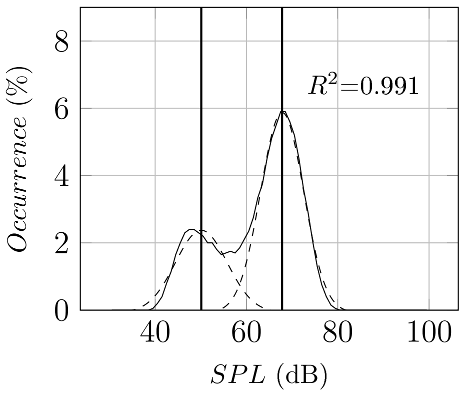


(e) Measured values during lesson I.
Student activity peak = 53.6 dB
Received speech level peak = 68.1 dB

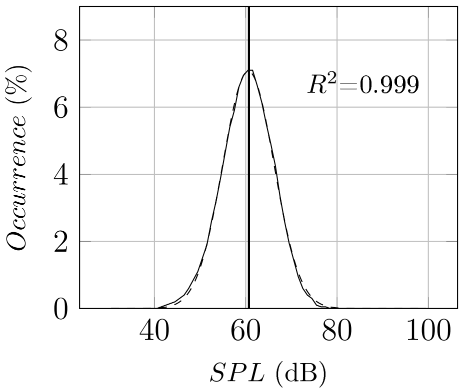
Figure 4.2: Aula V: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.



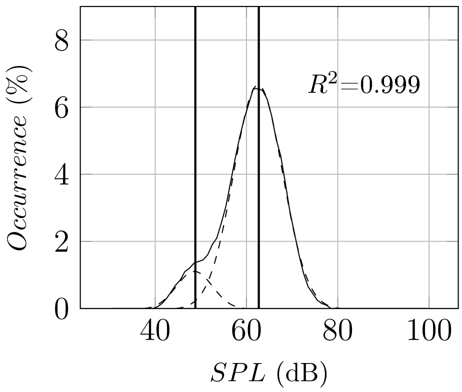
(a) Measured values during lesson L.
Student activity peak = 46.9 dB
Received speech level peak = 59.9 dB



(b) Measured values during lesson M.
Student activity peak = 50.4 dB
Received speech level peak = 68.2 dB



(c) Measured values during lesson N.
Student activity peak = not detected
Received speech level peak = 60.7 dB



(d) Measured values during lesson O.
Student activity peak = 48.8 dB
Received speech level peak = 62.7 dB

Figure 4.3: Aula VI: cumulative distribution of measured sound pressure levels during lessons activity with their occurrence. Curve fitting permits to highlight two peaks: the lower corresponding to student activity and the higher corresponding to the received speech level. In each graph the coefficient of determination R^2 is indicated to evaluate the goodness of fit.

rooms.

Measures show a *Lombard effect* during the various lessons and the signal-to-noise ratio highlights how there is a kind of auto-leveling of the speech and the student activity signal. In fact it assumes a value close to +15 dB in almost all lessons (see tables 3.5, 3.6 and 3.7). It means that with the public address support and the *Lombard effect* the teacher attempts to reach the minimum signal-to-noise level to have the maximum speech intelligibility possible as recommended by scientific literature [15] [21].

An interesting data concerns the lesson of medieval history, the lesson N of Aula VI. During this lecture, the teacher didn't use the microphone and so the public address support. Whereas the sound pressure level measured by the sound level meter is similar to that of the other lessons, it is easy to guess that the professor had to put a lot of strain on his voice. Leaving aside the vocal effort that the teacher has to bear, it is interesting to note that the lack of use of the microphone makes impossible to detect students activity in the regression of the curve. As shown in figure 4.3(c) the measured distribution is very symmetrical and with only one curve it's possible achieve a regression with a determination coefficient of 0.999. Although the little hump on the left side of the curve the occurrence is so low that is impossible to analyze its pick.

The tables 4.1, 4.2 and 4.3 show an high difference between predicted and measured values. In his work Hodgson highlights the low variance values, specifically 41% for the ventilation noise *VN*, 57% for the a) formulation (see equation 1.27) of student activity SA, depending by the number of student, the received speech level, his sex and the total-occupied room absorption surface, and 69% for the b) formulation (see equation 1.28) which is depending by the number of student and the total-occupied room absorption surface, 66% for the received speech signal *SL* and 69% for the instructor sound power *LW*. So this big difference wasn't unexpected. Another important consideration is about the better prediction of a) formulation of student activity SA values despite the b) ones. This could be explained by the *Lombard effect* that, as seen above, it's strongly present in measures. In fact the equation 1.28, despite the 1.27 has the dependency by the received speech level *SL*. This permits the evaluation of the *Lombard effect*, increasing the received speech level increase the student activity too. Nevertheless a similar dependency should be present between ventilation noise *VN* and student activity SA because every anthropic sound source is affected by the *Lombard effect* and student activity increase its sound level if the ventilation system increase too as suggested by Bradley [21]. Hodgson's prediction model besides doesn't specify how consider the absorption coefficient due to people in the room. This is an important issue since doesn't exist a unique database of absorption coefficients to use in design and research works so it's difficult understand how the prediction consider the effect of the people. Isn't clear why are two contributions due to people in the equations 1.26, 1.27 and 1.28 and why is positive in the equation 1.26 since the total-occupied absorption area includes it yet. Indeed calculating the prediction terms with the measured absorption area but in unoccupied state of the room an improve of about ±3 dB is obtained. The absorption area in fact is the only term which could be varied to get a better prediction.

