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Validation of predictive algorithm control of gatherings in enclosed places: simulation in the environment Mensa Circoop Polytechnic University of Turin

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0 | INTRODUCTION

During the first and second wave of the Covid-19 pandemic, it was hypothesized that the transmissibility of the virus increased significantly indoors where it is not possible to guarantee a correct and frequent ventilation, natural or forced, or in cases where it is not possible to maintain an interpersonal distance of at least 1-1.5 meters. Starting from these considerations, in Italy the government has proceeded to contain the Covid-19 pandemic by implementing numerous decrees including the D.P.C.M. of 26 October 2020 n. 265 called "*urgent measures to contain contagion throughout the national territory*" which establish the main restrictions and guidelines to follow in environments considered with higher contagious risk, among which the most interesting for the present thesis are catering environments such as school /university canteens. In the acoustic field, most of the academic literature has been based on investigating how to improve anthropogenic noise levels $L_{N,A}$ produced in rooms with different intended use, schools, canteens, auditoriums, etc., adopting instruments for noise control, as in the case of the SEM (speech and sound SEMaphore) designed by the DENERG of Polytechnic of Turin and used in school environments where it was possible to limit the noise levels L_{90} through visual feedback, or even proposing prediction models to calculate levels of $L_{N,A}$ expected starting from known physical characteristics of the analyzed environments, such as volume, reverberation time t_0 of empty environment, maximum occupancy N_{max} to ensure good SNR signal to noise ratio, and so on. The aim of this master's thesis is to present a prediction algorithm, which, starting from the anthropic noise levels $L_{N,A}$ measured by sound level meters, is able to return a reliable value of N=number of people present in the analyzed closed environment. To validate the prediction algorithm, it was necessary to monitor in 4 days, 25-26 November 2020 and 14-16 April 2021, real levels of anthropic noise $L_{N,A}$ generated inside the CIRCOOP Canteen of Polytechnic of Turin and simulate the environment through acoustic software Odeon version 16 Combined and Grasshopper application of Rhinoceros 7, to obtain some parameters required for a correct application of prediction algorithm. Starting from results, it was possible to underline a significant correlation between simulated data of prediction algorithm and the estimated ones during the measurement campaign.

1 | ANTHROPIC NOISE: WHAT IS AND HOW TO CHECK IT

Anthropic noise is the type of noise generated by talking people in an open or closed environment. Due to the difficulty of reproduction of noises of anthropogenic origin, which is a consequence of different variables and for this reason it doesn't follow a schematic sequence, it is necessary to refer to empirical models that attempt to assess the impact of noise generated by a variable number of people on total noise present in an environment. Those empirical models include:

- Hayne model, (Hayne M.J. et al., 2011) usefull in contexts where there are a range of 10-100 people and in which are summarizes all the factors that affect anthropogenic noise levels $L_{N,A}$ generated by a crowd (Figure 1):

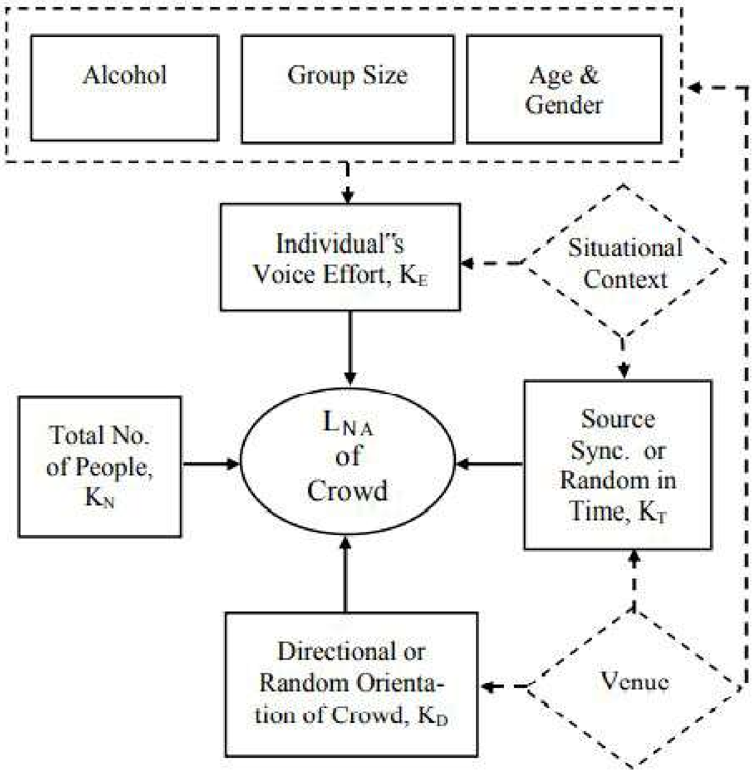


Figure 1 - Hayne M.J. et al., 2011

- model of Gallo and Shtrepi (Gallo E. et al., 2019) applicable in spaces with a range of 50-2000 people, in which it was possible to correlate the number of people present in a Turin's square, monitoring flows through the use of a WIFI scanner of MAC address that anonymously calculated the present number of devices, mobile phones and/or other wearable devices and correlate them to levels of $L_{N,A}$ measured from a class 1 sound level meters.

Depending on the nature of the noise, that weighs on anthropogenic noise levels $L_{N,A,}$ there are numerous ways to mitigate these noise sources, including adaptive user behavior. For example, in the most trivial cases, if the source of noise comes from outside, an adaptive behavior could be simply to close windows, or, if the noise starts from an internal source, it can be mitigated by reducing the volume of a sound playback device, and so on. In the wake of adaptive and/or proactive behaviour, Polytechnic of Turin designed a device called SEM (speech and sound SEMaphore) based on a LED feeedback system (Figure 2).

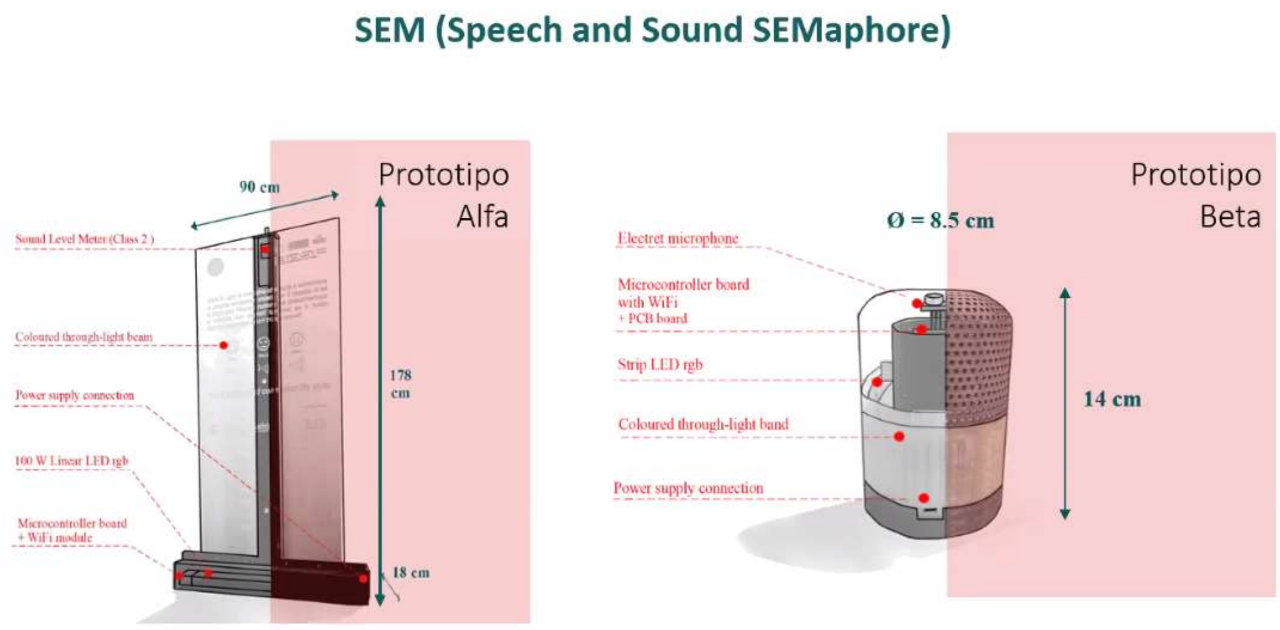


Figure 2 - Gallo E. et al., 2019

The SEM device is based on an adaptive algorithm that measures anthropogenic noise levels present in an close environment and responds by emitting LED signals, that change from green to red in case noise levels instantly measured is too higher than the values measured previously. For this reason, SEM is not based on a limit value that is pre-set before the device is switched on but works in adaptive mode so that limit value is resets every time automatically. The SEM experimentation took place in 13 primary schools in the city of Turin (Di Blasio S. et al., 2019) and in 2 phases; during the first phase the SEM was only placed in classes but turned off and only in second phase the device was really activated and students were actively involved in the simulation, explaining the objective of research and methodologies applied to improve the proactive behaviors of students. This motivational approach was conducted by showing graphs from day to day, where the research team registered improvements/deterioration achieved by each class in terms of daily noise levels.

1. 1 | LOMBARD EFFECT AND INDOORS ACOUSTIC QUALITY

Levels of speaker's vocal effort in a closed environment depends on multiple factors, including the physical characteristics of rooms, such as Volume, Room Cavity Ratio (Battaglia Paul L., 2018), the A_{tot} global absorption area and also factors of anthropogenic origin, among which the most influential are levels of noise generated by $L_{N,A}$ speakers and the intended use of the analyzed environment, such as a restaurant, a library etc. For this reason, when the level of anthropic noise in the environment increases, we tend to speak with greater vocal effort to ensure a good signal-to-noise ratio (SNR) between the speaker and his interlocutor. In general, speakers adapts the intensity of their voices in proportion to how they perceives the anthropogenic noise level $L_{N,A}$ surrounding them; for example, in presence of a noise of 50 dB(A), a normo-hearing speaker usually increases the intensity of his voice from 3-6 dB for each increment of 10 dB in noise levels that could mask the initial verbal message. This self-regulation, whether voluntary or involuntary (Pick Herbert L et al, 1989), is known as Lombard Effect and creates a vicious circle of noise pollution so that by increasing one's vocal effort, in order to overpower anthropic noise, it helps to make the $L_{N,A}$ worse. The Lombard effect phenomenon was first described in 1909 by Étienne Lombard (Lombard E., 1911), a French "otorino-laringoiatra" who studied how an interlocutor regulates his vocalization with audio-vocal reflexes to ensure good intelligibility responding to auditory and/or visual feedback. In this regard, we called "public loop" when, for example, a speaker tends to regulate his vocalization by observing how the audience reacts to his speech (Lane H. et al., 1971). Since Étienne Lombard's studies, it has generally been felt that the increase in the level of vocal exertion of a speaker in a noisy environment follows a linear and increasing trend, until we reach the maximum level of vocal effort that a human being is able to produce (Pickett J.M., 1958). In ISO 9921:2003 there's a description of how the vocal effort increases compared to succeeding increments of 6 dB in SPL(A) calculated one meter from the mouth of the speaker, from now on referred as $L_{S,A,1m}$. In ISO 9921:2003, a normal vocal effort is associated to a $L_{S,A,1m}$ of about 60 dB(A) (Figure 3), so it's easy to understand that a speech held in an environment with an anthropic noise level of about 75 dB can be difficult to understand if interlocutor's vocal effort remains unchanged.

$L_{S,A,\ 1m}$ dB	Vocal effort
54	Relaxed
60	Normal
66	Raised
72	Loud
78	Very loud

Figure 3 - ISO 9921:2003

It is also necessary to introduce the Gardner effect, according to which doubling the number of people in closed environment the corresponding ambient noise levels increases by 6 [dB] instead of 3 [dB]; this relationship was initially demonstrated by Gardner (Gardner M.B., 1971) and was implemented by Rindel (Rindel J.H., 2012) in a prediction equation by which it is possible to calculate the anthropic noise levels $L_{N,A}$ as a function of the total absorption area and number of people speaking simultaneously:

$$L_{N,A} = 93 - 20 \times \log\left(\frac{A_{tot}}{N_s}\right) \quad [dB(A)] \quad (1)$$

where:

- A_{tot}=area of absorption of the environment [m^2];
- N_s=the number of people speaking simultaneously [-].

However, generally it is only possible to know the N= total number of people present in a closed environment and certainly not how many of them will speak simultaneously; for this reason, we could introduce the value group size, g=N/Ns that represent the average value of people speaking simultaneously within a group. Replacing value g=group size in formula (1) we find out:

$$L_{N,A} = 93 - 20 \times \log\left(\frac{A_{tot} \times g}{N}\right) \quad [dB(A)] \quad (2)$$

Assuming that the total absorption area in a closed environment is given by:

$$A_{tot} = \frac{0,16 \times V}{t_0} + ap \times N\ [m^2] \quad (3)$$

where:

- t_0=reverberation time [s] of the unoccupied but furnished environment at middle frequencies of 500-1000 [Hz];
- ap=absorption per capita, depends on the type of clothing, winter or summer season, and also from the position of users, whether upright or seated, generally takes values between 0.2 – 0.5 [m^2].

We can combine the (2) and (3) formula to obtain the predictive equation by which it is possible to calculate the anthropogenic noise levels $L_{N,A}$:

$$L_{N,A} = 93 - 20 \times \log\left[\left(\frac{0,16 \times V}{t_0 \times N} + ap\right) \times g\right]\ \ [dB(A)] \quad (4)$$

An interesting consequence of this equation is that noise level in the environment increases by 6 [dB] instead of 3 [dB], for each doubling in the number of present, since the vocal effort as the anthropic noise level $L_{N,A}$ increases, doesn't remain constant, but also increases due to the Lombard effect (Figure 4).

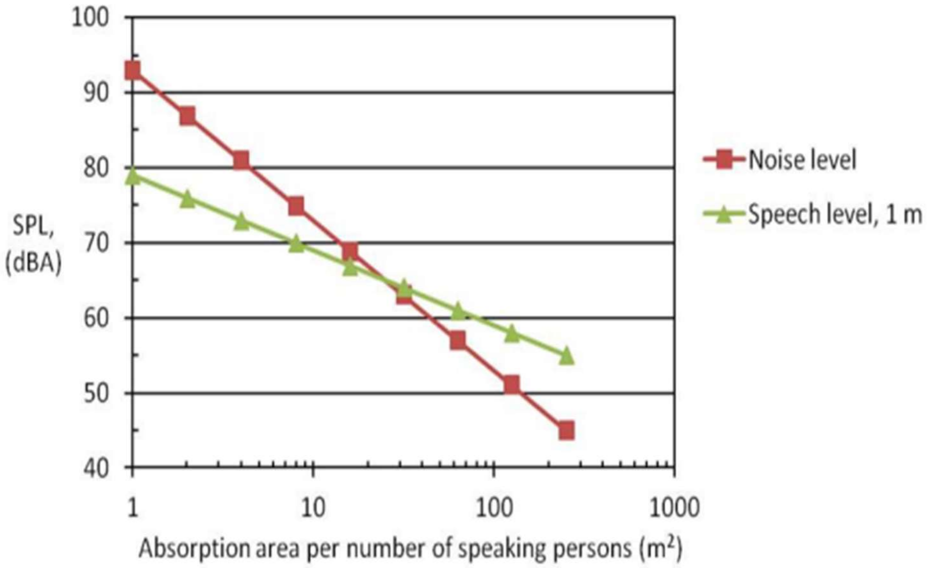


Figure 4 – Rindel J.H., 2012

Using this first predictive model it is therefore possible to calculate the expected anthropic noise level $L_{N,A}$ starting from known parameters, such as volume of the environment, reverberation time t_0 of furnished but unoccupied rooms and total number of seats or alternatively considering the parameter N_{max}=maximum number of people recommended to ensure a good SNR in the environment. Uncertainty remains over the value g= group size, since it is not possible to establish “a priori” the relationship between total number of people and number of people who will speak simultaneously; however, in models studied by Gardner it has prevailed that in most restaurants g=group size takes values between 3-4 [-] so it is recommended to use an average value of g=3.5 and, in order to obtain reliable data, it is appropriate to refer to environments in which the number of occupants is at least 50 people. Although these guidelines apply to normo-auditory people, ISO 9921 [4, section 5.1] states that "people with mild hearing disorders, in general the elderly or listeners not belonging to the same linguistic ethnicity, need a higher signal-to-noise ratio of about +3 dB". Consequently, in these cases, an SNR> 0 dB must be achieved to ensure conditions defined as "sufficient" for this category of persons, whereas in ordinary cases an SNR=0 is a value defined as "satisfactory" (Figure 5).

Quality of verbal communication	**SNR dB**
Very bad	< –9
Insufficient	(–9; –3)
Sufficient	(–3; 0)
Satisfactory	(0; 3)
Good	(3; 9)
Very good	> 9

Figure 5 – Lazarus, 1986

The N_{max} parameter mentioned above was introduced by Rindel (Rindel J.H., 2012) and is used to calculate maximum number of people that can be accommodated in a closed environment, in order to ensure good acoustic quality and a correct absorption coefficient A_{tot} operated by the environment itself; N_{max} parameter is estimated in a room where Sabine's hypotheses on the reverberating acoustic field are valid, starting from volume of the environment and reverberation time t_0 according to the following equation:

$$N_{max} = \frac{V}{20 \times t_0} \; [-] \quad (5)$$

where:

- V= volume [m^3];
- t_0= reverberation time [s] of unoccupied furnished environment at middle frequencies of 500-1000 [Hz].

Replacing the value t_0 with Sabine equation for reverberation time, we will have:

$$t_0 = \frac{0{,}16 \times V}{A_{tot}} \; [s] \quad (6)$$

By combining the formula (5) and (6) we could find that:

$$N_{max} = 0{,}3 \times A_{tot} \; [-] \quad (7)$$

In this study (Rindel J.H., 2012) the calculation continues by relating recommended number of customers defined C= acoustic capacity to maximum recommended number of people N_{max} to ensure a good SNR in the analyzed environment; considering that a ratio of C/N_{max}=1 [-] is sufficient to ensure a good quality of the conversation, then it is possible to reverse the C/N_{max} value by introducing the parameter Q= acoustic quality, calculated as:

$$Q = \frac{N_{max}}{C} = 0{,}3 \times \frac{A_{tot}}{C} \; [-] \quad (8)$$

In this equation for estimation of acoustic quality is possible to note that the parameter Q=acoustic quality is directly proportional to total amount of sound absorption carried out by the analyzed environment and inversely proportional to total number of occupants. This A_{tot}/C ratio corresponds to the parameter mentioned at the beginning of this chapter called ap=absorption per capita, which was first evaluated by J. H. Rindel considering it a strongly influential parameter that must be taken into account in predictive models used for a correct estimation of anthropogenic noise levels $L_{N,A}$.

1.2 | CIRCLE THEORY VERSUS GROUP SIZE

In a not too noisy environment usually to ensure sufficient SNR (sufficient=0) between speaker and his interlocutor, a voice effort defined as normal (ISO 9921-1) corresponding to an $L_{S,A,1m}$ of 65 dB(A) is used; if number of N_s speaking person within the same environment increases, the resulting sound pressure level in the diffuse field will be equal to:

$$L_{N,A} = L_{S,A,1m} + 10\log\left[4 \times \left(\frac{N_s}{A_{tot}}\right)\right] \quad [dB(A)] \quad (1)$$

where:

- A_{tot}= area of absorption of the environment [m^2];
- N_s= number of people speaking simultaneously [-].

However, as E. Ruiter (Ruiter E. Ph. J. 2015) explains, the validity of this formula is limited to relatively low range of $L_{N.A}$, while in reality noise levels also increases not proportionally to the growth of users in a closed environment, since a generic speaker tends to increase his vocal effort in the presence of background noise in order to ensure a good understanding of the speech with his interlocutor and vice versa. To understand this mechanism, E. Ruiter compared the noise levels of a restaurant where users occupy fixed positions, tables, with noise levels generated during a cocktail party where users can move freely within the environment. During a cocktail party the guests fill the environment little by little and tend to form small circles to converse with each other; let's assume that the distance between users within the same circle is constant, for example 1 meter, distance also considered by Rindel (Rindel J.H. 2010) in its predictive model of anthropogenic noise levels in environments characterized by diffuse field. For smaller groups, less than 4 people, the distance between a speaker and an interlocutor belonging to the same circle will be about 0.5-1 [m]. However, if the number of participants in the same conversation circle increases, it is easy to understand that the perimeter and consequently the diameter of circles will also increase until the distance between i-th speaker and j-th listener is such as to prevent the intelligibility of the word. The maximum distance D_{max} between two members of same circle, as Ruiter explains in his research (Ruiter E. 2011) corresponds to circle's diameter which can be calculated as:

$$D_{max} = \frac{N \times d}{\pi} \quad [-] \quad (2)$$

where:

- N= number of persons belonging to the same circle [-];
- d= interpersonal distance [m].

In circle theory it is hypothesized that only one person at a time speaks within the same conversation and if the intelligibility of the word is no longer guaranteed due to exceeding the maximum distance between speaker and diametrically opposite interlocutor, at this point the circle will break and speakers will form n+1 circles until the intelligibility of the word is sufficient (SNR=0) or until there are only 2 persons circles From these considerations, we can assume that the noise level in the environment $L_{N,A}$ generated during each interaction between the circles of cocktail party participants depends closely on the Lombard effect, that is the slope c between the anthropic noise level present in a closed environment $L_{N,A}$ and the sound pressure level calculated 1 meter from the mouth of the speaker $L_{S,A,1m,}$ and also that the Lombard effect is a function of the number of people present and the amount of sound absorption. This function of two variables is defined as Lombard 2-function (Figure 6).

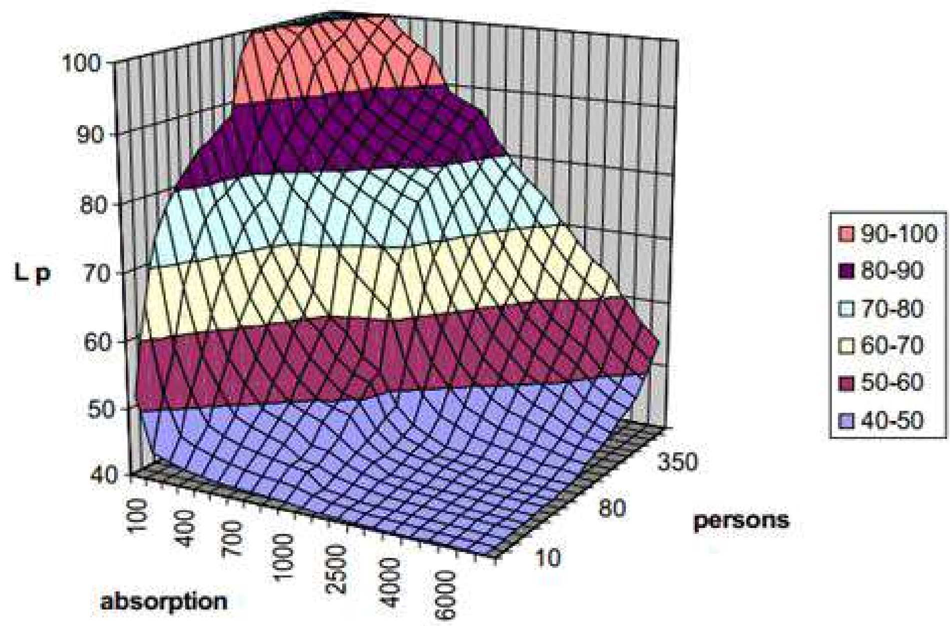


Figure 6 - Ruiter E. Ph. J., 2011

However, Lombard-2 function can be reduced to a function of only one variable, Lombard 1 function, by introducing the value $ap=A_{tot}/N$ i.e. absorption per capita (Figure 7).

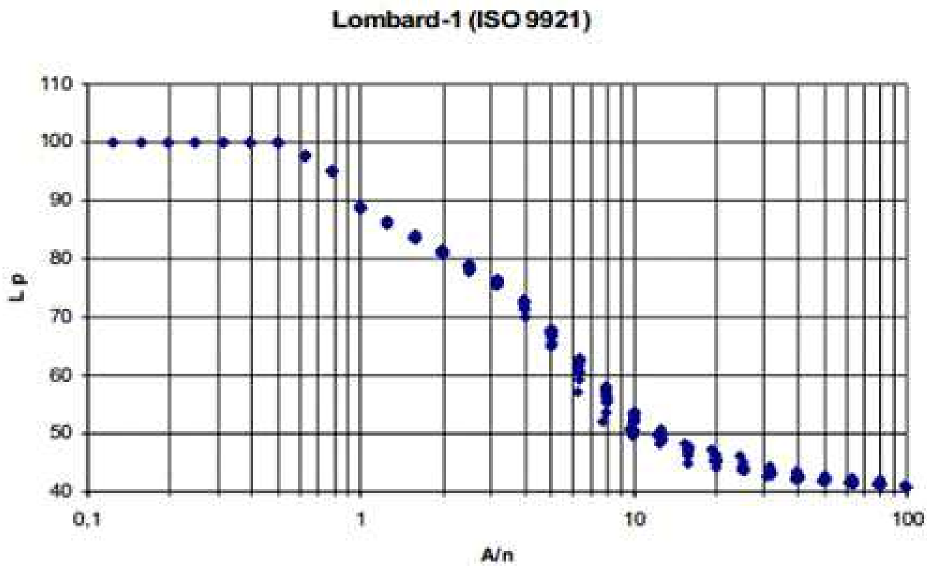


Figure 7 - Ruiter E. Ph. J., 2011

As is possible to see, for each value of $L_{N,A}$ there may be multiple corresponding values of A_{tot}/N_s due to the large statistical combination between these two values; for example, a restaurant may not always be at its maximum capacity and not all diners tend to speak in unison due to chewing. However, Lombard-1 function follows an observable trend. Typically, for values of A_{tot}/N_s <10 m^2 the slope "c" of the Lombard-1 function is accentuated and steep, while for values $A_{tot}/$ N_s>10 m^2 tends to be flatten and slow down. Considering 60-65 dB(A) as the limit value beyond which noise tends to be generated in a closed environment, it's possible to say that a value of A_{tot}/N_s =5 m^2 is a sufficient indicator from which to start in early stages of design or redesign of enclosed spaces for catering, in order to ensure a decent amount of sound absorption per capita when the levels of $L_{N,A}$ exceed the threshold value 60-65 dB(A); according to ISO 9921 the optimum value of A_{tot}/N_s is about 7 m^2. Starting from these considerations, we can assume an environment characterized by a diffuse field, in which 3 Sabine's hypotheses are therefore valid, uniform acoustic energy, density at every point in the environment, validity of statistical acoustics and continuity hypotheses; in this kind of situations, noise levels will be a function of the percentage of ap=absorption per person. In his study, E. Ruiter analysed, in a London restaurant, the relationship between the number of diners, the amount of sound absorption per capite and L_{eq_dt} levels monitored by sound level meters. The following graph compares data obtained previously from Gardner's studies (Gardner M.B., 1971), Tang (Tang S.K. et al., 1997) and Navarro (Porto Nuens Navarro M. et al., 2007) in restaurant-like environments analyzed by E. Ruiter, such as canteens and food courts (Figure 8).

In the graph we note that the trends of $L_{N,A}$ calculated with circle theory exposed by Ruiter, i.e. the two thick black lines, follow a similar trend to values found in previous studies of Gardner, Tang (canteen), Navarro (food court J and L).

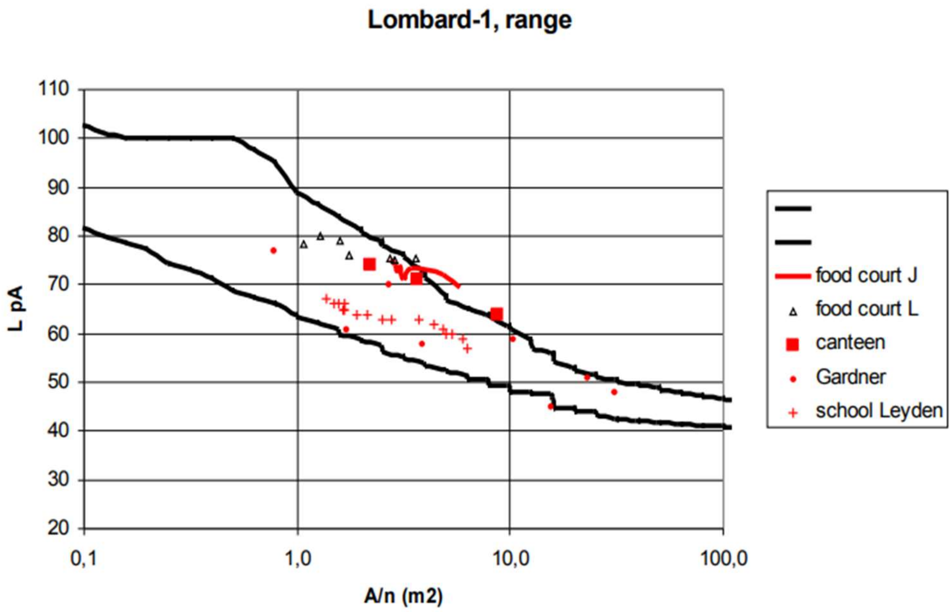


Figure 8 - Ruiter E. Ph. J., 2011

Rindel (Rindel J.H., 2010) in its predictive model for calculating $L_{N.A}$ levels in catering environments characterized by diffusive field, also assumes as a premise that diners want to guarantee a good SNR during a conversation and for this reason in its model considers:

- the Lombard slope c= 0.5 [dB/dB] to ensure a sufficient SNR;
- g=group size takes values between g=2-4 [-] in catering indoor rooms;

By adopting these values, the Lombard-1 function can be compared with model proposed by E. Ruiter (Figure 9), however a substantial difference between those two models is that in J. H. Rindel one's, g=group size is a parameter established a priori depending on the intended use of the analyzed environment, in case of catering rooms the most suitable values are g=2-4, while in circle theory the number of talking people is a variable within the model that as we have seen, depends on the levels of $L_{N,A}$ achieved in the environment as the number of speakers increases. This substantial difference is a consequence of the type of environment in which these predictive model is to be adopted, in fact in a restaurant diners tend to occupy fixed positions, tables and consequently tend to move less frequently from one circle to another; a diametrically opposite situation instead occurs when users attend a cocktail party and will be much more enticed to move within the environment to switch from "one circle to another" in order to maintain a conversation ensuring an optimal SNR.

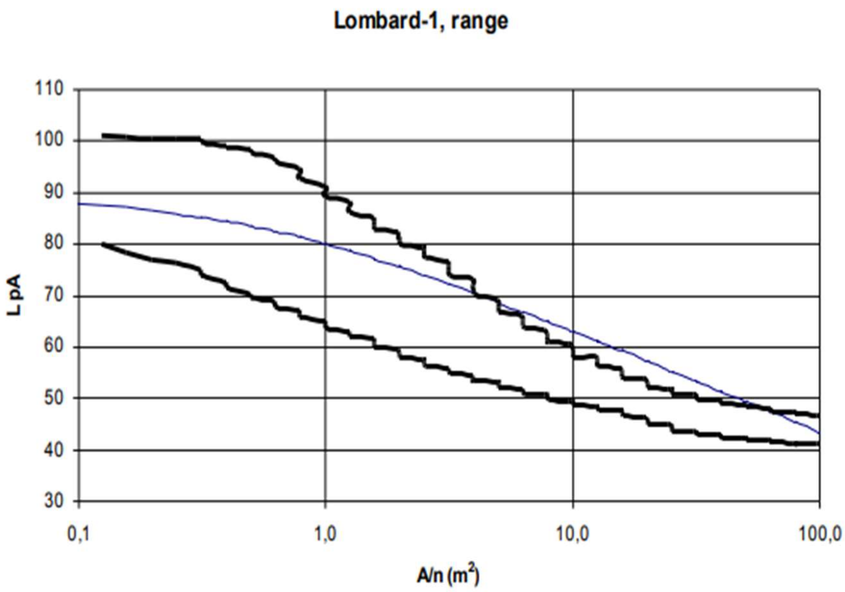


Figure 9 - Ruiter E. Ph. J., 2011

Another difference between the circle theory exposed by E. Ruiter and J.H. Rindel's predictive model is that it does not consider reverberation time as a variable that can affect the quality of the signal-to-noise ratio and consequently the intelligibility of the word. In fact, in circle theory the intelligibility of words is independent of reverberation time but is a factor that depends almost exclusively on the amount of $L_{N,A}$ achieved within catering environments. This simplification derives from the fact that catering environments tend to be characterized by medium-low ceilings but at the same time, built of material with high sound absorption power and consequently the reverberation time in such characterized environments is quite short, <1 [s].

2 | INTRODUCTION TO CALCULATION OF $L_{N,A}$ AND EVALUATION OF THE LOMBARD SLOPE

The ratio of the anthropic noise level generated in a closed environment $L_{N,A}$ to sound pressure level calculated 1 meter from the mouth of the speaker $L_{S,A,1m}$ is given by the slope "c" called Lombard slope, which according to Lazarus (Lazarus H., 1986) can take values between c=0.5-0.7 [dB/dB]. Lazarus has shown that the Lombard effect begins in an environment with a noise level of about 45 dB(A) and with a vocal effort of 55 dB. From these values we can derive the linear equation by which it is possible to calculate the expected level of $L_{S,A,1m}$ from a known level of anthropic noise $L_{N,A}$ using the formula:

$$L_{S,A,1m} = 55 + c \times \left(L_{N,A} - 45\right) \quad [dB(A)] \quad (1)$$

where:

- $L_{N,A}$= anthropogenic noise level measured in a closed environment [dB(A)].

However, as already described above, this equation is valid only for speech levels above 55 dB(A) or noise levels above 45 dB(A). In addition, we must remember that in acoustic free field the ratio of the sound power level $L_{W,A}$ and the sound pressure level $L_{S,A}$ produced by a speaker decreases as the distance r from the sound source increases. The sound pressure level $L_{S,A,r}$ produced by a speaker at a generic distance r in the free field can be calculated as:

$$L_{S,A,r} = L_{W,A} + 10 \log Q - 10 \log(4\pi r^2) \quad [dB(A)] \quad (2)$$

where:

- Q=directivity factor of the source [-];
- r= distance between the source i-th and j-th receiver [m].

The directivity factor is the ratio of the sound intensity in a specific direction the reference sound intensity I_0 that the acoustic field would assume if the sound source considered were omnidirectional. When the sound source is represented by a speaker, it is possible to assume a Q=2 in the case of sound propagation occurring frontally, i.e. in axis t to the speaker's mouth; the directivity factor considers that the source, in this case a speaker, emits sound waves in preferential directions than others, for example if a sound source is placed near an extremely reflective surface, the source will concentrate a lots of sound waves towards the free portion of space. For this reason, in certain directions it is possible to reach double the sound power that would be obtained if the source were placed in a totally free field, therefore without obstacles. Consequently, by choosing a Q=2 directivity factor, it’s possible to simulate a conversation that takes place between a speaker and an interlocutor sitting in a table, since it is considered that most of the sound waves propagated from the speaker's mouth are directed towards his interlocutor who is in a frontal position and in axis with the human sound source. In other situations, the Q=directivity factor, can take values >2 (Figure 10) but in any case, the sound pressure level will always decrease by 6 dB for each doubling of the distance r from the sound source considered. Alternatively, we can consider the D=directivity index calculable as:

$$D = 10 \log Q \quad [dB] \quad (3)$$

In addition, the source is not always in axis with respect to the receiver, so some sound waves will be attenuated in certain directions. By integrating the $L_{W,A}$ sound power levels produced by a speaker all over round corner, we will have that:

$$L_{W,A} = L_{S,A,1m} + D \quad [dB(A)] \quad (4)$$

where:

- $L_{S,A,1m}$ =sound pressure level of a speaker measured at 1 m in axis from his mouth and standardized by ISO 9921.

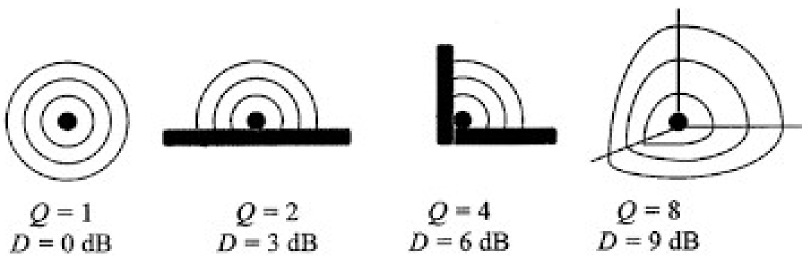


Figure 10 – Nicolini A.

2.1 | PREDICTIVE MODEL OF J.H. RINDEL IN ENVIRONMENTS CHARACTERIZED BY DIFFUSE ACOUSTIC FIELD

The predictive model developed by Rindel (Rindel J.H., 2010) considers that a conversation between an i-th speaker and j-th interlocutor in a catering environment, takes usually place at an interpersonal distance of about 1 [m]. Thus, assuming a directivity factor Q=2 and an interpersonal distance r=1 [m] the sound power level generated by a speaker can be calculated as:

$$L_{W,A} = L_{S,A,1m} + 8 \quad [dB(A)] \quad (1)$$

In a closed environment in which we find multiple active sound sources, ergo several people who speak simultaneously N_s each characterized by the same sound power level $L_{W,A,}$ it is possible to calculate the anthropic noise level $L_{N,A}$ expected through the formula:

$$L_{N,A} = L_{W,A,i} + 10logN_s - 10log\left(\frac{A_{tot}}{4}\right) \quad [dB(A)] \quad (2)$$

where:

- $L_{W,A,i}$ = sound power level of an i-th source [dB(A)];
- N_s= number of people speaking simultaneously [-];
- A_{tot}= total absorption area of the environment [m^2].

Reintroducing the value $L_{S,A,1m}$ as calculated previously according to ISO 9921 and combining it with (1)(2) formulas, it is possible to calculate the expected anthropic noise level $L_{N,A}$ generated by a series of simultaneously speakers N_s and how it affects the SNR=noise signal ratio that elapses between an i-th speaker and his j-th interlocutor, according to the formula:

$$L_{N,A} = \frac{1}{1-c} \times \left(69 - c \times 45 - 10\log\left(\frac{A_{tot}}{N_s}\right)\right) \quad [dB(A)] \quad (3)$$

Assuming a closed environment characterized by reverberating acoustic field, in which the 3 Sabine hypotheses are therefore valid, uniform acoustic density at every point in the environment, validity of statistical acoustics and continuity hypothesis, we can replace the value A_{tot} to make the equation a function of t_0=reverberation time and to value ap=absorption per capita, replacing A_{tot} as:

$$A_{tot} = \frac{0{,}16 \times V}{t_0} + ap \times N \quad [m^2] \quad (4)$$

where:

- V= volume [m^3];
- t_0= reverberation time [s] of furnished but unoccupied environment at middle frequencies of 500-1000 [Hz].
- ap=absorption per capita [m^2].

By combining formulas (3)(4) and reintroducing group size g=N/N_s i.e. the average value of people speaking simultaneously within a group, we will have that the expected level of anthropic noise $L_{N,A}$ in an environment characterized by reverberant acoustic field will be:

$$L_{N,A} = \frac{1}{1-c} \times \left(69 - c \times 45 - 10\log\left(g \times \left(\frac{0{,}16 \times V}{t_0 \times N} + ap\right)\right)\right) \quad [dB(A)] \quad (5)$$

From this predictive model it was possible to derive the value of N by extending the equation (5) as follows:

$$N = \frac{0{,}16 \times V}{t_0 \times \left\{10^{\left[\frac{69-45\times c - L_{N,A}\times(1-c)}{10} - \log_{10}(g)\right]} - ap\right\}} \quad [-] \quad (6)$$

This predictive model works good in catering environments with a reverberant acoustic field where the $L_{N,A}$ levels produced exceed the minimum value of 45 dB(A). The parameters to be input into the equation are mostly known parameters of the environment in which the forecast model is to be applied, such as volume, t_0= reverberation time and ap=absorption per capita that depends on the type of clothing. In this model, the only parameters not known a priori are the g=group size and the c=Lombard slope. Generally, the group size is the most complex parameter to estimate a priori, but we can hypothesize it by evaluating whether the analyzed environment is, very noisy and therefore "g" will take on rather low values, or in opposite situation if the environment is rather quiet, we can hypothesize a higher "g". However, J. H. Rindel analyzing the levels of $L_{N,A}$ recorded in previous studies, including Hodgson's studies (Hodgson M. et al., 2007) showed that the predictive model provides reliable results by assuming:

- c= 0.5 [dB/dB] to ensure sufficient SNR as ambient noise increases;
- g=group size, takes values between g=3-4 [-] in catering environment;
- absorption per capite ap= 0.2-0.5 [m^2], a value which is added to the α absorption coefficients of unoccupied chairs and strongly dependent on the type of clothing of diners, whether it is summer or winter one's and finally from their position, if erected or seated. Usually, diners tend to occupy seated positions in the dining rooms, a different reasoning is found in the case of cocktail parties where guests can move freely in the analyzed environment.

J. H. Rindel's model has also proven reliable when applied to environments with a different use from catering establishments, but where Sabine's hypotheses on the reverberating acoustic field were still valid. Pinho (Pinho P.G. et al., 2018) assessed some students' exposure to noise during lunchtime in a school cafeteria, where the team monitored the A-weighted equivalent continuous sound level. All measurements were made with a sound level meter (Type 1) and the J. H. Rindel predictive model was applied to compare the data measured by the sound level meters with those predicted by the model. The comparison showed that there is more correspondence by setting as input parameters an absorption per capita ap= 0.25 [m^2], a Lombard slope c= 0.4 [dB/dB] and a group size g= 2 [-] or a Lombard slope c= 0.5 [dB/dB] and g= 4 [-] (Figure 11).

