SCALED ACOUSTIC BARRIER INSERTION LOSS MEASUREMENT

CUPAIUOLO, CARLA [1] FREIDKES, JONATHAN [2] AND KAPLUN, TOMAS [3]

[1] Universidad Nacional de Tres de Febrero, Caseros, Argentina.
c.cupaiuolo@gmail.com
[2] Universidad Nacional de Tres de Febrero, Caseros, Argentina.
jonifreidkes@gmail.com
[3] Universidad Nacional de Tres de Febrero, Caseros, Argentina.
tomaskaplun@gmail.com

Abstract – This paper proposes an experimental study of the insertion loss (IL) provided by an acoustic barrier designed in acoustic scale mode. An office separation barrier was built in scale 1:3, 3 different structural modifications were made and the results are discussed in various aspects comparing with the theoretical calculations. Some conclusions could be obtained about the method and particularities.

1. INTRODUCTION - PREVIOUS STUDIES

The acoustic barrier can be a very good solution when a isolation needs to be done. This is commonly used in offices, industries, highways, and many other places in order to acoustically separate two zones. One of the most important parameter of an acoustic barrier is its insertion loss (IL). This parameter can be obtained by measurement *in situ*, once the barrier is placed. However, sometimes the barrier is not already constructed and the parameter needs to be predicted before the barrier is built in order to know how it will impact in the environment. In this case, a scaled simulation can be a good method because of its many benefits. This method provides a cheaper, faster, and easier way to determine the IL of the object being designed. This kind of simulation is often used in acoustics engineering and architecture to obtain the basic acoustical characteristic of the barrier in an early stage of the project, before the barrier is built.

There are many previous studies performed by researchers, engineers and students related to the subdivision of the acoustic science that analyzes scaled methods. The first author to analyze the scaled models was Spandock [1] in 1934. Since then, scale-modelling techniques have been applied to study concert halls, barriers, enclosures, factories, and outdoor sound propagation. Heading specifically to acoustical barriers, C. Arango and V. Molina [2] constructed a 1:16 scaled barrier to analyze the IL that will be obtained in the original proportions; the barrier was going to be used to isolate the traffic road noise, so the authors first analyzed the spectrum and energy of the source and then designed specific barriers. There is not a standard that provides a method for measuring scaled barriers, they based their method in the normative ISO 9613-2 that contemplate only the barriers measuring; they made many arrangements of barriers in free field, and obtained a IL of approximately 10 dB for low frequencies and 20 dB for the high frequencies. They also concluded that the variations of adding a ceiling on top of the barrier increased the IL in just 1 dB, almost negligible. Another author, R. Bullon [3], studied the scale models in a 1:10 scaled auditory; as the concert halls and auditoriums are also very common to be scaled, he analyzed the main acoustical parameters of the Physics Auditorium of the University of Peru. He first measured the real auditorium and then compared the results with the ones obtained from the scaled models. He found that the main acoustical parameters (RT, Clarity, Center time, EDT) were well correspondent between the real and the scaled model. Also, S. Voropayev et al. [4] evaluated different tops of scaled acoustic barriers using a 1:10 rate; he used a high voltage spark discharge as the source, in order to reach ultrasonic frequencies, with a omnidirectional pattern and high stability, needed for the scaled model. The measurements were taken in a regular room and then with post processing, the obtained signal was windowed in order to eliminate the reflection coming from the room boundaries. The 'T' shape was founded as the one that provide the higher IL. Another important study was the one realized by J. Farley and B. Anderson [5], who calculated IL of a barrier for ultra high frequencies (from 25 KHz to 96 KHz), but in this case they did not scale the barrier. Several materials were used: MDF, steel and polycarbonate and the IL analysis was in a wide band of high frequencies. The results show that the IL increase from 40 to 60 dB, approximately, as the frequency rises. Furthermore, T. busch and M. Hodgson [6] studied the behaviour of many scaled highway barriers applying different materials and heights; as the rate of 1:31 was adopted for the barriers a high frequency source and microphone were needed. They set an array of scaled barriers in an anechoic chamber and they also measured the IL. The source implemented was designed to reproduce ultrasonic frequencies and its core is an air-jet source, which can reproduce frequencies up to 100 KHz. The obtained results show a 5 dB IL for low frequencies and 15 dB for mid-high frequencies.

The purpose of this paper is to perform a measurement of the IL provided by a scaled office barrier and analyze how the scaled model can be an advantage for the analysis. Different considerations and theories are evaluated in section 2, 3 and 4, later, in sections 5 and 6, the design, constructruction and measurement of the scaled office barrier and its different modifications are precisely detailed. On section 7 the results and discussions are presented and afterwards, in section 8 the conclusions are obtained.

2. ACOUSTIC BARRIERS

When a barrier or obstacle interferes a sound wave, part of the energy of the wave will be reflected by the barrier, other will be absorbed, another one diffracted and the remaining will be transmitted through the barrier. The main effect that is represented on the acoustical barriers is the diffraction.