Table 4.1: Aula III: comparison between mean predicted and measured student activity (SA) and received speech level (SL) values and their difference indicated as *delta*. For student activity two alternative values are presented distinguishing the value (a) depending by the speech level, instructor sex, number of people and the total occupied-room absorption area (see equation 1.27) and the value (b) depends by the number of people and the total occupied-room absorption area (see equation 1.28).

Time	Predicted values			Measured values		Delta		
	SA (dB)		SL (dB)	SA (dB)	SL (dB)	SA (dB)		SL (dB)
	(a)	(b)				(a)	(b)	
09.00 - 11.00	44.1	41.6	55.1	48.0	65.1	3.9	6.4	9.9
11.00 - 13.00	45.6	43.0	55.9	47.4	63.5	1.8	4.4	7.6
13.00 - 15.00	42.1	40.0	51.6	51.1	65.8	9.0	11.1	14.2
15.00 - 17.00	44.3	41.8	55.2	50.8	67.4	6.5	9.0	12.2

4.2 Reliability of previsional models of sound energy

Measures in the tables 10 - 12 show an important increase of STI values obtained only with the installation of the new public address system. The negative values of the noticed differences are due to the directivity of the Syva loudspeakers and highlight the correct placement of them. The positions covered by the center of the sound area generated by the system even satisfy the standard requirements. In the Aula VI it's interesting to observe how the new loudspeakers made the received signal homogeneous.

In figures 3.9 - 3.11 values measured with direct method of STIPA are shown at various signal-to-noise ratio. With the sound level meter first of all

Table 4.2: Aula V: comparison between mean predicted and measured student activity (SA) and received speech level (SL) values and their difference indicated as *delta*. For student activity two alternative values are presented distinguishing the value (a) depending by the speech level, instructor sex, number of people and the total occupied-room absorption area (see equation 1.27) and the value (b) depends by the number of people and the total occupied-room absorption area (see equation 1.28).

Time	Predicted values			Measured values		Delta		
	SA (dB)		SL (dB)	SA (dB)	SL (dB)	SA (dB)		SL (dB)
	(a)	(b)				(a)	(b)	
09.00 - 11.00	47.0	44.2	56.3	47.9	68.2	0.9	3.7	11.9
11.00 - 13.00	44.7	42.3	52.5	50.1	66.7	5.4	7.8	14.2
13.00 - 15.00	43.4	41.0	51.8	62.0	75.8	18.6	21.0	24.0
15.00 - 17.00	44.4	42.0	52.4	55.5	76.1	11.1	13.5	23.7
17.00 - 19.00	46.0	43.2	55.7	53.6	68.1	7.6	10.4	12.4

Table 4.3: Aula VI: comparison between mean predicted and measured student activity (SA) and received speech level (SL) values and their difference indicated as *delta*. For student activity two alternative values are presented distinguishing the value (a) depending by the speech level, instructor sex, number of people and the total occupied-room absorption area (see equation 1.27) and the value (b) depends by the number of people and the total occupied-room absorption area (see equation 1.28).

Time	Predicted values			Measured values		Delta		
	SA (dB)		SL (dB)	SA (dB)	SL (dB)	SA (dB)		SL (dB)
	(a)	(b)				(a)	(b)	
09.00 - 11.00	45.1	41.9	53.2	46.9	59.9	1.8	5.0	6.7
11.00 - 13.00	43.7	40.6	52.4	50.4	68.2	6.7	9.8	15.8
13.00 - 15.00	45.4	41.9	55.8	–	60.7	–	–	4.9
15.00 - 17.00	47.2	44.0	54.3	48.8	62.7	1.6	4.8	8.4

the ventilation noise was measured, thus the signal input device was set to emit from the source a sound with the same level to achieve a zero signal-to-noise ratio. After that, the source output level was increased with steps of +10 dB. Not all the receiver positions seen in figure 2.3 were used but only those considered significant, specifically the positions along a linear direction to evaluate an important source-to-receiver distance.