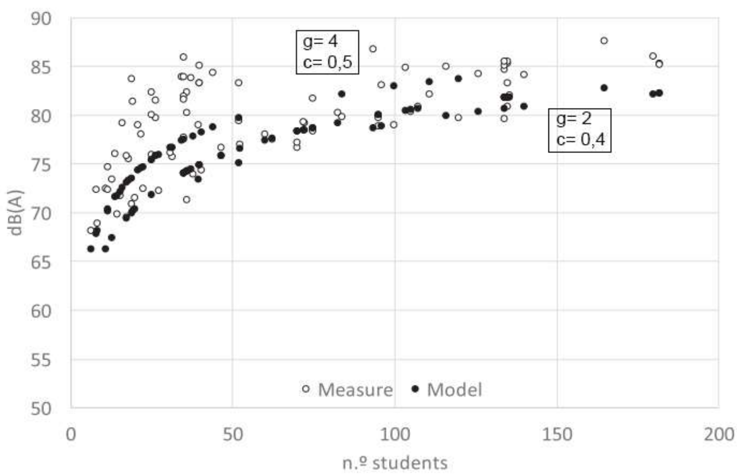


Figura 11 - Pinho P.G. et al., 2018

2.2 | PREDICTIVE MODEL OF D. D. D'ORAZIO IN ENVIRONMENTS CHARACTERIZED BY NON DIFFUSE ACOUSTIC FIELD

The predictive models to control of anthropic noise $L_{N,A}$ set out above refer to environments in which the Sabine hypotheses of the reverberating acoustic field are valid; this condition is not always verifiable especially if environments is full of screens or display, e.g. in museums or offices, which involve a attenuation of sound waves emitted from a generic source or even, in environments whose geometry causes sound waves to follow preferential directions during diffusion, e.g. industrial environments characterized by very large spaces in plants and machinery that hinder the propagation of sound waves. For this reason, D. D'Orazio in an article being published entitled "A predictive formulation of human noise in a non-diffuse environment" implemented the prediction model proposed by J. H. Rindel (Rindel J.H. 2010) monitoring the levels of $L_{N,A}$ produced within a food court, in which the ceiling height is far lower than the extension in plan. In this type of environment, the hypotheses of the diffusive Sabine field are not always verified so it is advisable to introduce a new parameter, D_2S= rate of spatial decay that replaces t_0= reverberation time of the furnished but unoccupied environment, the ap=absorption per capita [m^2] and A_{tot}= total area of absorption [m^2]. These parameters set out in Rindel's predictive model are valid in an environment characterized by diffuse acoustic field, instead the D_2S is a parameter that describes how the sound energy emitted by an i-th source decreases as the distance from the a forementioned source increases. According to ISO 14257, the attenuation of sound emitted by a generic source can be calculated through the parameter D_2S= rate of spatial decay, which represents the slope of the spatial distribution curve of a sound emitted from the source and calculated over a range of distances predetermined by the legislation. For this reason, in an environment characterized by a non-diffuse acoustic field and where there are more active sounds, the corresponding anthropic noise level $L_{N,A}$ will be:

$$L_{N,A} = 10\log\left(\sum_{i=1}^{N_s} 10^{(L_{W,A}-A_i)/10}\right) \quad [dB(A)] \quad (1)$$

where:

- $L_{W,A}$ = sound power level of the source [dB(A)];
- A_i= sound attenuation between the same source and the j-th receiver [dB];

The value A_i = sound attenuation, can be calculated as:

$$A_i = DL_2 \times 10 \log_2(r_i) \quad [dB] \quad (2)$$

where:

- r_i= mutual distance between an i-th source and a j-th receiver [m].

Even in this predictive model it must be considered that the human voice is strongly directive and that the source is not always in axis with respect to the receiver, so it is good to reintroduce the D= directivity index again. By integrating the $L_{W,A}$ sound power levels produced by a speaker all over the round corner, we have:

$$L_{W,A} = L_{S,A,1m} + D \quad [dB(A)] \quad (3)$$

where:

- $L_{S,A,1m}$= the sound pressure level of a speaker measured in axis at 1 m from his mouth as described in ISO 9921.

Remembering that the directivity index is D=10log(Q) and to simulate a conversation that takes place between a speaker and an interlocutor sitting at a table we can assumes a Q=2 [dB], a value also adopted by J.H. Rindel in his diffuse field forecasting model, we will have that a directivity factor Q=2 [dB] will be matched by a directivity index D=3 [dB]. Finally, it should be emphasized that in an environment, whether characterized by a diffuse or non-diffuse acoustic field, anthropogenic sound sources, speakers, tend to increase their vocal effort $L_{S,A,1m}$ as anthropogenic noise levels $L_{N,A}$ increase, in order to ensure a good signal-to-noise ratio. For this reason, it is necessary to reintroduce the linear equation by which it is possible to calculate the expected level of $L_{S,A,1m}$ from a known level of anthropic noise $L_{N,A}$ using the formula:

$$L_{S,A,1m} = 55 + c \times (L_{N,A} - 45) \quad [dB(A)] \quad (4)$$

Combining (1) (3) (3) (4) we will have that in an environment characterized by non-diffuse acoustic field, the expected anthropic noise levels will be:

$$L_{N,A} = \frac{1}{1-c} \times \left(55 - c \times 45 + 10\log\left(\frac{N}{g}\right) - 3.32 \times DL_2 \times r_i\right) \quad [dB(A)] \quad (5)$$

where:

- N= number of people in environment [-];
- $g=N/N_s$ is the average value of people speaking simultaneously within same groups;
- r_i= mutual distance between an i-th source and a j-th receiver [m].

Again, the value N= number of people could be extrapolated from the equation (5) as follows:

$$N = 10^{\left[\frac{L_{N,A} \times (1-c) - 55 + 4 \ \times c + 3,32 \times D_{2S} \times \log(r_{ij})}{10} + \log(g)\right]} \quad [-] \quad (6)$$

This predictive model works optimally in environments with a non-diffuse acoustic field where the levels of $L_{N,A}$ produced exceed the minimum value of 45 dB(A). The parameters to insert in input into the equation are mostly parameters that can be derived depending on the environment in which to adopt the model. The D_2S can be calculated by following ISO 14257 while the mutual distance between an i-th source and a j-th receiver was simulated by D. D'Orazio in Grasshopper for Rhino 6 through an algorithm that generates a series of random points, representing diners, randomly distributed within a useful portion of walkable space in the environment A. For each source-receiver pair, the algorithm calculates the mutual distance between pairs of points as the number of N diners increases within the simulated environment, until it reaches the actual occupancy rate estimated during the measurement campaign. The only parameters not known a priori are again the g=group size and the c= Lombard slope.

3 | MENSA CIRCOOP POLYTECHNIC OF TURIN

The environment in which the anthropic noise levels $L_{N,A}$ have been monitored to allow the validation of forecasting algorithms for the control of gatherings indoors, is the CIRCOOP Canteen of the Polytechnic of Turin (Figure 12), located beetwen Via Enrico D'Ovidio and Corso Duca degli Abruzzi. The canteen where sound levels meters were installed during the measurement campaigns, has a usefull area that extends longitudinally for 25.3 m and transversely for 18.3 m, with an average height of about 2.80 m, reporting a total volume of about V= 1115 m^3 and can be divided into two rooms bordering each other: the main room (Figure 13) is accessible just past the self-service and payment area (Figure 14), has a longitudinal extension of 25.3 meters and a transversal of 13.1 m, while the second room, here referred to as "coffee room" (Figure 15) has a longitudinal extension of 14.1 m and a transversal size of about 5.6 m and inside we can find the coffee area and two emergency exits. Based on the geometry, it was considered that two sound level meters were installed, one for each environment characterizing the CIRCOOP canteen, so that significant levels of $L_{N.A}$ could be collected in both rooms for a predetermined duration of about 2 hours during lunchtime.

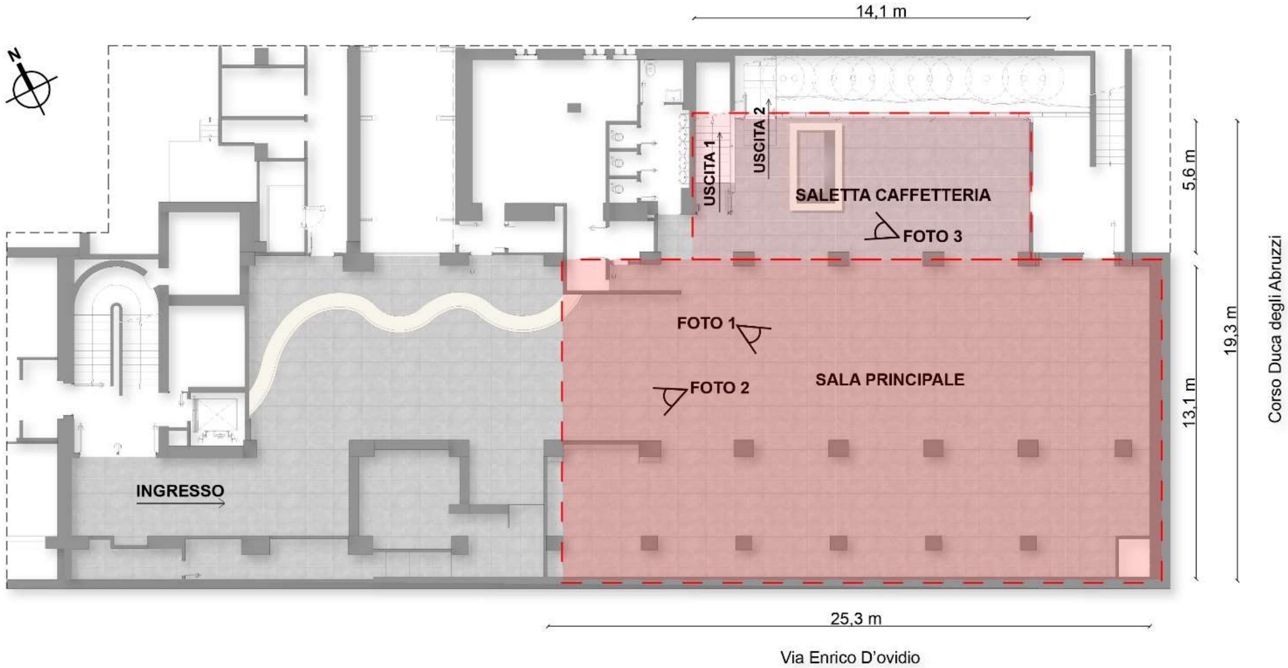


Figure 12 – Canteen floor plan

Volume	1115 [m^3]
Floor surface	350 [m^2]
Average Height	2.81 [m]
N_{max} people	155 [-]
Max occupancy density	2,26 [m^2]

Figure 13 – Main room

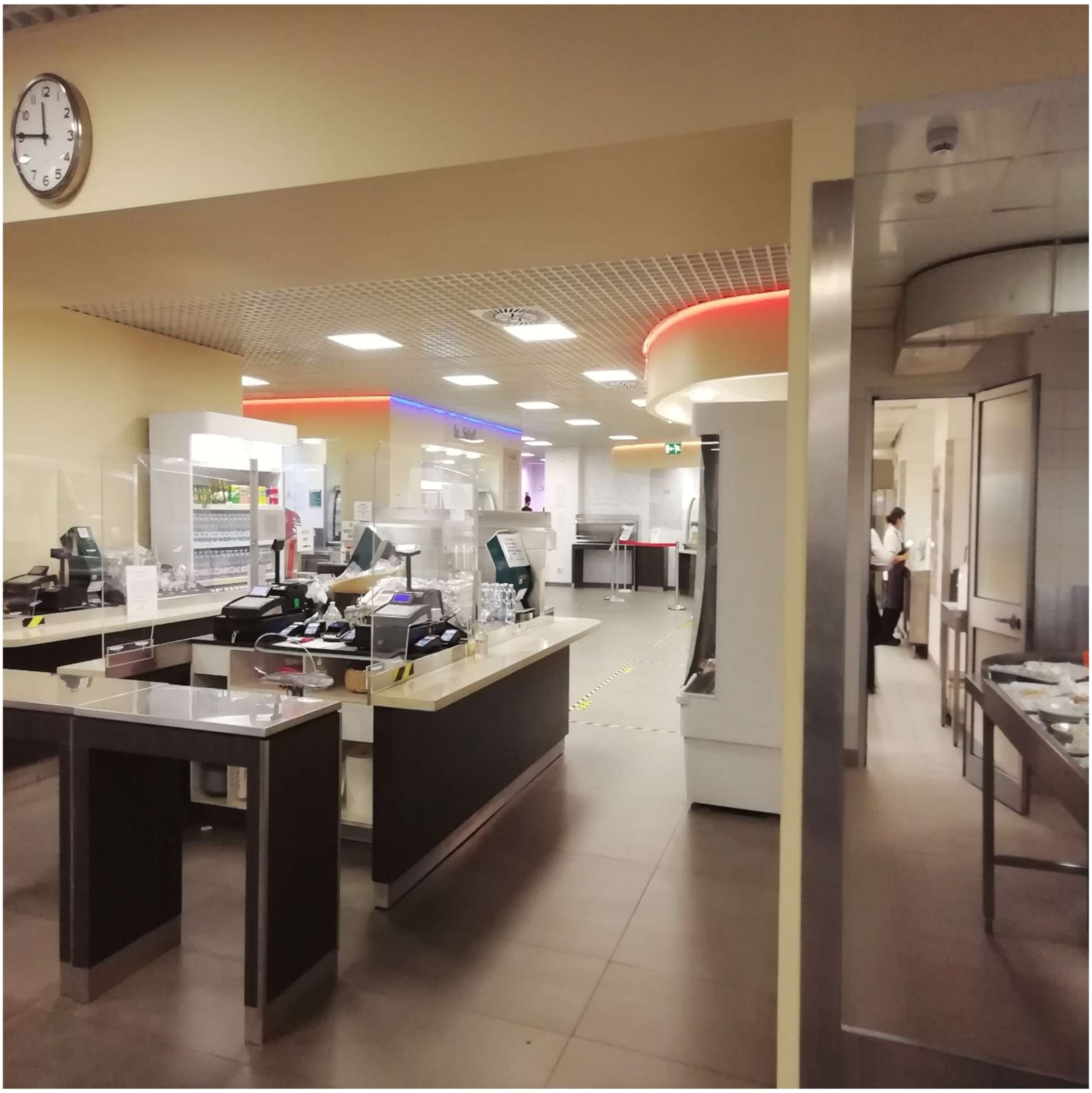


Figure 14 - Self-service and payment area

Figure 15 - Coffee area

3.1 | MEASUREMENT PROTOCOL

The monitoring campaign for the implementation of the predictive algorithm for the control of gatherings indoors, took place in Mensa CIRCOOP on 25-26 November 2020 and 14-16 April 2021 in the same time slot 12:00-14:00. It should be emphasized that during the first phase of monitoring held in November 2020, in Italy were already approved restrictions imposed by the D.P.C.M. 26 October 2020, n. 265 "*Urgent measures to contain contagion throughout the national territory*", section EE. and it was decreed that "activities of catering services (including bars, canteens, pubs, restaurants, pastry shops etc.) are allowed from 5.00 am until 6.00 pm; consumption at the table is allowed for two or maximum four people per table, if cohabiting; the activities of canteens and continuous catering on a contractual basis, which guarantee the interpersonal safe distance of at least one meters, shall be permitted". The measurement procedure followed throughout the monitoring campaign can be summarized in the following steps:

- for measurements of anthropogenic noise levels $L_{N,A}$ it was necessary to use class I instruments, as defined by the I.E.C. (International Electrotechnical Commission) standards; the measurements were carried out using two NTI Audio XL2 Sound Level Meter Class 1 integrated sound meters, capable of measuring SPL levels, actual. L_{eq}, L_{min}, L_{max}, LC_{peak} to which weighting curves A, C, Z can be applied; the main parameter used in this study is the LA_{eq_dt} to which an update frequency of t = 5 [s] has been applied and the signal has been recorded in bands of 1/3 octave and 1/1 octave (Figure 16);

Figure 12 - Sound level meters

Figure 17 - Microphone

- two omnidirectional condensation microphones M2230 NTI Audio (Figure 17) were calibrated before each monitoring cycle because due to temperature differences between rooms, the microphone may return untruthful values; calibration is done using a calibrator, i.e. an instrument that emits a pure sound of 94 dB at 1000 Hz. During the emission of pure sound at 94 dB, the sound level meter records the signal for about 10 seconds and compares the reference value set on the phonometer with the one emitted by the calibrator. Sound level meters data are valid if the two calibrations made before and after the measurement cycle differ by a maximum of ± 0.5 [dB];
- the 2 sound level meters have been mounted on two tripods in order to be positioned in the environment at a height of about +1.65 m to be able to pick up the direct and diffuse sound waves produced by the sources present in the environment, ergo be able to distinguish in the .wave rows the useful signal emitted by the speakers of the background noise. The phonometers were placed in the center of the "Main Hall" environment and in the passage area between the main room and the "Cafeteria area" pointing microphones towards the center of both rooms. Both sound level meters have been activated a "scheduler" with which it was possible to set the start-end time set for time slot 12:00-14:00;
- to calculate number of people present in the Mensa environment during the monitoring campaign, this research has used 3 infrared sensor people counting with single photocell calibrated by student Perrone Gabriele and the research fellow of Polytechnic of Turin Riente Fabrizio (Figure 18). People counters return an instantaneous value of the number of people corresponding to $N(t)= count_{out}(t)-count_{in}(t)$ so that they can monitor how many people are present in the environment while measuring the corresponding anthropogenic noise levels $L_{N,A}$ recorded by the two sound level meters. The people counters were placed at entrance doors of the Mensa environment (Figure 19) and near to the two exit doors (Figure 20-21);
- in order to synchronize the data recorded by sound level meters with the data returned by the counter, it was appropriate to re-set the time shown on the phonometer display by making sure that they correspond to the exact time (HH:MM:SS), for this real-time verification, reference was made to the https://www.oraesatta.co/ site;
- the data recorded by sound level meters was saved in a destination folder called MENSAPOLI where it's possible to find .wave files and text files in txt format, which can be open with all text editing applications (notepad, Wordpad, etc.).

Figure 18 – People counters

Figure 19 – Counter IN 1

Figure 20 – Counter OUT 1

Figure 21 – Counter OUT 2

3.2 | MEASUREMENT CAMPAIGN 25-26 NOVEMBER 2020

During the first measurement campaign that took place on 25-26 November 2020 the arrangement of the tables inside the CIRCOOP Canteen follows the pattern represented below (Figure 22) in which it is possible to appreciate that the tables respect the safety distance imposed by the D.P.C.M. number 265 of 1,5 interpersonal meter and for each table there are only 3 seats available for a total of 54 tables and 146 seats. On the first day of monitoring the flow of people inside the CIRCOOP canteen in the time slot 12:00-14:00 reached a maximum peak of 117 people while on the second day there was a peak of 151 people. Data collected during this first measurement campaign, was processed on the Excel program, taking care to synchronize the data of the anthropogenic noise levels $L_{N,A}$ recorded by sound level meters SLM 1 and SLM 2, with the data collected by the counters also set with an update frequency of 5 seconds. It was therefore crucial during the monitoring campaign, to try as much as possible to activate at the same time both phonometers and people counting, in order to associate with the instantaneous value $N(t) = count_{out}(t) - count_{in}(t)$ of the people counters, a corresponding anthropic noise level $L_{N,A}$.

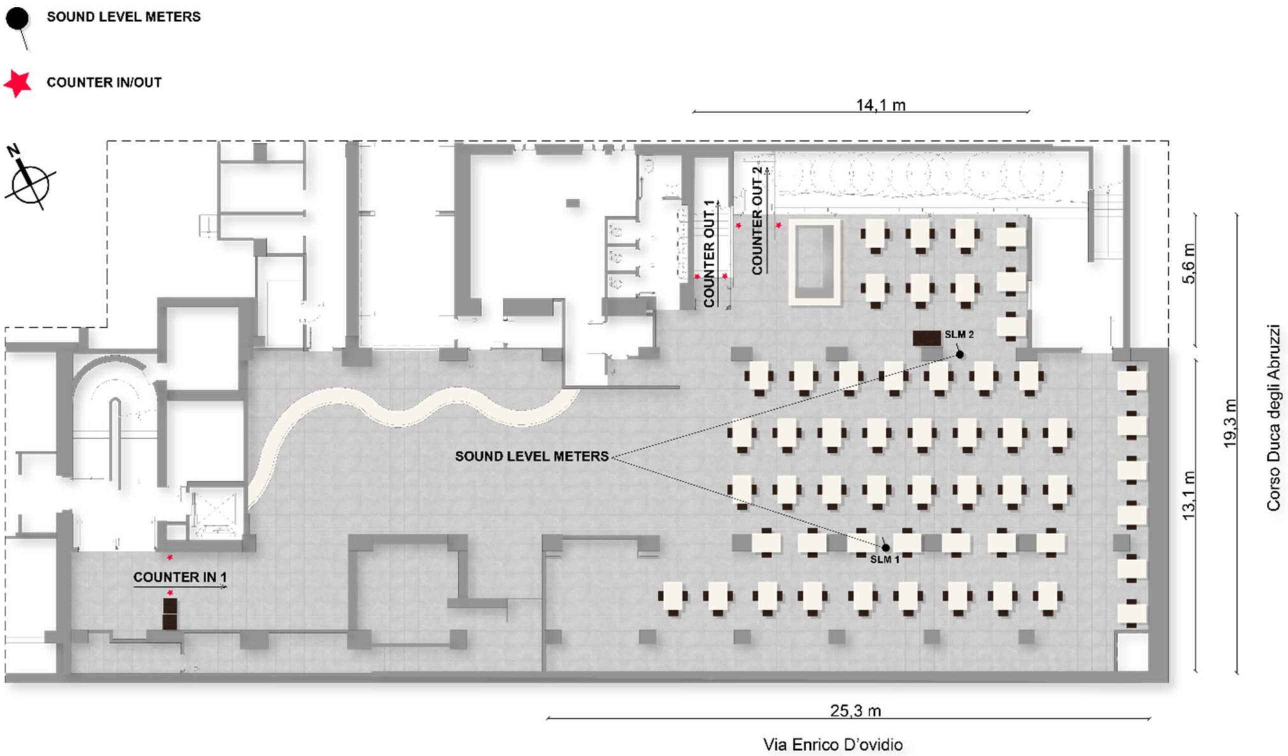


Figure 22 – November layout

Total seats	146 [-]
Total tables	54 [-]

3.3 | MEASUREMENT CAMPAIGN 14-16 APRIL 2021

Second part of the measurement campaign took place on 14-16 April 2021 in the same time slot 12:00-14:00 and it is possible to note in the image below (Figure 23), how the arrangement of the tables within the CIRCOOP canteen has been modified respect to scheme presented in November. The new arrangement of the tables was a consequence of the lower turnout recorded in the winter months due to the restrictions imposed by the D.P.C.M. number 265 and the tendency of diners to occupy an entire table alone or with a maximum of 2 users per table, as required by law. For this reason, the third chair available for each table has been eliminated to reduce the distance between two consecutive tables but still guaranteeing an interpersonal distance of 1,5 m between two successive diners (Figure 24). Within the CIRCOOP canteen there is a total of 74 tables and 155 seats, so the total number of seats in November and April remained almost unchanged although the number of tables increased by 20 between November 2020 and April 2021. However, although the total number of seated is the same as on November, it was possible to observe a higher turnout within the CIRCOOP canteen in which there is a maximum peak of 216 people on April 14, while on the second day there was a peak of 204 people. Another difference from the previous measurement campaign of November 2020 is the position of the second SLM 2 sound level meter (Figure 25), which in this case was placed near the coffee area with the microphone facing the main environment that has a larger extension in plan, while the SLM 1 sound level meter occupies the same position used in November 2020 monitoring campaign (Figure 26).

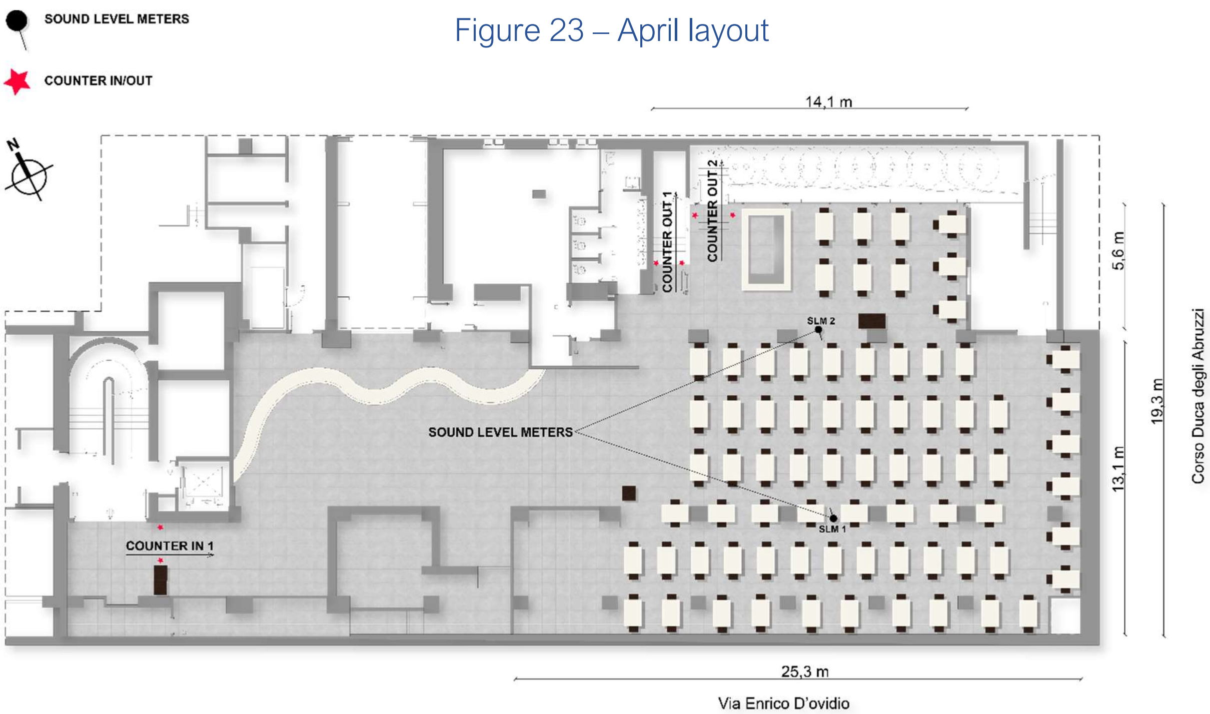


Figure 23 – April layout

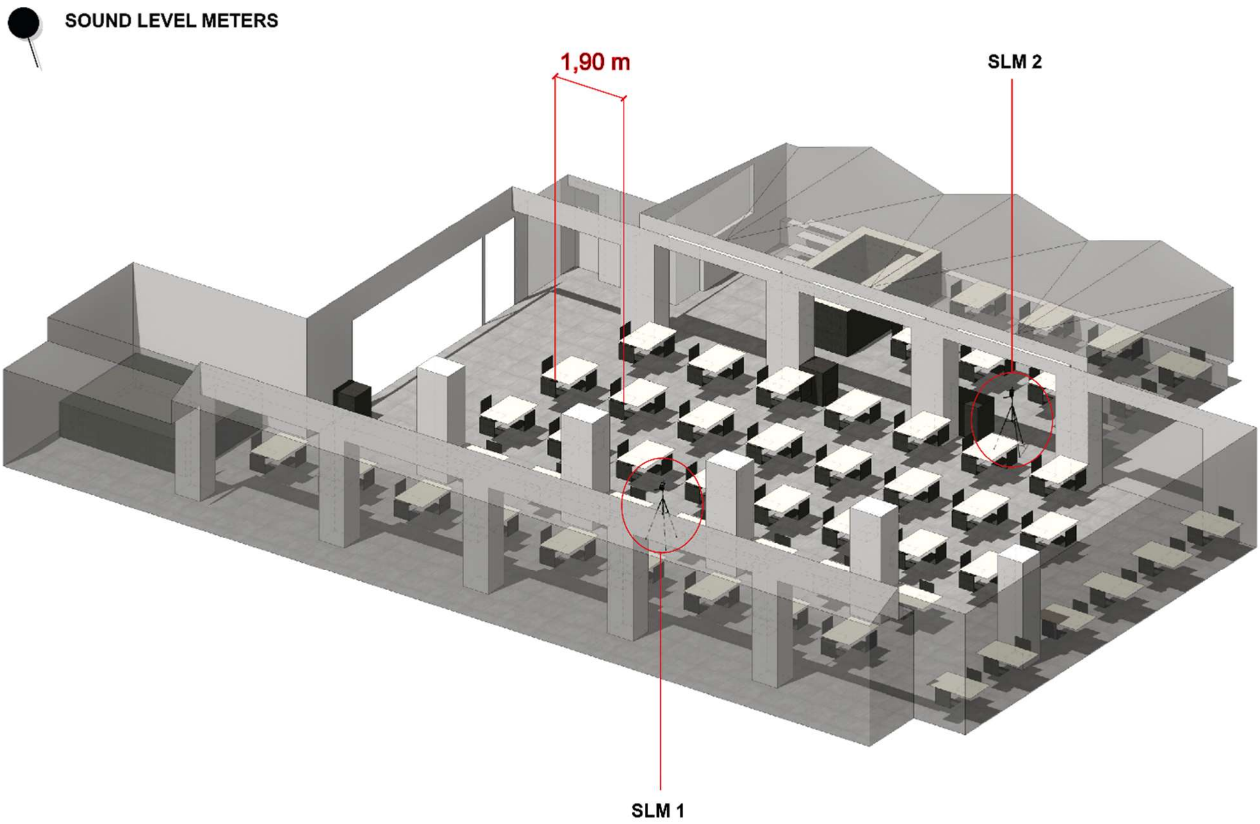
Total seats	155 [-]
Total tables	74 [-]

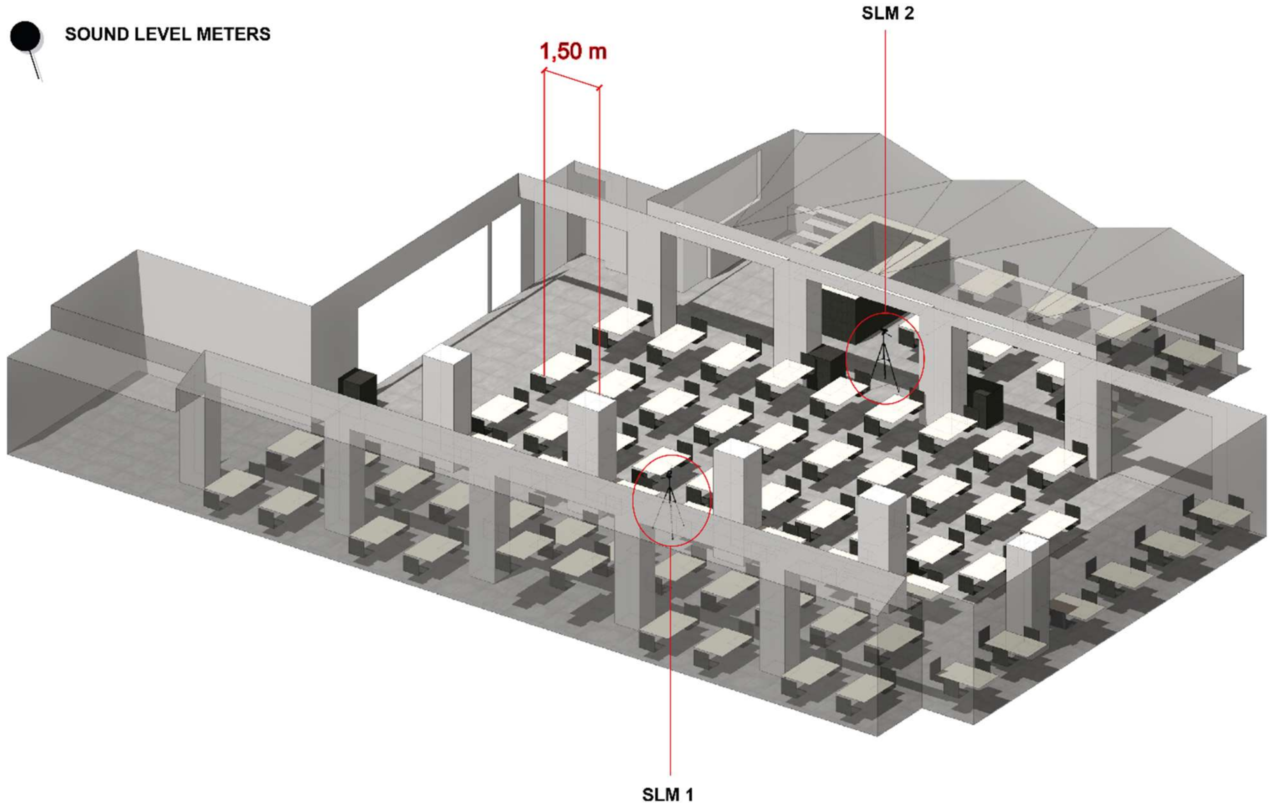
Figure 24

Figure 25 – Sound level meters SLM 2

Figure 26 – Sound level meters SLM 1

Canteen Layout November 2020 vs April 2021





3.4 | PROCESSING DATA IN EXCEL

All data extracted from the SLM 1 and SLM 2 sound meters during the 4 monitoring days were copied to computers in .txt format and then reworked within the Excel software using a column for $L_{A.eq\_dt}$ as measured by the SLM 1 and in another $L_{A.eq\_dt}$ collected by the SLM 2 for each of 4 days of monitoring (Figure 27). In addition to the levels of $L_{A.eq\_dt}$ levels recorded by the phonometers on Excel, a column called "N people counting" was created in which the data corresponding to $N(t)= count_{out}(t) - count_{in}(t)$. The data thus processed on Excel were then graphically represented to schematize the anthropogenic noise levels recorded by the SLM 1 and SLM 2 phonometers and to study their possible correlation with the N(t) values obtained from the people counters.

25/11/2020			26/11/2020			14/04/2021			14/04/2021		
N contapersone	SLM 1 LAeq_dt	SLM2 LAeq_dt	N contapersone	SLM 1 LAeq_dt	SLM2 LAeq_dt	N contapersone	SLM 1 LAeq_dt	SLM 2 LAeq_dt	N contapersone	SLM LAeq_dt	SLM LAeq_dt
9	57,4	57	7	61,5	62,5	16	56,1	62,3	26	62,7	58,1
9	61,1	58,4	7	53,5	54,9	16	59,8	64,1	29	56,4	63,6
9	60,1	64,5	7	55,3	54,7	16	61,1	69,7	31	59,4	61,9
9	51,8	52,1	7	60,7	62,5	16	62	62	32	60,9	60,8
9	53,3	55	7	60,4	59,8	16	61,4	65,5	32	56,3	59,4
8	60,9	58,3	7	63,3	66,8	16	57,3	60,9	32	61,2	57,5
8	54,2	56,5	7	54,2	54,7	16	64,4	68,6	32	56,8	57,4
8	55,5	58	7	55,8	57,4	16	63	62,6	32	60,6	63,9
8	57,1	58,4	7	57,7	55,5	16	63,9	69,1	32	60,3	61,7
8	62,6	62,2	7	59,8	57,5	16	61,6	71,2	32	57,9	63,1
8	57,8	58	7	76,1	56,9	16	65,4	61,6	32	56,4	62,5
8	68,8	66,4	7	66,4	66,7	16	61,2	66,5	32	60,7	61
8	65,4	63,3	7	60,7	59,3	16	70,7	60,3	32	59	62,7
8	64,1	66,1	7	67,1	65	16	67,1	64,6	32	60,3	65
8	58,5	62,2	7	59,3	60,6	18	64,4	68,3	32	66,4	65,1
8	59,7	59,2	7	65,3	66,7	19	58,4	69	32	60	59,5
8	64,2	61,6	7	60,5	62,6	19	74	63,4	32	67	68,7
8	58,6	61	7	60,4	61,4	19	61,6	64,8	32	63,8	60,1
8	61,9	64,6	7	63,7	62,7	22	62,9	64,5	32	59,2	64,5
9	55	58,5	7	58,3	59,5	22	71	67,1	34	55,4	58,4
9	58	59,9	7	59	57,7	22	68,3	64,3	34	62,7	61,5
9	56,8	57,5	7	60,9	60,9	22	68,9	64,9	34	63	61,9
11	56,1	56,8	7	60,4	57,4	22	65,9	66,7	34	59,6	58,1
12	62,3	65,2	7	61,5	61	22	60,7	62,9	34	59,8	62,6
12	59,7	59,6	7	58,4	60,3	22	71,4	68,5	34	62,6	66,4
12	59	63,9	7	54,1	57,5	22	68,5	67,6	36	57,9	58,4
12	59,7	60,4	7	60,6	54,9	22	64,4	63,4	37	57,3	57,9
12	59,6	60,2	7	61,7	59,1	22	62,9	66,4	39	63,9	62,4
12	54,3	54,6	7	63	57,2	22	63	63	40	58,2	61,2
12	59,6	58,9	7	59,5	57,1	22	58,4	65,2	41	59,3	61,9
13	58,3	59,4	7	58,5	58,2	23	62,7	63,6	41	60,5	61,4
14	57,9	59,2	7	55,3	57,5	23	67,2	59,3	41	58,5	61,2
14	62,9	60,2	7	60,8	58,7	23	63,1	61,8	41	69,4	58,3
14	62,4	61,1	7	68,7	67,3	23	64,2	63,2	42	62,1	63,6
14	61,4	60,9	7	58,5	64,8	23	70,6	61,2	45	59,7	62,7
14	59,6	58,1	7	62,6	62,9	23	58,3	65,4	46	61,6	57,2

Figure 27 – Excel data

Data analysis November 25, 2020

During the first monitoring day of November 25, 2020, it is possible to divide the N=people graph and corresponding anthropogenic noise levels $L_{N,A}$ measured in dB(A), into 3 phases. In the first part it is possible to notice a fairly significant change in $L_{N,A}$ levels, going from a minimum value of 55 dB(A) to a maximum value of almost 80 dB(A) which corresponds to a significantly rapid growth in the number of people, going from a minimum of 2 units to a maximum of 60 units in about 20 minutes. In the second part of graph we can see that the growth in the number of users tends to slow down by assuming values ranging from a minimum of 60 units to a maximum of 75 units and at some times the number of users also tends to decrease by about 20 units in a short time and then return to the range values. At this stage the corresponding anthropic noise levels $L_{N,A}$ also tend to remain constant although maintaining a fairly wide range ranging from a minimum of 60 dB(A) to a maximum of about 85 dB(A). In the last part of the graph we note that the number of people picks up from the previous 75 units to the peak of 117 people recorded during the first day of measurements but despite this, the measured $L_{N,A}$ levels remain constant in the same range from 60 dB(A) to 80 dB(A).

Data analysis November 26, 2020

A similar trend also occurred on November 26, 2020, the synthesis graph of which can be divided into two phases: during the first phase the number of people grows much less rapidly than in the previous day and this growth follows a constant trend for almost the duration of the measurement. Again, the anthropic noise levels $L_{N,A}$ cover a fairly wide range from a minimum of about 50 dB(A) to peaks of 80 dB(A) very quickly and maintaining this same trend until the second phase is reached during which we move from a number of people equal to 135 units to a minimum value of 105 units. During the second phase, the number of people at least 105 units resumes its growth until it reaches its maximum peak measured during this second day of 151 units, but this corresponds to a range of levels of $L_{N,A}$ which stands at between 65-75 dB(A) and therefore shows values much less variable than in phase 1.

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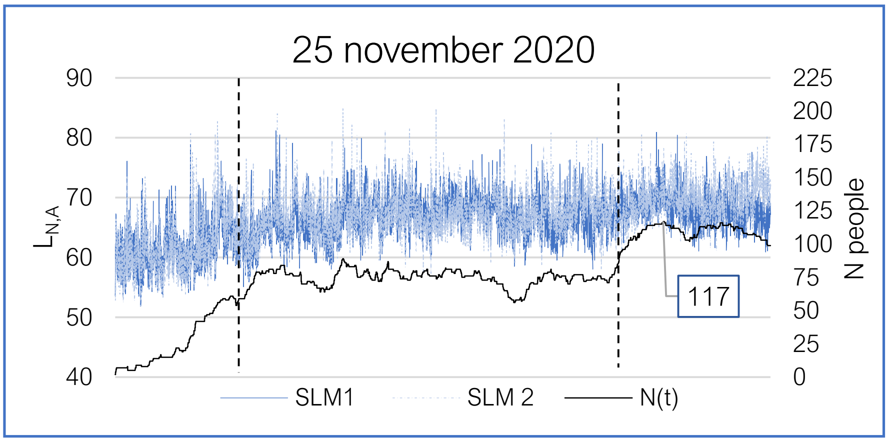
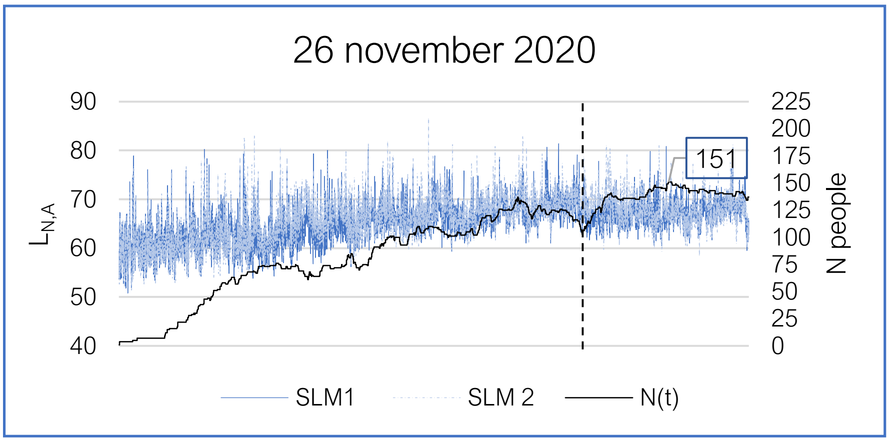
Data analysis April 14, 2021

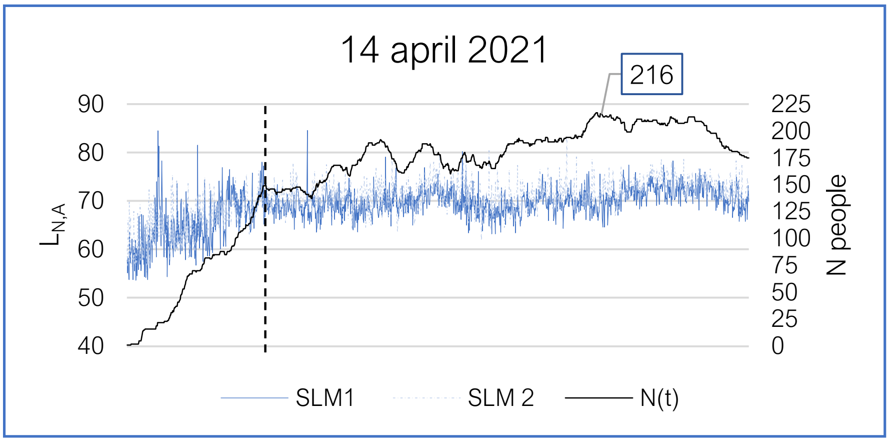
The measurement campaign was resumed on April 14 2021 and as described above, the only difference recorded compared to November is the arrangement of the tables inside the Canteen, which although modified did not affect the total number of seats available. During this first day of monitoring it was possible to divide the synthesis graph into 2 phases: during the first phase we notice a fairly rapid growth in the number of people going from a minimum of 2 units to a maximum of 147 in about 35 minutes from the start of monitoring, to which are corresponded by anthropogenic noise levels $L_{N,A}$ also increasing and within a range from a minimum of 55 dB(A) to a maximum of 75 dB(A) except for a peak of 84.5 dB(A) recorded at 12:03. In the second phase of graph we note that the number of users continues to grow less rapidly than in the first phase and with some decreases immediately followed by a new increase that brings the data back to follow the same stable growth trend until you reach the peak value recorded during this third day of measurements equal to 216 units. However, a fairly linear growth in the number of people corresponds to a range of anthropogenic noise levels $L_{N,A}$ which is in a range of between 65-75 dB(A) and therefore does not seem to have a direct correlation with the increase in the number of users. In fact, if you look closely at the graph, you can see that at the peak of the number of people at 216 units there is a decrease in the anthropogenic noise level $L_{N,A}$ which touches almost a minimum value of 60 dB(A).