Assuming that the materials of the obstacle are heavy enough in order to transmit as less as possible through it, the barriers base its acoustical property on the physical principle of the diffraction. This, is the property of a sound wave to envelop or surround an obstacle. The diffraction depends of the frequency of the wave and the size of the obstacle. This physic effect lays on the Huygens principle that determine that every specific point of a wave front can be considered as new spherical wave source that preserve the same velocity and frequency of the original wave.

As the diffraction is the main effect of the barriers, this implies that a it does not isolate at the same level for different frequencies, and that the lower frequencies can "involve" the obstacle with more efficiency than the higher frequencies. This mean that diffraction is much more accentuated for lower frequencies and thus induce that the barrier works "better" for higher frequencies.

When an acoustical barrier is analyzed three different influence zones can be determined. The lighting zone, where the influence of the barrier is despised, the shadow zone, where the effect of the barrier is appreciate, and the gloom (or transition) zone, where the effect of it is not very influential (between 0 and 5 dB_{SPL}). For this reason, when a barrier has to be located somewhere to isolate the emitter from the receiver as much as possible, this zone has to be taken into account.

2.1. CALCULATION METHODS

As it was already mentioned, the most important parameter of an acoustic barrier is the IL. There are different empirical methods to calculate the insertion loss: Keller [7], Kurze & Anderson [8] and Maekawa's [9] methods were used in this paper to theoretically calculate the IL provided by the barrier. In the following subsections these methods are explained.

2.1.1. KELLER

The geometrical theory of diffraction is an extension of geometrical optics which accounts for diffraction. It introduces diffracted rays in addition to the usual rays of geometrical optics. These rays are produced by incident rays which hit edges, corners or vertices of boundary surfaces or which graze such surfaces. In terms of the new rays, diffracted wave fronts can be defined. A Huygens wavelet construction can also be devised to determine them.

In the case of the ordinary rays, the field on a ray emerging from a source is specified at the source. But on a reflected or transmitted ray, the initial value is obtained by multiplying the field on the incident ray by a reflection or transmission coefficient. By analogy, the initial value of the field on a diffracted ray is obtained by multiplying the field on the incident ray by a diffraction coefficient, which is a matrix for a vector field. There are different coefficients for edge diffraction, vertex diffraction, etc.

Dimensional considerations show that edge-diffraction coefficients are proportional to $\lambda^{1/2}$. Diffraction coefficients depend on geometrical and physical properties.

According to Keller, the reduction of sound pressure level due to the insertion of a barrier between source and receiver can be calculated by equation 1.

$$\Delta L = -20\ log \frac{d}{2\ sin\beta\ 2\pi/\sqrt{\lambda}\ \sqrt{AB(A+B)}} \left| sec\frac{\theta-\alpha}{2} + csc\frac{\theta+\alpha}{2} \right| dB \quad (1)$$

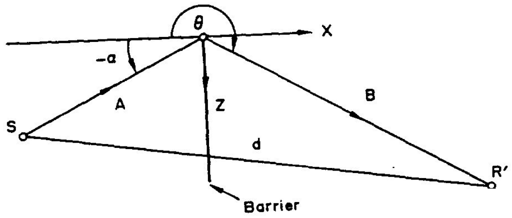


Figure 1: Keller's distances.

The distances A, B, and d and the angles θ and α are determined as shown in Figure 1. λ is the wavelength; β is the angle between the incident ray and the edge.

2.1.2. KURZE-ANDERSON

The attenuation of a sound barrier is determined by edge diffraction over or around the ends of the barrier and by sound transmission through the barrier. Kurze and Anderson derived an analytical approximation for the attenuation of spherical waves due to the edge diffraction over a rigid barrier, by a review of various theoretical and experimental results. This method takes into account the Fresnel number as shown in equation 2.

$$N = \frac{2\delta}{\lambda} \quad (2)$$

where λ is the wavelength of the sound and δ is the path-length difference which is defined by equation 3.

$$\delta = \pm (A + B - d) \quad (3)$$

As shown in Figure 2, (A+B) is the shortest path over the edge, from the source to the receiver in the shadow zone; d is the direct path distance between source and receiver through the barrier. δ is positive in the shadow zone of the barrier and negative in the bright zone.

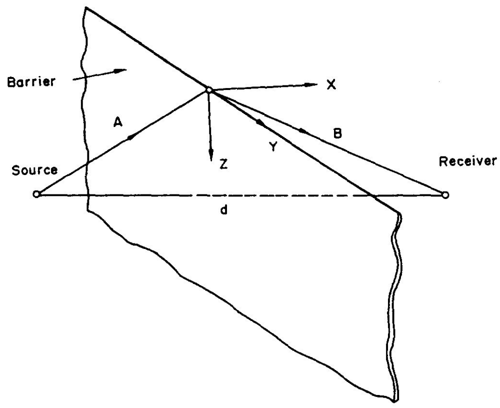


Figure 2: perspective view of a barrier.