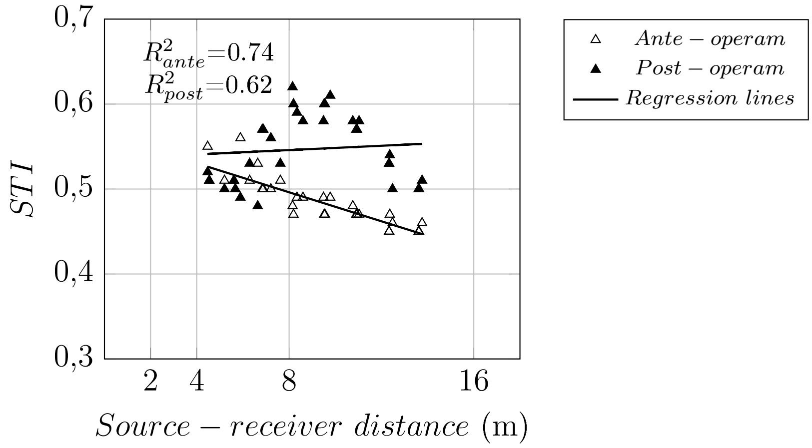
The measures in Aula III (see figure 3.9) show various fluctuations of values in the distances corresponding to the first rows of the audience area which are not involved in the coverage area of the new loudspeakers. Starting from a certain distance, in fact, STIPA values are more homogeneous with the highest values in the central receiver positions. Instead the last rows over the last truss aren't covered effectively by the public address system and the values experience a sharp fall. The difference of the increase due to the better signal-to-noise ratio highlights how the benefit grows with a logarithmic tendency so, as noticed by the technical literature, over a certain SNR value, corresponding to +20 dB, the speech intelligibility doesn't increase.

In Aula V (see figure 3.10) some differences are noticeable unlike the previous one. Its behaviour is more flat. Despite this, like in Aula III, the lowest values correspond to the first rows then, with the increase of the distance, they grow. A little decrease is visible on the farthest rows but not with a significant relevance. STIPA values, in this room, were evaluated for a signal-to-noise ratio of +30 dB too. As explained above, over an SNR value of +20 dB is impossible to detect a benefit and this is demonstrated by the overlap of the two curves.

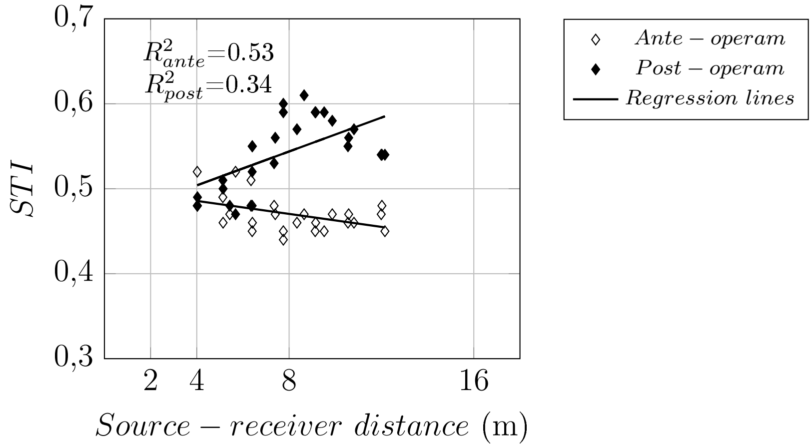
In the case of Aula VI it was impossible to measure STIPA values with a certain signal-to-noise ratio set since the ventilation was turned off. Thus a typical output signal, called SPL_{10} was emitted from the source and then increased of +10 dB, called SPL_{10}. Differences between the two signals aren't evident (see figure 3.11). A strong uniformity is noticeable, this highlights the effectiveness of the new public address system installation.

Measured values in all receiver positions of each room in function of source-to-receiver distance and signal-to-noise ratio are visible in detail in tables 7- 9. A comparison between ante-operam and post-operam measured values, with the indirect method, is shown for each room in figure 4.4.