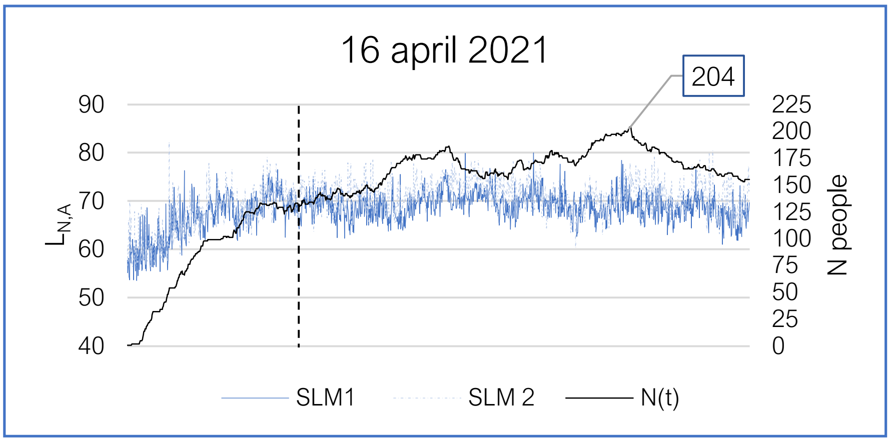
Data analysis April 16, 2021

Even during this fourth and final day of measurements, it is possible to notice a similarity with data recorded on April 14, if not more marked and significant. Again, the graph can be divided into 2 phases; during the first phase there is an exponential growth in the number of users going from a minimum of 1 unit to a maximum of 137 units in just 20 minutes from the beginning of the simulation which corresponds to an increase in anthropogenic noise levels $L_{N,A}$ also increasing and linear except for a peak point of 82.2 dB(A) recorded at 12:05. However, in the second phase it can be seen that in increments and/or decreases in the number of people, there is a exactly inverse trend in anthropogenic noise levels $L_{N,A}$;in fact, at the maximum number of people recorded during this last day of measurements of 204 units, there is a level of $L_{N,A}$ of 63.1 dB(A).


25 november 2020
90
80
70
60
50
40
$L_{N,A}$
225
200
175
150
125
100
75
50
25
0
N people
117
SLM1
SLM 2
N(t)


26 november 2020
90
80
70
60
50
40
$L_{N,A}$
225
200
175
150
125
100
75
50
25
0
N people
151
SLM1
SLM 2
N(t)


14 april 2021
216
90
80
70
60
50
40
$L_{N,A}$
225
200
175
150
125
100
75
50
25
0
N people
SLM1
SLM 2
N(t)


16 april 2021
204
90
80
70
60
50
40
$L_{N,A}$
225
200
175
150
125
100
75
50
25
0
N people
SLM1
SLM 2
N(t)

4 | INTRODUCTION TO ACOUSTIC SIMULATION

The analysis of the sound field in a closed environment and the corresponding sound pressure p(x,y,z,t) at any point in space at any given time t can be estimated through the Laplace–d'Alambert equation. The complexity of this equation, however, limits its application to very simple cases such as, for example, the study of a cavity, since in it we will have considerably small dimensions, regular geometry, homogeneous surface and the size of the cavity are comparable with the wavelengths produced by the sound within the cavity itself. On the contrary, in real practice, the dimensions of the rooms are not comparable with the wavelengths under consideration, the surfaces are particularly complex due to the presence of fixtures, niches and protruding objects that cause the diffraction or concentration of sound in certain areas of the closed environment. In addition, the bounding surfaces of the environments, have different absorption coefficients depending on the material of which the partitions are of and this generates uneven sound reflection modes on all surfaces. As an alternative to the Laplace–d'Alambert equation in environments with complex geometry to approaches based on a simplifying hypotheses are used:

- geometric approach - geometric acoustics method;
- energy-statistical approach - statistical acoustics method.

Both approaches, while simplifying the environments to be analyzed with respect to their real configuration, offer numerous benefits, including the possibility of using the same basic model for theevaluation of different indices that describe the sound phenomenon within the environment such as reverberation time $t_{0,}$ EDT (Early Decay Time), C_{50}, C_{80} etc. On the market there are numerous acoustic simulation software that allow to use both methods listed above for the acoustic evaluation of indoor environments, through numerical simulations that although simplify the real models allow a description at different levels of accuracy of the sound phenomenon.

4. 1 | GEOMETRIC ACOUSTICS METHOD

Geometric acoustics, as in the field of optics, consider sound power in the form of rays whose wavy nature is neglected. This simplification is desirable only when the wavelength under consideration is much smaller than the predominant size of the environment analysed and/or the obstacles present in it and the areas delimiting the environment are mostly smooth and homogeneous. The idealized sound rays are transmitted from the source to the perimeter surfaces of the environment on which they are then reflected in mirror mode from a fictitious sound source beyond the surface on which the beam affects. This mode of sound diffusion is called the "image source method" (Figure 28), can be summarized as follows:

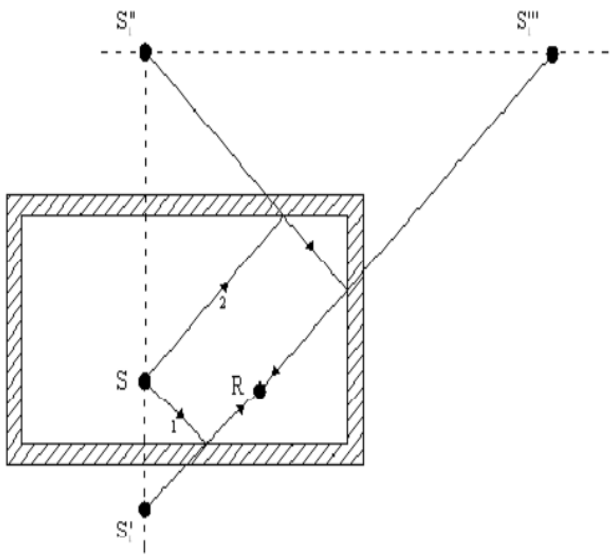


Figure 28 – A. Nicolini

- the approximations of geometric acoustics are valid, infinitely large surfaces with respect to the size of the sound wave under examination. For example, if the sound analyzed has a frequency f=1000 Hz corresponding to a wavelength of 0.34 m since the sound propagates in the air (with a temperature of 20 degrees Celsius) at a speed of 342 m/s, surely the size of the environment will be much larger than the wavelengths analyzed;
- sound rays are reflected specularly on surfaces delimiting the environment, so that the incident radius, reflected radius and surface normal lie on the same plane and the angle between the incident radius and the normal is equal to the angle between the normal and the reflected radius; each mirrored ray is associated with an image source beyond the surface on which the beam affects and it behaves as if it were actually emitted from the original source;
- each source, real or imaginary, emits spherical wave fronts that propagate from the source to receivers;
- the sound power of each reflected beam is equal to that emitted by the real source to which the effect of absorption by the surfaces delimiting the environment and the

attenuation by the means of propagation is subtracted, in general case the means of propagation is air.

In addition, in the case of complex geometries, for the correct application of the image source method it is necessary to ensure that the visibility criteria between the emitting source and the position of the receiver are met: first, the point of reflection of the incident radius must belong to the surface on which the radius itself is reflected (image on the left) and the radius must not be interrupted by other surfaces or objects present within the analyzed environment (image on the right). The calculation algorithm must then perform for each source, receiver, surface delimiting the environment and any order of reflection of the sound rays, a visibility test to check that the reflection point of the incident radius belongs to the reflection surface and that the same is not interrupted by another surface not involved in the reflection process (Figure 29). There are also two other controls which make it possible to exclude a priori the impossible combinations of subsequent reflections:

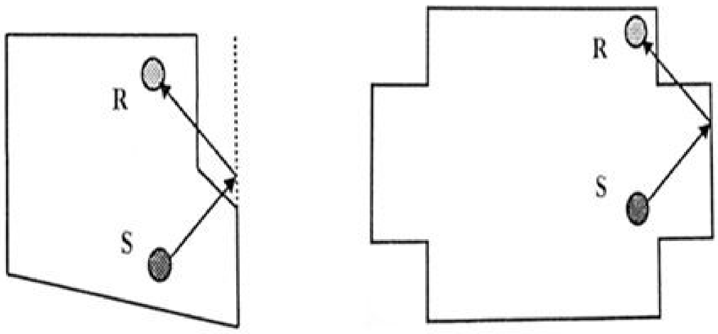


Figure 29 – A. Nicolini

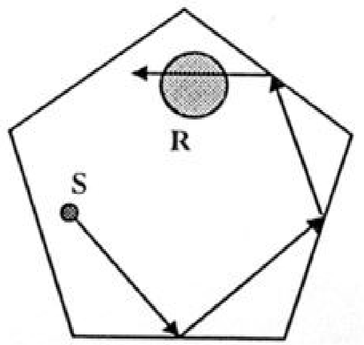
- exclude paths containing sequences of mutually invisible surfaces;
- exclude for order reflection (I), surfaces that are not visible from the position of the order virtual source (I-1).

The ray tracing method is named after the way the propagation of the sound wave in the environment is schematized. In this methodology, sound rays, instead of propagating from the emitting source in the form of spherical wave fronts, are represented as real rays containing fractions of the sound energy emitted along straight trajectories. In addition to the assumptions already listed in the image source method, the following assumptions apply:

- the sound energy of the source is fractionated into a finite number of sound rays propagating omni-directly from the emitting source;
- rays has an ideally infinitesimal and constant section;
- the sound rays lose energy as a result of absorption of the surfaces delimiting the environment and due to the attenuation of the propagation element.

Ray generation can occur in deterministic mode, following a geometric rule of partition of a unit radius sphere, or in statistical mode, orienting directional vectors randomly. Unlike the image source method, in ray tracing the calculation time depends on the number of rays and the maximum order of reflections manually set at the beginning of the calculation; there is therefore, a great saving of time since there is no need to carry out any visibility tests. For this reason, it is important to set a sufficient number of rays to geometrically cover the entire environment in which to perform acoustic simulations and it is necessary to set for each radius a sound power proportional to the W power of the emitting source and its q directivity factor. Contrary to the image source method, in this case the receiver is no longer represented pointwise since its ability to pick up rays would be almost n nothing: in this methodology receivers are represented instead by finite volumes representing their zone of sensitivity; they are usually represented by spheres of constant diameter to emphasize that the effective area of the receiver is the same by any direction of incidence. For each receiver, the sound level it picks up can be obtained by dividing the overall sound energy picked up by each by the volume of its sensitivity area, represented by a sphere (Figure 30) and calculates its sound energy density and corresponding sound level.

Figura 30 – A. Nicolini



The aim of this methodology is to be able to reconstruct at a given number of points the response to the impulse of the environment analysed from which it is then possible to determine acoustic descriptors. Hybrid methods are simulation techniques that combine the image source method and ray tracing. This methodology is born because in daily practice very few sound sources meet the criteria of visibility between source-receiver required in the process of image sources and visibility tests are long and expensive.

4.2 | METHOD OF STATISTICAL ACOUSTICS

The statistical method does not consider sound in the form of a spherical radius or wave fronts, but it is assumed that the acoustic energy calculated at a certain point in the environment can come from all directions more or less likely depending on the average free path. The term free mean path is the path that run across acoustic particles between two successive reflections which, in environments of regular and compact form, is equal to 4V/S as shown in the Sabine hypotheses of the reverberated acoustic field in which V= volume of the hall and S=total surface that delimits it.

This methodology implies that sound is represented by infinitesimal particles that, once the bounding surfaces of the environments reach, are not reflected in specular mode by the surfaces but in diffuse mode (Figure 31). The main hypothesis of statistical acoustics is in fact the perfect sound diffusion following the law of the little thing or Lambert:

$$P(\alpha) = \frac{1}{\pi \times \cos\alpha} \quad (1)$$

where:

- P= probability function of acoustic energy as a function of the solid angle α.

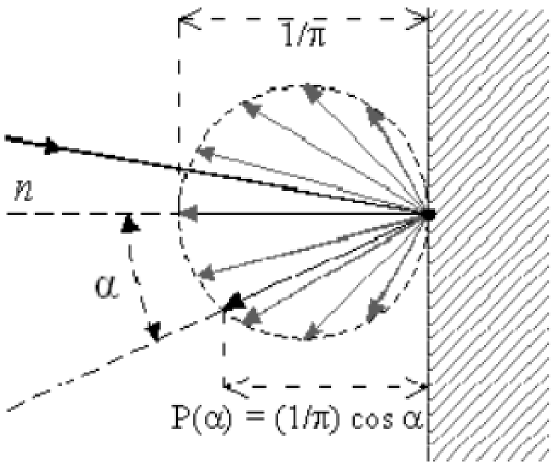


Figura 31 – A. Nicolini

4.3 | SIMULATION IN ODEON VERSION 16 COMBINED

To evaluate how implement algorithms derived from models of J. H. Rindel and D. D'Orazio for the calculation of the value N=number of people present in closed environment, starting from different levels of anthropic noise $L_{N,A}$,Odeon 16 ,has been used in this thesis for acoustic simulations. Simulation software such as Odeon exploits laws of geometric acoustics and simulation is divided into 2 main phases, early reflection that are recreated according to the method of image sources (red rays) and second-order reflections (blue rays), also called late reflections that are reproduced through the ray-tracing method. This type of software therefore allows to virtually recreate the environment to be analyzed and with the calibration of geometric model it is possible to obtain all the acoustic parameters necessary for a redesign of spaces in which improve acoustic performances. In a real room, there are objects, furnishings, users and geometric features, e.g. shadow zones, that makes analytical calculation too complex and expensive, for this reason numerical simulation models try to simplify real without neglecting the physical characteristics of the sound phenomenon, dissipation, reflections/refractions, attenuation and so on. The steps followed in this thesis for the calibration of the Mensa CIRCOOP environment can be summarized in the following paragraphs.

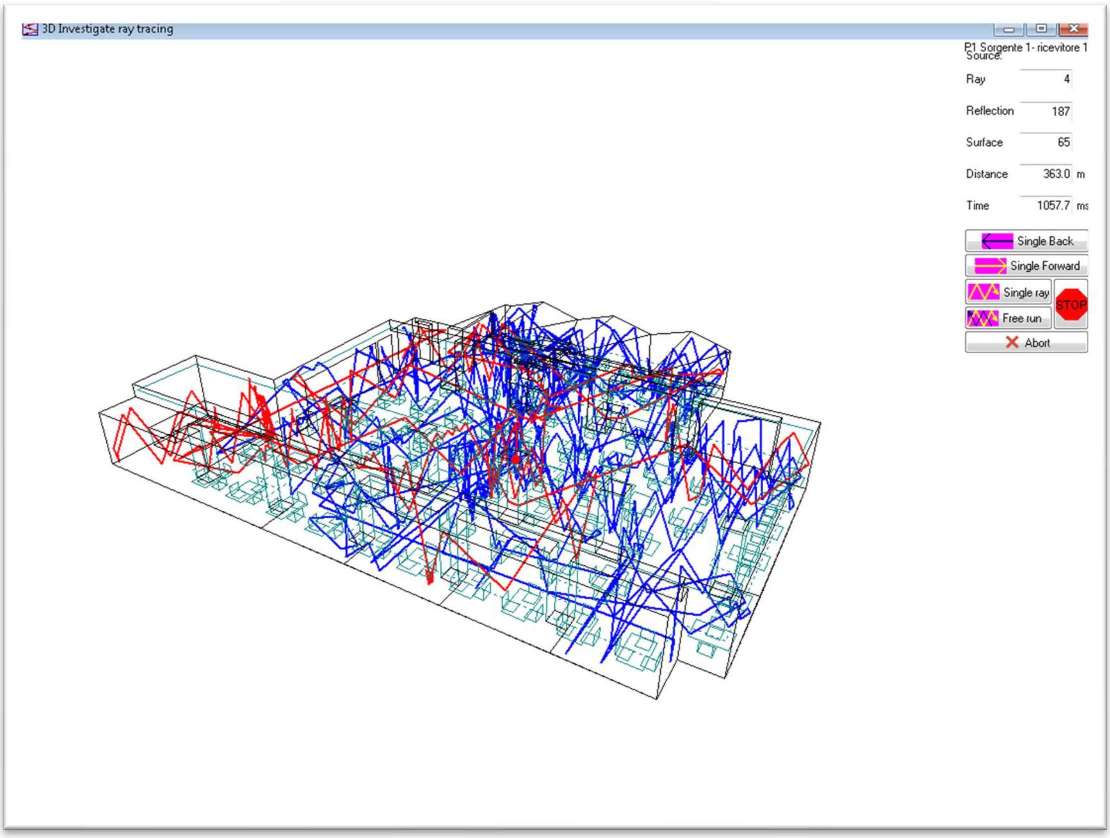


Figura 32 – 3D Geometry debugger ODEON

3D modeling and importing to Odeon 16

The geometry of the CIRCOOP Canteen of Corso Castelfidardo has been modeled in meters on the Sketch- Up software assigning to each type of surface delimiting the environment a layer with a specific name, e.g. walls, metal furnishings, windows etc., in order to simplify the recognition of surfaces and the subsequent assignment of materials on Odeon 16 Combined. The 3D model must be watertight, i.e. there must be no holes in the geometry as such openings would cause rays to leak during acoustic simulations on Odeon. For this reason, at the areas of passage between the analyzed environment and neighboring environments, the surface has been closed and delimited by assigning to Odeon material 1 corresponding to an absorption coefficient of 100% to simulate a portion of the space in which there is an opening. When the template is complete, you must import it in .par on Odeon 16 combined and check for problems with 3D geometry by clicking on the "3D Geometry debugger" (Figure 32) where you can view any overlapping surfaces and correct them because this situation does not allow Odeon to understand which absorption coefficient to refer to in the simulation. Further verification can be done by assigning all surfaces a random absorption coefficient, placing a source in the center of the environment by clicking on the Source - receiver list and checking if there is a loss of rays by clicking on "3D Investigate ray tracing" and "free run" to display the propagation of the rays starting from the sound source in the 3D model. This model verification phase is essential so as not to incur a wrong calibration of the real environment.

Room setup for initial calculation

A first setting of the environment to be calibrated to get a general idea of how sound is transmitted inside the "black box," was possible using a generic source inside the environment without necessarily placing receivers; by clicking on "Global Estimate" the software calculates how much the sound is attenuated in the simulation duration time frame. During the calculation, the values of t_{20} update from second to second for each of the frequencies in octave bands until they reach a static value for each of them. On the other hand, if the values of $t_{3}0$ are not correctly derived, a symbol "**" appears at certain frequencies indicating that the number of rays emitted by the source is not sufficient to cover the distances between the set source and the area of delimitation of the environment. For this reason it is necessary to set in "Setup room" a sufficient number of rays depending on the type of environment analyzed, It is always recommended to try to import into Odeon an environment that is geometrically not complex, therefore that it is composed of surfaces that tend to be flat; For example, in environments where columns or arcs exist, the curvature of those surfaces can be simplified by using a series of flat surfaces that reproduce their convexity. In "Room setup" Odeon has already preset 3 precision categories "Survey", "Engineering" and "Precision" and each of them corresponds to an increasing number of Late rays and consequently greater precision during simulations that correspond to an incremental analysis time. The second value to be set is the "Impulse Response Length" which represents the distance that the sound rays can travel from the emitting source. To set this value it is sufficient to refer to the average values returned by the "Global Estimate", in the case of Mensa CIRCOOP the t_{30} measured in the early stages of the simulation is 2 [s], so it was possible to set an initial value of I.R.L. of 2000 [m/s]. Again, if the value of I.R.L. it is not large enough to cover the entire environment considered, during the "Global Estimate" certain frequencies in particular the bands from 63 to 1000 [Hz] will return null values, this is because the low frequencies have a lower propagation rate than the high frequencies and consequently if the value of I.R.C. is incorrect, most sound rays will be attenuated before reaching the areas of delimitation of the environment.

Material assignment and simulated VS reverberation time

Once setting first parameters in the "Room setup", you must assign each surface a material to characterize the model, make sure that it is as likely as possible to the real environment. To assign the materials you must click on Toolbar, Material List and select in the cards that open the materials already present in Odeon divided by type (Figure 33).

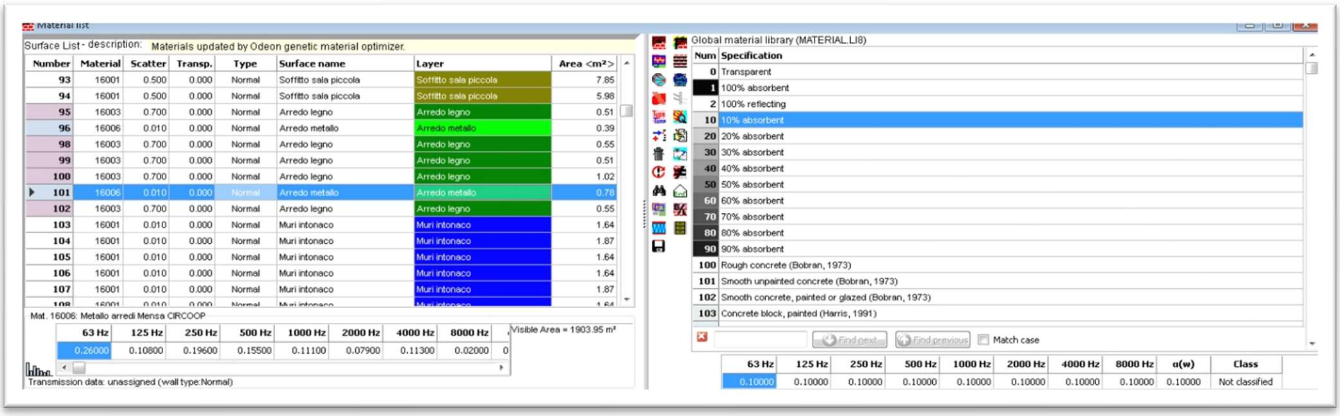


Figura 33 – Material List

This material attribution phase is a first general approach to evaluate how the combination of the different absorption and scattering coefficients of the modeled surfaces returns parameters more or less similar to those actually measured; in a second phase the attribution of materials was refined with the calculation of the "Genetic Material Algorithm, a function present only in the Odeon Combined and Auditorium versions created ad hoc thanks to a research by Christensen C.L. Genetic Material Algorithms a process of optimization in the calibration of materials set in the 3D model that allows to improve the combination of the different absorption coefficients, instead of proceeding manually and iteratively with a trial and error approach.

For this reason, it is necessary to introduce into the simulation of the parameters really measured in environment with which to compare the values of the same parameter returned by the simulation on Odeon. In this thesis the parameter really measured within the Mensa CIRCOOP environment is the reverberation time t_{20}. The steps taken to optimize the materials used in the model can be summarized as follows:

- it was necessary to carry out an inspection, which took place on November 25, 2020 before the opening hours of the Canteen, then to a furnished but unoccupied environment. During the inspection it is necessary to evaluate the type of surfaces present and the materials of which they are composed; if catalogues or information on stratigraphy of the surfaces are not available, in Odeon you have to set materials similar to those observed, but it is also possible and insert a new material created ad hoc by the user by inserting the different absorption coefficients for the octave bands 63-8000 Hz and the scattering coefficient that by default assumes on Odeon a value 0.1 for smooth surfaces, on which therefore the reflection of the rays is almost mirrored. In the Mensa CIRCOOP environment, the pulse response was measured using an Audio XL2 Sound Level Meter Class 1 supplement sound level meter mounted an M2230 NTI Audio condensation omnidirectional microphone positioned on a tripod at a height of +1.50 m from the floor (Figure 34), while sound sources have been simulated using a clapper (Figure 35), from the English term "to clap", an instrument capable of producing high-frequency noise at high speeds, similar to those produced by the burst of a large balloon.

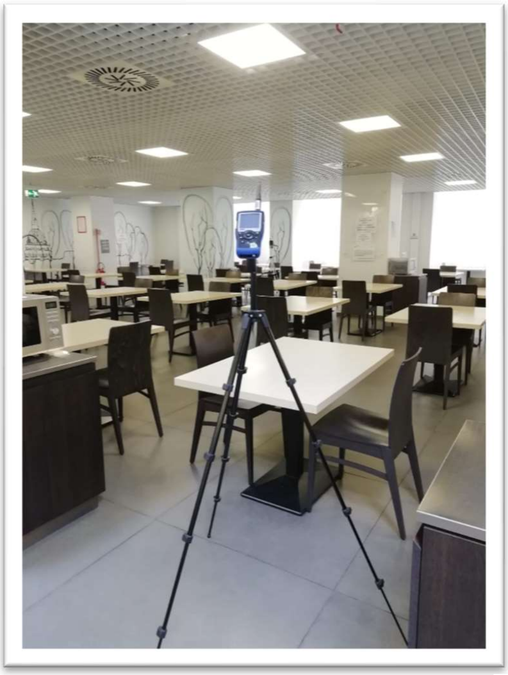


Figura 34 – SLM

Figura 35 – Clapper

- the phonometer and the clapper were placed in the Mensa environment in different combinations of source-receiver positions, recording the t_{20} for the duration of the procedure. Once the measurement was complete, the files were saved on the sound level meters in a fold called Mensapoli and named RT60 and then extracted in .txt as follows:

RT60 Cycle Results

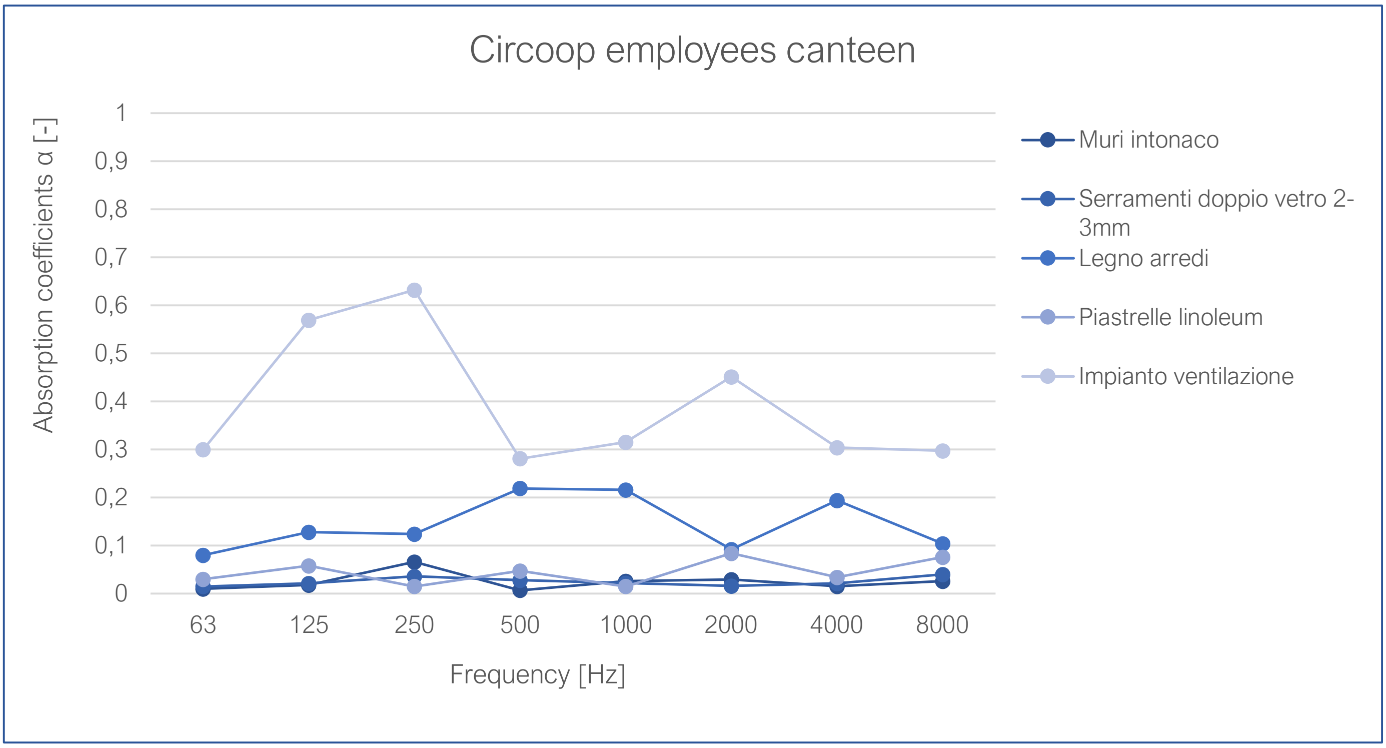
Band [Hz]:	63		125		250		500		1000		2000		4000		8000	
CYCLE	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]	T20 [s]	Correltn [%]
01	-.--	-.--	0.45	96.50	0.51	98.27	0.86	99.87	0.75	99.89	0.79	99.95	0.74	99.93	0.55	99.94
02	-.--	-.--	-.--	-.--	0.77	99.07	0.88	99.71	0.81	99.82	0.86	99.87	0.81	99.94	0.63	99.92
03	-.--	-.--	0.69	97.84	0.84	99.88	0.78	99.80	0.82	99.86	0.86	99.95	0.78	99.96	0.58	99.97

- the impulse response was measured at 5 points in the environment and the respective source-receiver positions were marked because in the next simulation phase in Odeon, the receiver and sources must be repositioned at the same points really measured. The measure of the impulse response in the real environment must follow the guidelines present in ISO 3382:2008 and the environments in which to perform the measurements must be furnished, partially or completely, but unoccupied. The phonometer should be positioned at a height of +1.55 m from the floor and at distance of 1 meter from the surfaces delimiting the environment and/or from rather reflective surfaces and at about 0.5 m from the furnishings present inside the closed environment. These guidelines also apply to the sound source which in our case is represented by clapper;
- in the imported 3D model on Odeon, you must set the receiver and sources in the same locations where calculated the reverberation time of the real environment. Again, must follow the standards in ISO 3382:2008 by setting the sources/receivers to a height of +1.5 0 m from floor if the source is upright or +1.2 m if the source, for example a speaker, is seated. In addition, it is advisable to place the sources/receivers at a distance of 0.5 m from the furnishings that may be present in the 3D model and at a distance of 1 m from the area of delimitation of the environment and/or from very reflective surfaces;
- for each source/receiver combination, Odeon calculates the "multi-points response" on the Job List tab and enter the data really measured by the phonometer to compare the simulated values with those measured. In this case study the files generated by the phonometers are not in .wave format, so it was not possible to load the data directly through the "Load Impulse response tool" option; however, being in .txt format it can be imported by clicking on "Measured versus Simulated" and "Open/Edit measured data file", writing the parameters measured in the following form to the .par file:

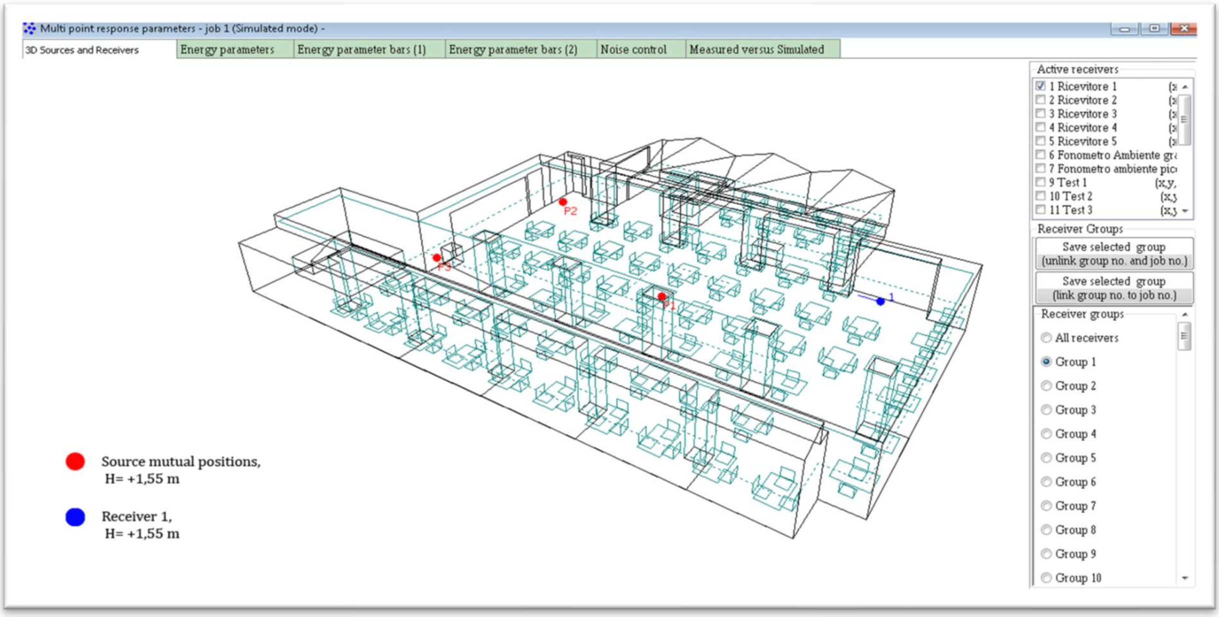
E:\2_Tesisti\Giulia Calia\3D Odeon\Mensa\Mensa.MeasuredEx01

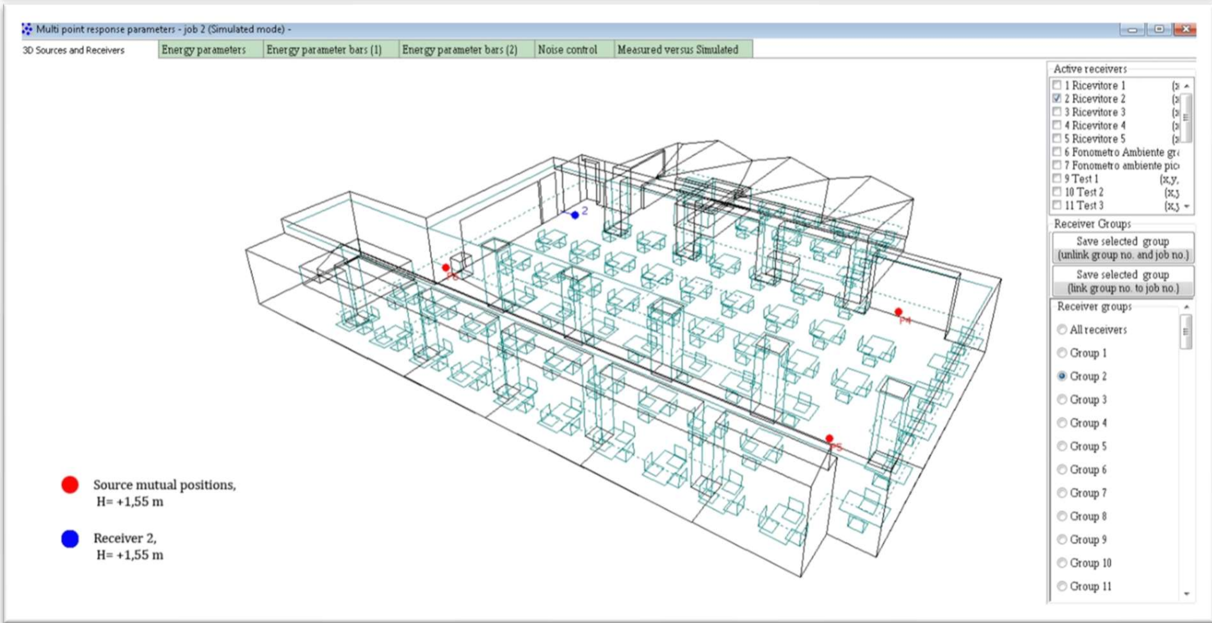
```
1  RECEIVER 1
2  T(20) - 0,57 0,70 0,84 0,79 0,84 0,78 0,59
```

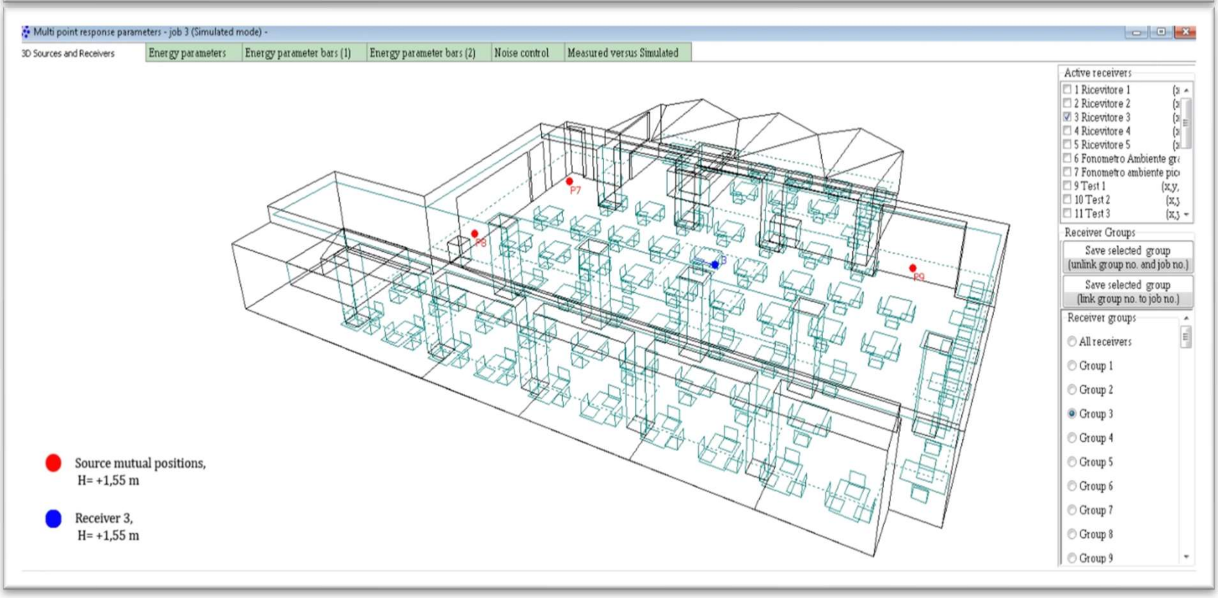
- once the analysis had been completed with the Genetic Material Algorithm it was possible to export an .txt file containing for each octave band the absorption coefficients α of each material assigned to the surfaces and compare the t_{20} for each octave band obtained from the simulations with the data really measured by Sound level meters during the inspection.