Based on Rathe's [10] work and Keller's [7] geometrical theory of diffraction and equation for calculating the reduction of sound pressure level due to the insertion of a barrier between source and receiver, Kurze and Anderson arrived to an expression (shown in equation 4) that can be used for the calculation of the excess attenuation of sound rays that graze, or pass slightly above the top of the barrier, if one sets ΔL=0 for N<0.2.

$$\Delta L = 5\ dB + 20\ log \frac{\sqrt{2\pi N}}{tanh\sqrt{2\pi N}} dB \quad (4)$$

This expression is a good approximation to Keller's formula in the following particular cases:

- Receiver position is close to the barrier: $\delta \approx A + B$ and $\theta \rightarrow 3\pi/2$
  $$\Delta L = 5\ dB + 10\ log\frac{2\pi\delta}{\lambda}\ dB \quad (5)$$
- Small barrier height: $\delta \approx A + B$ and $\theta - \alpha \rightarrow \pi$
  $$\Delta L = 5\ dB + 10\ log\frac{8\pi\delta}{\lambda}\ dB \quad (6)$$
- N=0
  $$\Delta L = 5\ dB \quad (7)$$

2.1.3. MAEKAWA

Maekawa presented experimental data on the diffraction of sound rounding a semi-infinite plane screen in a free field and described a method for calculating the shielding effect of a real screen employed for the purpose of noise reduction.

The results of the experiments are shown approximately by one curve of attenuation depending on Fresnel's number N in the Figure 3. It also shows Redfrearn's [12] theory (— — —), Kirchoff's theory (- - -), and experimental values measured by Maekawa (o). When the receiving point is far enough from the ground to neglect the effect of the reflected sound, Figure 3 is very useful to estimate noise reduction by the barrier with more reasonable accuracy.

There have been many attempts to fit this results with a simple formula. One of the simplest is shown in equation 8 [9].

$$Att = 10\ log\ (3 + 20N)\ dB \tag{8}$$

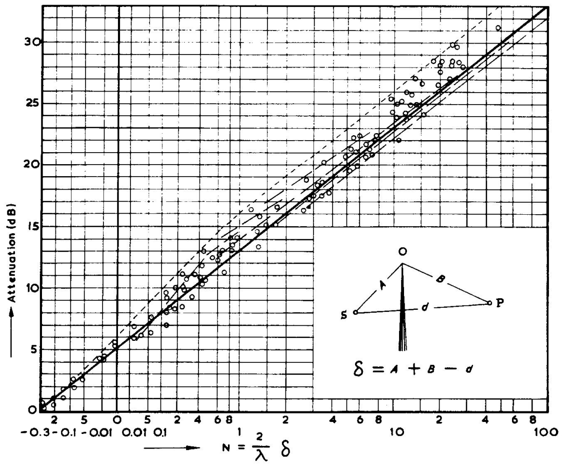


Figure 3: Maekawa's abacus.

2.2. OFFICE BARRIERS

According to a research of available office barriers in the market, it is possible to affirm that they are not manufactured meeting a standard contemplating sizes or materials. They can be made of MDF, plaster, wood, polymer, acrylic or mixed. The thicknesses are variable and can be double barriers, filled with air or absorbent. Generally they are used to separate visually or for aesthetic purposes.

3. SCALED METHOD

The scaled method is a very important tool when a simulation is carried on. It allows to make a relatively fast and easy setup of the model under test, and once the measurements in the scale model are done, the way back to the real model is as easy as applying an 'n' factor multiplying or dividing in each case. The following table shows the relative magnitudes of a scaled and the real ones:

Magnitude	Real model	Scaled model
Length	L	L/n
Frequency	F	F*n
SPL	Lp' - 20 log(n)	Lp'
Speed of sound	C	C

Table 1: Magnitude comparison between scaled and real model.

It has advantages such as an easy construction and the possibility of applying the n factor, but there are also some disadvantages. One of the most important is the absorption coefficient, which depends on frequency and when a scaled model is carried out the frequencies have to be multiplied by the 'n' factor. This means that the surfaces of the object under test has to present an equal absorption coefficient for the frequency 'F' (in the real model) and 'F*n' (in the scaled model); sometimes this is almost impossible and many corrections on the material have to be consider. Another problem that present the scaled models is the limitation of the source and microphones of the measurement, as the frequencies are multiplied n times, sometimes they are above the human hearing range and special ultrasonic transducers and sources are needed, for example, the ones used by T Busch et al [6] and Farley et al [5] that were explained in the previous section. Furthermore, another problem is the nonlinear absorption of the air, and many different gases, such nitrogen or dry air [2], are sometimes needed. Despite of this, many authors despise this correction because they considered it non-significant. Many others like J. H. Rindel [14] conclude that scaled models, mainly because of the air absorption, are not very accurate for very high frequencies (higher than 40 KHz). Another problem of the sources used in scaled models is the directivity pattern, and as they have to present a same directivity for 'F' and for 'F*n' sometimes this could be a big problem to reach.