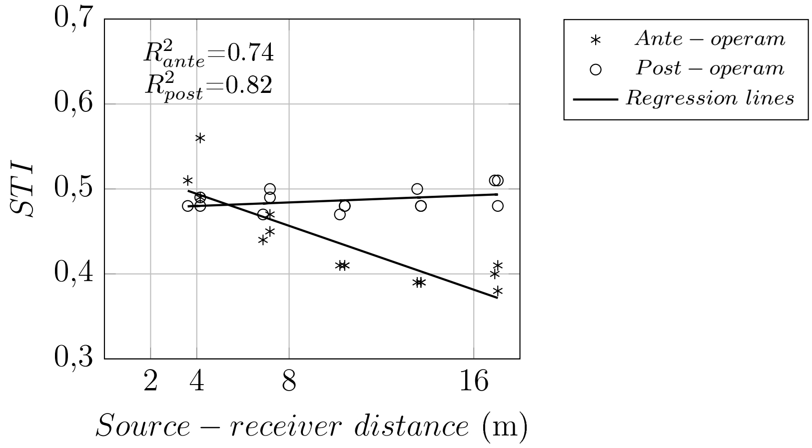
The figure 4.5 shows comparison of the ante-operam and post-operam sound strength G values and the prevision models, the Barron and Lee's revised theory and the Bradley and Sato's correction. The source used is the new public address system. Measures show how, especially in Aula V and Aula VI, the sound strength doesn't decrease with the distance. In Aula III the curve slope decreases less than expected. This highlights the goodness of the diffusion of the loudspeakers. The mini line array, in fact, permits to have an high homogeneity of the diffusion of sound. This particular configuration of loudspeakers emits cylindrical waves. These have a longer decay, so it's possible arrive to the farther rows with an higher sound pressure levels.



(a) Aula III: measured *STI* values in function of source-receiver distance.

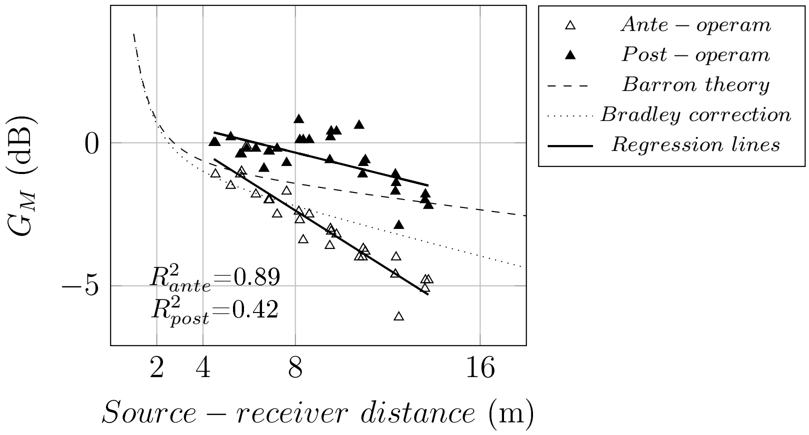


(b) Aula V: measured *STI* values in function of source-receiver distance.

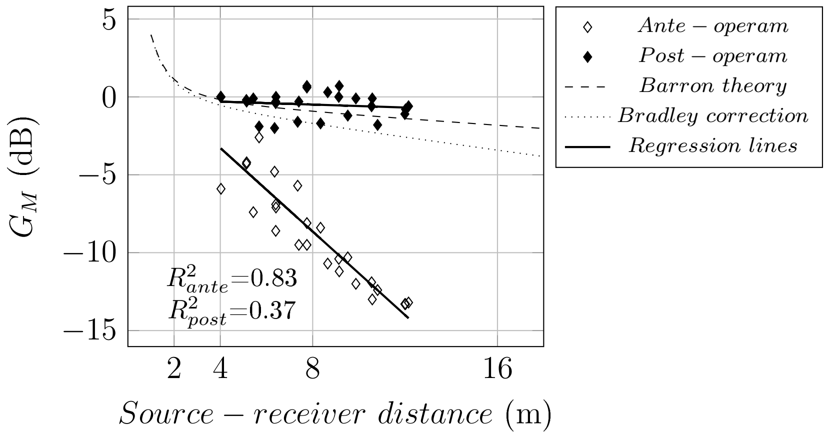


(c) Aula VI: measured *STI* values in function of source-receiver distance.

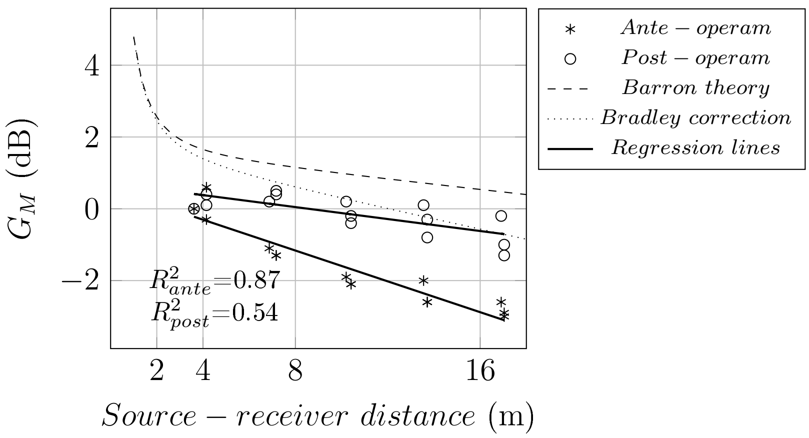
Figure 4.4: Comparison ante-operam and post-operam values of speech transmission index STI in function of source-receiver distance. Both of measurements were conducted in unoccupied state.