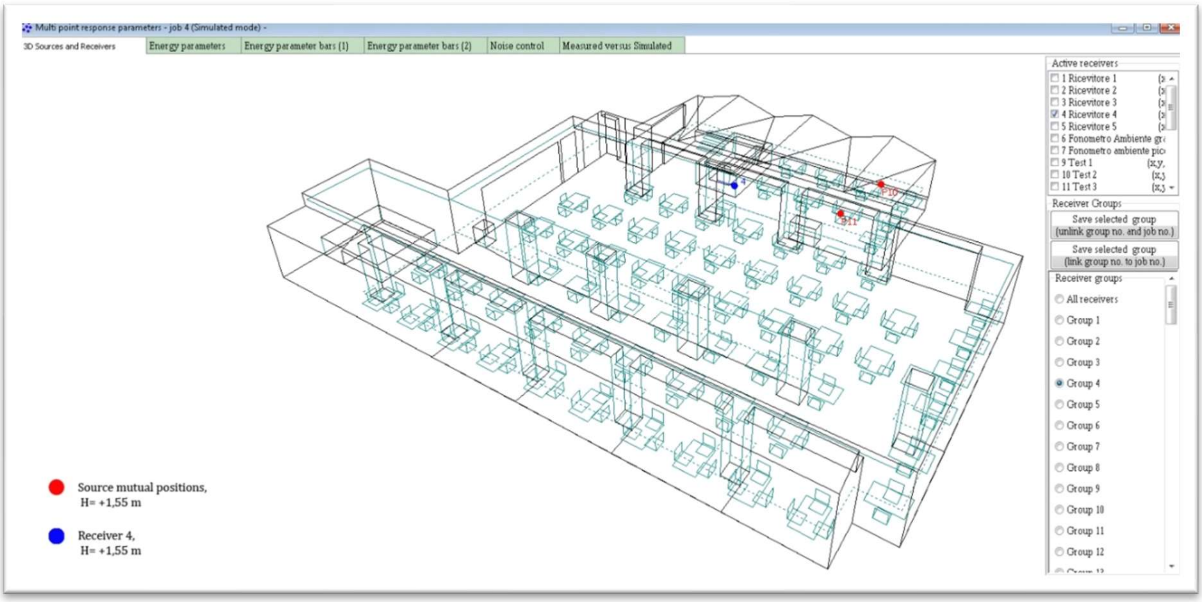


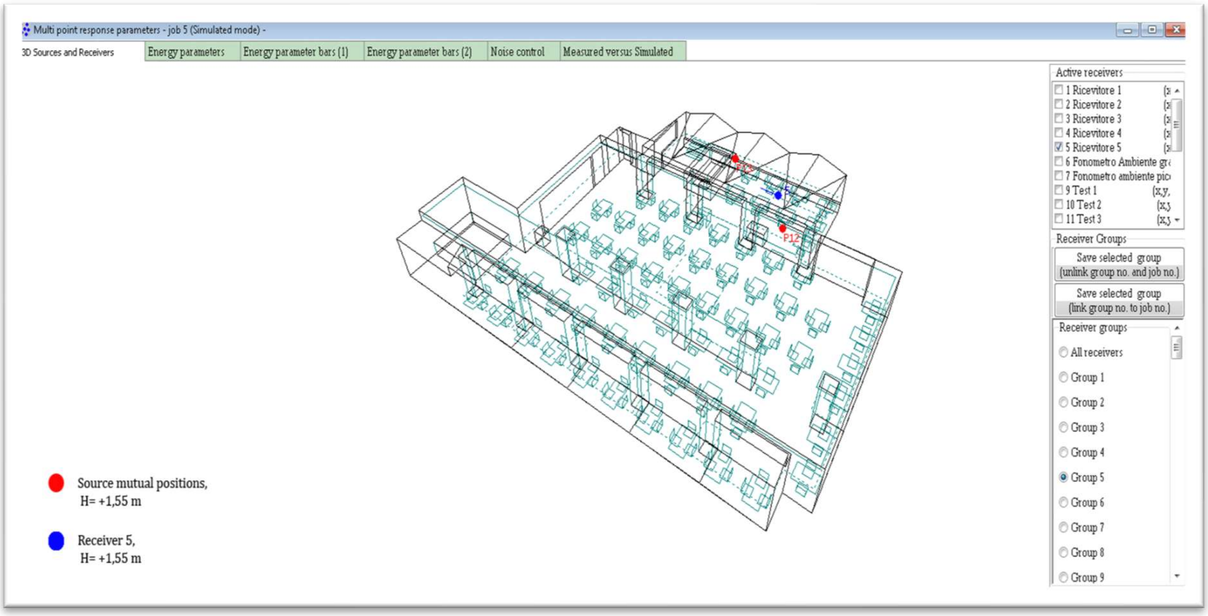
Circoop employees canteen								
Materials	Absorption coefficients as a function of frequency [Hz]							
	63	*125*	*250*	*500*	*1000*	*2000*	*4000*	*8000*
16001 - Muri intonaco	0,01	0,02	0,07	0,01	0,03	0,03	0,02	0,03
16002 - Serramenti doppio vetro 2-3mm con 10mm di camera d'aria	0,02	0,02	0,04	0,03	0,02	0,02	0,02	0,04
16003 - Legno arredi	0,08	0,13	0,12	0,22	0,22	0,09	0,19	0,10
16004 - Piastrelle linoleum	0,03	0,06	0,02	0,05	0,02	0,08	0,03	0,08
16005 - Marmo gradini scale	0,01	0,02	0,02	0,03	0,03	0,05	0,03	0,02
16006 - Metallo arredi	0,26	0,21	0,13	0,12	0,08	0,10	0,07	0,04
16007 - Porta REI	0,14	0,08	0,10	0,09	0,09	0,05	0,11	0,10
16008 - Impianto ventilazione	0,30	0,57	0,63	0,28	0,32	0,45	0,30	0,30

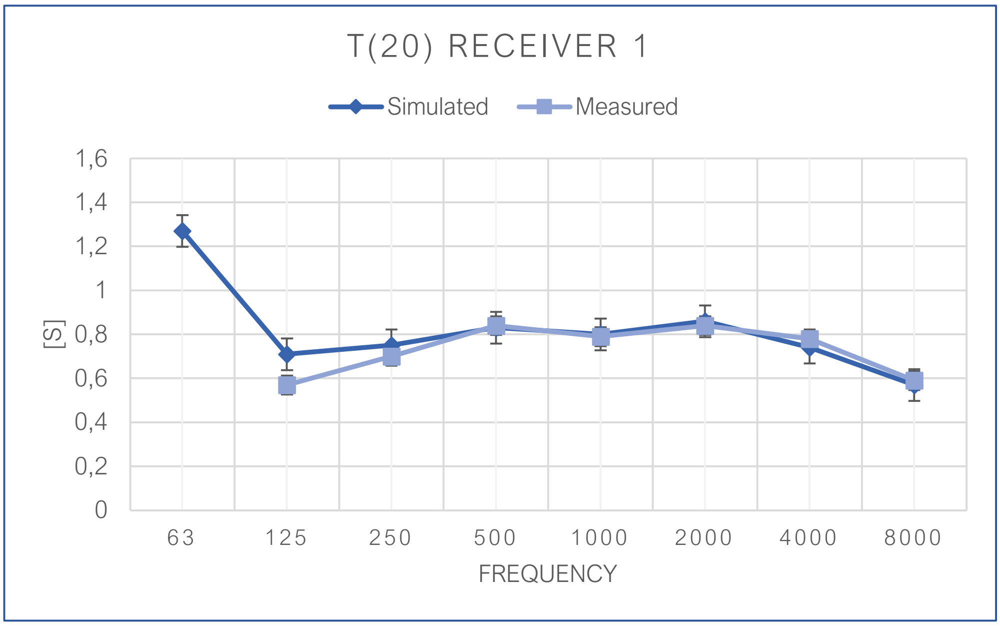

Multi point response parameters - job 1 (Simulated mode) -
3D Sources and Receivers
Energy parameters
Energy parameter bars (1)
Energy parameter bars (2)
Noise control
Measured versus Simulated
Active receivers
1 Ricevitore 1
2 Ricevitore 2
3 Ricevitore 3
4 Ricevitore 4
5 Ricevitore 5
6 Fonometro Ambiente gr
7 Fonometro ambiente pic
9 Test 1
10 Test 2
11 Test 3
Receiver Groups
Save selected group (unlink group no. and job no.)
Save selected group (link group no. to job no.)
Receiver groups
All receivers
Group 1
Group 2
Group 3
Group 4
Group 5
Group 6
Group 7
Group 8
Group 9
Group 10
P2
P3
P1
1
Source mutual positions,
H= +1,55 m
Receiver 1,
H= +1,55 m

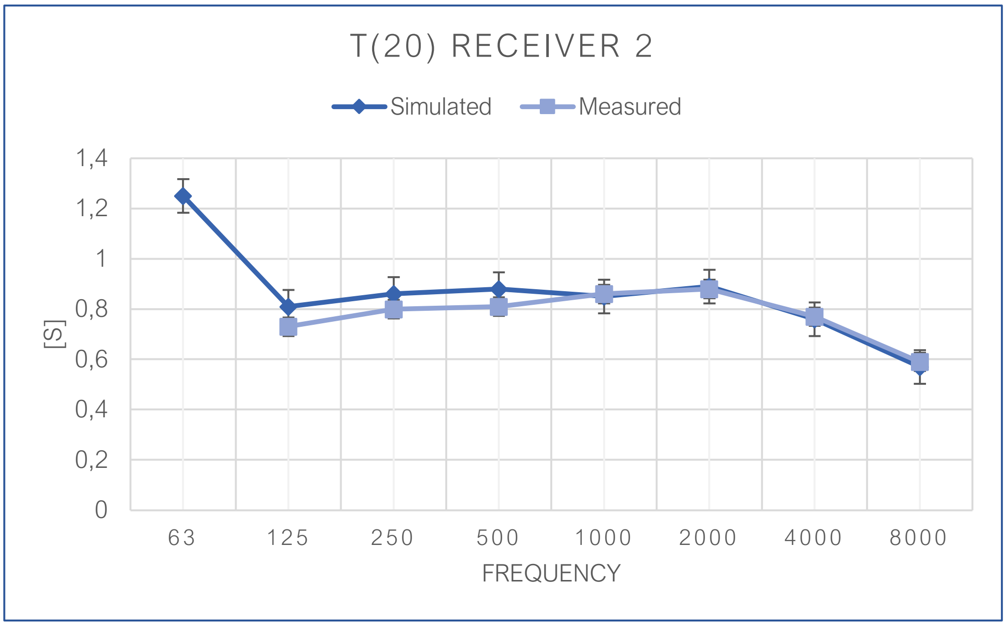

Multi point response parameters - job 2 (Simulated mode) -
3D Sources and Receivers
Energy parameters
Energy parameter bars (1)
Energy parameter bars (2)
Noise control
Measured versus Simulated
Active receivers
1 Ricevitore 1
2 Ricevitore 2
3 Ricevitore 3
4 Ricevitore 4
5 Ricevitore 5
6 Fonometro Ambiente gr
7 Fonometro ambiente pic
9 Test 1
10 Test 2
11 Test 3
Receiver Groups
Save selected group (unlink group no. and job no.)
Save selected group (link group no. to job no.)
Receiver groups
All receivers
Group 1
Group 2
Group 3
Group 4
Group 5
Group 6
Group 7
Group 8
Group 9
Group 10
Group 11
2
P4
P5
Source mutual positions,
H= +1,55 m
Receiver 2,
H= +1,55 m

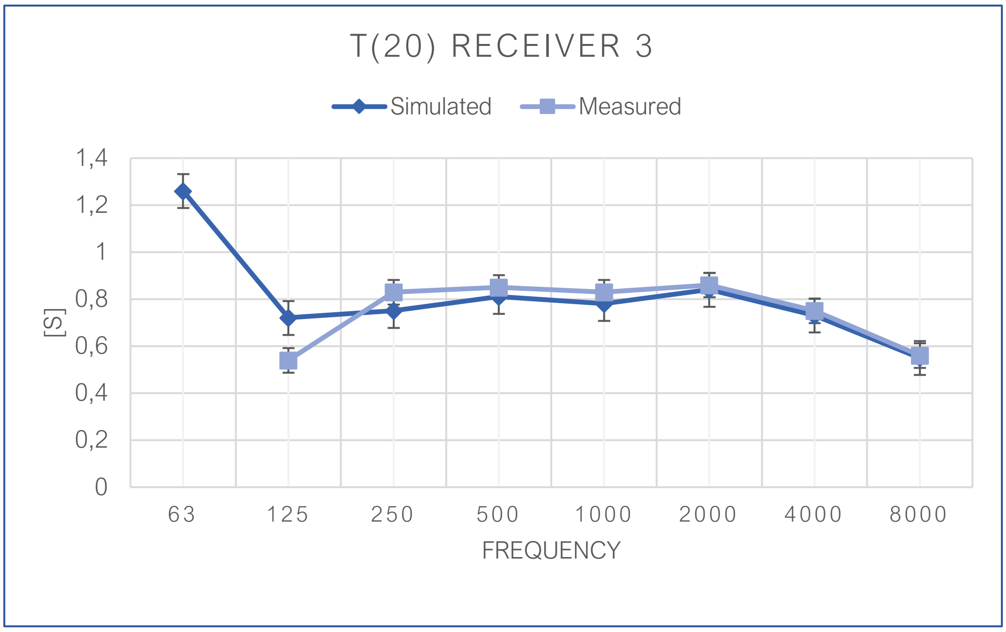

Multi point response parameters - job 3 (Simulated mode) -
3D Sources and Receivers
Energy parameters
Energy parameter bars (1)
Energy parameter bars (2)
Noise control
Measured versus Simulated
Active receivers
1 Ricevitore 1
2 Ricevitore 2
3 Ricevitore 3
4 Ricevitore 4
5 Ricevitore 5
6 Fonometro Ambiente gr
7 Fonometro ambiente pic
9 Test 1
10 Test 2
11 Test 3
Receiver Groups
Save selected group (unlink group no. and job no.)
Save selected group (link group no. to job no.)
Receiver groups
All receivers
Group 1
Group 2
Group 3
Group 4
Group 5
Group 6
Group 7
Group 8
Group 9
P7
P8
P9
Source mutual positions,
H= +1,55 m
Receiver 3,
H= +1,55 m

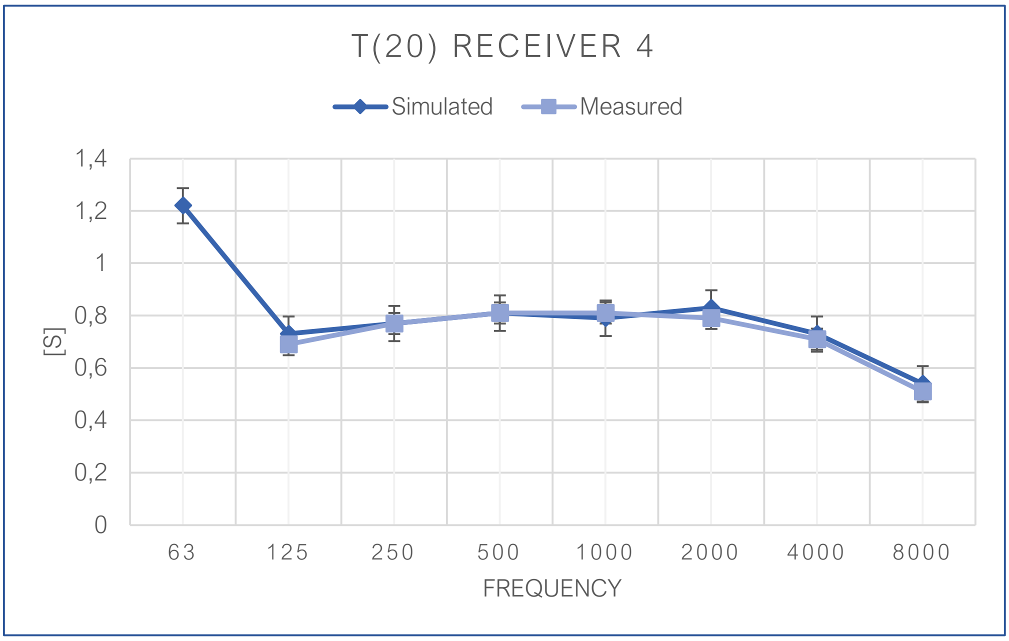

Multi point response parameters - job 4 (Simulated mode) -
3D Sources and Receivers
Energy parameters
Energy parameter bars (1)
Energy parameter bars (2)
Noise control
Measured versus Simulated
Active receivers
Receiver Groups
Save selected group (unlink group no. and job no.)
Save selected group (link group no. to job no.)
Receiver groups
All receivers
Source mutual positions,
H= +1,55 m
Receiver 4,
H= +1,55 m

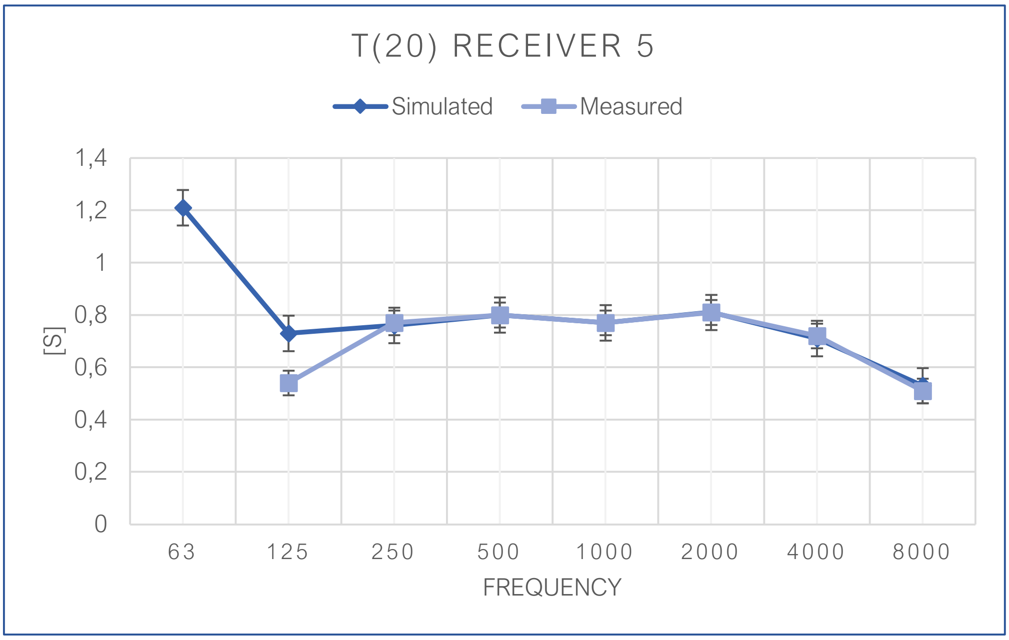

Multi point response parameters - job 5 (Simulated mode) -
3D Sources and Receivers
Energy parameters
Energy parameter bars (1)
Energy parameter bars (2)
Noise control
Measured versus Simulated
Active receivers
Receiver Groups
Save selected group (unlink group no. and job no.)
Save selected group (link group no. to job no.)
Receiver groups
All receivers
Source mutual positions,
H= +1,55 m
Receiver 5,
H= +1,55 m


T(20) RECEIVER 1
Simulated
Measured
1,6
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(20) RECEIVER 2
Simulated
Measured
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(20) RECEIVER 3
Simulated
Measured
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(20) RECEIVER 4
Simulated
Measured
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(20) RECEIVER 5
Simulated
Measured
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY

Simulazione del Lombard Effect e multi surface sources

Noise levels produced within a closed environment such as a restaurant or event room, are difficult to represent within an acoustic modeling software since the noise sources are multiple and as the noise in the environment increases comes into play the Lombard effect so that the sources, the speakers, increase their sound power to ensure a good signal-to-noise ratio SNR. Within the ODEON 16 Combined software, a new method has been developed to simulate the sound power produced by multiple sources through a transparent surface covering the entire area of the analyzed environment located just above the heads of diners in a sitting position, about +1.50 m; this surface, characterized by a series of source points, emits in random mode a large number of sound rays in all directions. The receivers are also placed inside a grid that covers the same area of the surface source and positioned in this case at the ears of diners, about +1.30 m (Figure 34).

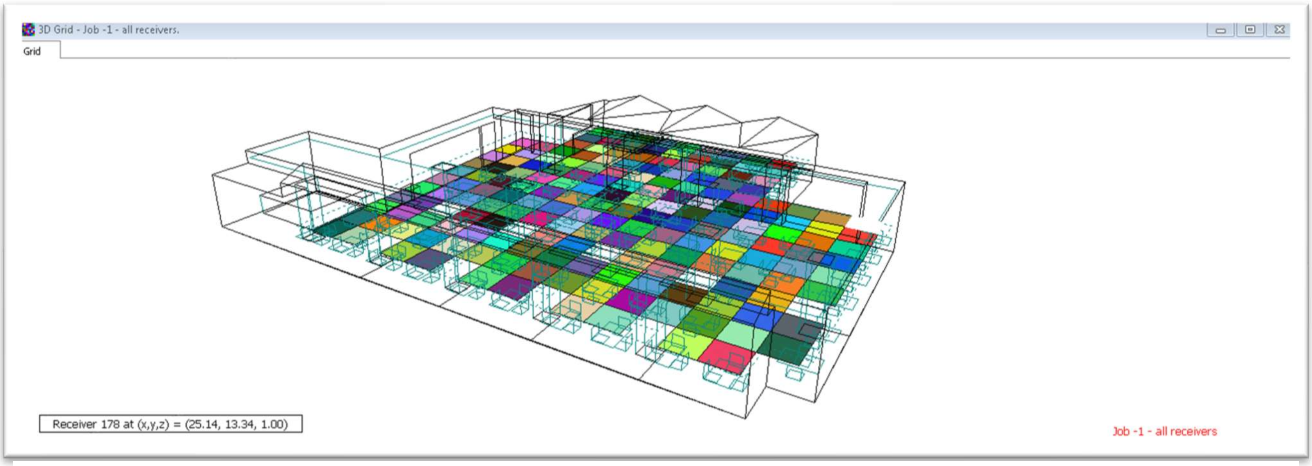


Figura 34 – Receivers grid

The sound power of the emitting surface and the vocal spectrum meet the standards of ANSI 3.5 - 1997 *American National Standard – Methods for Calculation of the Speech Intelligibility Index*, (1997). In addition, calculating the ambient noise levels produced by one or more sound sources, the Lombard effect is taken into account by assigning to the emitting surface an Overall gain value calculated from the reverberation time of the environment to which is added the contribution of each talking person N_s. It should also be considered that the human voice covers a wide spectrum of dynamic frequencies, while in ANSI 3.5 the values of $L_{S,A,1m}$ are represented in only 4 cases: normal, raised, loud and shouted speech. The Surface source method simulates the cocktail effect that is created in densely occupied environments, with at least 50talking people to obtain realistic values, where participants speak simultaneously or in groups in non-schematic mode. This situation is reproduced within the ODEON software using Multi surface-sources, in which the point sound sources are activated in random mode, placing it above the heads of diners at +1,30 m when seated and +1. 50 m if standing. This surface must also be transparent, so material 0, which represents a completely transparent material, must be assigned within Odeon. The direction of radiation and sound must be set as "Both"; for this reason, single surface source should not be used because it is not possible to assign the double direction of surface radiation. Each point source emits sound rays, the ray-tracing method, in different directions following Lambert's law. Within ODEON it is possible to choose whether the diffusion of sound rays from point sources takes place in spherical mode or following lambert's coseno law (Figure 35).

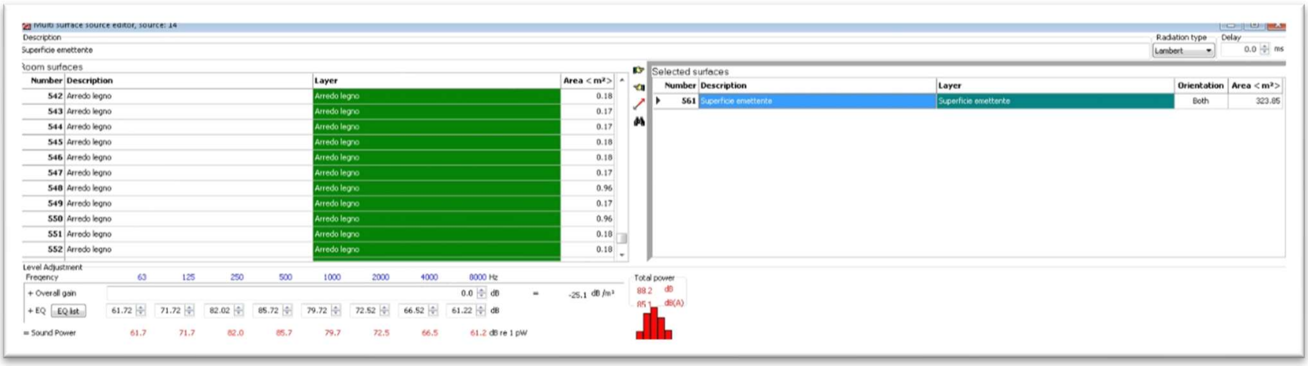


Figura 35 – Multi surface-sources parameters

In the case of Lambertian diffusion, the intensity will be proportional to the angle between the direction of propagation of the radius and the normal to the emitting surface. For this reason, the rays parallel to the surface will have no intensity, while the perpendicular rays will have maximum intensity. In the case of spherical diffusion, the intensity of the rays will be constant regardless of the direction of propagation. As Rindel explains (Rindel J.H. et al., 2012) inside ODEON it was tried to use sound rays that propagated in spherical form around the sound sources, however Lambert's Law returns better results and a possible interpretation of what happened when people are sitting around a table and engage in a conversation with other diners. In this case, the sound rays are all directed towards the inside of the table and not outwards due to the strong directionality of the human voice. For this reason, Lambertian diffusion must be set up. The second step is to choose the vocal effort that you want to assign to the emitting surface to represent the real conditions present within the environment to be analyzed. In the case of a conversation taking place in a not too noisy environment, a "normal" level can be assumed, present in ANSI 3.5 which reports for each vocal effort the corresponding vocal spectra for each octave band from 250-8000 Hz and the weighted sound power levels (A) calculated one meter from the speaker's mouth, $L_{S,A,1m}$ in ODEON this value is referred to as $L_{W,A,1m}$.

The receiving grid must cover the same area as the emitting surface but must be positioned at about -0. 20 from the latter then to +1.10 m in the case of seated diners, +1.30 m if standing. In addition, the receivers inside the grid must be set to a distance between receivers of about 1-1.5 meters interpersonal to ensure anti-Covid distancing. After setting up the receiving grid, it's possible to start a Job in the "Job List" by activating the emitting surface and the receiving grid thus set. After the calculation has been completed, the SPL(A) value at X(50) percent , equal to 51.1 dB(A), must be read on the tab corresponding to the multi surface source. This value represents the noise level $L_{N,A,1}$ generated within the environment by a single talking person (Figure 36).

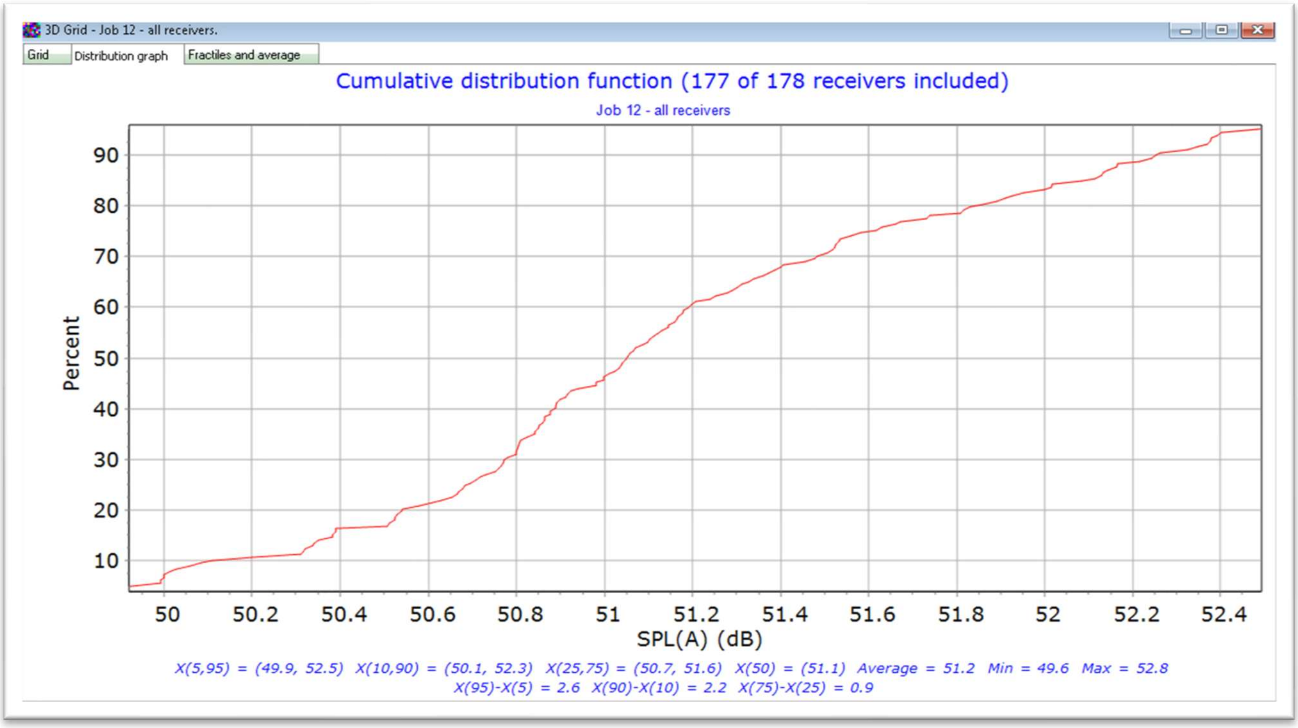


Figura 36 – $L_{N,A,1}$ transfer function

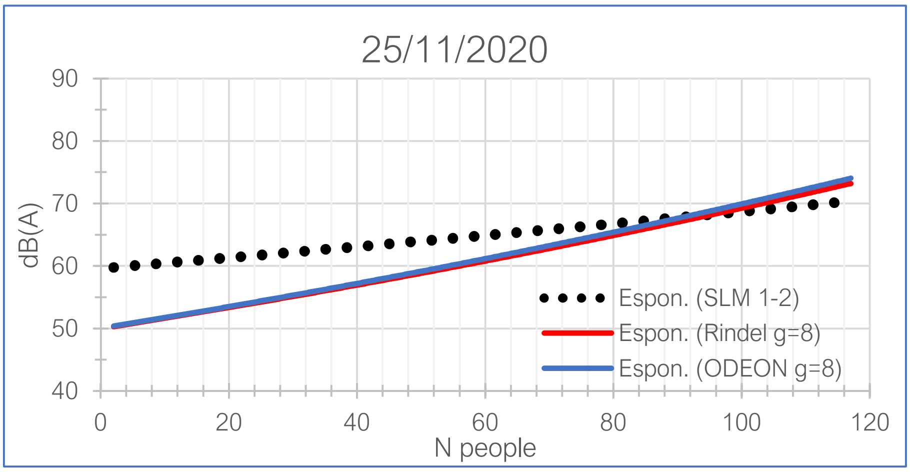
Depending on the type of environment analyzed we can hypothesize the value of g=group size, in the case of Mensa CIRCOOP having tables of up to 2-3 seats but given the low turnout especially in the winter months, a value of about g=6-8 is expected to guarantee a good SNR among all the present diners. To calculate the number of talking people we reintroduce the value N_s=N/g where N=total number of people present within the environment and g=group size taken according to the type of environment analyzed. After assigned values of N, g and the corresponding anthropogenic noise level generated by a single speaker $L_{N,A,1}$ in this case equal to 51.1 dB(A) the simulations must be proceeded by calculating the overall gain (ΔL) to be attributed to the emitting surface in order to simulate the noise levels produced by several speakers at random points in the environment to be analyzed:

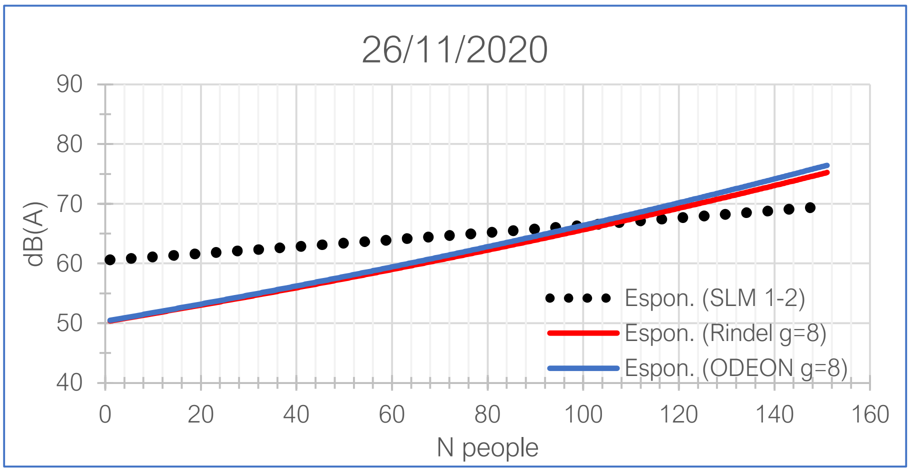
$$\Delta L = 81 + 20\log\frac{N}{g} + L_{N,A,1} - 2 \times L_{S,A,1m}\ [dB(A)] \quad (1)$$

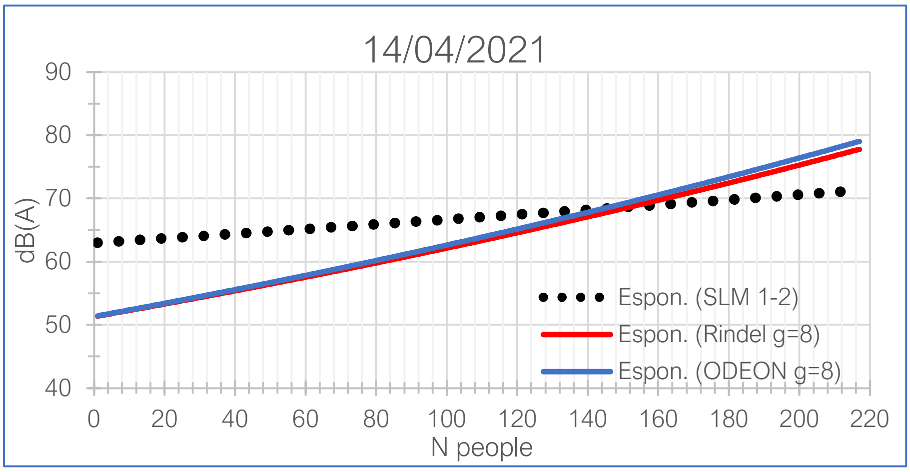
The ΔL value should be entered on the definition board of the emitting surface as an Overall gain value. Once set up it's possible to recalculate the Job and after the calculation will have a new value of $L_{N,A}$ which corresponds to the sum between the initial value $L_{N,A,1}$ obtained from a single speaker to which is added the value ΔL calculated considering a generic number of speakers. The value thus obtained will be:

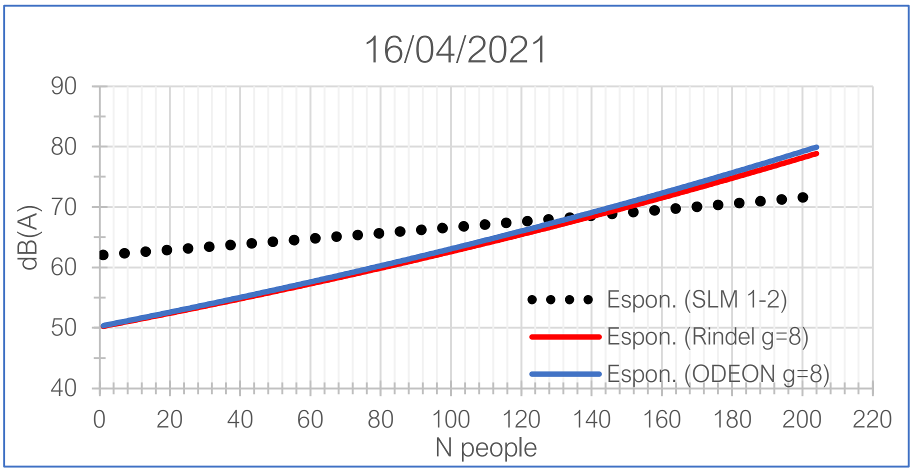
$$L_{N,A} = L_{N,A,1} + \Delta L\ \ [dB(A)] \quad (2)$$

The simulation with multi-surface sources allows to calculate the anthropic noise levels $L_{N,A}$ generated within a closed environment by multiple sound sources activated in random mode and was used as the first method of comparison with the data actually measured during the measurement campaign in the CIRCOOP Canteen on 25-26 November 2020 and 14-16 April 2021. However, it was found that the anthropic noise levels $L_{N,A}$ simulated with multi-surface sources corresponded almost perfectly to the levels calculated with the J. H. Rindel prediction model exposed in the previous chapter, assuming a Lombard slope of c= 0.5 [dB/dB] and the same group size g=8 [-], as can be seen in the following graphs. Its use has therefore been limited in the early stages of this research to compare the data measured by the two phonometers with the simulated data on the Odeon software to confirm or not, the correct calibration of the 3D model.


25/11/2020
90
80
70
60
50
40
dB(A)
0
20
40
60
80
100
120
N people
Espon. (SLM 1-2)
Espon. (Rindel g=8)
Espon. (ODEON g=8)


26/11/2020
90
80
70
60
50
40
dB(A)
0
20
40
60
80
100
120
140
160
N people
Espon. (SLM 1-2)
Espon. (Rindel g=8)
Espon. (ODEON g=8)


14/04/2021
90
80
70
60
50
40
dB(A)
0
20
40
60
80
100
120
140
160
180
200
220
N people
Espon. (SLM 1-2)
Espon. (Rindel g=8)
Espon. (ODEON g=8)


16/04/2021
90
80
70
60
50
40
dB(A)
0
20
40
60
80
100
120
140
160
180
200
220
N people
Espon. (SLM 1-2)
Espon. (Rindel g=8)
Espon. (ODEON g=8)

Calculation of D_2S, rate of spatial decay of sound pressure for distance doubling: ODEON Application Note – ISO 3382:3

The D_2S "rate of spatial decay of sound pressure per distance doubling" is a parameter indicating the rate of decay of the sound pressure level emitted by a speaker for each doubling of the distance from the emitting source. This parameter serves to characterize the acoustic performance of open-space workplaces as required by ISO 3382-3:2012 "*acoustics-measurement of room acoustic parameters – Part 3: open plan offices*"; however, this parameter can also be applied to environments with other uses but in which there are furnishings, e.g. tables, screens etc. that help to reflect the sound rays emitted by the source in some preferential directions of the analyzed space. The calculation principle of D_2S is to measure how sound propagated by an Omnidirectional source with the spectrum "Normal speech" is then perceived by a series of receivers positioned in a straight line at a constant distance from the emitting source. The D_2S can be simulated or measured in the real environment, using an Omnidirectional source positioned at a height of +1.20m to mimic a sound source in a sitting position, or +1.50 if upright, while the row of receivers must also be positioned with the same criterion depending on the situation to reproduce. In this thesis, parameter D_2S was simulated within the Odeon 16 combined version software after characterized the model as described in the previous chapters. To simulate the D_2S, the sources need a "Normal speech" vocal effort, but a standardized directivity pattern named "ISO3382-3_OMNI. SO8" already available in Odeon 16 files (Figure 37).

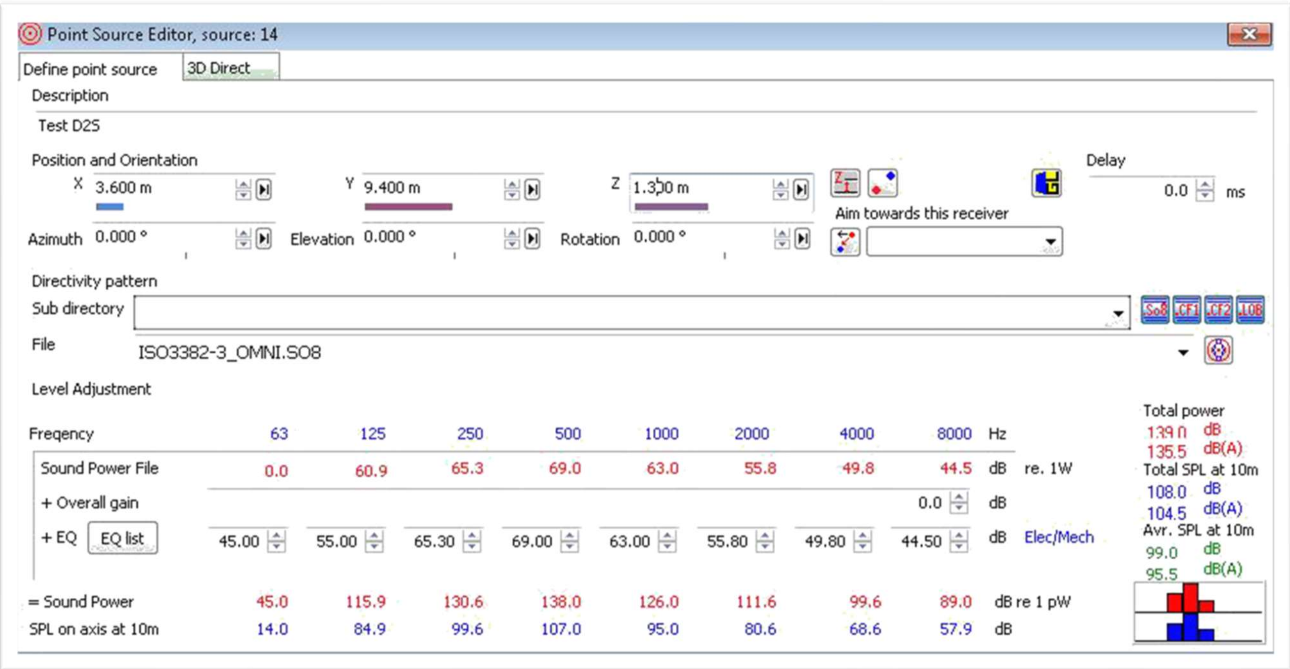


Figura 37 – ISO3382-3_OMNI. SO8

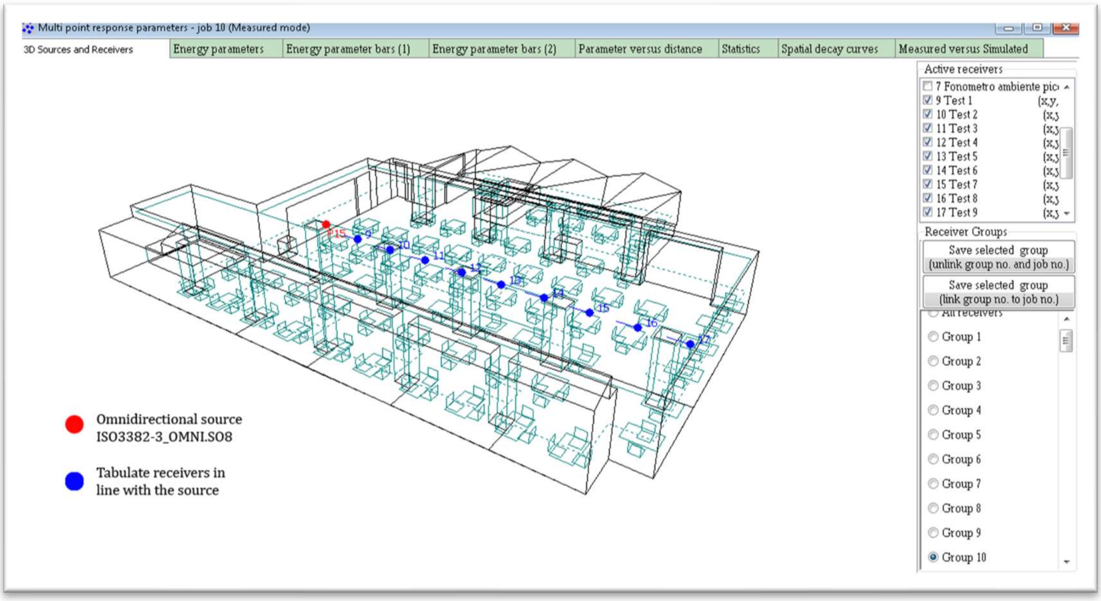


Figura 38 – Source and receivers position for D_2S simulation

The source has been positioned at a height of +1.20 meters to simulate a talking person in a sitting position, being in a Canteen it is expected that the sound sources, the diners, are mainly in a sitting position during meals. The receivers, following the same logic, were also positioned at a height of +1.20 meters in a linear position with respect to the source and arranged in series at a constant interpersonal distance of 2 meters. Following ISO 3382:3 for the calculation of the D_2S it is necessary to place receivers between minimum 4 and maximum 10, a positioning them in line with the source considered. Given the geometry of the environment characterized by a longitudinal extension of about 25 meters, it was possible to place a total of 9 receivers (Figure 38).

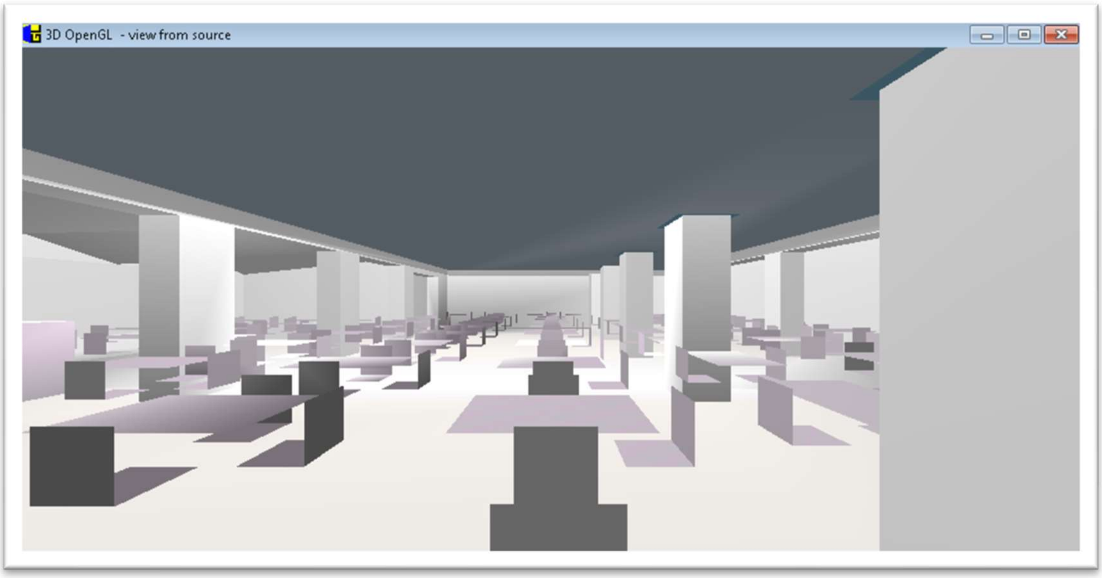


Figura 39 – Ominidirectional Source 3D view

The receivers were then positioned spaced 0.5 meters from the furnishings present in the 3D model and at a distance of 1 [m] from the area bounding surfaces of the environment and/or from very reflective surfaces (Figure 39). Once the source and receivers have been set, it's possible to start a new "Job" in the "Job List" tab by selecting the "Multi-points response" and after the calculation read the value of D_2S at the bottom of the "Energy parameters" tab as attached below (Figure 40).