4. APPLIED STANDARDS

In order to design the construction and measurement of the acoustic barrier, some standards had to be meeted. In one hand, the scaling method was based on the ISO-series standard for scattering and diffusion: ISO 17497-2:2002 [15]. On the other hand, based on the election of office barrier, the measurement was based on the specific regulation: ISO 10053:1991 [16].

Taking into account that there was not a complete standard compiling scaled acoustics and office separators, some fundamental structure was taken from each of the mentioned standards. ISO 17497 provided the guide to measure in scale, while ISO 10053 set the conditions of the measurement and the calculations performed.

5. BARRIER DESIGN

The measurement equipment was a main limitation for the design stage; 100 to 5000 Hz analysis needed to be performed and the available source for the measurement was a regular 2-way small loudspeaker Edifier R1900TII (*serial 191125656984*), which has a limited frequency response in audio-wide-band (about 20 Hz to 20 KHz). First of all a transfer measurement of this system needed to be done in order to explicitly view the limits of the source. This was performed with Smaart software in transfer mode with an Earthworks M50 as measurement microphone placed at 1m in front of the loudspeaker. The result is shown in Figure 4.

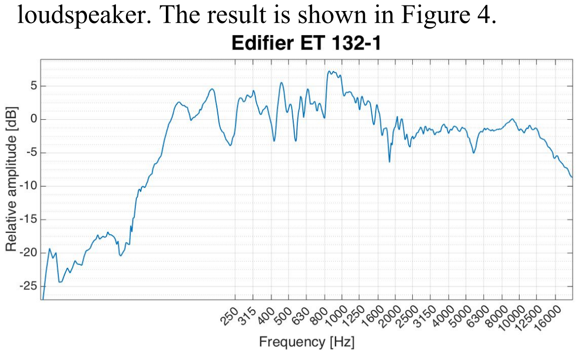


Figure 4: Frequency transfer of Edifier ET 132-1.

Taking in mind the upper limit (5 KHz) and the scale transformation (N*5 KHz), analysing the Edifier system, the scale limit is 1:3 because higher frequencies than 15 KHz can not be optimally reproduced.

The measurement was made with 4 Earthworks M50 (*serials 4212H, 4213H, 4219H, and 4222H*) through an RME Fireface UX+ (serial *23771444*) recording in a Macbook pro early 2010 (*serial WQ024SWUATM*).

Based on the investigation of similar offered products, the original dimensions of the barrier were chosen as 1.08 m wide and 1.98 m tall. In scale, these measures transform to 36 cm wide and 66 cm tall.

The chosen material was MDF for its easy management while cutting and mounting to form the acoustic barrier. The density chosen is medium with 1 cm thick, resulting in a total superficial density of 5.5 kg/m^2. In order to meet the recommendations of the ISO standard of a minimum of 20 kg/m^2, two MDF panels had to be used together, resulting in total 20 mm thick (in scale, in original size, this would transform to 60 mm).

The design is based on double-wall principle, with an air gap of 60 mm between the individual walls. No window, opening or door was included in the design in order to meet the ISO requirement of a totally closed surface with no acoustic filtration through the barrier. Both double walls were supported by a bracket on each side to make the barrier 'stand' by itself.

6. MEASUREMENT

The applied standard, ISO 10053, has indications for measurement in outdoors facilities or indoor semi-anechoic chamber, which was prefered in this case for the availability. The chosen measurement laboratory was one available in University's building in the 5th floor of 'Caseros II' facilities. It is chosen because of its low RT value[1], making it close to a semi-anechoic chamber and because of its size: it meets the minimum ISO requirement of minimum effective length, once the dimensions are re-scaled. Figure 5 shows the room sizes and Table 2 shows the RT values.

[1] This RT values were provided by Luciano De Bortoli, Elouan Trichard and Alejandro Suarez's measurement on his study of this laboratory's isolation.

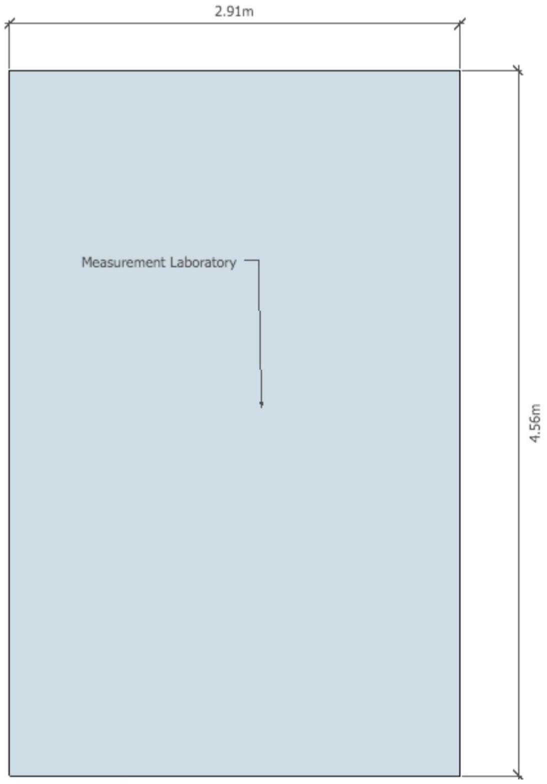


Figure 5: Laboratory’s plan with dimensions.