(a) Aula III: measured G values in function of source-receiver distance.



(b) Aula V: measured G values in function of source-receiver distance.



(c) Aula VI: measured G values in function of source-receiver distance.

Figure 4.5: Comparison between measured values of sound strength G ante-operam and post-operam in function of source-receiver distance. Both of measurements were conducted in unoccupied state.

Conclusions

Learning process is anchored to the communication quality. An high acoustic comfort is fundamental to get the suitable condition for an intense teaching and learning flow between teachers and students. A good speech intelligibility permits to minimize the vocal effort of the teacher and maximize the students' concentration, improving the educational work. The case studies presented in this work are three historical lecture halls of the Faculty of Letters and Philosophy of University of Bologna. Taking into account their historical charisma, it's easy to understand the importance of achieving a good acoustic comfort in them. According to ISO 3382-1 [1] a measurements campaign was carried out in order to describe with objective parameters the acoustic characteristics of the case studies. In all the lecture halls the reverberation time was too high to achieve the optimal speech intelligibility. This latter had lower values than the standard requirements in every measured position. Further investigations were made to study the spatial energy distribution in each room and their background noise levels due to the ventilation system and, eventually, to the outdoor factors.

The design process followed normative guidelines, specifically DIN 18041 [7], UNI 11532 [8] and BB 93 [9]. Two types of interventions were defined: active and passive treatments. The active treatments include the installation of a new public address system with a new technology: mini line array loudspeakers which spread cilindrical wave instead of the usual spherical one. This permits to reduce the spatial decay of the sound and make the sound field more homogeneous. The passive treatments concern the installation of sound absorbing panels in two lecture halls (Aula III and Aula V) and slat absorbing panels in the other one (Aula VI) in order to reduce the reverberation time, and thus also the late reflections energy, without damaging the early reflections paths. Decreasing the reverberation time is not enough for increasing the speech intelligibility but the placement of the sound absorbing panels is a fundamental aspect. Numerical models were created for the acoustic simulations in order to study in detail the issues and to estimate the effect of the intervention for achieving the supposed requirements. Many softwares were used, specifically SketchUp [2], Autocad [3], 3ds Max design [4] for three-dimensional modelling and rendering process, Odeon Room Acoustics [5] and Soundvision [6] for simulation process, respectively for the passive and active interventions.

A deep analysis was conducted, after design process, in order to evaluate the noise which affects the speech intelligibility during a lecture due to the student activity. Two sound level meters measured one entire day of lessons in each room. With the extracted data, a cumulative distribution and a statistical peaks analysis permitted to highlight the student activity noise and the speech levels. Measurements pointed out how the *Lombard effect* automatically works and how it's important trying to keep the speech levels low with the aim of minimizing the vocal effort of the teacher.

Finally, a further measurements campaign was carried out to evaluate the effects of the installation of the public address system. Spatial energy distribution and speech intelligibility parameters were measured and compared with the ante-operam ones. The results show an high increase of the speech intelligibility in every position covered by the directivity of the loudspeakers and a decrease of the sound energy decay with the distance. Even though in some positions the standard requirements were not satisfied with only the effect of the public address system, the passive treatments are expected to be crucial in fulfilling the enhancing process.

Bibliography

[1] ISO 3382-1:2008, Acoustics - Measurement of Room Acoustic Parameters - Part 1: Performance Spaces. Geneva: International Organization for Standardization.

[2] SketchUp. Trimble Inc., 2016.

[3] AutoCAD. Autodesk Company, 2017.

[4] 3ds Max Design. Autodesk Company, 2017.

[5] Odeon 12 Combined Edition. ODEON Room Acoustics Software, 2012.

[6] Soundvision 3.0.8. L-Acoustics, 2017.

[7] DIN 18041:2016, Acoustic quality in rooms - Specifications and instructions for the room acoustic design, DIN-Normenausschuss Bauwesen (in German).

[8] UNI 11532:2014, Acustica in edilizia - Caratterizzazione acustiche interne di ambienti confinati. Milano: Ente Nazionale Italiano di Unificazione (in Italian).

[9] Building Bulletin 93: acoustic design of schools - performance standards.

[10] IEC 60268-16:2011, Sound system equipment - Part 16: Objective rating of speech intelligibility by speech transmission index. Geneva: International Electro-technical Commission.

[11] ISO 3382-2:2008, Acoustics - Measurement of room acoustic parameters - Part 2: Reverberation time in ordinary rooms. Geneva: International Organization for Standardization.