DL2 parameters - ISO 14257

Band (Hz)	63	125	250	500	1000	2000	4000	8000
DL2 (dB)	2.91	3.77	3.63	3.54	3.61	3.44	3.71	4.31
Corr.	0.99	0.97	0.98	0.98	0.98	0.99	0.99	0.99

```
DL2(A) (125 - 4000 Hz)   =      3.57 (Correlation:       0.99)
DL2(Lin) (125 - 4000 Hz) =      3.60 (Correlation:       0.98)
(Rmin; Rmax) = (       2.02 m;      18.00 m)

ISO 3382-3 parameters              (source is requiered to have speech spectrum)
STI(nearest) = 0.77                (STI at nearest work station)
rD = 37.04 metres                  (distraction distance)
rP = 82.63 metres                  (privacy distance)
D2,S = 3.59 dB                     (Spatial decay rate of A-weighted SPL of speech, 125-8000 Hz)
Lp,A,S, 4 m = 118.61 dB            (A-weighted speech at 4 metres, 125-8000 Hz)
Lp,A,B = -992.04 dB                (A-weighted background noise, 125-8000 Hz)

Looking towards Source: Aiming towards direction -X
Active sources:         1
Rays used:              19440    (Lost: 2 = 0.0 %)
Total ray flights:      2747673
Mean free path:         1.85 m
```

Figura 40 – Energy parameters tab

The significant value of D_2S is the median value between frequencies of 500-1000 Hz, in this thesis a value of D_2S=3.6 [dB] has been considered. This simulated parameter has been adopted in the forecast model derived from the D. D'Orazio model described in the previous chapter, to calculate in the non-diffuse field, what is the rate of sound decay as you move away from the issuing i-th source.

4.4 | ESTIMATION OF VALUE r_{ij} AND SIMULATION IN GRASSHOPPER FOR RHINOCHEROS

The last unknown parameter required to apply the forecast models presented in Chapter 2 is the mutual distance between a simulated i-th point within the environment and one of the two receivers, the SLM 1 and SLM 2 phonometers, used during the measurement campaign. In real cases the value r_{ij} cannot be measured because in a closed space the points, in this case the speakers, are distributed in unpredictable mode so that this value must necessarily be simulated. However, an estimate of r_{ij} values can be made by considering 2D space as a chessboard in which each i-th row and j-th column represent the possible positions of speakers in the space considered. Remembering that in a closed environment the weighted anthropic noise level A, generated by a number N_s of sound sources, can be calculated as:

$$\mathrm{L_{N,A}} = \mathrm{L_{W,A,i}} + 10\log \mathrm{N_s} - 10\log\left(\frac{\mathrm{A_{tot}}}{4}\right) \quad [\mathrm{dB(A)}] \quad (1)$$

From which it appears that the corresponding A-weighted anthropic noise level calculated in the position of a generic j-th occupant will be:

$$L_{N,A,j} = 10log\left[\sum_{i=1}^{n} \frac{W_i}{W_0}\left(\frac{Q_i}{4\pi r_{ij}^2} + \frac{4}{A_{tot}}\right)\right] \quad [dB(A)] \quad (2)$$

where:

1. i= number of active sources;
2. j= number of receivers;
3. r_{ij}= mutual distance between the source i-th and j-th receiver.

Taking up the idea of imagining space as a chessboard it is possible to calculate the distance r_{ij} with the formula of the circumference radius for which we will have:

$$\mathrm{r_{i,j}} = \sqrt{\left\{\left[\mathrm{j} - \left(\frac{\mathrm{m_c} - 1}{2}\right) \times \Delta_\mathrm{c}\right]^2 + \left[\mathrm{i} - \left(\frac{\mathrm{m_r} - 1}{2}\right) \times \Delta_\mathrm{r}\right]^2\right\}} \quad [\mathrm{m}] \quad (3)$$

where:

- m_r x m_c= boardsize [m];
- Δ_r=line i-th;
- Δ_j=line j-esima

Sawhereas the values of the anthropic noise level $L_{N,A}$ changes depending on the distance between the source and the receiver in question, being r_{ij} inversely proportional to $L_{N,A}$ and imagining space as a chessboard, it's possible to get that the noise level $L_{N,A}$ increases faster when first fill the spaces of the board closest to the center, on the other hand a more moderate accretion occurs when is firstly fill the spaces closest to the perimeter being the major source-receiver distance. Applying these concepts Tang (Tang S. K. et al., 1977) showed that the average values between the case when the space fills up along the edges and the case when it fills first in the center, returned values comparable to the truly measured data of $L_{N,A}$ in a staff canteen at Polytechnic University in Hong Kong. However, a simplification of what has been described has been possible by writing a special algorithm that would allow to calculate the different values of r_{ij} as the number of people within the environment in which to apply the predictive models increases/decreases. In this thesis the parameter r_{ij}= mutual distance between the same source and j-th receiver was simulated using the Grasshopper application developed within the Rhinoceros 7 software. The environment of CIRCOOP canteen was imported in 2D on Rhinoceros and inside a series of points were placed at seated really available within the environment. The 2 points representing the SLM 1 and SLM 2 phonometers were also placed at the same points where the phonometers had actually been placed during the measurement campaign. In Rhinoceros model it's possible to simulate how seats are gradually occupied in random mode by increasing the number of people in the simulated environment. The algorithm for simulating the growth/decrease of places occupied in Grasshopper has been written as follows (Figure 41):

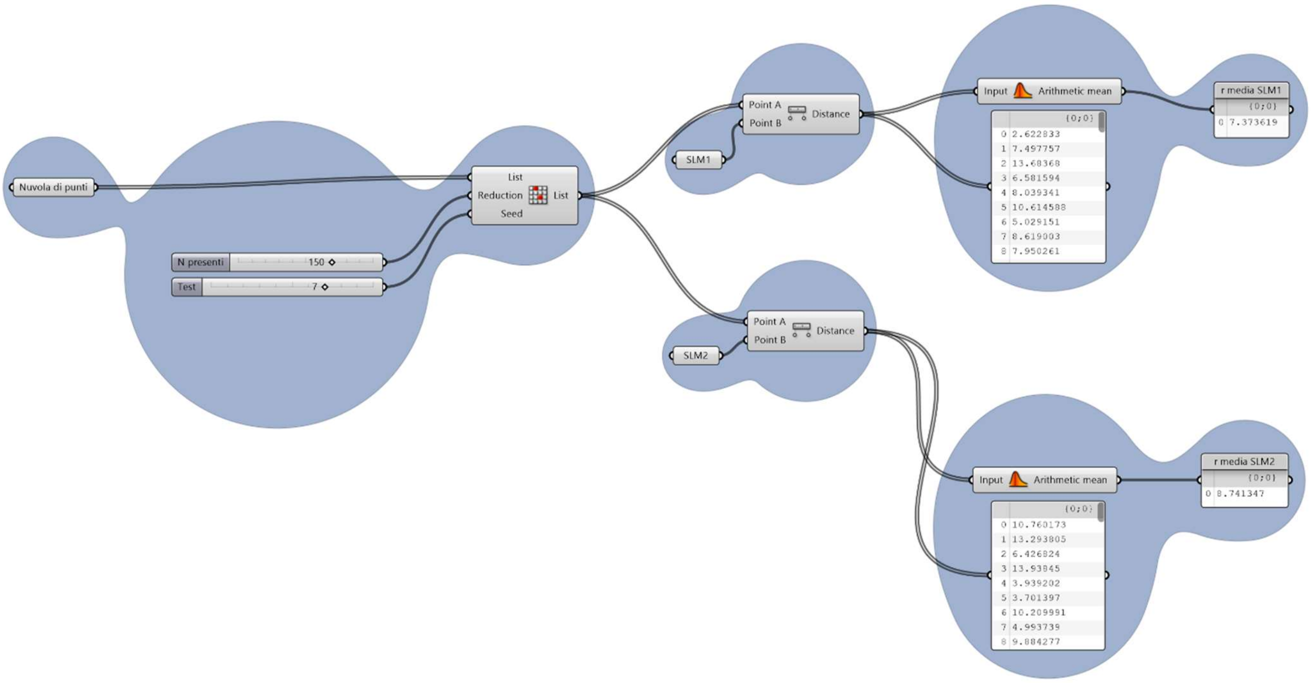


Figura 41 – Grasshopper algorithm for r_{ij} simulation

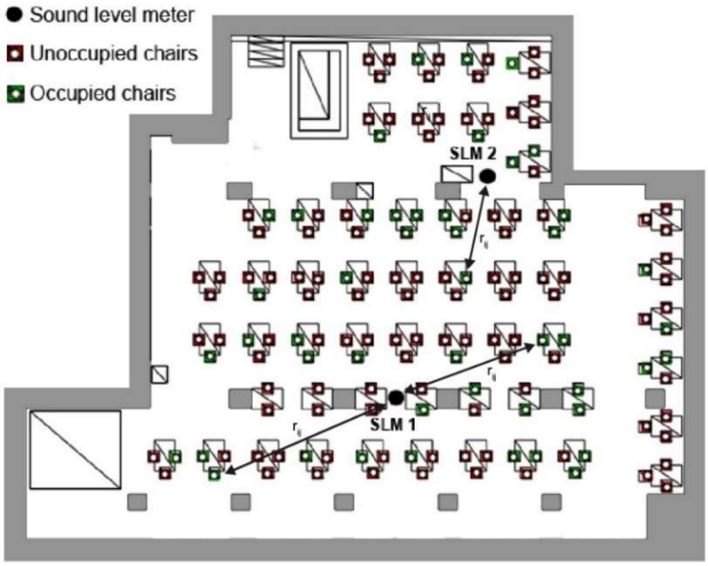


Figure 42 – Simulation of N in Grasshopper

The point cloud represents the total number of places available within the Mensa environment and by moving the slider "N present" it is possible to increase or decrease the occupancy rate of the environment by setting min and max values according to the capacity of the analyzed environment. In the case of Mensa CIRCOOP, the maximum value corresponds to the total number of seats. The third command indicated by a checkerboard icon is the "random reduce" and allows to fill/empty the space in non-schematic mode so as the number of people increases the points in the space fill randomly, changing color from red=unoccupied to green=occupied (Figure 42-43). At the same time, the algorithm calculates for each i-th point the mutual distance between points and the two phonometers SLM 1 and SLM 2. Finally, for convenience in reading data, the algorithm calculates for each mutual distance r_{ij} by arithmetic mean.

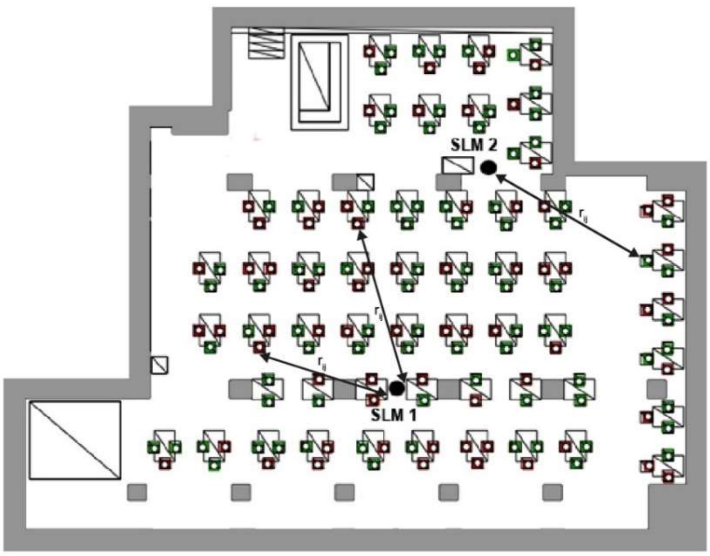


Figure 43 – Simulation of N in Grasshopper

5 | VALIDATION OF PREDICTIVE ALGORITHMS

The data of L_{N,A,dt_eq} collected by the two NTI Audio XL 2 with NTI Audio M2211 microphones during the 4 days of monitoring on 25-26 November 2020 and 14-16 April 2021 and the corresponding $N(t)= count_{out}(t) - countin\ (t)$ processed through the infrared sensor people counting with single photocell calibrated by the student Perrone Gabriele and the research fellow of the Polytechnic of Turin Riente Fabrizio, were ordered and elaborated on Excel as described in Chapter 4 to hypothesize a possible correlation between the anthropic noise levels $L_{N,A\ and}$ the corresponding values of N=number of people. The data was then analyzed using the free R version 3.1.2 software invented by Canadian mathematician Robert Gentlemen for statistical data analysis. Linear models have been plotted by inserting the values of L_{N,A,dt_eq} on abscisses and on ordinates the corresponding values of N(t) (Figure 44).

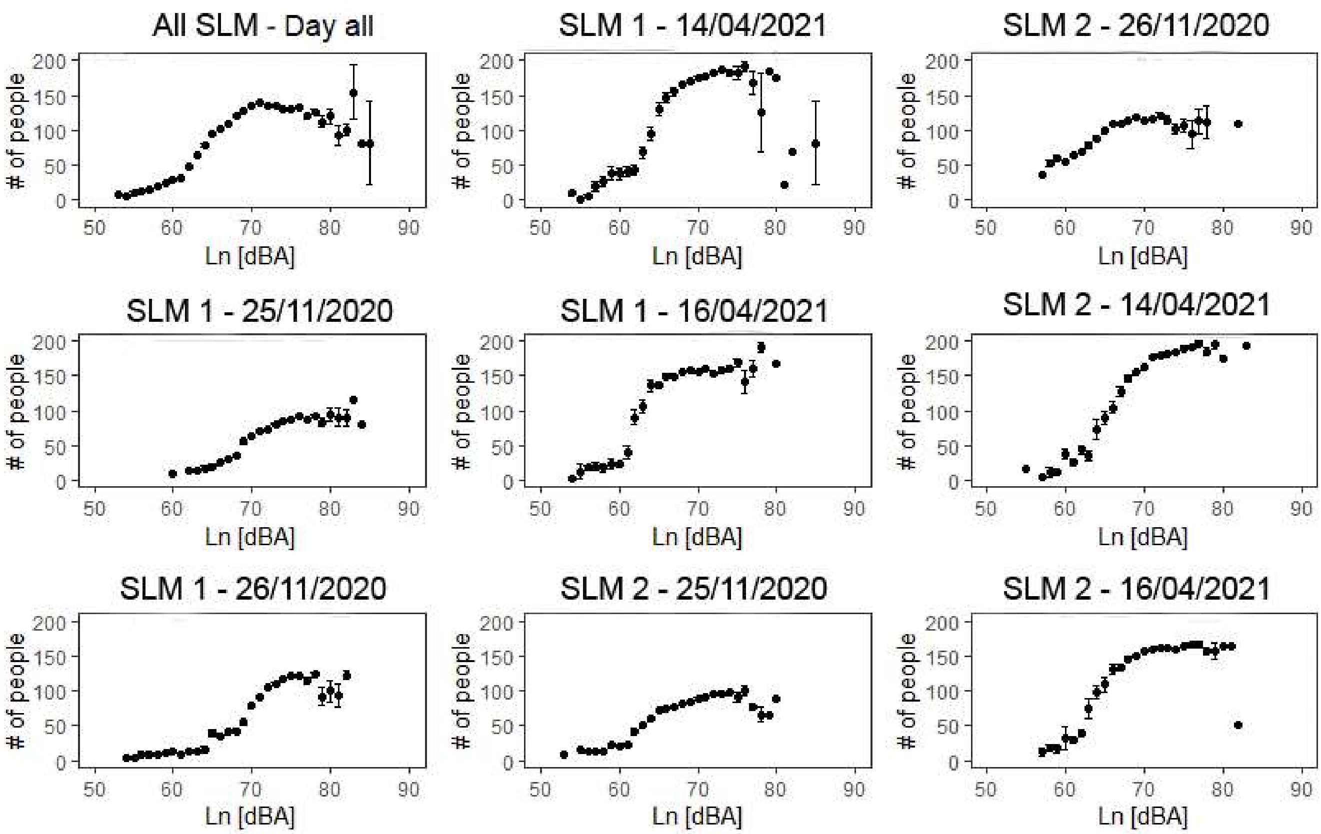


Figure 44 – Linear models with L_{N,A,dt_eq} and N(t) counter

Segmented and break-point function

As can be seen in the previous graphs, to a given value of L_{N,A,dt_eq} corresponds an occurrence of less variable N values up to about 75 dB(A) as had already been indicated in Chapter 4, while for values above this threshold a same value of L_{N,A,dt_eq} corresponds to many values of N(t). For this reason, within the R software version 3.1.2 the simulation proceeded by applying the "segmented or broken-line" function that allows to create a "regression model with segmented" model with which can find the two, or more lines, that better approximate the model data. In segmented models, data approximation lines must be constrained to one, or more points called break points, which correspond to the points where the relationships between the response variable and the predictor variable change. The break-point are change-points in which there's a variation in the relationship between the response variable and the predictor variable. At the point where we find a break-point it is assumed that the average value between the two segmented and for this reason the average between the two slopes is constant. If the slope of the first segmented is zero and consequently the first break-point is a point with zero value on the axis of ordinates, this break-point point can be considered as a starting point in the approximation of the linear regression model. Applying the segmented function to the general linear regression model containing the 4-day monitoring data, only one significant break-point was found at the L_{N,A,dt_eq}=70.25 dB(A) value where a net change in slope of the two linear regression lines is visibly (Figure 45).

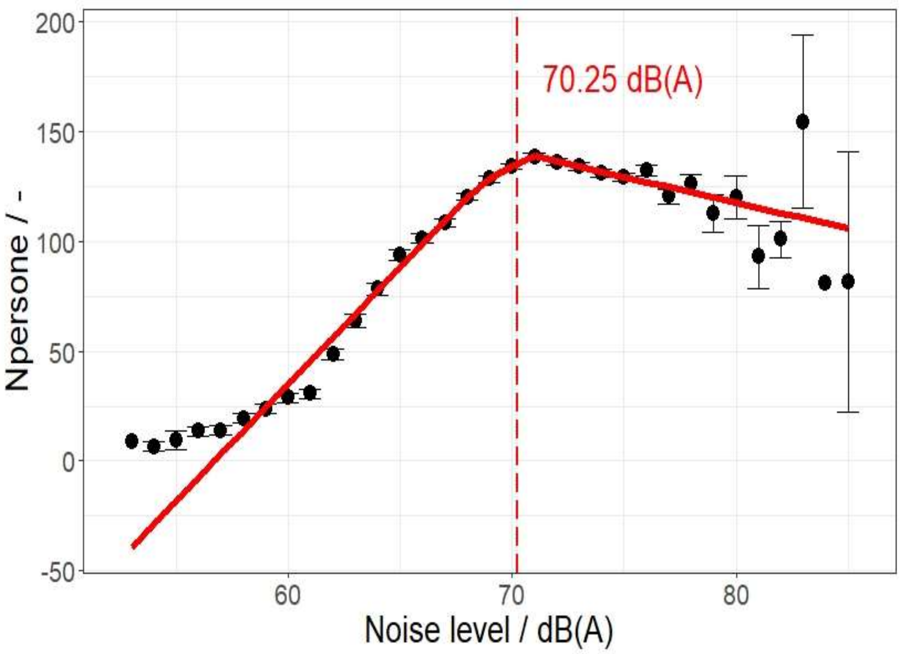


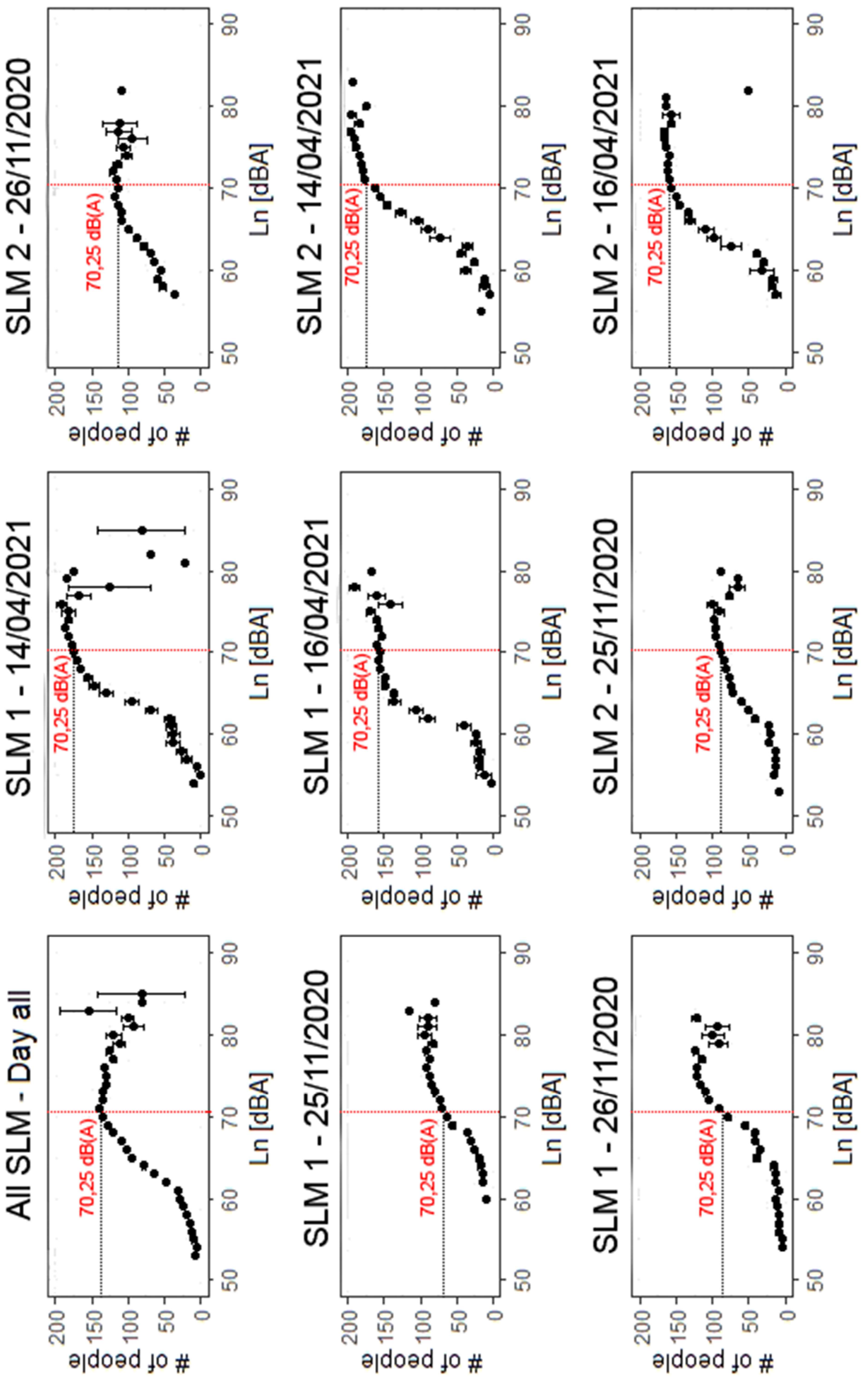
Figure 45 – Application of segmented function

In the segmented function report it's possible to read that the intercept of the function is -570.4 and the angular coefficient is Ln=10.12 so will have that for each increase of one unit of the anthropic noise levels L_{N,A,dt_eq} the number of people N(t) will increase by 10. This relationship is valid up to the break-point beyond which it occurs that for each increase in L_{N,A,dt_eq} the corresponding value of N(t) will be given by the sum of (Ln+U1Ln), being U1Ln= -12.45 for each increment of L_{N,A,dt_eq} so it will be a decrease in the number of people by about -2 units (Figure 46).

```
***Regression Model with Segmented Relationship(s)***

Call:
segmented.lm(obj = my.lm, seg.Z = ~Ln, npsi = 1)

Estimated Break-Point(s):
          Est. St.Err
psi1.Ln 70.243  0.122

Meaningful coefficients of the linear terms:
             Estimate Std. Error t value             Pr(>|t|)
(Intercept) -570.3903    12.4786  -45.71 <0.0000000000000002 ***
Ln            10.1198     0.1870   54.11 <0.0000000000000002 ***
U1.Ln        -12.4587     0.3789  -32.88                   NA
```

Figure 46 – Segmented function report

It was therefore found that, beyond the break-point despite the increase in noise levels L_{N,A,dt_eq} recorded during the 4 days of monitoring, the number of people speaking simultaneously N s(A) tends to decrease perhaps due to the high background noise which results in an ever-decreasing number of simultaneously speaking people N_s. The break-point at 70.25 dB(A) corresponds to a value of about N=140 people, beyond which a shear filter has been applied for all values above that threshold.



All SLM - Day all
SLM 1 - 14/04/2021
SLM 2 - 26/11/2020
SLM 1 - 25/11/2020
SLM 1 - 16/04/2021
SLM 2 - 14/04/2021
SLM 1 - 26/11/2020
SLM 2 - 25/11/2020
SLM 2 - 16/04/2021
70,25 dB(A)
# of people
Ln [dBA]

Poisson distribution

The linear model thus filtered was approximated using the Poisson distribution since our case study is based on countable data represented by the number of people and the statistical model that best approximates the cumulative data is the Poisson distribution (Figure 47).

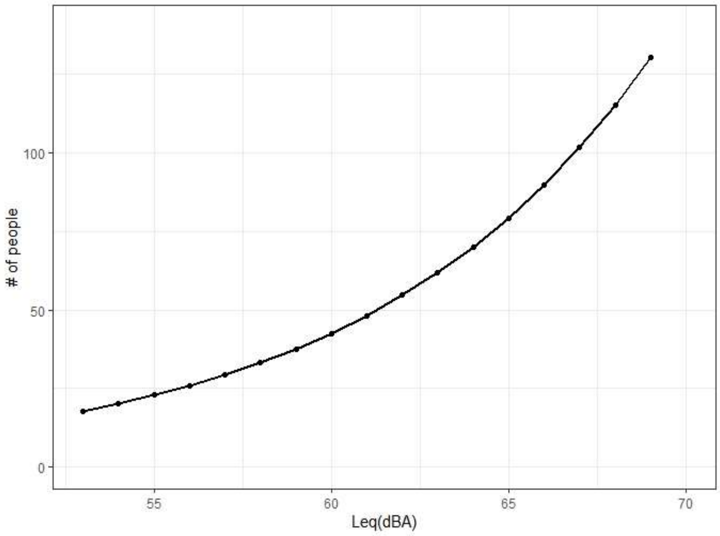


Figure 47 – Poisson distribution of all SLM

In the Poisson generalized model, the distribution of data is exponential, so the Ln value in the model report thus created no longer represents the angular coefficient as in the case of the linear regression of the segmented function. but it tells us that the generalized Poisson distribution model with data filtered beyond the break-point at 70.25 dB(A) is significant, so the relationship between the recorded L_{N,A,dt_eq} levels and the number of N(t) monitored through people counters is significant.

Summary of parameters from Odeon and r_{med} in Grasshopper

The values to insert in input into the two predictive models for the control of enclosed spaces, as described in Chapter 4 on acoustic simulations and grasshopper software, are in the following table:

c=Lombard slope	ap	t_0	Volume	D_2S	r_{med}
0,50	0,40	0,82	1115,00	3,60	3,66

Application of predictive models to Poisson distribution

By applying to the generalized Poisson distribution model the two prediction models for the control of the gatherings derived from the predictive models of J. H. Rindel (1) and D. D'Orazio (2) it was possible to compare the trend of the values actually measured with the values returned by the 2 models.

$$N = \frac{0{,}16 \times V}{t_0 \times \left\{10^{\left[\frac{69-45\times c-L_{N,A}\times(1-c)}{10}-\log_{10}(g)\right]} - ap\right\}} \quad [-] \quad (1)$$

$$N = 10^{\left[\frac{L_{N,A}\times(1-c)-55+45\times c+3{,}32\times D_{2S}\times\log(r_{ij})}{10}+\log(g)\right]} \quad [-] \quad (2)$$

As explained in the previous chapters, in the two forecast models, the only unknown values are the g=group size value and the c=Lombard slope that depend heavily on the type of environment in which the forecast model is to be applied. In the case of the CIRCOOP Canteen where the levels of L_{N,A,dt_eq} and the corresponding N(t) were monitored during the 4 days of the monitoring campaign, it was found that the values closest to the generalized Poisson distribution model were:

- c=0.50 falls within the range of values found by J. H. Rindel for the application of the forecast model in school cafeterias, where a c=0.4-0.5 dB/dB was adopted;
- g=8 is a higher value than the range of values found by J. H. Rindel, in fact in his studies conducted in school cafes there is usually a g=2-4.

The value of g=8, which best approximates the forecasting models for the control of gatherings indoors with the generalized Poisson distribution model, is an anomalous value compared to the range of values found in the previous studies of J. H. Rindel, Navarro and Pimentel and Hodgson, which shows that in catering environment the group size takes values of about g=2-4. It can be assumed that such a high value of g=group size is a direct consequence of the restrictions imposed from the D.P.C.M. 26 October 2020, n.265 " *Urgent measures to contain contagion throughout the national territory*", EE. Remembering that the group size $g=N/N_s$ derives from the Theory of Circles set out in Chapter 1 of this thesis and represents the average value of people speaking simultaneously within a group, it is easy to guess that a value greater than g, then a g=6-8, implies fewer people speaking simultaneously within a closed environment. In this study the value g=8 implies that within the Mensa CIRCOOP environment the number of N_s speakers is a small number compared to the values normally measured in the catering environments analyzed in the studies before the Covid-19 pandemic. As a result, a g=8 tells us that in the 4 monitoring days there was a lower turnout than in ordinary regimes

and at the same time, due to the restrictions imposed by the D.P.C.M. 265/20 and in particular the impossibility of sitting in more than 2-3 people per table has meant that the levels of L_{N,A,dt_eq} recorded were much lower than the levels normally produced in the same environment and consequently the group size takes on higher values, doubling them, than the range of g=3-4. Applying these g=group size and c=Lombard slope values to the prediction models for gathering control, it's possible to see that the J. H. Rindel is better approximated to the data actually measured, while the D. D'Orazio model has a good approximation for data up to 60 dB(A) corresponding to about 45 speakers simultaneously within the closed environment, while beyond that value the data returned by the prediction model thus set tend to deviate from the data actually measured (Figure 48).
However, it is possible to refine the prediction model derived from the D. D'Orazio model by applying a correction to Lombard slope value, assuming a c=0.48 that increases the slope of the curve so as to make it almost superimposed on the Poisson curve of the data actually measured by the two sound level meters (Figure 49).

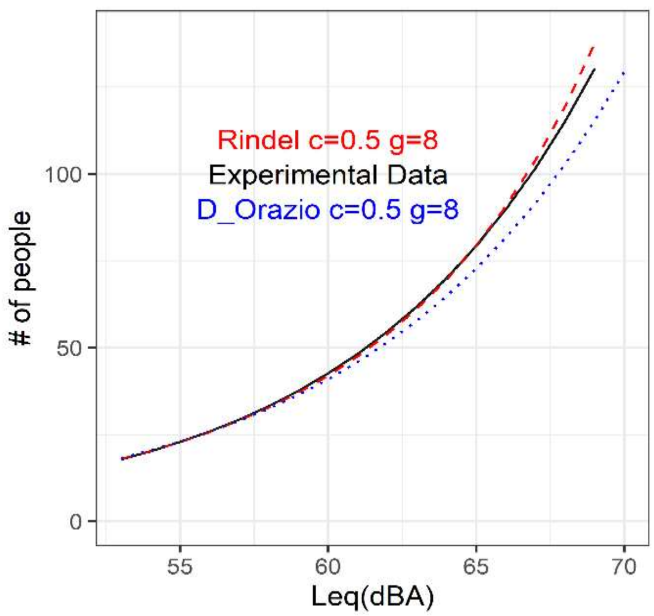


Figure 48

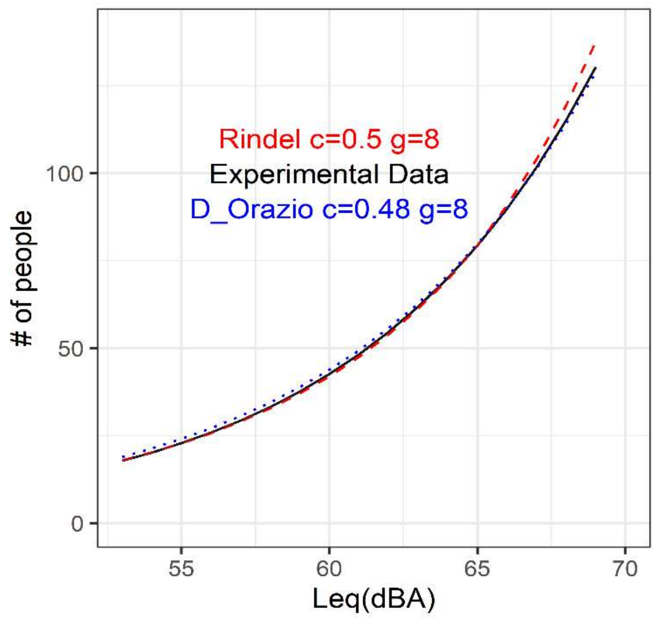


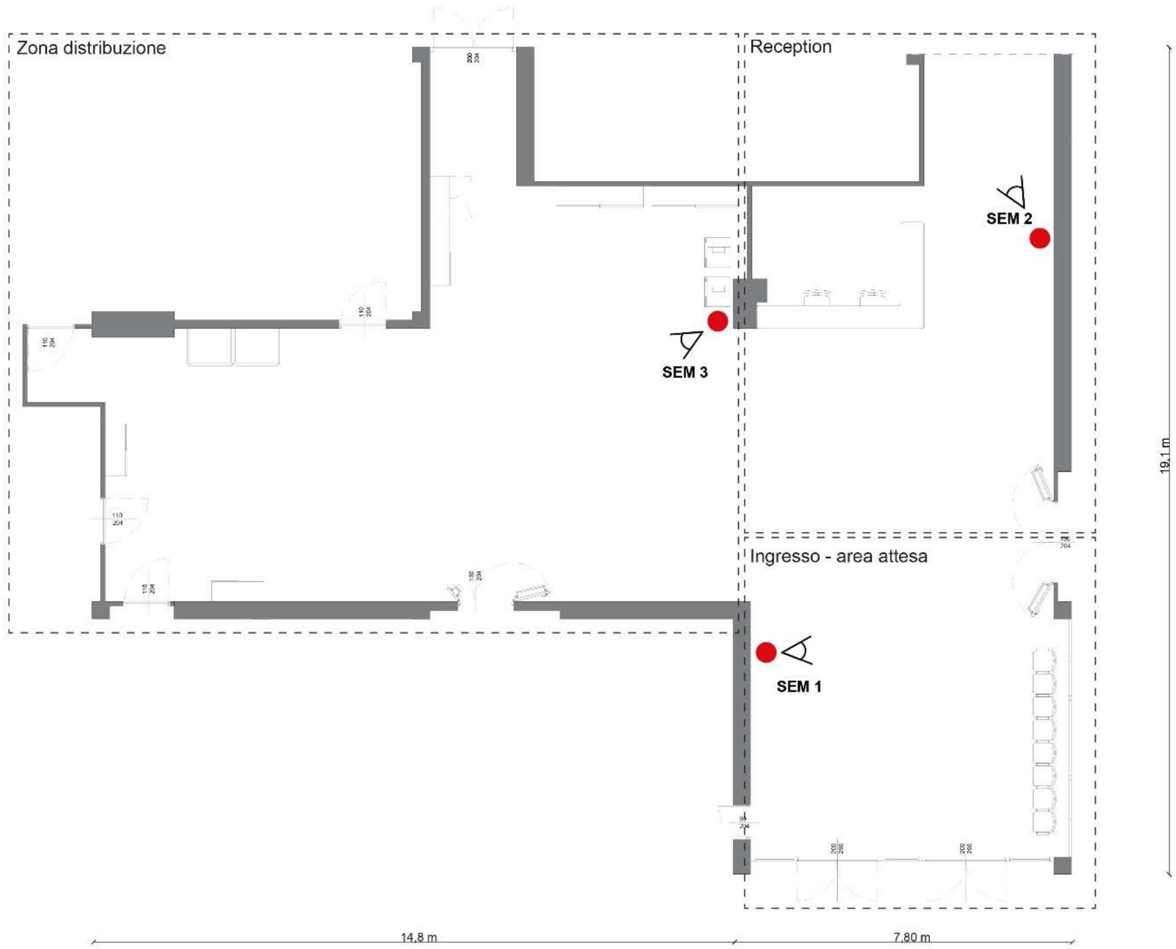
Figure 49

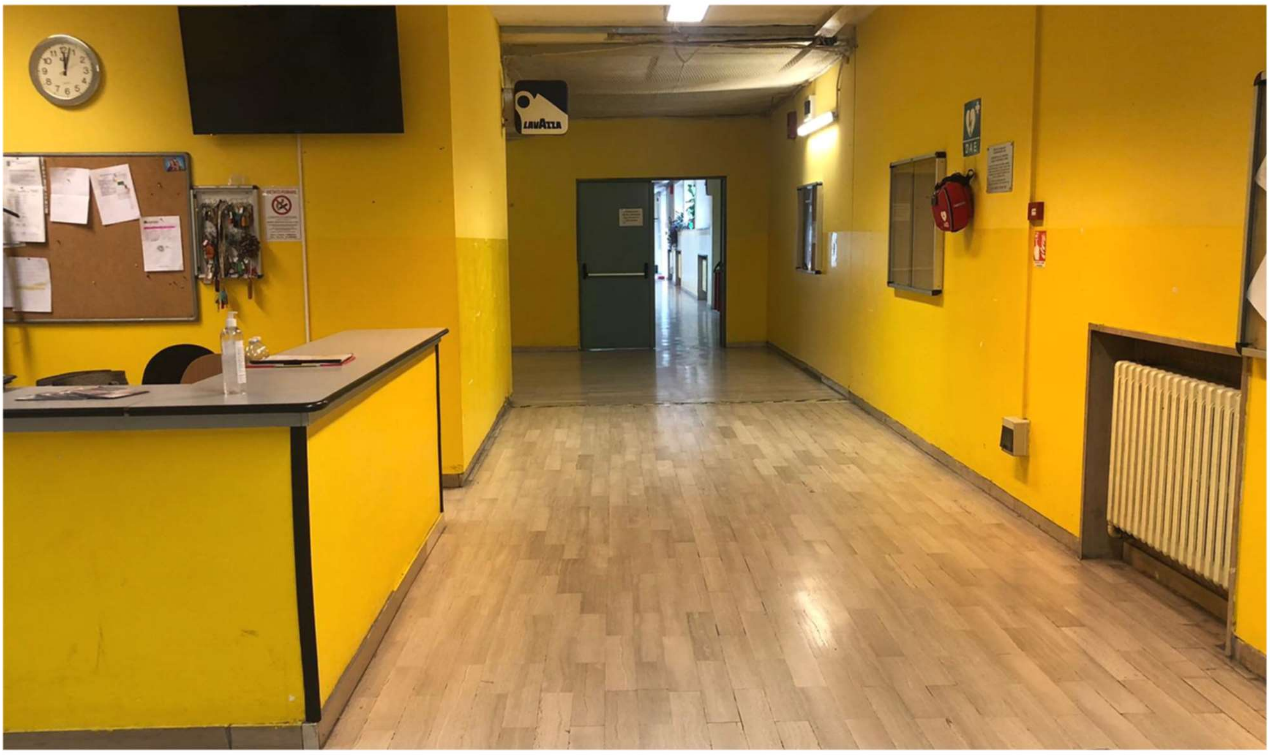
6 | APPLICATION OF FORECASTING ALGORITHM IN SEM ENVIRONMENTS

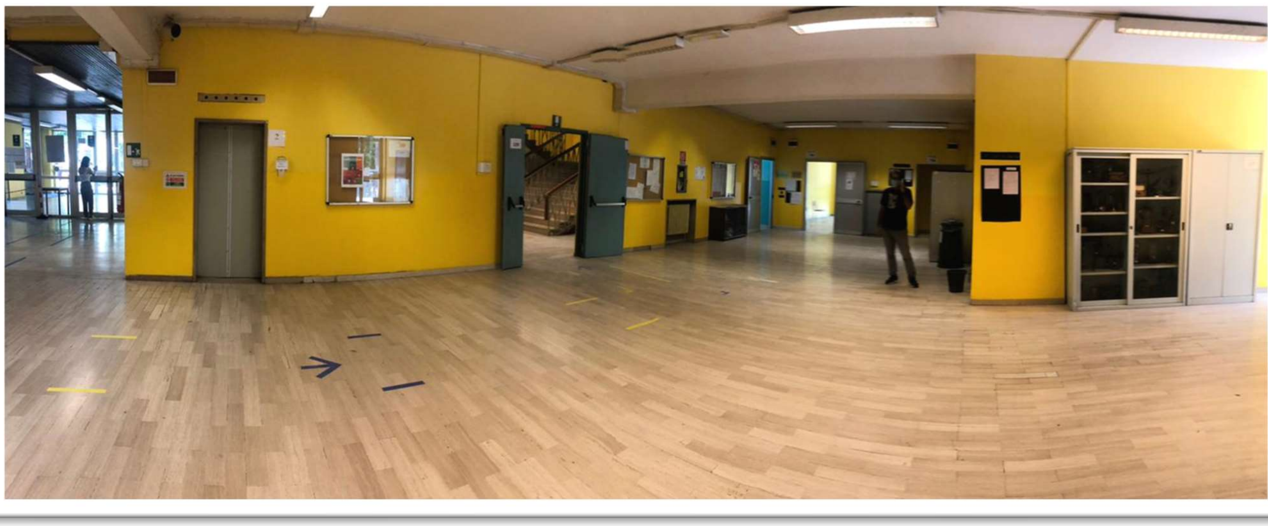
The work carried out in this thesis aims at the future application of the 2 forecasting models derived from the models of J.H. Rindel and D. D'Orazio in environments with a different intended use from the catering premises, including 2 high schools and 2 museums of the city of Turin where it is planned to install the SEM (speech and sound SEMaphore) devices designed by the DENERG of Polytechnic of Turin introduced in chapter 1 of this thesis, updating the calculation algorithms in order to allow the devices to monitor in real time the number of people present in closed environments starting from the levels of anthropic noise $L_{N,A}$ measured, in order to prevent gatherings in closed places and counteract the spread of the SARS Covid-19 virus. Contrary to Circoop canteen environment in which it was possible to actively monitor the levels of $L_{N,A}$ despite the restrictions imposed by the D.P.C.M. of 26 October 2020 n. 265 called "urgent measures to contain the contagion throughout the national territory", in the other 4 environments in which it is expected to install the SEM the levels of $L_{N,A}$ have been simulated within the ODEON software using the multi-surface sources method that allows to reproduce the levels of $L_{N,A}$ generated by a series of sound sources that are activated in random and non-schematic mode within the emissive surface. The acoustic simulation procedure is the same used for the calibration of the CIRCOOP Canteen and is summarized as follows until the necessary parameters are obtained for each environment to be included in input within the forecasting algorithms for the control of gatherings and to evaluate the reliability or otherwise of the algorithms in environments with different geometries and uses.