f [Hz]	63	125	250	500	1 K	2 K	4 K	8 K
RT	0.38	0.27	0.23	0.12	0.12	0.07	0.06	0.05

Table 2: Laboratory’s RT values.

The ISO 10053 details that floor should be reflective, so a blackboard was used to provide the new floor into the laboratory.

Microphone and source positions were scaled as the complete model in 1:3 and an average of 30 seconds was used (instead of 16 s as asked by the standard). The data was recorded in a sample rate of 192 KHz in order to later resample the audio files to 64 KHz (192 divided by 3) to transform the measurement to the original (non-scaled) size.

Air absorption was despised because previous studies showed that it did not have further influence in IL results.

The test signal was pink noise, as asked by the ISO standard. The noise bandwidth is from 20 to 20 KHz but the analysed windows was between 100 and 5000 Hz in original size, meaning 300 to 15000 Hz in the scaled measurement.

Figures 3 and 4 show the disposition of the measurement equipment.

Figure 3: Measurement from Ex view.

Figure 4: Measurement from Rx view.

6.1. POST PROCESSING

After the measurement, the recorded audio files, that were recorded with a sample rate of 192 KHz, had to be resampled to 64 KHz. Most of the commercial softwares offer this option, but they have an internal algorithm to adapt the frequency change and do not convert the frequency scale. However, this was needed to revert the scale to original size, so a Matlab function seemed to be the best option. A simple script (shown in annex) was developed in order to load the files and resample them without pitch correction, creating a new file with the sample rate changed.

The converted audio files are analysed with another Matlab script (also shown in annex) that calibrates the audio file, filters it with third octave filters (meeting ANSI S1.42-2001, Class 1) and then converts the Pascal values to dB units, integrating during its whole length.

The obtained values were loaded into an Excel spreadsheet to calculate the IL as described in ISO 10053:

$$IL = SPL'po - SPL'p - 20Log\left(\frac{R}{r}\right) \quad (9)$$

Where SPL'po is the sound pressure level at the reference position, SLP'p the sound pressure level at the receiver position, R the distance between the source and the standard position, and r the distance between the source and the reference position.

The reference microphone was measured in two different configurations: as asked by the ISO 10053, without the presence of the barrier and on the other way, simultaneously with the other microphones. This was in order to compare the results with and without the barrier, if this indication should be met.

Three receiver microphones were placed in order to measure in three different points, changing the position relatively to the barrier. All the microphones were at the same distance to the barrier (in an imaginary axis orthogonal to the floor): one next to the floor, other as asked by ISO 10053 (same height than the source) and other in a higher position, directly 'viewing' the source. This last position was chosen to analyse the transition zone of the IL (transition zone) provided by the barrier's top acoustic diffraction.

6.2. BARRIER MODIFICATIONS

As required, 3 modifications of the original acoustic barrier were performed. In the first case, the air gap between the two walls was removed and the system was re-measured, as shown in Figure 6.

Figure 6: Measurement during first modification.

Later, a little top ceiling was added as shown in Figure 7.

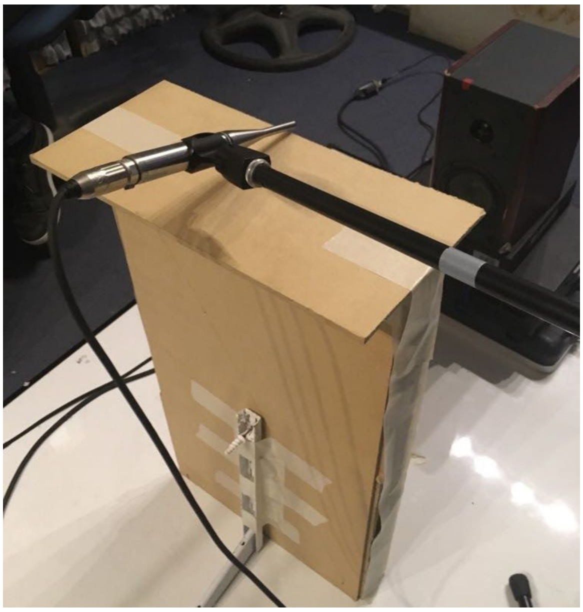


Figure 7: Measurement with second modification.

Then, the third modification of the original design was a change of building material. Now two pieces of pine wood were used replacing the MDF panels. This is shown in Figure 8.

Figure 8: Measurement with third modification.