[12] Pelegrin Garcia D., Rasmussen B., Brunskog J., Classroom acoustics design for speakers' comfort and speech intelligibility: a European perspective, Forum Acusticum, 2014.

[13] UNI 11367:2010, Acustica in edilizia - Classificazione acustica delle unità immobiliari, Procedura di valutazione e verifica in opera. Milano: Ente Nazionale Italiano di Unificazione.

[14] Beranek L., Analysis of Sabine and Eyring equations and their application to concert hall audience and chair absorption, J. Acoust. Soc. Am., 120(3) (2006), 1399–1410.

[15] Hodgson M., Nosal E., Effect of noise and occupancy on optimal reverberation times for speech intelligibility in classrooms, J. Acoust. Soc. Am., 111(2) (2002), 931–939.

[16] Hodgson M., Experimental investigation of the acoustical characteristics of university classrooms, J. Acoust. Soc. Am., 106(4) (1999), 1810–1819.

[17] Hodgson M., Case-study evaluations of the acoustical designs of renovated university classrooms, Applied Acoustics, 65 (2004), 69–89.

[18] Hodgson M., Wong G., Ray-tracing prediction of optimal conditions for speech in realistic classrooms, Applied Acoustics, 70 (2009), 915–920.

[19] Hodgson M., When is diffuse-field theory applicable?, Applied Acoustics, 49 (1996), 197–207.

[20] Reich R., Bradley J., Optimizing classroom acoustics using computer model studies, Canadian Acoustics, 26(4) (1998), 15–21.

[21] Sato H., Bradley J., Evaluation of acoustical conditions for speech communication in working elementary school classrooms, J. Acoust. Soc. Am., 123(4) (2008), 2064–2077.

[22] Barron M., Lee L-J., Energy relations in concert auditoriums. I, J. Acoust. Soc. Am., 84 (1988), 618–628.

[23] Marshall G., Speech intelligibility prediction from calculated C_{50} values, J. Acoust. Soc. Am., 98(5) (1995), 2845–2847.

[24] Bradley J., Acoustical design of rooms for speech, Construction Technology Update, 51 (2002).

[25] Hodgson M., Rempel R., Kennedy S., Measurement and prediction of typical speech and background noise, J. Acoust. Soc. Am., 105(1) (1999), 226–233.

[26] Nijs L., Rychtarikova M., Calculating the optimum reverberation time and absorption coefficient for good speech intelligibility in classroom design using U_{50}, Acta Acustica United With Acustica, 97 (2011), 93–102.

[27] Bradley J., Predictors of speech intelligibility in rooms, J. Acoust. Soc. Am., 80(3) (1986), 837–845.

[28] ISO 3741:2010, Acoustics - Determination of sound power levels and sound energy levels of noise sources using sound pressure - Precision methods for reverberation test rooms. Geneva: International Organization for Standardization.

[29] Rindel H.J., The use of computer modeling in room acoustics, Journal Of Vibroengineering, 3(4) (2000), 219–224.

[30] Christensen CL., ODEON Room Acoustics Software-version 12. Lyngby, Denmark: Industrial, Auditorium and Combined Editions, 2013.

A. Tables

Table 4: Aula III: predicted values of ventilation noise, student activity noise, received speech levels and instructor sound power levels according to Hodgson's model averaged to the two receiver's positions [25]. For student activity two alternative values are presented distinguishing the value (a) depending on the speech level, instructor sex, number of people and the total occupied-room absorption area (see equation 1.27) and the value (b) depending on the number of people and the total occupied-room absorption area (see equation 1.28).

Lesson time	Ventilation noise (dB)	Student activity (dB)		Speech level (dB)	Instructor sound power (dB)
		(a)	(b)		
09.00 - 11.00	40.8	44.1	41.6	55.1	69.9
11.00 - 13.00	42.2	45.6	43.0	55.9	70.6
13.00 - 15.00	39.1	42.1	40.0	51.5	66.5
15.00 - 17.00	40.9	44.2	41.8	55.2	70.0

Table 5: Aula V: predicted values of ventilation noise, student activity noise, received speech levels and instructor sound power levels according to Hodgson's model averaged to the two receiver's positions [25]. For student activity two alternative values are presented distinguishing the value (a) depending on the speech level, instructor sex, number of people and the total occupied-room absorption area (see equation 1.27) and the value (b) depending on the number of people and the total occupied-room absorption area (see equation 1.28).