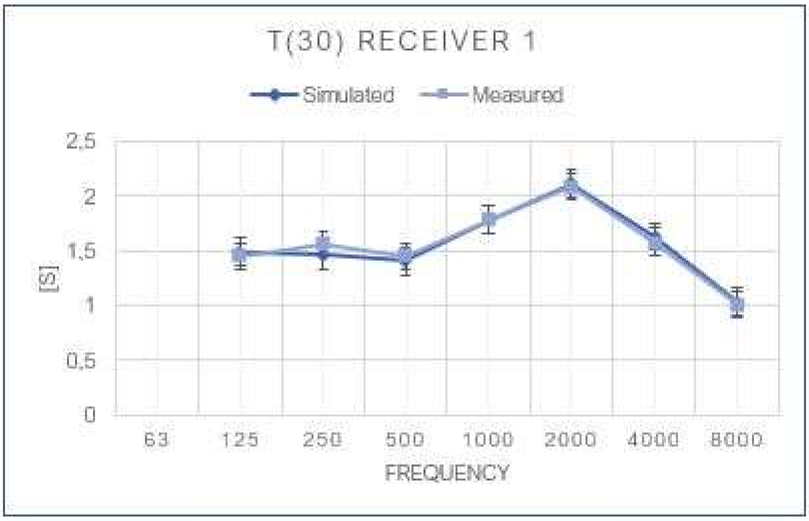
6.1 | VITTORIO ALFIERI CLASSICAL HIGH SCHOOL, TURIN

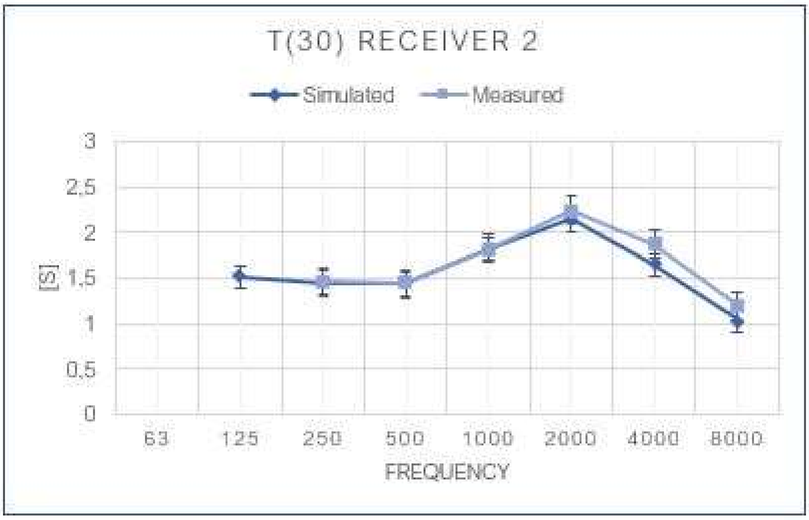
The first environment in which it was considered to install the SEM device for the control of flows and gatherings, is Alfieri Classical High School in Turin, located in Corso Dante 80. The environment represented below, has an extension in plan of 22.6 meters longitudinally by 19.1 transverse meters. The stations where it was decided to install the SEM Totem version (Alpha prototype) are represented in plan by a red dot that is always located at the existing electrical outlets or channels to which it is possible to connect, since for its operation the SEM must necessarily be connected to a power outlet. The first SEM was positioned in the entrance area of Liceo Alfieri where there are also waiting sessions. The second SEM has been positioned at the reception counter where it is assumed there is a greater probability of crowding represented both by students in transit in the school environment and by staff / teachers during class hours to reach the different classes. The third SEM has been positioned in the area called distribution area as in this space there are two vending machines where students can buy food and drinks and there are also two professional photocopiers as well as 5 lockers for the exhibition of trophies and teaching material of various kinds. In addition, in this environment there are also 6 doors connecting with other school environments and the elevator compartment connecting with the upper floors of high school.

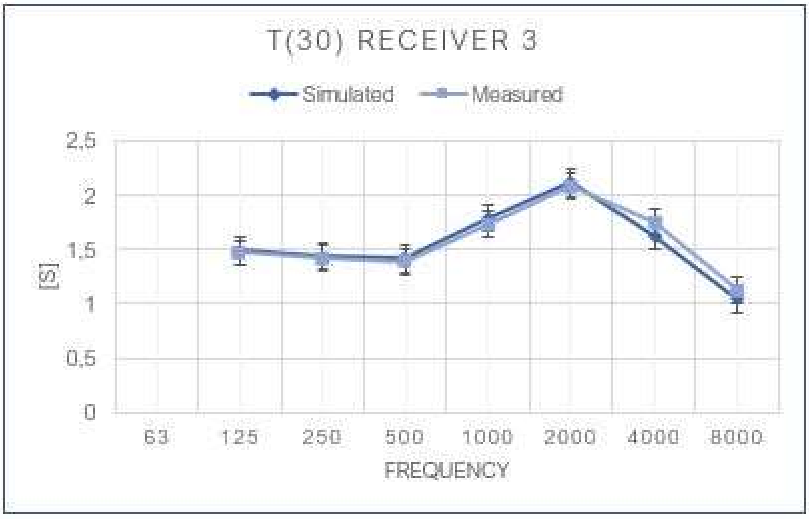



LAVAZZA
DAE

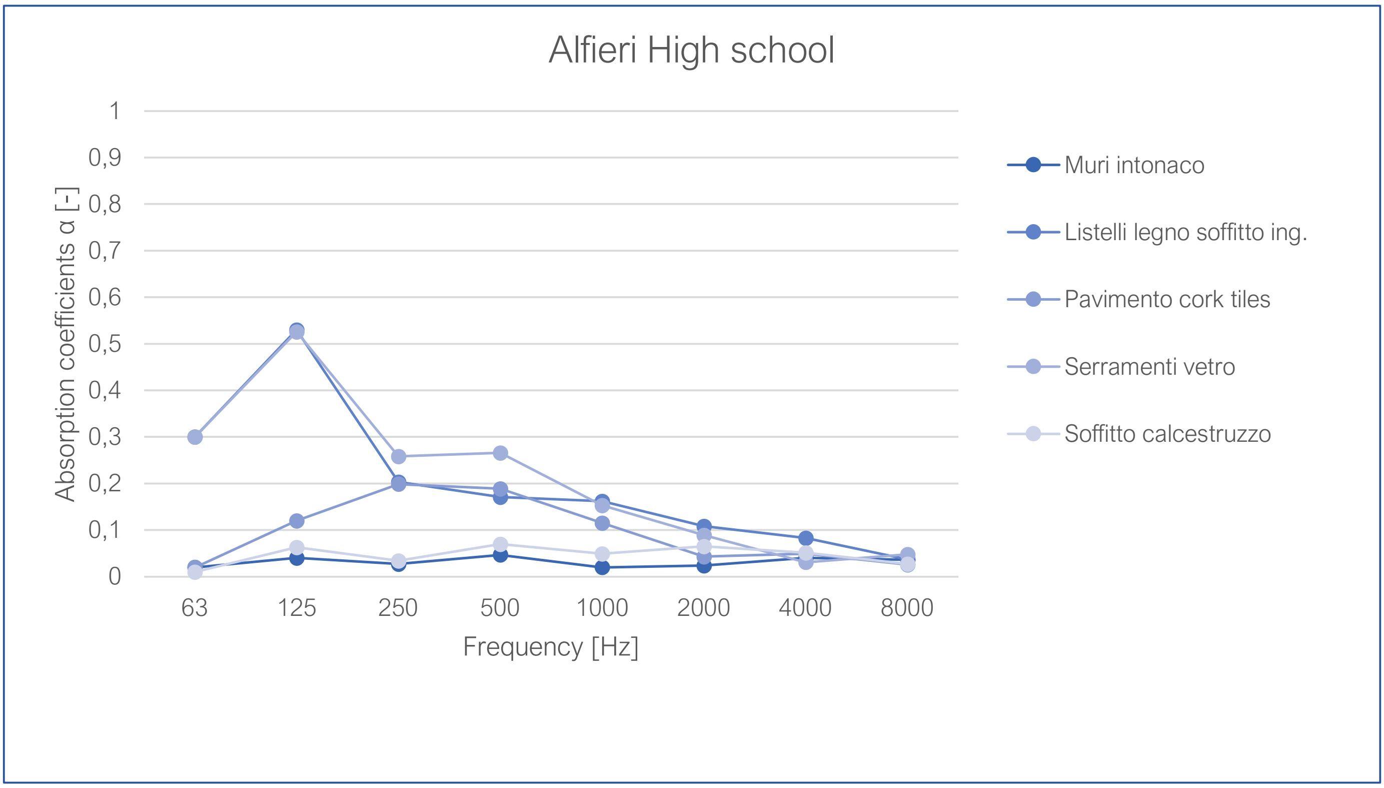



T(30) RECEIVER 1
Simulated
Measured
2,5
2
1,5
1
0,5
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(30) RECEIVER 2
Simulated
Measured
3
2,5
2
1,5
1
0,5
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY

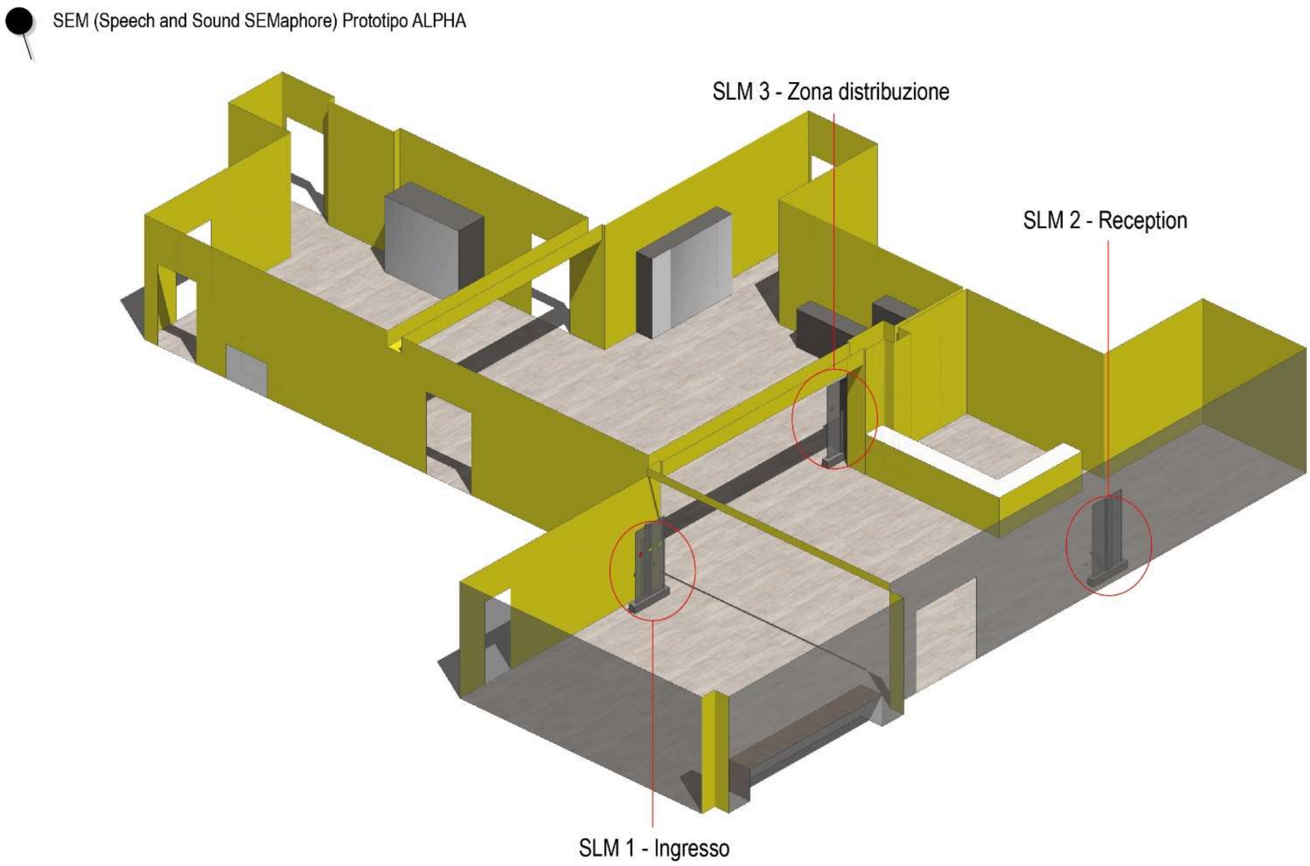

T(30) RECEIVER 3
Simulated
Measured
2,5
2
1,5
1
0,5
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY

Alfieri High school								
Materials	Absorption coefficients as a function of frequency [Hz]							
	63	*125*	*250*	*500*	*1000*	*2000*	*4000*	*8000*
16000 - Plastica fotocopiatrici	0,06	0,08	0,08	0,13	0,29	0,66	0,12	0,15
16045 - Vetro singolo arredi	0,20	0,20	0,27	0,06	0,09	0,20	0,05	0,05
16046 - Alluminio arredi	0,13	0,10	0,24	0,11	0,12	0,10	0,06	0,17
16047 - Muri intonaco	0,02	0,04	0,03	0,05	0,02	0,02	0,04	0,04
16007 - Porta REI	0,14	0,25	0,13	0,04	0,21	0,02	0,15	0,05
16050 - Listelli legno copertura ingresso	0,30	0,53	0,20	0,17	0,16	0,11	0,08	0,04
16051 - Pavimento cork-tiles su cemento	0,02	0,12	0,20	0,19	0,12	0,04	0,05	0,03
16052 - Serramenti vetro	0,30	0,53	0,26	0,27	0,15	0,09	0,03	0,05
16053 - Sedie legno vuote	0,02	0,03	0,04	0,06	0,08	0,07	0,06	0,09
16055 - Soffitto calcestruzzo con travi	0,01	0,06	0,03	0,07	0,05	0,07	0,05	0,03



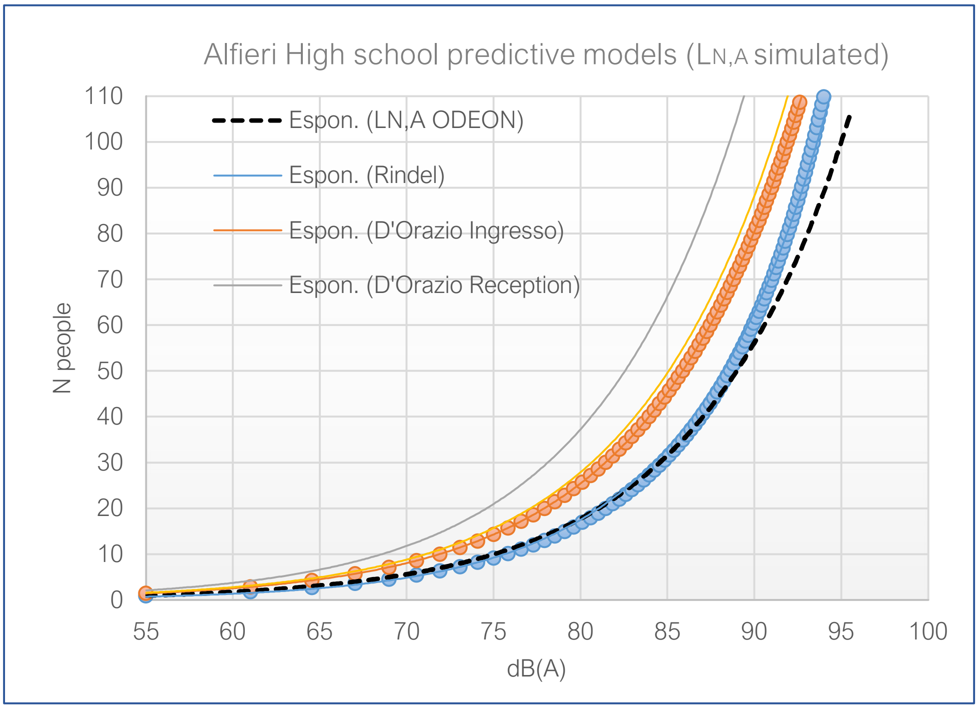
Parameter summary from acoustic simulation Odeon and r_{ij} in Grasshopper

c=Lombard slope	ap	t_0	Volume	N max	D_2S	$L_{N,A,1}$ funzione di trasferimento	r_{ij} SLM1	r_{ij} SLM2	r_{ij} SLM3
0,50 [dB/dB]	0,25 [m^2]	1,73 [s]	763 [m^3]	107 [-]	2,2 [dB]	55,0 [dB(A)]	7,6 [m]	12,7 [m]	8,6 [m]

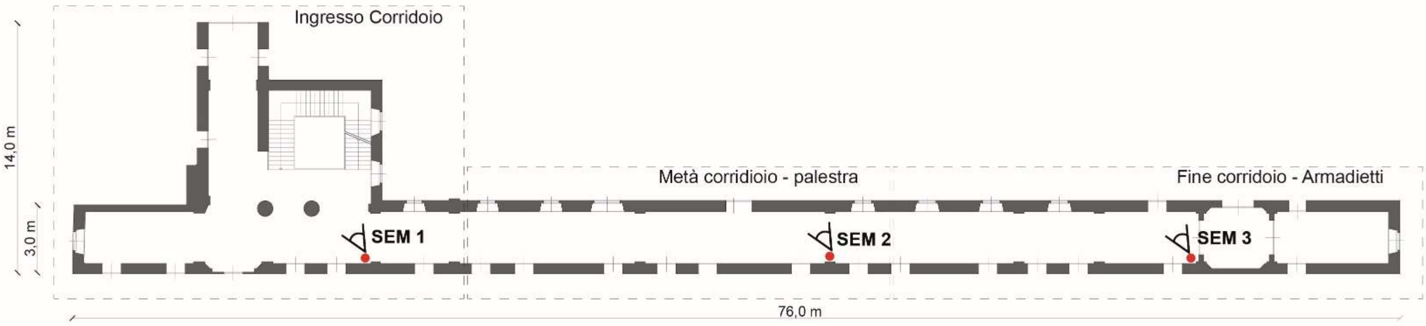


Application of previsional algorithms

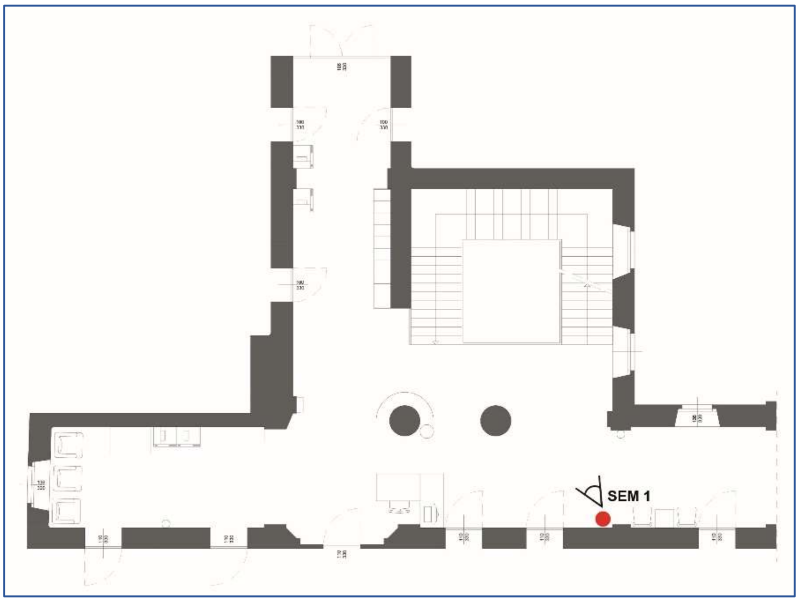
The noise levels $L_{N,A}$ have been simulated in Odeon through the use of multi-surface sources that as explained in the chapter dedicated to acoustic simulations allows you to simulate the level of noise $L_{N,A}$ generated by several speaking people, since within the multi surface sources are activated in random and non-schematic mode a series of source points that simulate the talk of multiple users. The way in which sound is transmitted in the environment starting from the aforementioned surface, can be calculated through the parameter $L_{N,A,1}$ called "transfer function" which corresponds to the level of noise generated by a speaking person starting from the physical/geometric characteristics of the environment considered. Once the "transfer function" was calculated, it was possible to obtain on Excel the corresponding levels of $L_{N,A}$ generated by an incremental number of users, until reaching the maximum capacity allowed in the environment in order to guarantee a distance of 1,5 meters interpersonal. In the case of the Liceo Alfieri, the maximum number of users that can be accommodated so that interpersonal distancing is guaranteed, is a maximum of 107 people. Levels of $L_{N,A}$ simulated in Odeon were represented graphically by assigning on the axis of the abscisses the levels of $L_{N,A}$ and on the axis of ordinates the corresponding number of users. Within Excel, other columns have been created in which to apply the forecast model of J.H. Rindel and the forecast model of D. D'Orazio for each SEM considered. In fact, the model of D. D'Orazio takes into account the average distance between an i-th speaker and a j-th receiver, therefore wanting to install 3 SEM devices it was necessary to calculate the distance between the simulated point cloud and the fixed position of the receivers using the Rhinoceros Grasshopper application again. The parameters obtained through the simulations, have been included in the two forecast models considering also in this case a Lombard slope of c = 0.50 [dB /dB] while as regards the parameter g = group size it was not possible to apply the same value used in the simulation of the CIRCOOP Canteen because in this case the values of $L_{N,A}$ are not real so they do not take into account the n+1 conditions that influence the levels of anthropogenic noise generated by a crowd in an internal environment and consequently it would not be true to assume a g= 8 [-] with which it is considered that within a group the speakers are 1/8 of the total number of people. Starting from these considerations, it was necessary to assume a g= 1 [-] with which it is simulated that all people can speak with the same frequency within the considered environment and generate increases in the level of $L_{N,A}$. The values of N= number of people obtained by applying the models of J.H. Rindel and the model of D. D'Orazio for each position of the SEM device have been schematized in the graph below and it is possible to evaluate that in this environment the model that returns results of N more similar to the "real" ones is the model of J.H. Rindel whose curve is superimposable with the curve of the real data until you reach the 85 dB (A) that in this environment correspond to only 30 people. Such a high level of anthropogenic noise $L_{N,A}$ generated by a rather small number of people is a consequence of the reverberation time of the environment in this case equal to t_0= 1.73 [s] which represents a rather high value compared to the optimal reverberation time that in environments dedicated to listening to the word is calculable as $t_{60,optimal}$ = 0.5 +0,0001*V =0,5+0,0001*763= 0,58 [s].

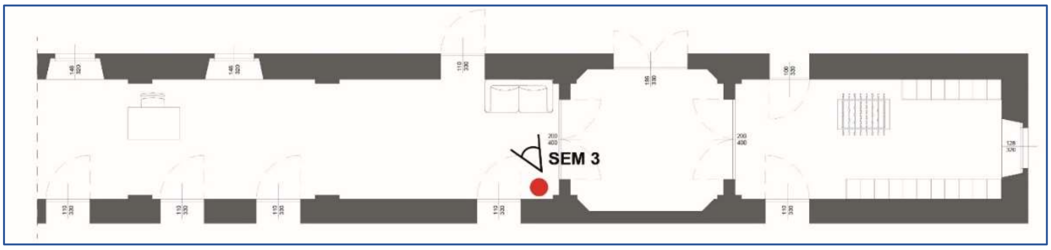


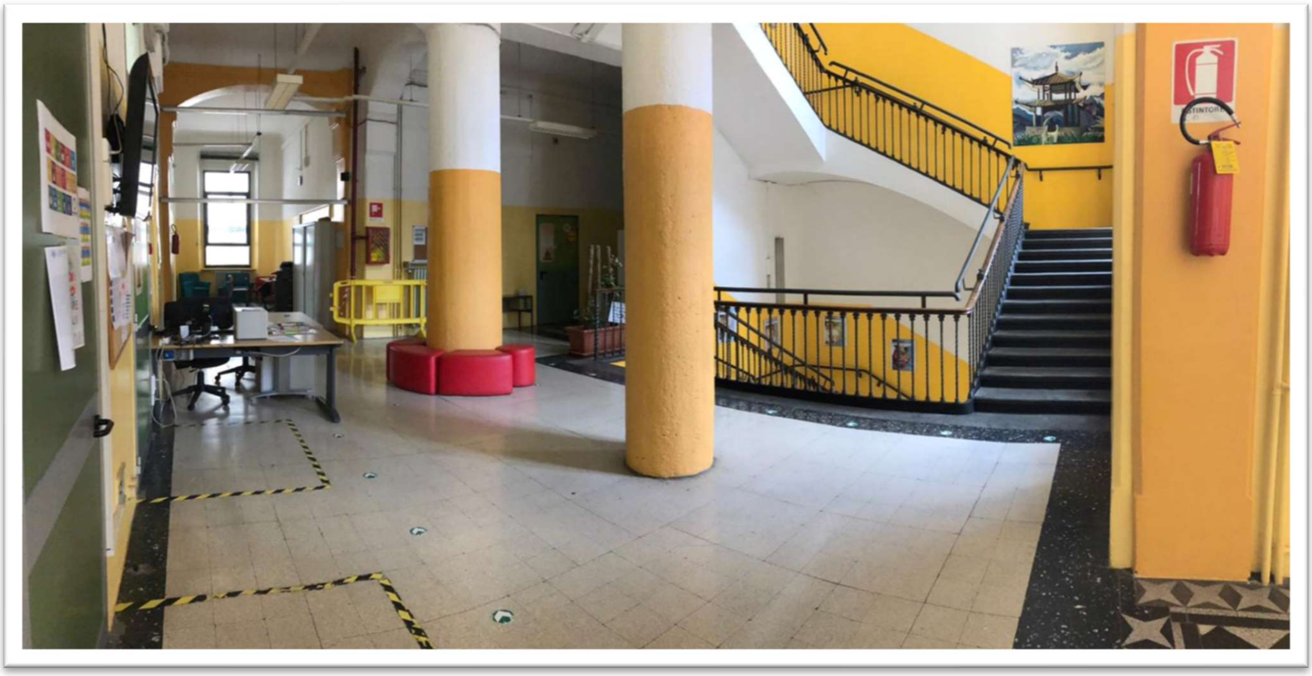
6.2 | CONVITTO NAZIONALE UMBERTO I HIGH SCHOOL, TURIN

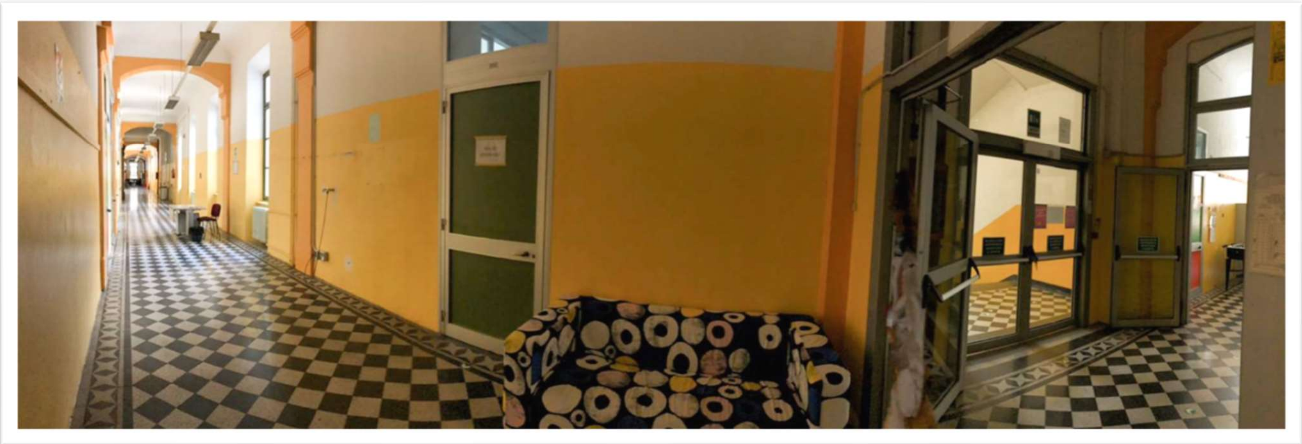


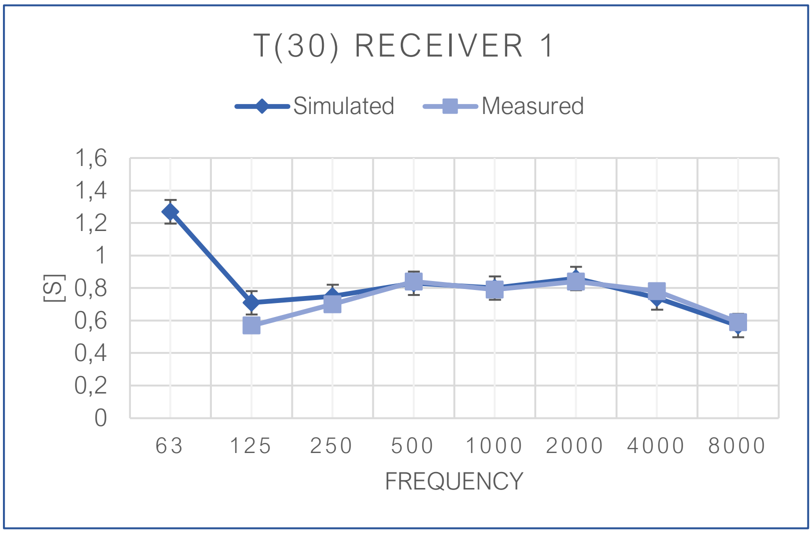
The second environment in which the device was evaluated is Convitto Nazionale Umberto I High School in Turin, located in via Bertola 10. The environment represented below, has a plan extension of 76 meters longitudinally for 14 transverse meters. The stations where it was decided to install the SEM Totem version (Alfa prototype) are represented in plan by a red dot that is positioned in correspondence with the existing electrical current sockets or channels to which it is possible to connect. The first SEM was positioned in the environment that is accessed immediately after climbing the stairs and reaching the first floor, inside which we find a small reception on the floor and access to teachers room. In addition, there are 4 photocopiers, a series of sofas on which you can stop during the hours of recreation and a large number of lockers where to store school supplies. The second SEM was positioned in the middle of the corridor near the access door to the gym and finally the last SEM device was placed in an area certainly susceptible to crowding since there are the female bathrooms, a sofa for relaxation and in the neighboring environment we find the emergency stairs and a small room containing other lockers in which to store the personal belongings of the students and a table football.

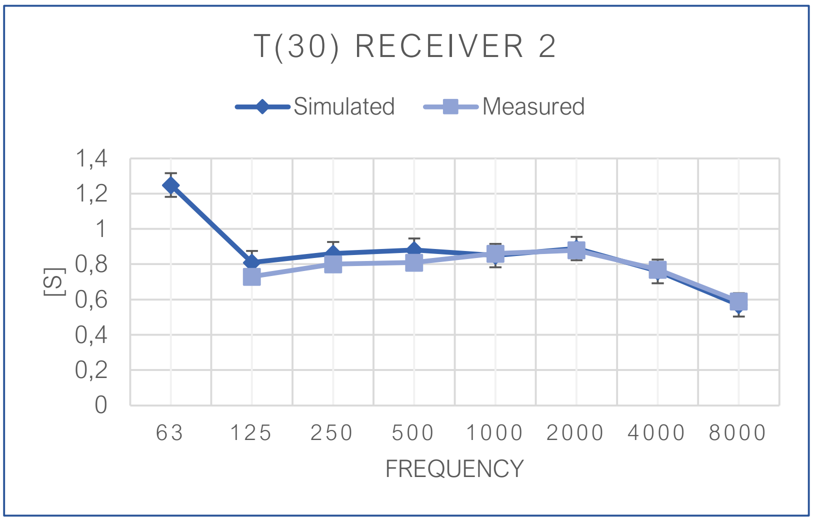


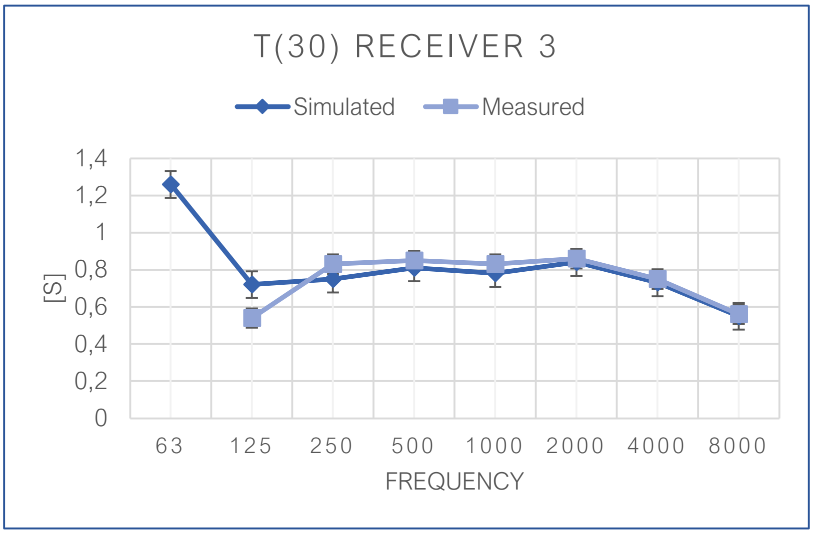




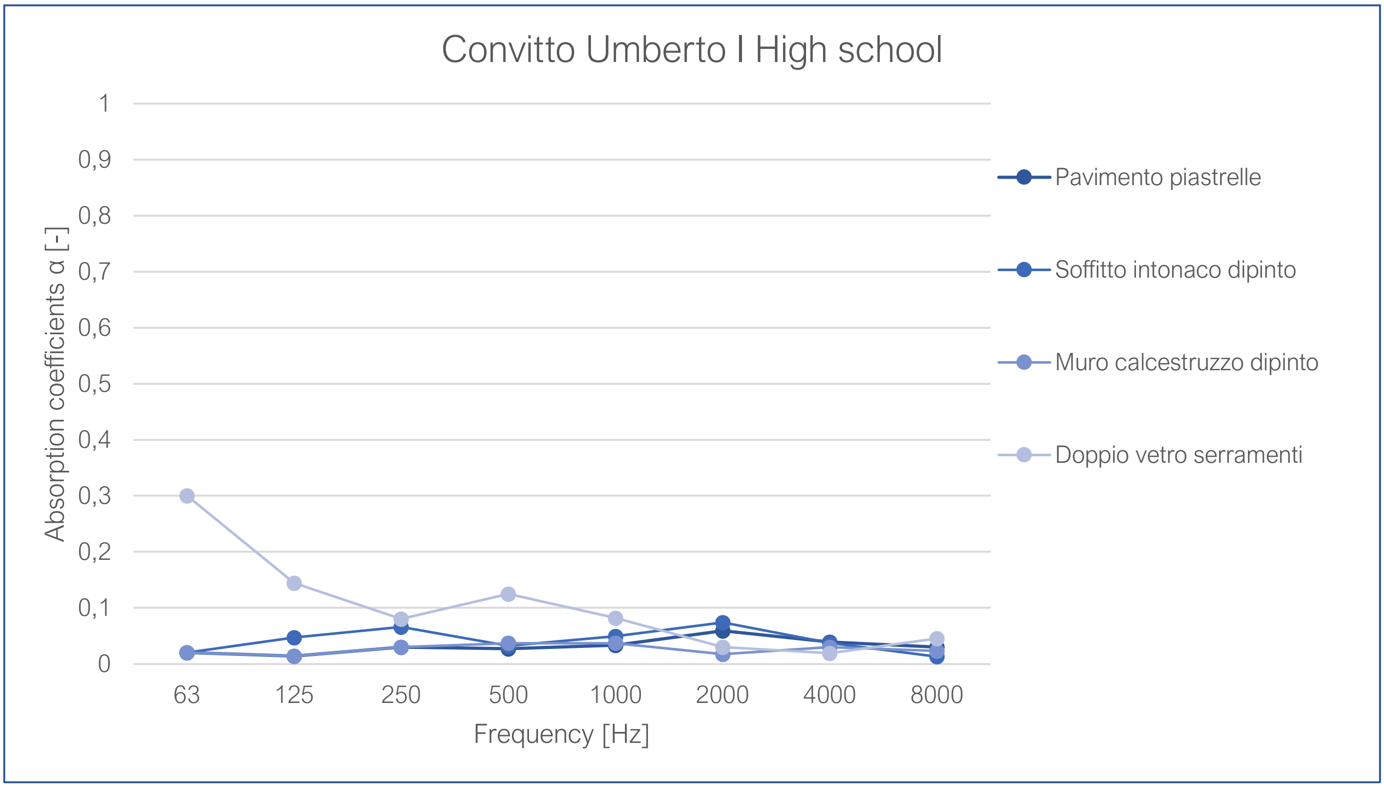


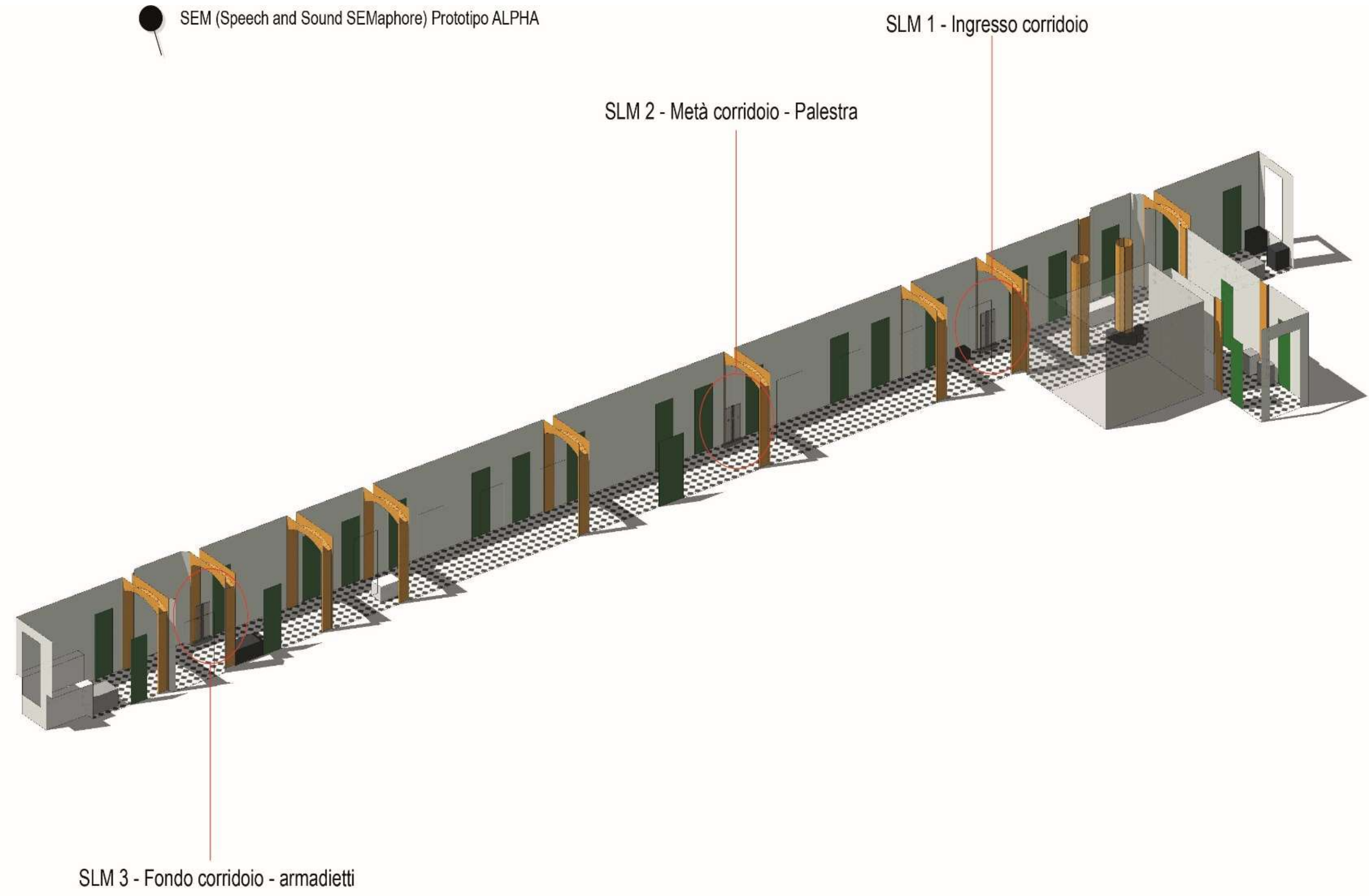

T(30) RECEIVER 1
Simulated
Measured
1,6
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(30) RECEIVER 2
Simulated
Measured
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(30) RECEIVER 3
Simulated
Measured
1,4
1,2
1
0,8
0,6
0,4
0,2
0
[S]
63
125
250
500
1000
2000
4000
8000
FREQUENCY

Convitto Umberto I High school								
Materials	Absorption coefficients as a function of frequency [Hz]							
	63	*125*	*250*	*500*	*1000*	*2000*	*4000*	*8000*
15270 - Pavimento in piastrelle	0,02	0,01	0,03	0,03	0,03	0,06	0,04	0,03
15271 - Soffitto intonaco	0,02	0,05	0,07	0,03	0,05	0,07	0,04	0,01
15273 - Porte legno alluminio	0,30	0,19	0,08	0,11	0,08	0,08	0,05	0,05
15274 - Muri intonaco	0,02	0,01	0,03	0,04	0,04	0,02	0,03	0,02
16059 - Divanetti pelle	0,40	0,40	0,52	0,25	0,48	0,79	0,67	0,08
16060 - Plastica fotocopiatrice	0,02	0,01	0,03	0,03	0,03	0,06	0,04	0,03
92 -Sedie vuote con tessuto	0,44	0,34	0,65	0,67	0,92	0,73	0,64	0,85
16061 - Arredi legno	0,08	0,19	0,10	0,15	0,12	0,14	0,39	0,19
15270 - Doppio vetro serramento	0,30	0,14	0,08	0,13	0,08	0,03	0,02	0,05
16062 - Pannello vetro arredo	0,18	0,16	0,08	0,04	0,06	0,03	0,04	0,01