7. RESULTS AND DISCUSSION

Figure 9 shows a comparison of the level in each microphone position, including background noise.

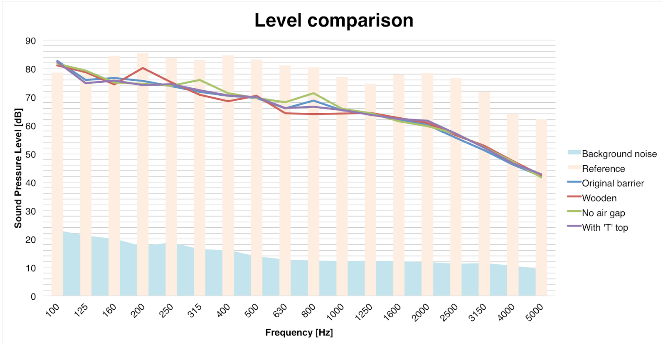


Figure 9: SPL in each microphone position and background noise.

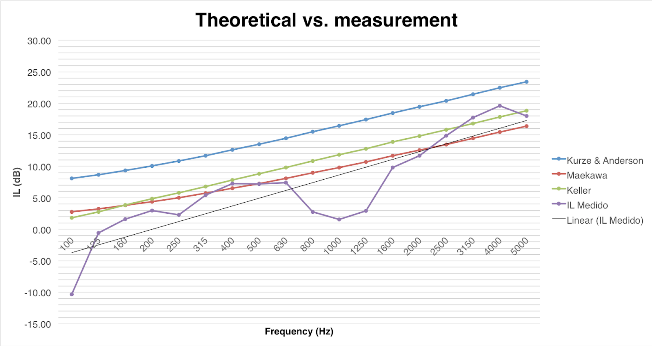
Table 3 shows these results expressed as global values.

Position	Global Level
Background noise	29.17 dB
Reference	91.88 dB
Rx with Original barrier	85.93 dB
Rx with Wooden	86.19 dB
Rx with No air gap	86.14 dB
Rx With 'T' top	85.46 dB

Table 3: Global values of Sound Pressure Level.

This data shows some important information: the average difference between background noise and receiver microphones is about 60 dB, bringing coherence to the results; the global differences do not seem to be very important, none of the modifications perform better than the original barrier (based on the less than 1 dB difference); despite having difference in ⅓ octave bands of around 15 dB, the global levels are closer, around 6 dB lower than the reference spot; for third octave bands higher than 1 KHz, none of the modifications present a better performance: all the values are almost the same; as expected, the IL values tending to higher frequencies are much more important than the low frequency ones (this might by effect of diffraction, which depends on wavelength and then frequency).

In Figure 10 a comparison between the theoretical methods and the measurement of the original barrier (with its linear regression) is shown.



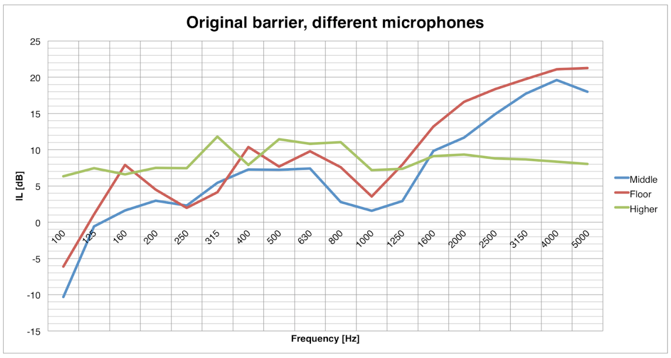


Figure 10: Comparison of measured IL (with its linear regression) and the three calculation methods (Keller, Maekawa and Kurze & Anderson).

The regression line turns evident that the tendency of the measurement is the same as expected by the calculation methods, increasing the IL value in high frequency. The IL based on the measurement information has a negative value in 315 Hz band and around 0 dB in 400 Hz band. It is important to take in mind that in the measurement, this corresponds to 945 and 1200 Hz. This means that the level in the *standard* position is higher than in the reference position, probably provided by some reflection or because of sound waves surrounding the acoustic element easily. This comparison also shows that Maekawa's calculations are closer in this case and Kurze & Anderson's method provide a very optimistic calculation for the IL. Despite this, the slope of the regression line is similar to Keller's calculation. The tendency of the IL in the measurement is broken between 2500 and 4000 Hz that might be due to a reflection in the room, the SPL are very close in reference and standard positions at this frequencies.

The Figure 11 shows the comparison of the insertion loss at three measurement points.

Figure 11: IL in three microphone positions.

Table 4 shows these results expressed as global values.

Position	Global Level
Middle (10053)	24.76 dB
Floor	27.38 dB
Higher	21.50 dB

Table 4: Global values of IL in three positions.