Lesson time	Ventilation noise (dB)	Student activity (dB)		Speech level (dB)	Instructor sound power (dB)
		(a)	(b)		
09.00 - 11.00	44.8	47.0	44.2	56.3	70.3
11.00 - 13.00	42.8	44.7	42.3	52.5	66.7
13.00 - 15.00	41.5	43.4	41.0	51.8	66.1
15.00 - 17.00	42.5	44.4	42.0	52.4	66.6
17.00 - 19.00	43.8	46.0	43.2	55.7	69.8

Table 6: Aula VI: predicted values of ventilation noise, student activity noise, received speech levels and instructor sound power levels according to Hodgson's model averaged to the two receiver's positions [25]. For student activity two alternative values are presented distinguishing the value (a) depending on the speech level, instructor sex, number of people and the total occupied-room absorption area (see equation 1.27) and the value (b) depending on the number of people and the total occupied-room absorption area (see equation 1.28).

Lesson time	Ventilation noise (dB)	Student activity (dB)		Speech level (dB)	Instructor sound power (dB)
		(a)	(b)		
09.00 - 11.00	40.3	45.1	41.9	53.2	67.1
11.00 - 13.00	38.8	43.7	40.6	52.4	66.4
13.00 - 15.00	40.3	45.4	41.9	55.8	69.7
15.00 - 17.00	42.3	47.2	44.0	54.3	68.1

Table 7: Measured A-weighted equivalent sound level of background noise and STIPA values in function of position, the corresponding source-to-receiver distance and signal-to-noise ratio SNR for Aula III using public address system as source. Receiver positions are the same used for the measurement campaign (see figure 2.3).

Receiver	Source-receiver distance (m)	$L_{eq,A}$ (dB)	2.2<SNR<4.3 (+dB)	STIPA	9.3<SNR<12.4 (+dB)	STIPA	18.6<SNR<22.1 (+dB)	STIPA
1	5.89	47.3	3.5	0.25	11.7	0.42	21.6	0.49
2	4.47	47.6	3.5	0.28	11.5	0.45	21.1	0.51
3	4.54	47.0	3.8	0.28	12.1	0.47	21.8	0.52
4	5.20	48.5	3.5	0.26	11.4	0.45	21.1	0.50
5	6.64	47.0	3.7	0.27	11.9	0.46	21.6	0.53
6	5.61	46.6	4.3	0.27	12.4	0.45	22.1	0.49
7	5.67	47.2	3.7	0.29	11.6	0.45	21.2	0.52
8	6.29	47.9	3.3	0.30	11.3	0.50	20.5	0.54
9	7.62	47.7	4.0	0.28	11.5	0.49	21.3	0.53
10	6.84	47.4	4.0	0.34	12.0	0.53	21.7	0.56
11	6.88	47.2	3.7	0.31	11.5	0.49	21.4	0.57
12	7.21	48.2	3.3	0.33	11.4	0.50	21.1	0.57
17	9.78	47.6	3.7	0.34	11.6	0.52	21.4	0.57
18	9.52	48.1	2.9	0.36	10.8	0.55	20.5	0.59
19	9.55	48.3	2.6	0.33	10.0	0.52	19.8	0.59
20	9.49	48.6	3.0	0.31	10.0	0.49	19.7	0.56
24	10.76	48.3	3.2	0.34	11.0	0.51	21.0	0.60
25	12.33	47.8	3.3	0.28	10.7	0.48	20.4	0.55
26	12.33	47.6	2.7	0.22	9.9	0.43	19.4	0.51
27	12.36	46.8	3.7	0.27	11.1	0.46	20.6	0.55
28	13.76	47.2	3.0	0.25	9.9	0.44	19.5	0.51
29	13.61	47.5	2.2	0.22	9.3	0.39	18.6	0.50
30	13.63	46.8	3.1	0.21	10.0	0.42	19.7	0.50
31	12.48	46.9	3.1	0.21	10.3	0.40	19.8	0.50

Table 8: Measured A-weighted equivalent sound level of background noise and STIPA values in function of position, the corresponding source-to-receiver distance and signal-to-noise ratio SNR for Aula V using public address system as source. Receiver positions are the same used for the measurement campaign (see figure 2.3).

Receiver	Source-receiver distance (m)	$L_{eq,A}$ (dB)	$2.7 < \mathrm{SNR} < 4.1$ (+dB)	STIPA	$10.0 < \mathrm{SNR} < 12.1$ (+dB)	STIPA	$19.7 < \mathrm{SNR} < 21.8$ (+dB)	STIPA	$29.7 < \mathrm{SNR} < 31.9$ (+dB)	STIPA
3	4.03	38.7	4.1	0.20	12.1	0.39	21.8	0.49	31.9	0.50
7	5.14	39.3	2.7	0.18	10.0	0.35	19.7	0.48	29.7	0.48
11	6.41	39.5	3.1	0.23	10.7	0.42	20.2	0.52	30.4	0.53
15	7.75	38.8	4.5	0.27	12.7	0.47	22.6	0.59	32.6	0.60
19	9.15	38.5	3.7	0.25	11.5	0.46	21.1	0.59	31.2	0.57
23	10.58	39.2	3.3	0.24	10.7	0.43	20.4	0.56	31.5	0.56
26	12.02	39.6	3.5	0.20	10.1	0.41	19.8	0.53	29.7	0.55