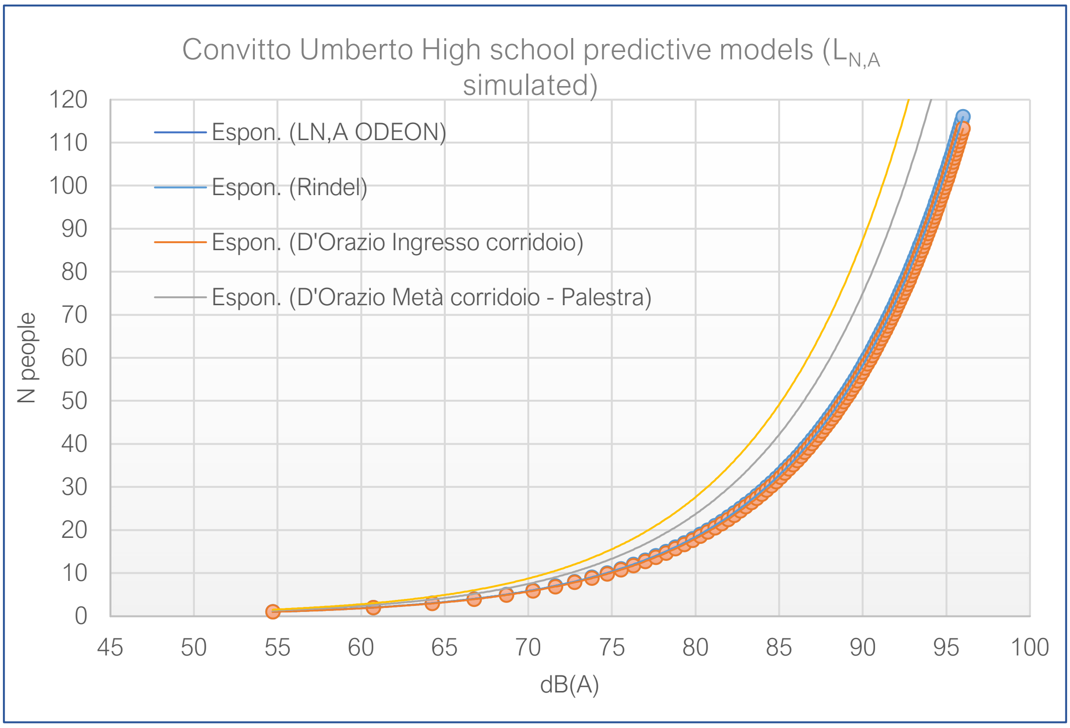


Parameter summary from acoustic simulation Odeon and r_{ij} in Grasshopper

c=Lombard slope	ap	t_0	Volume	N_{max}	D_2S	$L_{N,A,1}$ funzione di trasferimento	r_{ij} SLM1	r_{ij} SLM2	r_{ij} SLM3
0,50 [dB/dB]	0,25 [m^2]	2,10 [s]	1280 [m^3]	116 [-]	1,4 [dB]	54,7 [dB(A)]	12,4 [m]	22,6 [m]	31,5 [m]

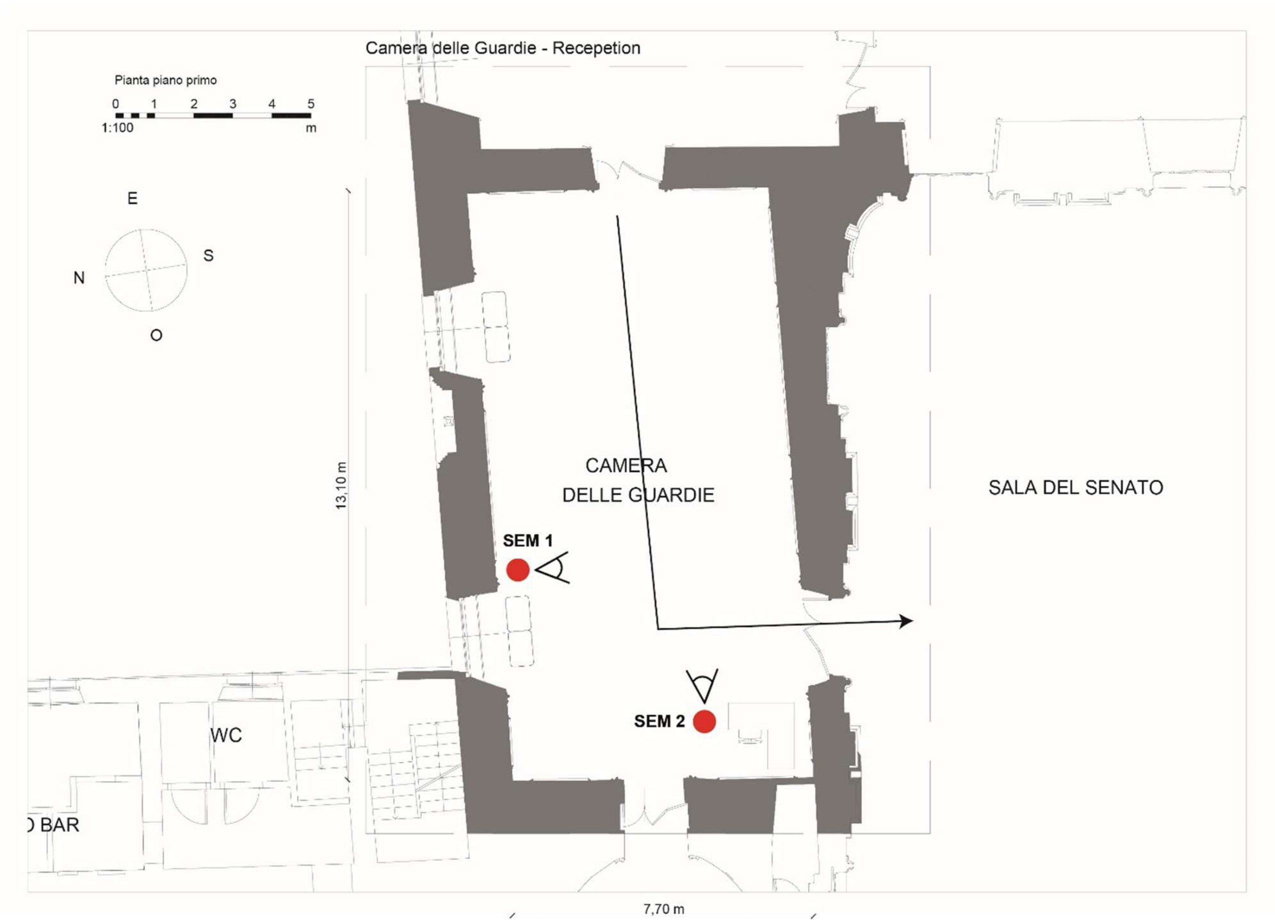
Application of previsional algorithms

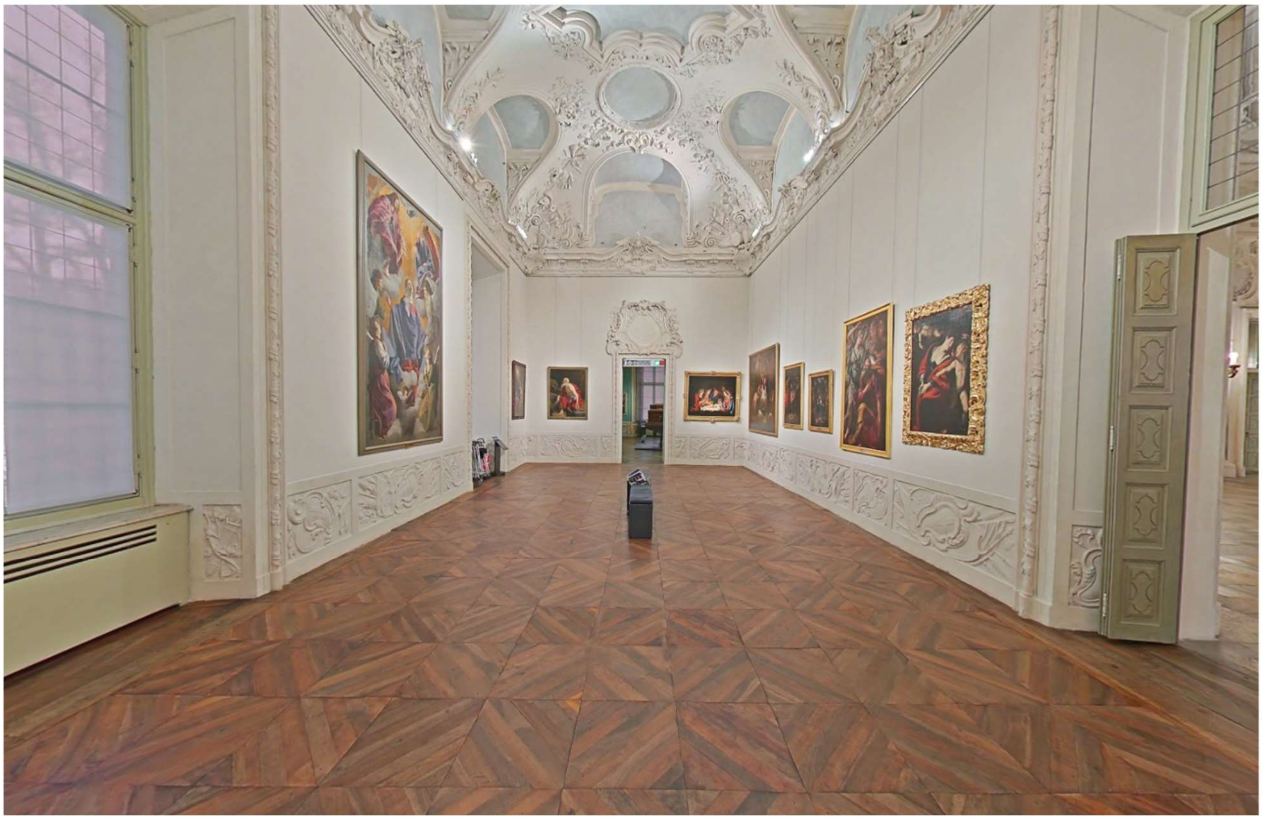
After calculating the "transfer function", the corresponding levels of $L_{N,A}$ generated by an incremental number of users were again calculated on Excel, until the maximum capacity of 116 people was reached in order to guarantee an interpersonal distance of 1,5 meters. The levels of $L_{N,A}$ simulated in Odeon were represented graphically by assigning on the axis of the abscisses the levels of $L_{N,A}$ and on the axis of ordinates the corresponding number of users. Within Excel, other columns have been created in which to apply the forecast model of J.H. Rindel and the forecast model of D. D'Orazio for each SEM considered, wanting to install 3 SEM devices it was necessary to calculate the distance between the cloud of simulated points and the fixed position of the receivers using the Rhinoceros Grasshopper application again. The parameters obtained through the simulations, have been included in the two forecast models considering also in this case a Lombard slope of c= 0,50 [dB/dB] while the group size parameter is always g= 1 [-]. In the graph it is possible to observe how the curve of the model of J.H. Rindel overlaps exactly with the "real" values of N pervisti starting from the levels of $L_{N,A}$ simulated in Odeon and in turn overlaps perfectly with the model of D. D'Orazio calculated at the SEM positioned in the corridor entrance, while the models of D'Orazio applied to the SEM 2 located in the middle of the corridor and the third SEM placed at the end of the corridor return values of N more distant from the values simulated on Odeon, this is because the SEM 2-3 are positioned rather distant from the main environment that is accessed immediately after climbing the stairs and in which , having a greater extension in the plan it is possible to place more points representing people, always maintaining an interpersonal distance of 1.5 meters, compared to the corridor which for its conformation is rather narrow and during the simulation on Grasshopper the points were less dense in this area. Consequently, by increasing the value of r distance from the i-th receiver source, the sound decay is less and less accentuated so that the D'Orazio model tends to move away from the simulated values of N and there is a prevalence of diffuse sound field with respect to the direct sound field.

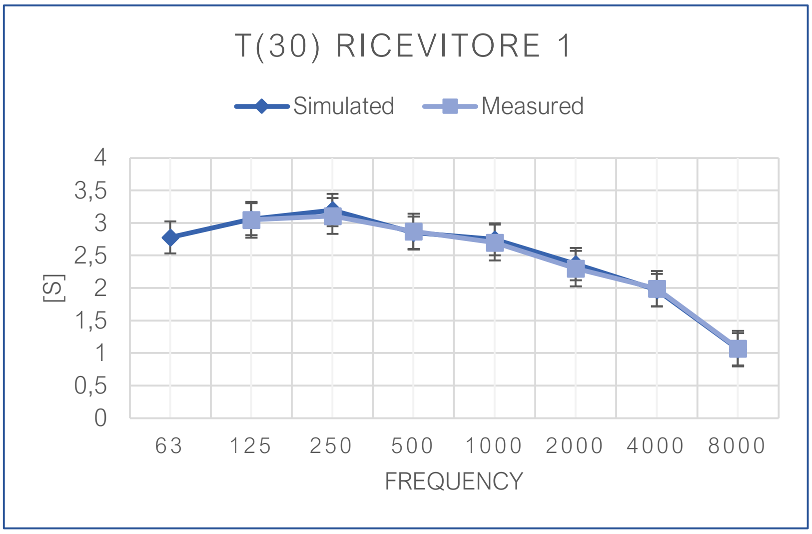


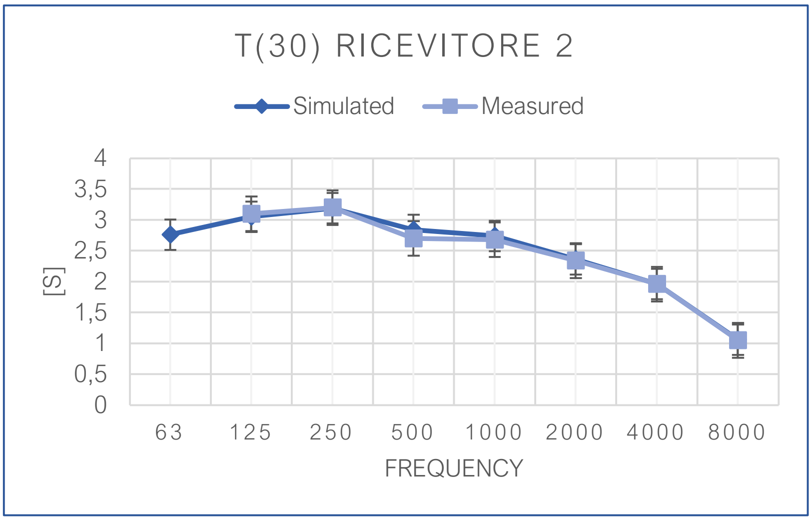
6.3 | PALAZZO MADAMA MUSEUM – CAMERA DELLE GUARDIE

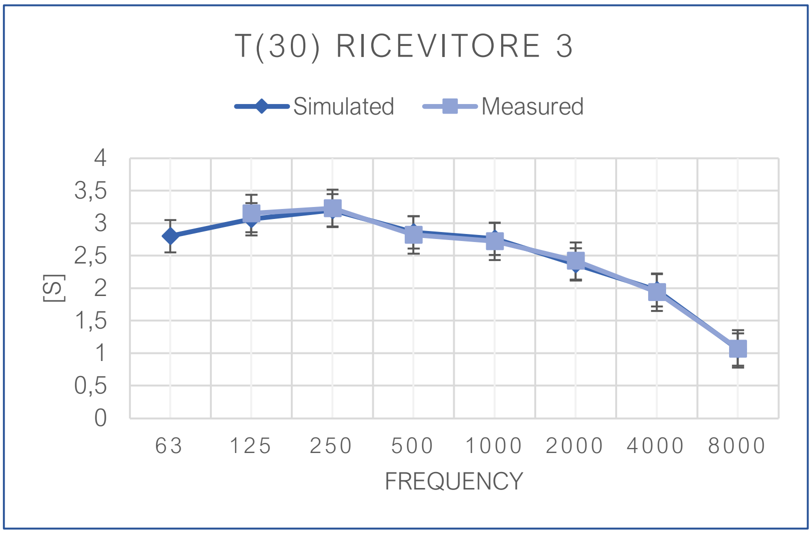
The third environment in which the device was evaluated is Sala della Guardie of Palazzo Madama, located in Piazza Castello in Turin which houses the Civic Museum of Ancient Art. The environment represented below, has a plan extension of 7,70 meters longitudinally by 13,1 transverse meters. In this environment there is an information desk and audio-guide distribution for the exhibitions set up in the adjacent Senate Hall. There are also 2 sofas and the environment borders to the west with the Gabinetto Rotondo from which you can reach the toilets and the Bar located in the Gallery Room. The only power outlets available in the room are located at the reception counter and near the seat immediately to the left of the counter, where you can connect the SEM to a cockpit not visible externally. It is therefore assumed to place the SEM device either next to the seat in a median position or alternatively near the reception counter which is assumed to be susceptible to crowding during the distribution of audio-guides and during the passage of visitors with the neighboring environments, especially to take advantage of the toilets.



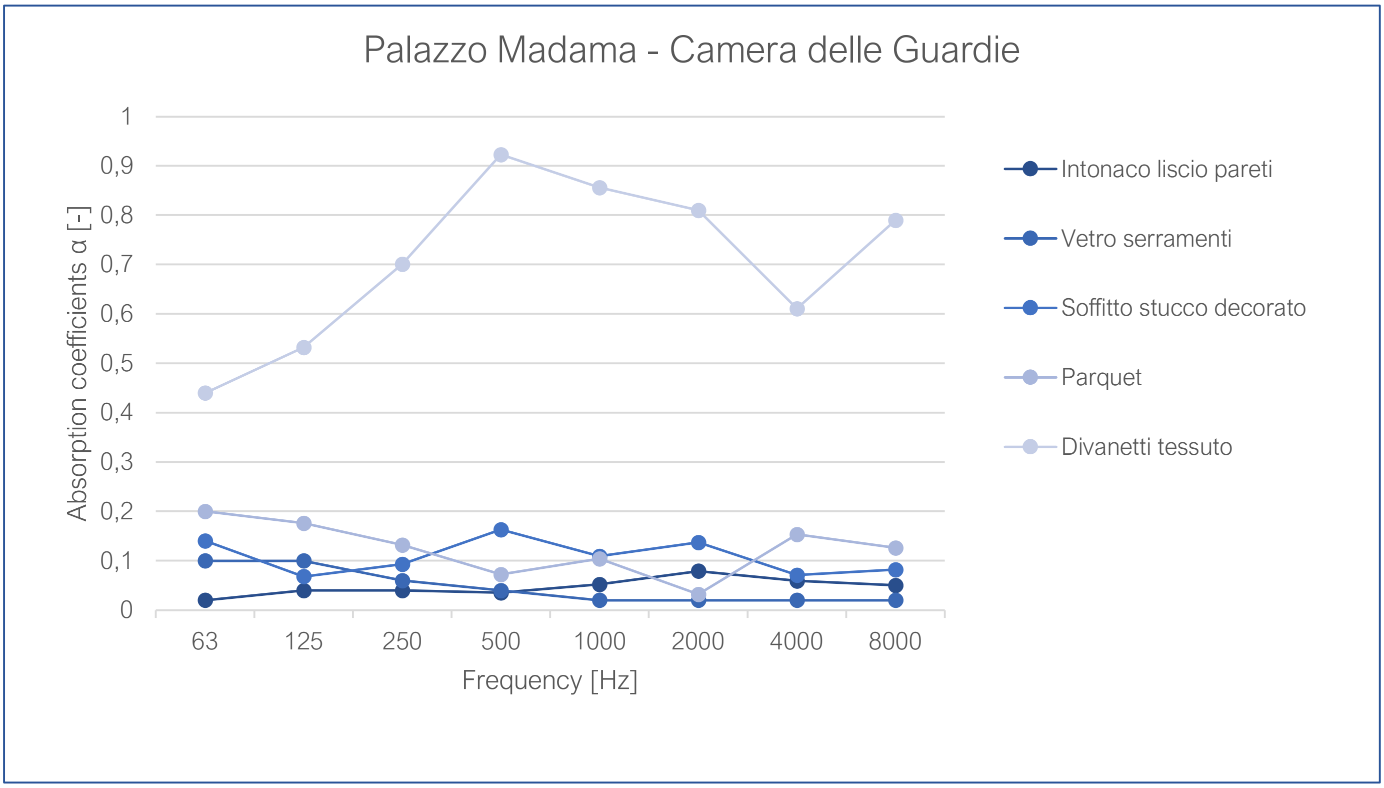


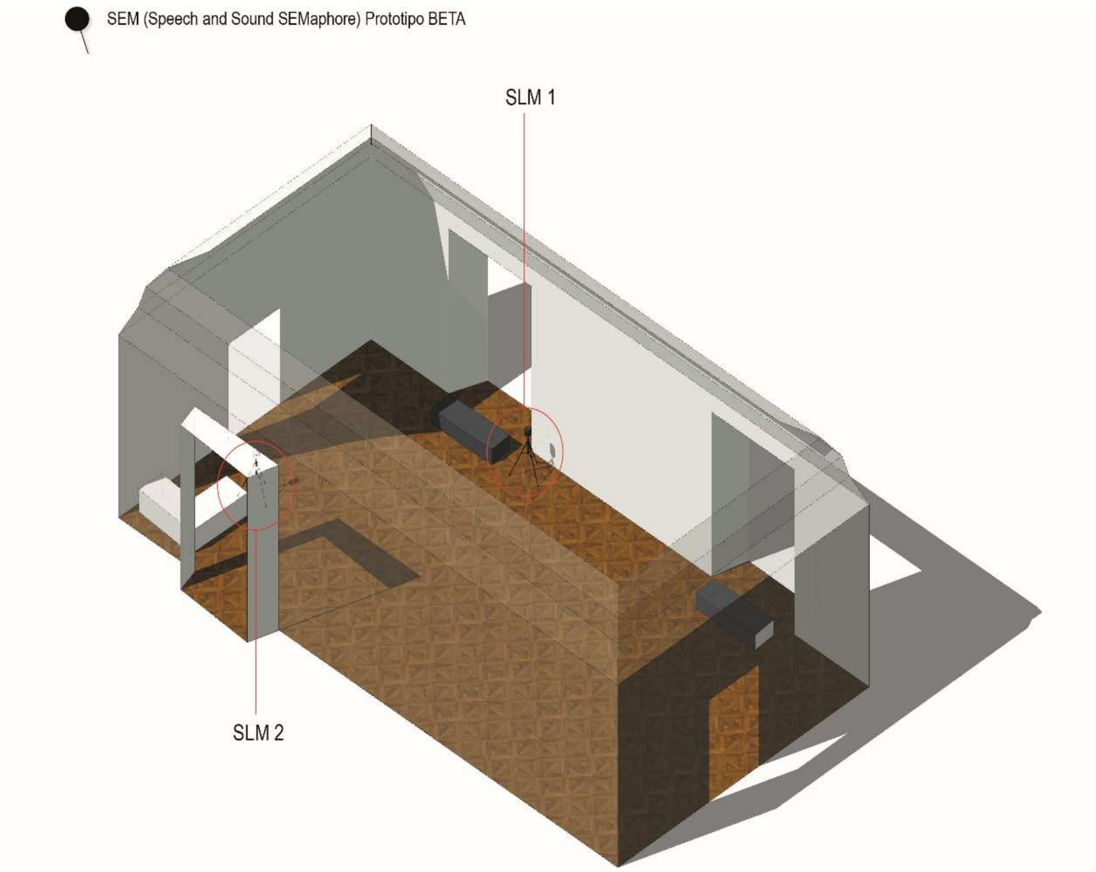

T(30) RICEVITORE 1
Simulated
Measured
[S]
4
3,5
3
2,5
2
1,5
1
0,5
0
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(30) RICEVITORE 2
Simulated
Measured
[S]
4
3,5
3
2,5
2
1,5
1
0,5
0
63
125
250
500
1000
2000
4000
8000
FREQUENCY


T(30) RICEVITORE 3
Simulated
Measured
[S]
4
3,5
3
2,5
2
1,5
1
0,5
0
63
125
250
500
1000
2000
4000
8000
FREQUENCY

Palazzo Madama - Camera delle Guardie								
Materials	Absorption coefficients as a function of frequency [Hz]							
	63	*125*	*250*	*500*	*1000*	*2000*	*4000*	*8000*
15297 - Fascia stucco altamente decorato	0,14	0,07	0,09	0,16	0,11	0,14	0,07	0,08
15298 - Intonaco liscio	0,02	0,04	0,04	0,04	0,05	0,08	0,06	0,05
15305 - Vetro serramenti	0,10	0,10	0,06	0,04	0,02	0,02	0,02	0,02
15300 - Soffitto stucco decorato	0,14	0,07	0,09	0,16	0,11	0,14	0,07	0,08
15301 - Porte legno	0,14	0,17	0,05	0,11	0,17	0,20	0,12	0,04
15302 - Parquet	0,20	0,18	0,13	0,07	0,10	0,03	0,15	0,13
15304 - Divanetti tessuto	0,44	0,53	0,70	0,92	0,86	0,81	0,61	0,79
15303 - Scrivania legno	0,05	0,07	0,05	0,09	0,16	0,10	0,12	0,10

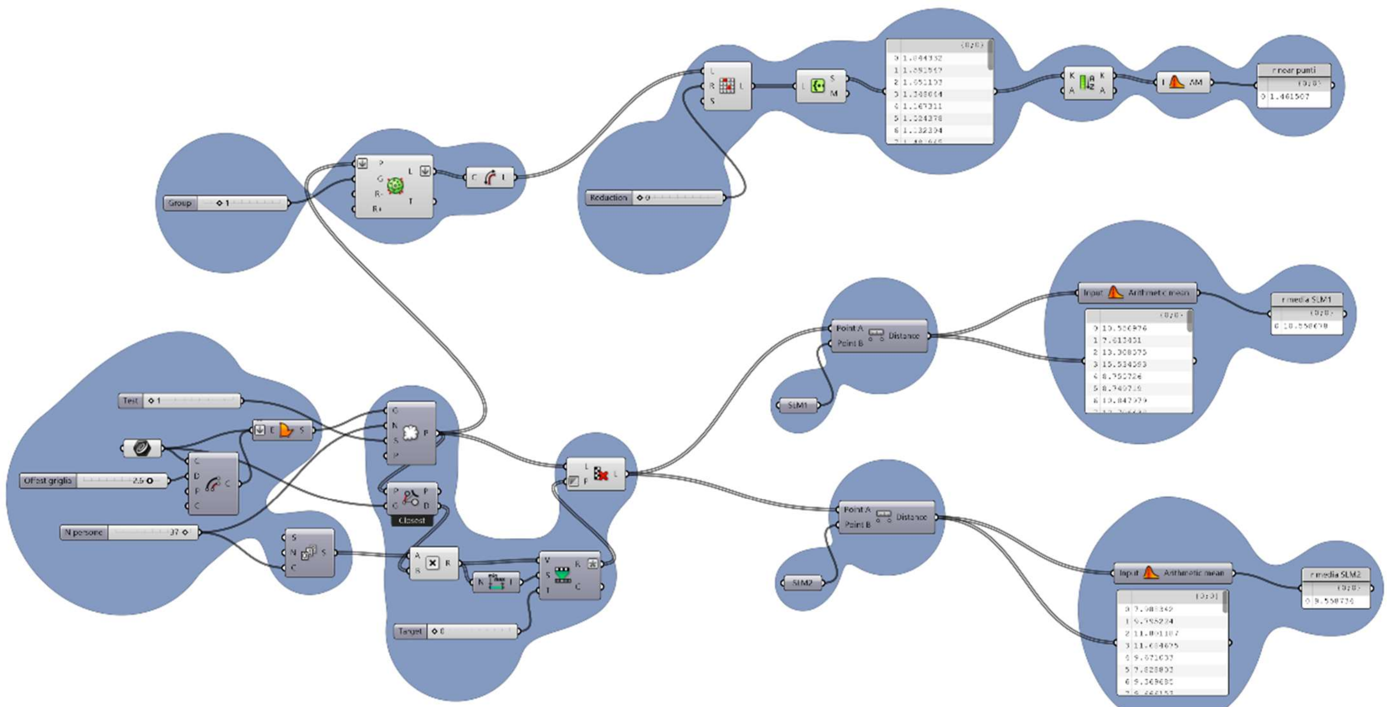




Parameter summary from acoustic simulation Odeon and r_{ij} in Grasshopper

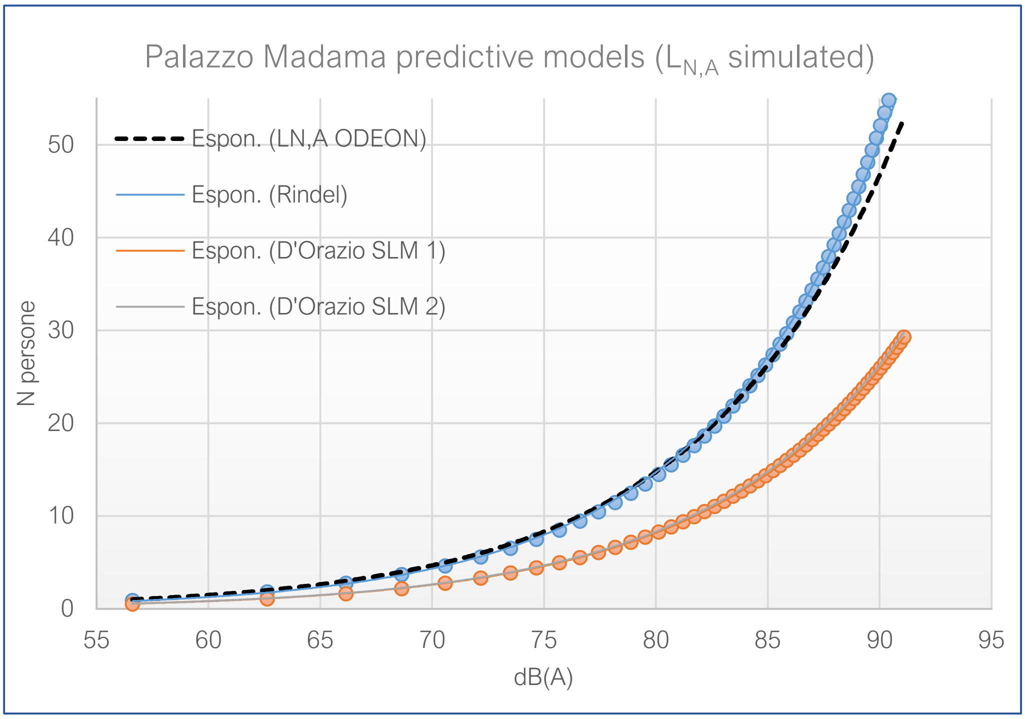
c=Lombard slope	ap	t_0	Volume	N_{max}	D_2S	$L_{N,A,1}$ funzione di trasferimento	r_{ij} SLM1	r_{ij} SLM2
0,50 [dB/dB]	0,25 [m^2]	2,79 [s]	1050 [m^3]	53 [-]	0,5 [dB]	56,6 [dB(A)]	9,55 [m]	10,55 [m]


SEM (Speech and Sound SEMaphore) Prototipo BETA
I-esime sorgenti sonore simulate con Grasshopper
SLM 1
Distanza dalle superfici
0,70 m
d_i = 1,9 m
d_i = 1,5 m
SLM 2
Area superficie utile

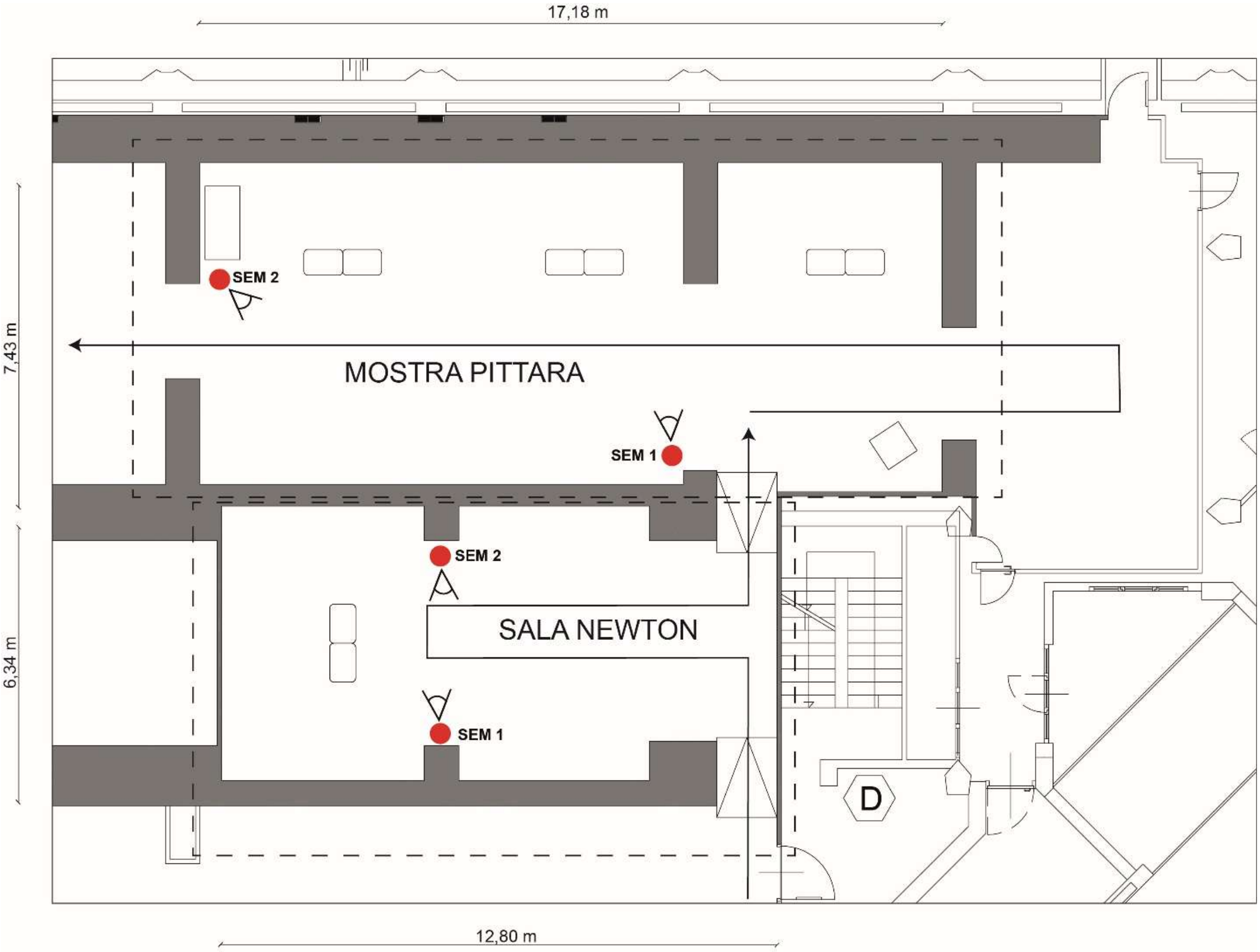


Application of previsional algorithms

Also in this case we see in the graph below that the model that best approximates the simulated data on Odeon is the model of J.H. Rindel since the environment has a reverberation time t_0= 2,79 [s] so there is certainly a prevalence of diffuse acoustic field compared to the direct acoustic field. In fact, the model of D. D'Orazio that takes into account the sound decay as the distance between an i-th source and a j-th receiver increases, in this case the D_2S = 0,5 [dB], returns values that deviate from the "real" model and this applies both to the SEM 1 positioned near the seat located almost in the middle of the room and to the SEM 2 positioned about a 1,5 m distance near the counter for the distribution of audio-guides. Both models overlap since the distance that the two devices is too small. Starting from these considerations, assuming that only one device should be placed in the Guards Room, it would be appropriate to try to place the SEM in a central position so preferably where SEM 1 has been positioned, so that the average distance between the i-th source and a j-th speaker is evenly distributed and at the same time, placing the SEM too close to the reception counter would result in over-estimated N data since the $L_{N,A}$ levels recorded by the device could be affected by the speech of visitors during the distribution of audio-guides.

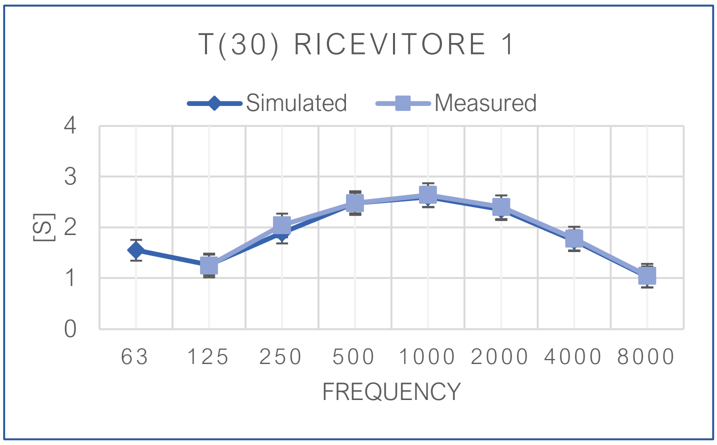


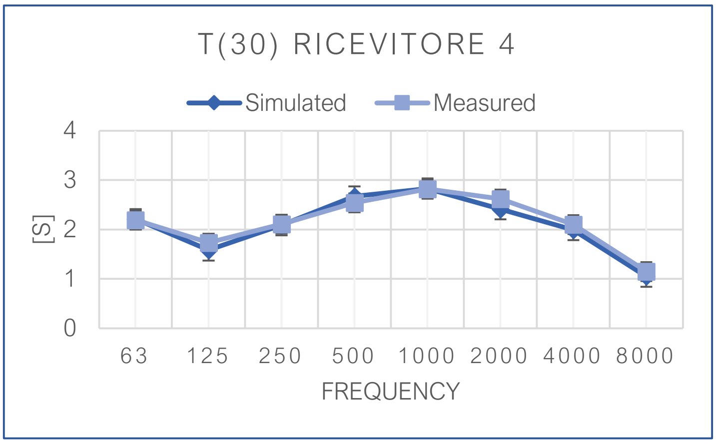
6.4 | GAM MUSEUM – SALA NEWTON e MOSTRA PITTARA

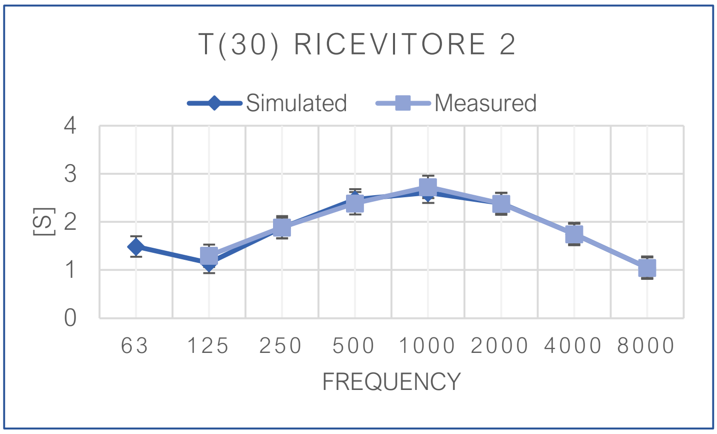


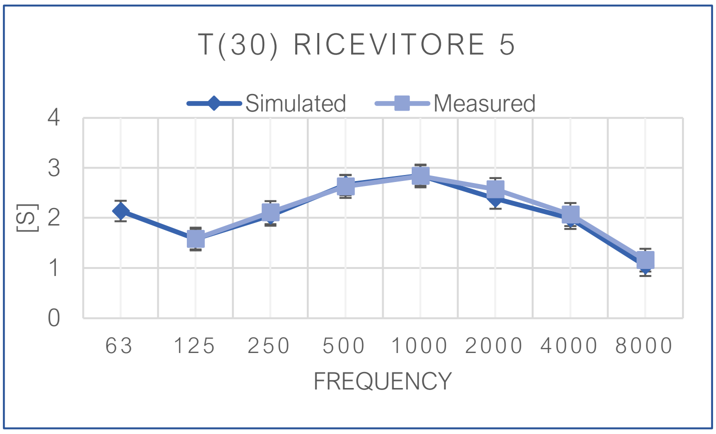
The last environment in which it was considered to install the Beta prototype SEM device, is the GAM -Galleria d'Arte Moderna di Torino, located in via Magenta 31. In particular, the choice of application of device is based on two rooms adjacent to each other, since they are considered the most interesting in the Museum. The first room often hosts temporary exhibitions and in the period December 2019 - November 2020 and in the period in which the reverberation time measurements were carried out, the room hosted an exhibition by the photographer Helmut Newton. The second room, which is accessed through a small wooden platform, is the Pittara Exhibition, dedicated to the painter Carlo Pittara and in this environment we find the painting "Fiera di Saluzzo" which entirely covers the border wall with the Newton room. In both environments, the reverberation time t_0 was measured on July 14, 2020 and starting from the geometry of the environments and the position of the available power outlets to which the Beta prototype SEM device could be connected, it was evaluated to install a device for each environment by preparing them as represented in the plan below.