This information is the base to determine the different zones of attenuation provided the barrier. This can be seen for frequencies higher than 1250 Hz, where the three lines take individual routes. The low frequency is still the weakest point of the attenuation, presenting some increased level below 125 Hz (375 Hz for the measurement), probably due to diffraction or acoustic surrounding of the barrier. In global levels the difference is almost by steps of 3 dBs between floor and middle point (*standard* position as described by ISO 10053) and also between the middle and highest measurement point.

The IL of the original barrier and three modifications are shown in Figure 12 and then global values can be read from Table 5.

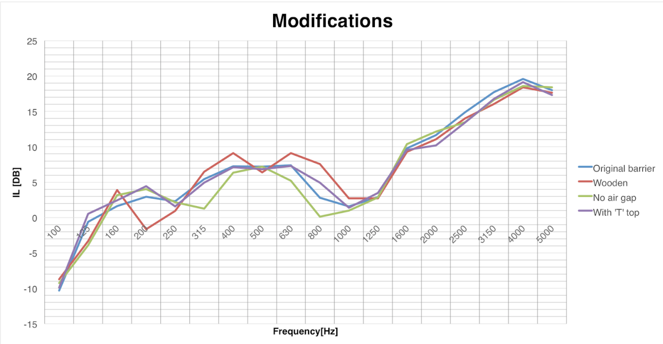


Figure 12: IL Comparison of the original barrier and its three modifications.

Modification	Global Level
Original barrier	24.76 dB
Wooden	24.05 dB
No air gap	24.19 dB
With ‘T’ top	24.10 dB

Table 5: Global values of IL in three modifications and the original barrier.

The first analysis of the IL, comparing the modifications is that the difference shown is really worst than expected. The global values are almost the same (varying in a range of 0.71 dB) and the three curves do not show further difference. Between 315 and 1250 Hz, the wooden barrier seems to have the best IL, but in the extremes, the values are almost equal. This might mean that none of the modifications affect importantly the insertion loss provided by the acoustic obstacle.

An additional analysis is made to check if the requirement of the ISO 10053 to take out the barrier to measure the response in reference position is a real necessity. Some of the results comparing the measurement level of the microphone reference with and without the barrier are shown in Table 6.

Modification	Maximum difference (⅓ octave bands)	Global difference
Original Barrier	5.17 dB	15.27 dB
Wooden	4.24 dB	14.96 dB
No air gap	4.06 dB	15.05 dB
With ‘T’ top	4.59 dB	15.24 dB

Table 6: Difference between measuring reference with and without barrier.

This clearly demonstrates the importance of meeting the requirements of the ISO standard, the error caused by its ignoration is important. This effect might be because of the reflection provided by the barrier. The only absorption is the one provided by MDF, which also is not scalable, so it is not a representation of the real-size behaviour. These reflections increase the apparent level in reference measurement position by considering direct sound summed to the reflected energy.

Another further study of this measurement was the comparison between calculating the IL by the ISO 10053 formula (previously transcripted) and by the usual difference at receiver spot with and without the barrier’s presence. Figure 13 shows the comparison between both IL and Table 7 shows both global values.

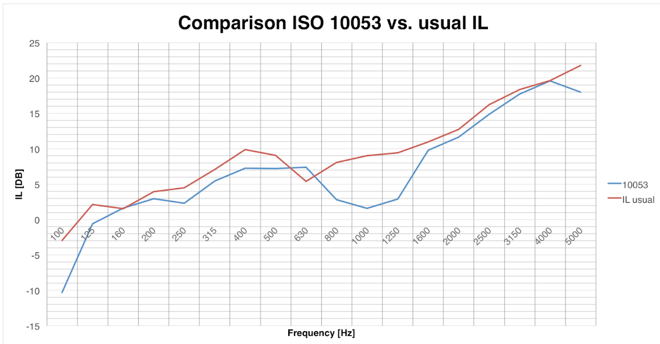


Figure 13: IL calculation comparison.

Calculation	Global Level
IL 10053	24.76 dB
Usual IL	26.48 dB

Table 7: Global values of IL in both calculation modes.

The IL calculated by the ISO 10053 standard is less optimistic than the usual method, increasing the difference around 1 KHz (5 dB of difference). The global levels show some difference of 2 dB but both tendencies are similar. The difference could be attributed to the factor in the standard that considers the barrier thickness and the distances.

8. CONCLUSIONS

The analysed model provides the same attenuation as an average barrier, about 24 dB of insertion loss.

The differences between the modifications and the original barrier do not seem to be very important. The easiest construction might be the model with no air gap, which can be easily achieved.

The acoustic scale provides an easy and quick measuring method. The lack of a standard that could join acoustic scale concept and office separators make that the analysed results might not be representative of the big-sized model.

The measurement should follow the instruction of ISO 10053 to remove the barrier while measuring the reference position. Not doing this might introduce an error up to 4 dB in a third octave band.

Some non-clear of failed results might be consequence of the source. Its size is really big compared to the scale model and acoustic surrounding of the barrier was easily achieved.