Table 9: Measured A-weighted equivalent sound level and STIPA values in function of position and the corresponding source-to-receiver distance using public address system as source. Since the system was turned off in the time of measures these values don't consider the SNR (signal-to-noise ratio) but a first level, which is a typical received speech level in the room, considered as 0 and a level augmented of +10 dB. Receiver positions are the same used for the measurement campaign (see figure 2.3).

Aula VI					
Receiver	Source-receiver distance (m)	0		+10 dB	
		$L_{eq,A}$ (dB)	STIPA	L_{Aeq} (dB)	STIPA
2	3.61	66.1	0.48	76.7	0.47
5	6.87	66.5	0.47	76.5	0.49
8	10.20	65.5	0.46	75.8	0.47
11	13.55	65.5	0.49	75.4	0.51
14	16.91	64.9	0.47	74.9	0.52

Table 10: Aula III: comparison between measured *STI* values ante-operam and post-operam after the installation of Syva loudspeakers. The negative values of the first receivers, corresponding to the first rows of the audience area, are due to the directivity of the source. The difference between values are indicated as *delta*.

Receiver	STI		Delta
	Ante-operam	Post-operam	
1	0.56	0.49	-0.07
2	0.55	0.52	-0.03
3	0.51	0.51	0.00
4	0.51	0.50	-0.01
5	0.53	0.48	-0.05
6	0.51	0.51	0.00
7	0.50	0.50	0.00
8	0.51	0.53	0.02
9	0.51	0.53	0.02
10	0.50	0.57	0.08
11	0.50	0.57	0.08
12	0.50	0.56	0.06
13	0.49	0.58	0.09
14	0.48	0.62	0.14
15	0.47	0.60	0.13
16	0.49	0.59	0.10
17	0.49	0.61	0.13
18	0.47	0.60	0.13
19	0.47	0.60	0.13
20	0.49	0.58	0.09
21	0.47	0.58	0.11
22	0.47	0.57	0.10
23	0.47	0.57	0.10
24	0.48	0.58	0.10
25	0.45	0.53	0.08
26	0.45	0.53	0.08
27	0.47	0.54	0.07
28	0.46	0.51	0.05
29	0.45	0.50	0.05
30	0.45	0.50	0.05
31	0.46	0.50	0.04

Table 11: Aula V: comparison between measured STI values ante-operam and post-operam after the installation of Syva loudspeakers. The negative values of the first receivers, corresponding to the first rows of the audience area, are due to the directivity of the source. The difference between values are indicated as *delta*.

Receiver	STI		Delta
	Ante-operam	Post-operam	
1	0.52	0.47	-0.05
2	0.52	0.48	-0.04
3	0.48	0.49	0.02
4	0.47	0.48	0.01
5	0.51	0.48	-0.03
6	0.49	0.51	0.03
7	0.46	0.50	0.05
8	0.48	0.52	0.04
9	0.48	0.53	0.05
10	0.45	0.55	0.10
11	0.46	0.55	0.10
12	0.47	0.56	0.10
13	0.46	0.57	0.11
14	0.44	0.60	0.16
15	0.45	0.59	0.14
16	0.47	0.61	0.14
17	0.45	0.59	0.14
18	0.45	0.59	0.14
19	0.46	0.59	0.13
20	0.47	0.58	0.12
21	0.46	0.57	0.11
22	0.46	0.55	0.09
23	0.47	0.56	0.09
24	0.45	0.54	0.09
25	0.47	0.54	0.07
26	0.48	0.54	0.06

Table 12: Aula VI: comparison between measured *STI* values ante-operam and post-operam after the installation of Syva loudspeakers. The negative values of the first receivers, corresponding to the first rows of the audience area, are due to the directivity of the source. The difference between values are indicated as *delta*.

Receiver	STI		Delta
	Ante-operam	Post-operam	
1	0.56	0.48	-0.08
2	0.51	0.48	-0.03
3	0.49	0.49	0.00
4	0.47	0.50	0.04
5	0.44	0.47	0.04
6	0.45	0.49	0.04
7	0.41	0.48	0.08
8	0.41	0.47	0.06
9	0.41	0.48	0.07
10	0.39	0.48	0.09
11	0.39	0.50	0.11
12	0.39	0.48	0.09
13	0.41	0.51	0.11
14	0.40	0.51	0.11
15	0.38	0.48	0.10