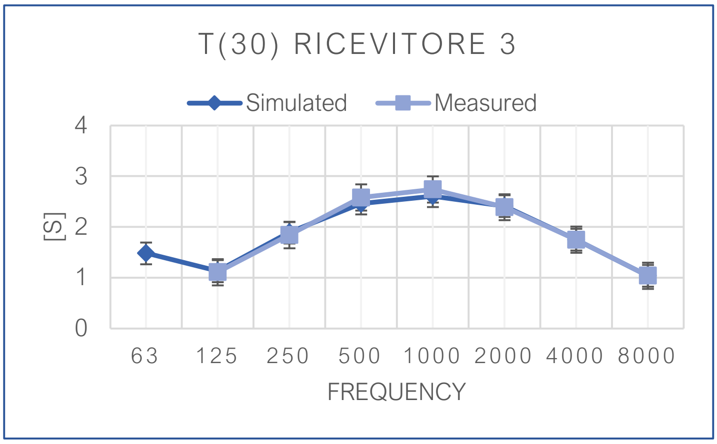
Model calibration on Odeon 16 and assignment of materials from t_{30} estimation

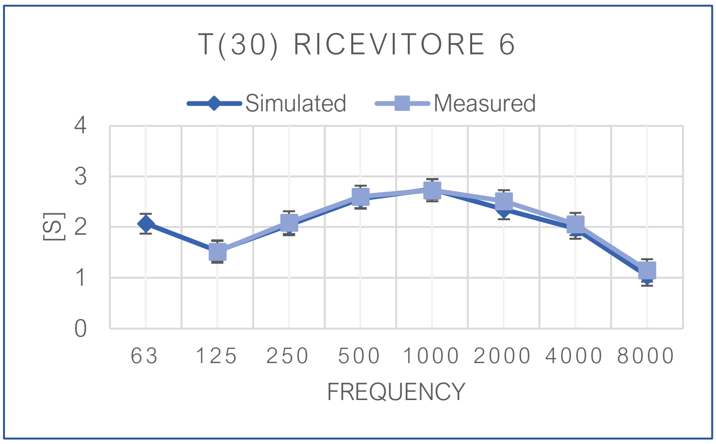




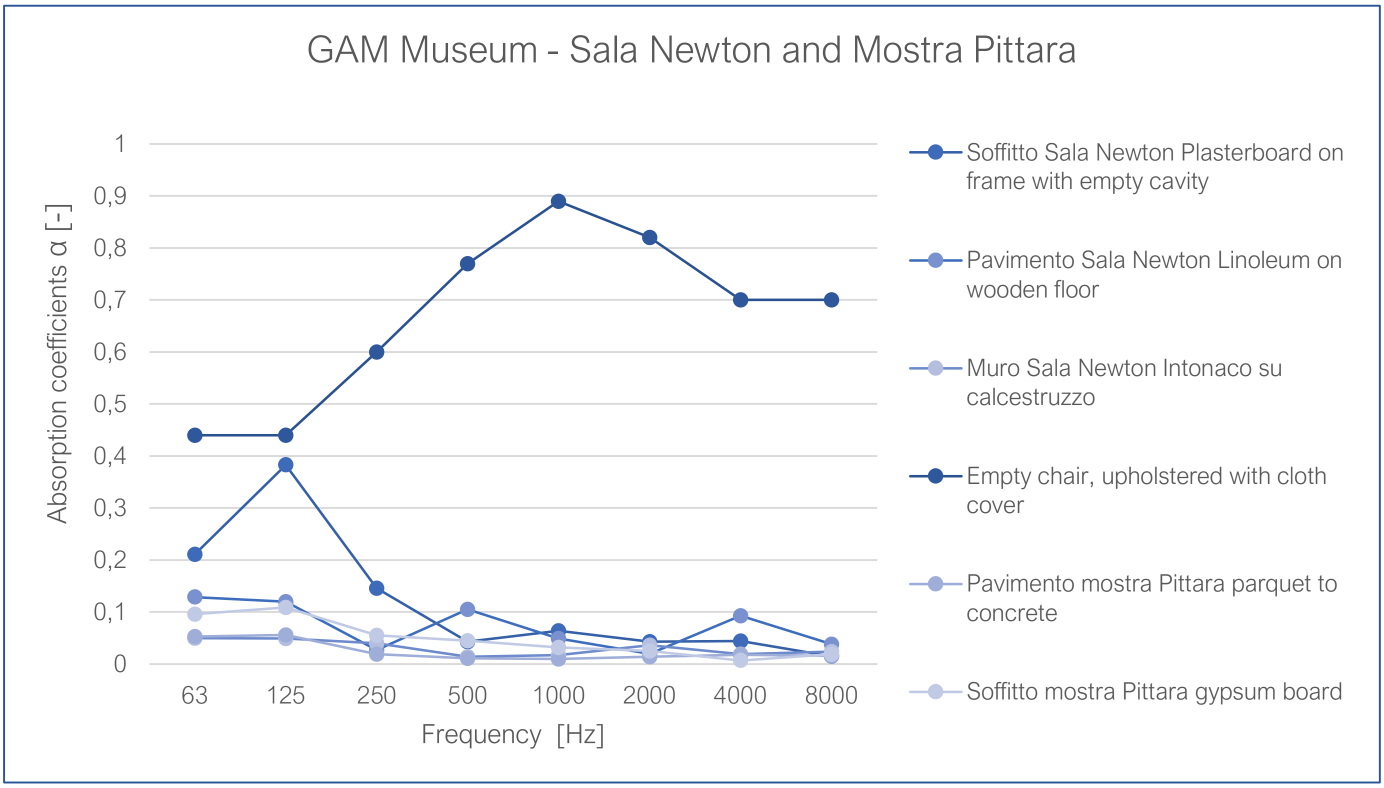






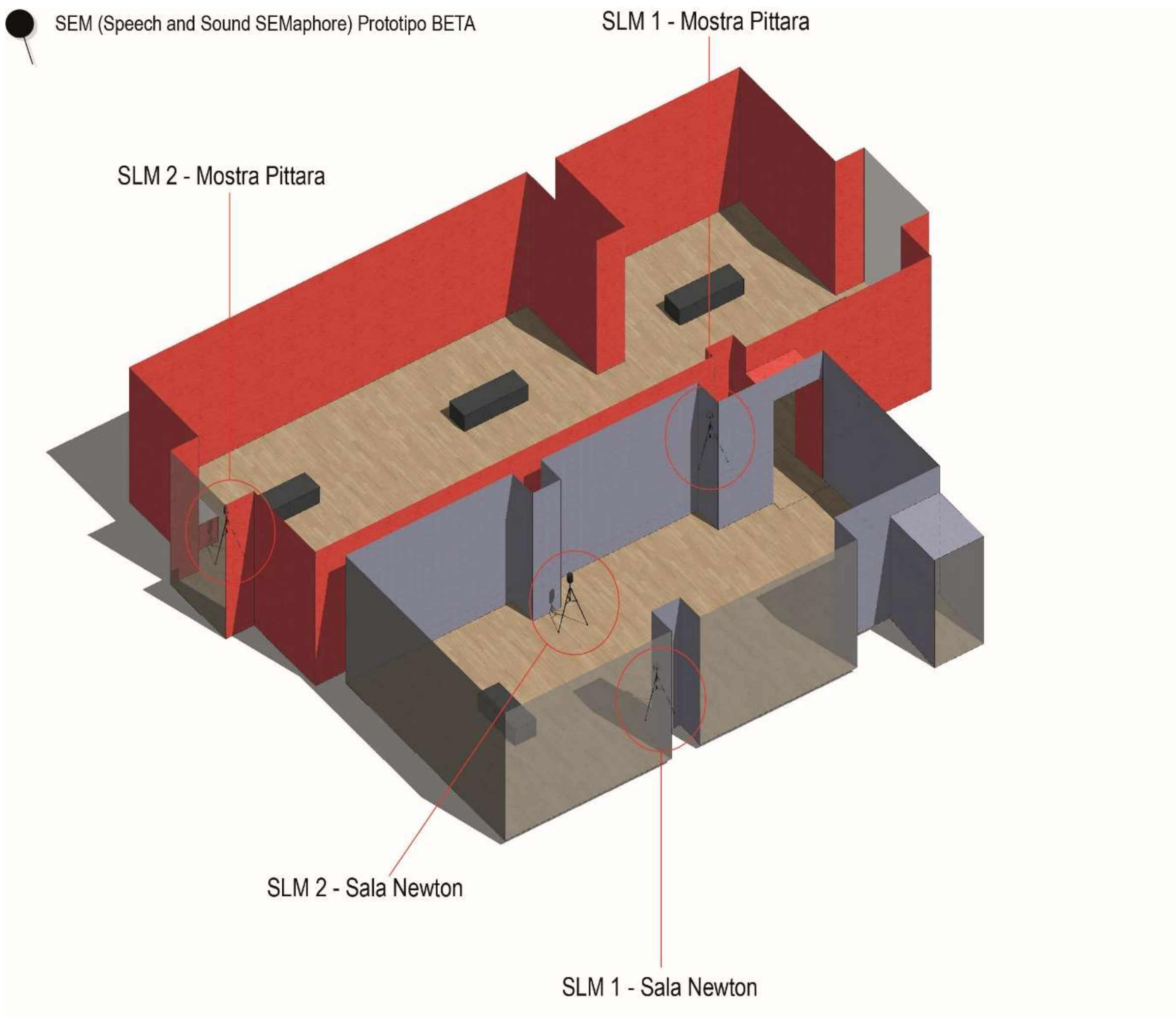


GAM Museum - Sala Newton and Mostra Pittara								
Materials	Absorption coefficients as a function of frequency [Hz]							
	63	125	250	500	1000	2000	4000	8000
4000 - Lime cement plaster (Bobran, 1973)	0,01	0,02	0,04	0,02	0,03	0,05	0,08	0,05
5600 - Soffitto Sala Newton plasterboard on frame with empty cavity	0,21	0,38	0,15	0,04	0,06	0,04	0,04	0,02
56001 - Pavimento sala Newton linoleum on wooden floor	0,13	0,12	0,03	0,11	0,05	0,02	0,09	0,04
56004 - Muro Sala Newton intonaco su calcestruzzo	0,05	0,05	0,04	0,01	0,02	0,04	0,02	0,02
56005 - Muro mostra Pittara intonaco su calcestruzzo	0,03	0,10	0,07	0,03	0,02	0,03	0,01	0,01
56002 - Soffitto mostra Pittara gypsum board	0,10	0,11	0,06	0,05	0,03	0,03	0,01	0,02
56006 - Pavimento mostra Pittara parquet to concrete	0,05	0,06	0,02	0,01	0,01	0,01	0,02	0,02
10002 - Single pane of glass, 3 mm (Fasold & Winkler, 1976)	0,08	0,08	0,04	0,03	0,03	0,02	0,02	0,02
92 - Empty chairs, upholstered with cloth cover	0,44	0,44	0,60	0,77	0,89	0,82	0,70	0,70
15229 - Marmo	0,01	0,02	0,02	0,02	0,02	0,03	0,03	0,01



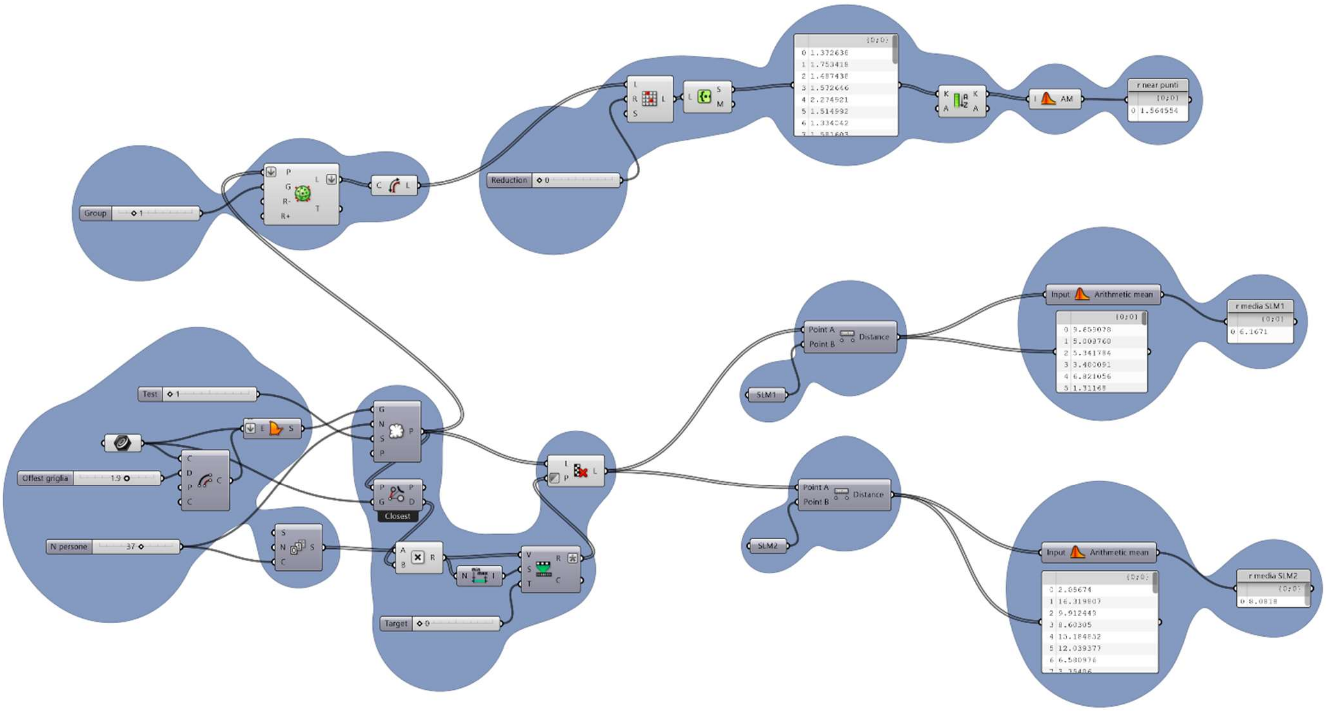
Parameter summary from acoustic simulation Odeon and r_{ij} in Grasshopper – Sala Newton

c=Lombard slope	ap	t_0	Volume	N_{max}	D_2S	$L_{N,A,1}$ funzione di trasferimento	r_{ij} SLM1-2
0,50 [dB/dB]	0,25 [m^2]	2,62 [s]	308 [m^3]	33 [-]	0,9 [dB]	60,3 [dB(A)]	4,68 [m]



Parameter summary from acoustic simulation Odeon and r_{ij} in Grasshopper – Mostra Pittara

c=Lombard slope	ap	t_0	Volume	D_2S	$L_{N,A,1}$ funzione di trasferimento	r_{ij} SLM1	r_{ij} SLM2
0,50 [dB/dB]	0,25 [m^2]	2,70 [s]	560 [m^3]	1,2 [dB]	58,7 [dB(A)]	6,20 [m]	8,10 [m]

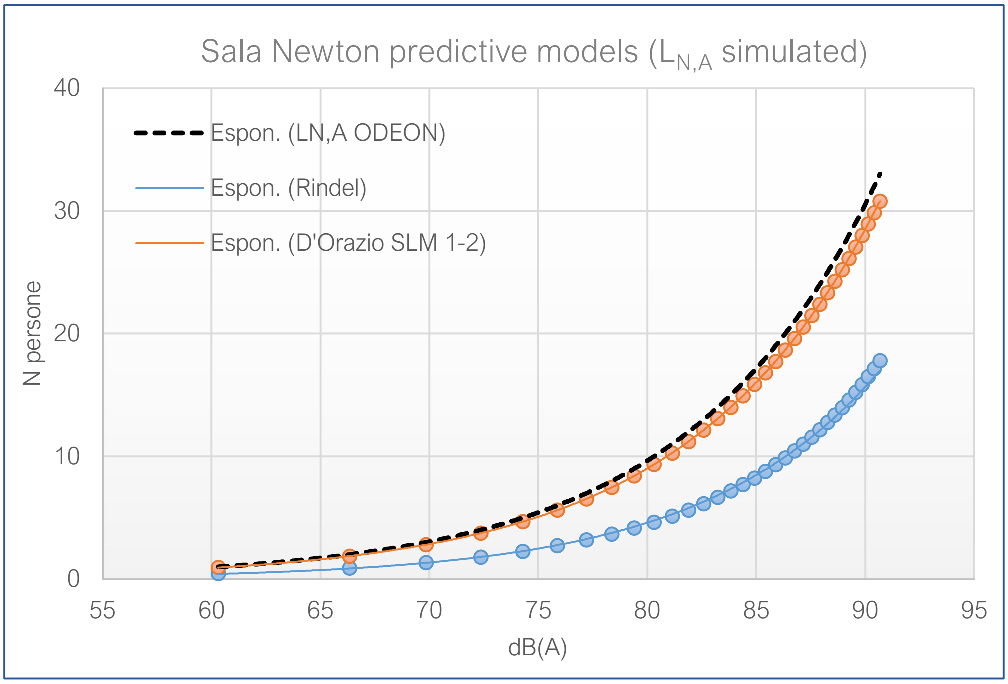


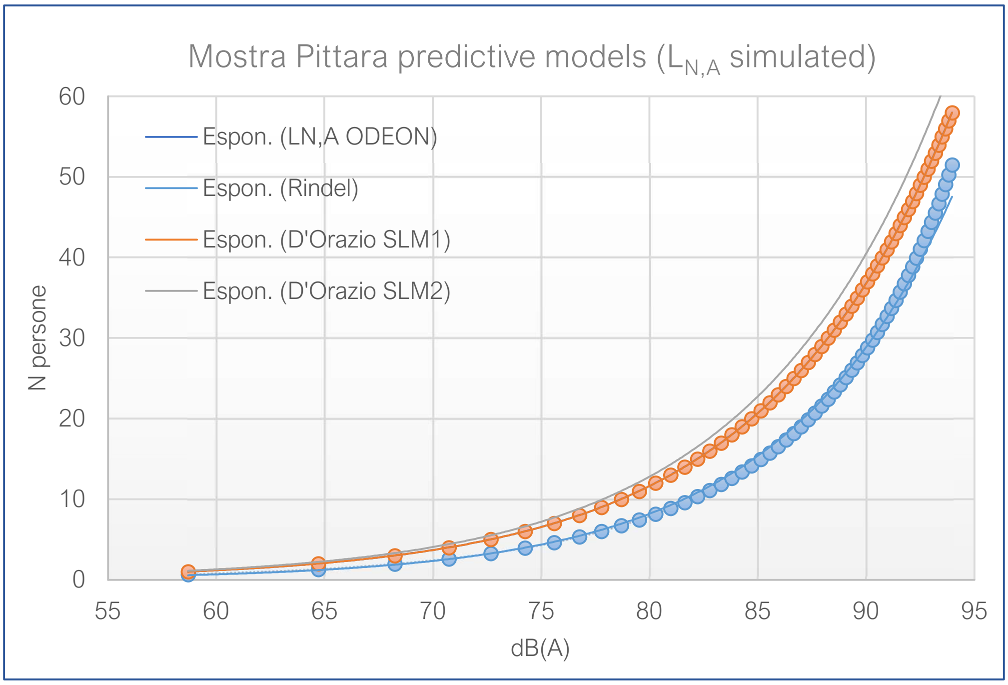
SEM (Speech and Sound SEMaphore) Prototipo BETA
I-esime sorgenti sonore simulate con Grasshopper
Distanza dalle superfici
SLM 2
Area superficie utile
0,50 m
d = 1,5 m
SLM 1 - 2
Area superficie utile
d = 1,8 m
SLM 1
0,20 m
Distanza dalle superfici

Application of previsional algorithms

In Newton Room characterized by a Volume = 308 [m^3], a t_0 = 2,62 [s] and a D_2S = 0,90 [dB] it is possible to evaluate as represented in the graph of the following page, that the forecast model that best approximates the N data simulated with Odeon is the model of D. D'Orazio while the model of J. H. Rindel deviates from the simulated data. This result is in contrast to what has been verified in the previous paragraphs, in fact with a t_0> 2 [s] a better approximation with the Rindel model is expected and this hypothesis could also be verified by the value of the D_2S which in this environment is <1 [dB] confirming the fact that the considered environment is characterized by a prevalence of diffuse acoustic field for which the hypothesis of uniformity is valid and between two successive points from the sound source there is a decay of the sound pressure for which in almost all the points of the space considered there is a comparable level of $L_{N,A}$. However, looking at the comparison graph between the forecast models, there is a countertrend with respect to what has just been stated because despite the D_2S = 0.90 [dB] the model of D. D'Orazio returns values of N= number of people, more reliable than the model of J. H. Rindel.

On the other side, in the Pittara Exhibition room immediately adjacent to the Saletta Newton, the model that returns data comparable with those simulated on the Odeon is the Rindel model, therefore the trend already verified in the environments of the previous paragraphs is reconfirmed. In fact, the environment of the Pittara Exhibition is characterized by a t_0 = 2,7 [s] and a D_2S = 1,2 [dB] consequently has all the characteristics so that there is a prevalence of diffuse acoustic field with respect to the direct field. In fact, as you can also appreciate in the following page chart, the model that best approximates the simulated data on the Odeon is the Rindel model.


Sala Newton predictive models (LN,A simulated)
Espon. (LN,A ODEON)
Espon. (Rindel)
Espon. (D'Orazio SLM 1-2)
N persone
dB(A)
0
10
20
30
40
55
60
65
70
75
80
85
90
95


Mostra Pittara predictive models (LN,A simulated)
Espon. (LN,A ODEON)
Espon. (Rindel)
Espon. (D'Orazio SLM1)
Espon. (D'Orazio SLM2)
N persone
dB(A)
0
10
20
30
40
50
60
55
60
65
70
75
80
85
90
95

7 | CONCLUSIONS

Application of predictive algorithms in the 4 environments described in the previous chapter in which it is expected to install the SEM (speech and sound SEMaphore) device has returned reliable results of the value N= number of people for 50% of cases applying only the model of J.H. Rindel, for 25% considering the model of D. D'Orazio and for the remaining 25% mixing between the two forecasting algorithms as schematized in the following table :

Environment	D_2S	t_0	Volume	H ceiling	Predictive algorithm
Liceo Alfieri	2,25	1,70	763	3,1	Rindel
Scuola Convitto Umberto I	1,40	2,10	1280	4,6	Rindel + D'Orazio ingresso
GAM - Mostra Pittara	1,20	2,70	560	4,3	D'Orazio SLM1
GAM - Sala Newton	0,90	2,60	308	4,1	D'Orazio SLM1+SLM2
Palazzo Madama - Sala delle Guardie	0,50	2,80	1050	8,5	Rindel

As is possible to see in previous table, when the reverberation time t_0 of furnished but unoccupied environment takes on higher values, the environment is consequently more reverberant and for this reason the sound waves propagated starting from a generic source are infinitely reflected by surfaces, generating points of the environment where there is a uniform acoustic density and for this reason Sabine's hypotheses on the reverberated field is verified. A consequence of what has been said is found in D_2S value that represents the sound decay curve for each doubling of the distance from the sound source considered; in fact in environments with a reverberation time t_0> 2 [s] the D_2S <1,5 [dB] since moving further and further away from the source in a reverberant environment the sound decays rather slowly as a direct consequence of the hypothesis of uniformity of the diffuse field, so between 2 successive points of the considered space we will have a similar sound pressure level. This relationship between t_0 and D_2S we find in all simulated environments, in fact as the reverberation time decreases we will have an increase in the value of the sound decay as if the source were placed in free field, therefore in spaces without reflective delimiting surfaces. It is also good to remember that in the free field the D_2S= 6 [dB] so the closer we get to this value the less reverberating the analyzed environment will be. However, although this relationship is valid in all the analyzed environments, it is possible to note in the previous table, how the applicability of the forecasting models follows an inverse trend in some cases; in the Sala delle Guardie of Palazzo Madama you have a t_0= 2,80 [s] and a D_2S= 0,50 [dB] and refuting what has just been said, the model that best approximates the values of N = number of people simulated on Odeon is the model of J.H. Rindel. In the other environments, such as in the case of the Pittara Exhibition of the GAM in which a t_0= 2,70 [s] and a D_2S= 1,20 [dB] was measured, the model that best approximates the values of N is the model of D. D'Orazio, despite that the

environment is characterized by a t_0 similar to the case of the Sala delle Guardie of Palazzo Madama. A similar trend also occurs in the case of the Newton Room bordering the Pittara Exhibition, in which despite a t_0= 2,60 [s] and a value of D_2S= 0,90 [dB], which therefore indicate how the environment is strongly reverberating, the model that best approximates the values of N also in this case is the model of D. D'Orazio. However, if we also take into account the Volume characterizing each environment, it is possible to note that the applicability of forecasting models is strongly influenced by this value. In fact, for environments characterized by a Volume <600 [m^3] it is advisable to apply the model of D. D'Orazio, while for environments with a Volume> 600-700 [m^3] the model that more reliable N values is certainly the model of J.H. Rindel.
The future objective of the present research work is to apply these predictive models within the SEM device to be able to measure in real time levels of anthropic noise $L_{N,A}$ produced by users present in analyzed 2 schools and 2 museum of Turin, so that SEM device is able to calculate through the predictive algorithms in real time the number of people present within the studied environments so as to prevent people from gathering indoors contributing to spread the Sars-Covid 19 virus. The application of predictive algorithms is possible starting from known parameters of analyzed environments such as the reverberation time t_0, the Volume of the environment, D_2S that can be both measured within the real and simulated environment on Odeon as described in this thesis and the mutual distance between an i-th source and fixed position of the receivers, in our case represented by SEM using the algorithm set on the Grasshopper for Rhinoceros application. The only parameters not known a priori are the c= Lombard slope that we have verified to provide reliable results in environments with different uses when assuming c= 0.5 [dB] and the g= group size, that can be estimated a priori considering the intended use of analyzed environment; in catering rooms takes values equal to g= 3-4 [-] but in the present thesis it has been verified that to estimate the value of g is very important to consider also the boundary conditions that can impact on levels of anthropogenic noise $L_{N,A}$ produced by speakers in closed places; in our case values of $L_{N,A}$ have been strongly influenced by the context of Sars-Covid 19 pandemic which has forced to halve entrances in closed places to comply with impositions of the D.P.C.M. of 26 October 2020 n. 265, for which the g=group size assumes values doubled compared to standard values, so that if under normal conditions in catering places g=3-4 [-] during the Covid period g= 8 [dB].

Appendix

1 | NOTES ON ACOUSTIC PHYSICS

The term sound comes from the Latin "sonus" is represented by the feeling generated by the vibration of a body oscillating in a means of propagation. This vibration is only possible if the source emitting the sound is immersed in an elastic medium, be it a gas such as air, a liquid such as water and so on. The elastic means of propagation must also have inertia in order to allow, during oscillations, the transfer of mechanical energy from one particle to another; in fact, during an oscillation the solicited particles transmit their movement to other particles placed in the vicinity until they generate sound waves, i.e. localized pressure variations caused by an alternation between phases of pressure increase/decrease at the same point in space. Sound waves are also called longitudinal waves because the particles of which the medium is composed move in the same direction of propagation as the pressure wave. The propagation mode of a sound wave can be reproduced by a mechanical piston that moves in harmonic continuous motion, producing progressive plane waves that can be summarized as follows:

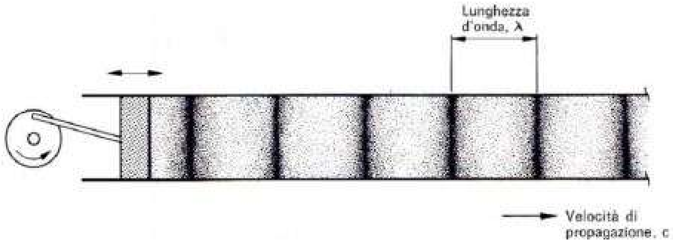


Figure 1 - M. Lainati

- $f = \frac{1}{T} = \frac{c}{\lambda}$ $[Hz = s^{-1}]$ the frequency of a sound wave represents the number of cycles of the same waveform calculated in a period T= 1 [s];
- c= speed of propagation of sound in the medium in question, in the air c= 343 [m/s] at a temperature of 20 °C and at atmospheric pressure at sea level;
- λ=wavelength [m].

Since the frequency of a wave is directly proportional to the rate of propagation in the medium considered and inversely proportional to the wavelength itself, it will be that a sound propagating in the air at a temperature of 20°C characterized by a f= 20 [Hz] will have a λ= 17 [m] while a sound with f= 2000 [Hz] will have a λ= 0.017 [m].

1.1 | BASIC ACOUSTIC QUANTITIES

Sound pressure

During the propagation of a sound wave, the particles contained in the elastic propagation medium oscillate causing local pressure variations. This pressure change can be calculated as follows:

$$p = \sqrt{\frac{1}{t_2 - t_1}\int_{t_1}^{t_2} p(t)dt} \quad [\mathrm{Pa}]$$

The value p thus calculated represents the mean quadratic deviation RMS of the pressure calculated over a sufficiently long time interval t_2-t_1 with respect to the oscillation period of the wave in question.

Sound intensity and audibility thresholds

Considering a generic sound source, be it pointy, linear or superficial, we will have that the sound intensity of the source corresponds to its sound power W transmitted through surface unit A perpendicular to the direction of propagation of the longitudinal wave considered, expressible as:

$$I = \frac{dW}{dA} \quad \left[\frac{\mathrm{W}}{m^2}\right]$$

In the case of point sources, surface unit A corresponds to the surface of a sphere for which the resulting sound intensity will be equal to $I = \frac{dW}{4\pi r^2} \quad \left[\frac{\mathrm{W}}{m^2}\right]$

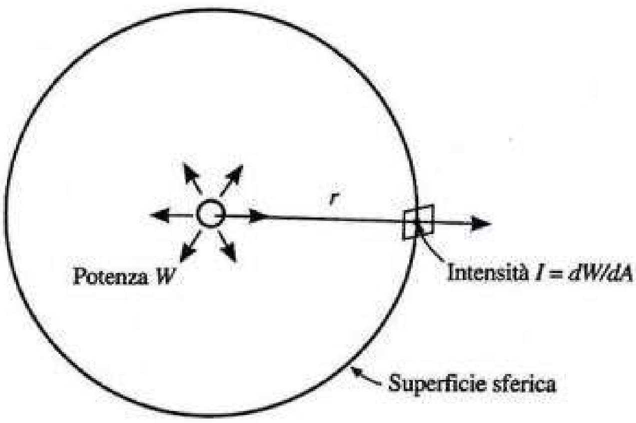


Figure 2 - M. Lainati

The human ear is able to hear sounds, or changes in sound pressure, which fall within a range called the threshold of audibility and pain threshold; the first corresponds to the minimum intensity I_{min} that the ear is able to pick up equal to:

$$I_{min} = 10^{-12} \left[\frac{\mathrm{W}}{m^2}\right]$$

The audibility threshold corresponds to a pressure change with respect to the reference atmospheric pressure of only 20μPa at 1000 Hz, while the pain threshold will be:

$$I_{max} = 1 \left[\frac{\mathrm{W}}{m^2}\right]$$

The pain threshold corresponds to the maximum intensity that the ear is able to withstand beyond which sound is perceived as a feeling of pain and involves permanent damage even in values just below that threshold. The changes in sound pressure perceived by the ear therefore occupy as many as 12 orders of magnitude for this reason the intensity of a sound is expressed in logarithmic scale:

$$I(dB_{SIL}) = 10 \ \log_{10}\left(\frac{I}{I_{min}}\right)$$

Free-range pressure, power and sound intensity levels

The sound intensity level is a pure number, so a dimensionless magnitude obtained by comparing the calculated sound intensity to the reference sound intensity. It is, however, clearly attached to a unit of measurement the decibel referred to as [dB]. The sound pressure level referred to as SPL indicates the dB measurement of atmospheric pressure deviation as a result of oscillation caused by a sound wave. Since the sound wave causes local and instantaneous pressure variations, an effective level corresponding to the mean quadratic deviation of the sound pressure level in a given time interval is referred to as:

$$SPL = L_p = 10 \ \log_{10}\left(\frac{p^2}{p_0^2}\right) = 20 \log_{10}\left(\frac{p}{p_0}\right) \ [\mathrm{dB}]$$

where:

- p= standard deviation of the pressure in question;
- p_0= reference atmospheric pressure of 20μPa at 1000 Hz.

Similarly, it is possible to calculate the sound power level also measured in dB as follows:

$$L_W = 10\ \log_{10}\left(\frac{W}{W_0}\right)\ [\mathrm{dB}]$$

where:

- W_0= reference sound power of10^{-12} [W/m^2].

Finally, the sound intensity level can be calculated by the same process by taking:

$$L_I = 10\ \log_{10}\left(\frac{I}{I_0}\right)\ [\mathrm{dB}]$$

where:

- I_0= reference sound intensity of10^{-12} [W/m^2].

In the case of propagation of a sound in the air at a temperature of 20 ° C we will have that the corresponding sound intensity will be of which the term to the denominator represents the acoustic impedance of the medium and replacing this term in the equation for the calculation of the sound intensity level we will have that:$I = \frac{p^2}{c\, p_0}$

$$L_I = L_p\ = 10log\frac{I}{I_0} = \ 10log\frac{p^2}{c\, p_0} \times \frac{c\, p_0}{p_0^2} = 10log\frac{p^2}{p_0^2}$$

So, when a sound propagates in the air at a temperature of 20°C the sound intensity level and sound pressure level will match.

1.2 | INDOOR SOUND PROPAGATION

Depending on the environment in which the sound source is immersed, the sound rays propagating from the above source follow different modes and directions, since the propagation is strongly conditioned by the physical characteristics of the surrounding space. Starting from these simple indications it is possible to define two different types of acoustic field:

- direct sound field: it is the type of sound propagation that occurs when the source and receiver are positioned rather close together and the acoustic quantities take values comparable to those assumed if the source is placed in a free field, that is, an environment free of obstacles that modify the path of the sound rays emitted by the source;
- diffuse sound field: this is the type of propagation that occurs when the source is in an environment whose bounding surfaces are not very absorbent and for this reason most sound waves are constantly reflected in the environment itself generating numerous and continuous overlaps of sound waves.

In a closed space, whether characterized mainly by a direct or diffused acoustic field, the bounding surfaces help to absorb part of the energy emitted by the source due to the absorption coefficients α of surfaces. The overall absorption coefficient of an environment can be calculated as the summation of each α multiplied by the corresponding areas of area S j-th as follows:

$$\alpha_{med} = \frac{\alpha_1 S_1 + \alpha_2 S_2 + .. + \alpha_n S_n}{S_{tot}} = \frac{1}{S}\sum_i \alpha_i S_i$$

Hence that the total absorption area of the environment also called the equivalent sound absorption area will be:

$$A_{tot} = \alpha_{med}\, S_{tot}\, [m^2]$$

Acoustic energy density

Remembering that the acoustic energy density is the amount of sound energy localized in the volume of the around one point i-th of the propagation medium can be calculated as:

$$\mathrm{D} = \frac{I}{c} = \frac{p^2}{p_0 c}\left[\frac{W}{m^2}\right]$$

In the case of a point source radiating its sound power omnidirectionally in an environment characterized by a diffuse acoustic field, the energy density resulting will be given by the sum between the sound density of the direct field D_d to which is added the energy density of the reverberated field D_r:

$$D(r) = D_d + D_r \left[\frac{W}{m^2}\right]$$

Having hypothesized a point source we can say that the D_d factor d is also equal to:

$$D_d = \frac{W}{c4\pi r^2} \times Q \left[\frac{W}{m^2}\right]$$

where:

- Q= directivity factor of the source considered [-].

Sabine hypothesis and reverberated field energy density

To calculate factor D_r, however, reference should be made to the Wallace Clement Sabine hypotheses (1868-1919) which first demonstrated the relationship between the reverberation time of an environment with its size and the total amount of sound absorption. The reverberation time is defined by ISO 3382 as the time it takes for the acoustic energy density at a point to under 10^6 times the value it had at moment when the direct wave ceased to reach the same point considered. Sabine's hypothesis is based on 3 main assumptions:

- uniform acoustic energy density: the acoustic density D(x,y,z) is assumed to be uniform throughout the eiclosed space considered, although we know that the acoustic density is variable at every point in space due to oscillations caused by sound wave propagation and local pressure variations with respect to the initial reference pressure p_0. This assumption is as true as the environment is reverberating and therefore the sonor rays are distributed evenly and densely within the space;
- validity of statistical acoustics: Sabine introduces the concept of the average free path L_{med} and free average time t_{med} which correspond to the distance, or time, that a sound wave travels in the interval between two successive reflections that can be calculated as:

$$L_{med} = \frac{4V}{S}\ [\mathrm{m}] t_{med} = \frac{L_{med}}{c} = \frac{4V}{Sc} [\mathrm{s}]$$

- continuity hypothesis: it is assumed that in any time interval between two successive reflections, there is absorption of acoustic energy by the elements delimiting the environment.

It can therefore be said that the energy density at full speed in the diffuse field D_r will be:

$$D_r = \frac{4W}{Sc\alpha_{med}} = \frac{4W}{A_{tot}c}\ \left[\frac{W}{m^2}\right]$$

Taking up the definition of reverberation time provided by ISO 3382 in which the time interval that D_r takes to decrease by 10^6 times from the moment the direct field is no longer present is taken, it can be assumed that t_{60} is equal to:

$$t_{60} = -\ln(10^{-6})\frac{4V}{cA_{tot}} = 0{,}16\frac{V}{A_{tot}}\ \ [\mathrm{s}]$$

8 | BIBLIOGRAPHY

- ODEON Room Acoustics Software, User's Manual Version 16, marzo 2020;
- ODEON APPLICATION NOTE – *ISO 3382-3 Open plan offices* – Part 2 Measurements, 2014;
- ODEON APPLICATION NOTE –*Restaurants*, 2012;
- SPAGNOLO R, *Manuale di Acustica*, Torino, UTET, Marzo 2001;
- NICOLINI A., *Dispense di Acustica degli ambienti chiusi*, Perugia, CIRIAF, A. A. non indicato.

Normativa

- Norma ISO 1999:1990 "*Acoustics -- Determination of occupational noise exposure and estimation of noise-induced hearing impairment*" sostituita dalla ISO 1999:2013 "*Acoustics -- Estimation of noise-induced hearing loss*";
- Decreto Legislativo 9 aprile 2008, n. 81, art. 188 "*Protezione dei lavoratori contro i rischi di esposizione al rumore durante il lavoro*";
- ISO 9921:2003, "*Ergonomics – Assessment of speech communication (International Organization for Standardization)*", Geneva, Switzerland, 2003).
- CEI EN 61672:2003 "*Elettroacustica - Misuratori del livello sonoro*" Parti 1,2,3;
- ISO 14257:2001 "*Acoustics – Measurement and parametric description of spatial sound distribution curves in workrooms for evaluation of their acoustical performance*", International Organization for Standardization, Geneva, Switzerland, 2001;
- ANSI 3.5 1997, "*American National Standard – Methods for Calculation of the Speech Intelligibility Index*", (1997).

Articoli

- Lombard E. *Le signe de l'élévation de la voix*. Ann. Mal. Oreil. Larynx 1911, 37, 101-119;
- Pickett J. M. *Limits of direct speech communication in noise*. J. Acoust. Soc. Am. 1958, 30(4), 278–281;
- Gardner M.B. *Factors Affecting Individual and Group Levels in Verbal Communication*. J. Audio. Eng. Soc. 19, 1971, 560-569
- Lane H., Tranel B. *The Lombard sign and the role of hearing in speech*. J. Speech, Language and Hearing Research 1971, 14, 677-709;
- Lazarus H. *Prediction of Verbal Communication in Noise - A Review: Part 1*. Applied Acoustics19, 1986, 439-464;
- Pick Herbert L., Siegel Gerald M., Fox Paul W., Garber Sharon R., Kearney Joseph K. *Inhibiting the Lombard effect*. J. Acoust. Soc. Am. 1989, 85, 894;
- Tang S.K., Daniel W.T. Chan, K.C. Chan. *Prediction of sound-pressure level in an occupied enclosure*, JASA 101 (5), 1997, pp. 2990-2993;
- Muggeo V. M., *Estimating regression models with unknown break-points*, Stat. Med. 22(19), 3055–3071 (2003);
- Hodgson M, Steiniger G, Razavi Z. *Measurement and prediction of speech and noise levels and the Lombard effect in eating establishments*. JASA 2007;121:2023–33;
- Navarro M.P.N., Pimentel R.L., *Speech interference in food courts of shopping centres*. Appl. Acoust. 68, 2007, 364-375;
- Rindel, J.H. *Verbal communication and noise in eating establishments*. Appl. Acoustics 2010, 71, 1156-1161;
- Hayne M.J., Taylor J.C., Rumble R.H., Mee D.J., *Prediction of Noise from Small to Medium Sized Crowds*, Proceedings of ACOUSTICS 2011, Gold Coast, Australia, 2-4 November 2011;
- Ruiter, E.Ph.J. de. *Lombard effect, speech communication and the design of (large) public spaces*, Forum Acusticum 2011, Aalborg;
- Rindel J. H. Christensen C.L., Gade A.C.: *Dynamic sound source for simulating the Lombard effect in room acoustic modeling software*. Proceedings of Inter-Noise 2012, New York, USA, (2012);

- Rindel, J.H. *Acoustical capacity as a means of noise control in eating establishments*. Proc. of BNAM 2012, Odense, Denmark, 18-20 June 2012;
- Ruiter E.Ph.J. de. *Feedback from the foodcourt*. The 22nd International Congress on Sound and Vibration July 2015, Florence, Italy;
- Battaglia Paul L. *Achieving acoustical comfort in restaurants*. J. Acoust. Soc. Am. 2014, 136, 2018;
- Bottalico P., *Lombard effect, ambient noise, and willingness to spend time and money in restaurants*, J. Acoust. Soc. Am. 144 (3), EL209–214, September 2018;
- Pinho P.G., Pinto M., Almeida R.M.S.F., Lemos L.T., Lopes S.M., *Aspects concerning the acoustical performance of school cafeterias*, Appl. Acoust.,136,2018,36–40;
- Di Blasio S., Vannelli G.,Shtrepi L., Puglisi Giuseppina E., Calosso G. Minelli G., Murgia S., Astolfi A., *Long-term monitoring campaigns in primary school: the effects of noise monitoring system with lighting feedback on noise levels generated by pupils in classrooms*, Inter-noise 2019, Madrid, 16-19 giugno 2019;
- Gallo E., Shtrepi L., *Long term monitoring of noise pollution in social gatherings places: time analysis and acoustic capacity as support of management strategies*, Proceedings of ACOUSTICS 2019, Aachen, Germany, 9-13 settembre 2019;
- D'Orazio D., Montoschi F., Garai M., *Acoustic comfort in highly attended museums: A dynamical model*, Build. Environ., 183, 2020, 107176;

Tesi di laurea

- E. MARGESIN, *Progetto e sonorità dello spazio: una proposta di modellazione avanzata per la sede del Dipartimento di Lettere dell'Università di Trento*, Università degli studi di Trento, Facoltà di Ingegneria Civile Ambientale Meccanica, relatori: Prof.ssa Arch. Giovanna A. Massari, Prof. Ing. Paolo Baggio, A. A. 2016-2017;
- M. LAINATI, *Contributo dell'effetto Lombard nell'aumento del confort acustico in un ambiente*, Politecnico di Milano, Facoltà di Ingegneria dell'Informazione, relatore: prof. Giuseppe Bertuccio, A.A. 2011-2012;
- C. VIAZZO, *L'acustica delle mense scolastiche*, Politecnico di Torino, Facoltà di Architettura, relatori: Prof. Chiara Aghemo, Prof. Arch. Arianna Astolfi, A. A. 2004-2005.

Sitografia

- Forum Rhinoceros, rhino3d.com/, consultato per implementare il modello previsionale in ambienti caratterizzati da campo acustico diretto;
- Forum Grasshopper, grasshopper3d.com/, consultato per implementare il modello previsionale in ambienti caratterizzati da campo acustico diretto;

Elenco dei software utilizzati

- Adobe Illustrator e Photoshop, per l'elaborazione delle immagini e delle mappe;
- Office Word, per l'impaginazione della tesi;
- Office Excel, per l'analisi delle misurazioni dei livelli di rumore antropico $L_{N,A}$ e dati counter;
- ODEON, per la simulazione acustica;
- Rhinoceros con il suo plugin Grasshopper per per implementare il modello previsionale in ambienti caratterizzati da campo acustico diretto;
- R versione 3.1.2 per l'analisi statistica dei dati.