The prediction is very optimistic, the measurement is a demonstration that they might not be adequated to this barrier group.

The IL calculated by ISO 10053 contemplate the barrier thickness and its results are more pessimistic than the usual method.

As a future work, an extension for higher frequencies can be analyzed. However, as this barrier is designed for an office space, the main frequencies are the analyzed ones, being those the most important frequencies of the vocal human range. Another work that can be made in the future is the construction of the barrier in a the real size in order to compare the results of the scaled and real models; this might help to create a new standard combining acoustic scale with office barriers.

9. REFERENCES

[1] F. Spandock. *“Akustiche Modellversuche”*, Annalen der physik. 1934.

[2] C. Arango and V. molina. *“Construcción de una barrera acústica a escala para disminuir la contaminación sonora producida por el alto flujo vehicular”*. University of San Buenaventura. 2010.

[3] R Bullon. *“El modelo físico a escala como herramienta de diseño de la acústica de un recinto”*. Pontificia Universidad católica de Peru. 1998.

[4] S Voropayev et al. *“Finding optimal geometries for noise barriers tops using scaled experiments”*. University of Notre Dame. JASA 141, 722. 2017.

[5] J Farley and B Anderson. “*Ultrasonic airborne insertion loss measurements at normal incidence”*. Bringham young university. ASA. 2010

[6] T. Busch and M. Hodgson. *“Scale-model study of the effectiveness of highway noise barriers”*. University of British Columbia. ASA. 2003

[7] J. B. Keller. *“Geometrical Theory of Diffraction.”* Journal of the Optical Society of America. Volume 52, Number 2. 1962.

[8] U. J. Kurze and G. S. Anderson. *“Sound attenuation by barriers.”* Applied Acoustics (4). 1971.

[9] Z. Maekawa. *“Noise reduction by screens.”* Applied Acoustics, 1, pp. 157-173. 1968.

[10] E. J. Rathe. *“Note on two common problems of sound attenuation”*. J. Sound Fib, pp. 472-479. 1969.

[11] C. M. Harris. *“Handbook of Noise Control”* McGraw-Hill, New York, pp. 3-4. 1957.

[12] S. W. Redfearn. *"Phil. Mag."*. Vol. 7, pp. 223. 1940.

[13] R. B. Tatge. *“Barrier-wall attenuation with a finite-sized source”*. J. Acoust. Soc. Am. 53, pp. 1317–1319. 1973.

[14] J. H. Rindel. *“Room acoustic modelling techniques: A comparison of a scale model and a computer model for a new opera theatre’’*. ISRA. 2010.

[15] ISO 17497-2. *“Measurement of sound scattering properties of surfaces — Part 2: Measurement of the directional diffusion coefficient in a free field”*. International Organization for Standardization. 2002.

[16] ISO 10053. *“Acoustics - Measurement of office screen sound attenuation under specific laboratory conditions”*. International Organization for Standardization. 1991.

10. ANNEX
10.1 RESAMPLING SCRIPT

```
clear
clc

[filenames, pathnamecal] = uigetfile(
'*.wav', 'Select audio files to resample',
'MultiSelect', 'on');   %User selects the
files

M = cell(1,length(filenames));  %Basic cell
array for all files
nm = length(filenames); %Total of files
loaded
calroute = cell(1,nm);    %Reset route cell
array

for u=1:nm  %Writes each element on route
cell array
        calroute(u) = strcat(pathnamecal,
filenames(u));  %Path and name file in a
vector
end

for K = 1:nm  %Efectively loads each  file
    pause(0.01)
               cal  =audioread(calroute{K});
%Loads each data and then each  file
        M{K} = cal; %Writes the audiofile in
the cell array

audiowrite(strcat(mat2str(filenames{K}),'.wa
v'),cal,  64000,  'BitsPerSample',  24);
%Writes the resampled audio
end
```

10.2 LEVEL OBTAINING SCRIPT

```
clear
clc

[filenameaudio, pathnameaudio] = uigetfile(
'*.wav', 'Select audio');   %UI to locate
the audio file
rutaaudio    =    strcat(pathnameaudio,
filenameaudio);  %Makes the string to load
the calibration file
[audio, fs] = audioread(rutaaudio); %Load
audio

[filenamecal, pathnamecal] = uigetfile(
'*.wav', 'Select calibration'); %UI to
locate the audio file
rutacal = strcat(pathnamecal, filenamecal);
%Makes the string to load the calibration
file
[cal, fs] = audioread(rutacal); %Load audio

pascal = rms(cal);  %Gets the pascal value

audiocal = audio./pascal;   %Calibrates the
audio

audiodb = filtanddb(audiocal);   %Uses a
custom function to filter the audio signal
and get the dB values

csvwrite    (strcat(num2str(filenameaudio),
'.csv'),audiodb);  %Writes a .csv file with
the information
```

10.3 LEVEL OBTAINING SCRIPT

```
function    [    filteredindb    ]    =
filtanddb(signal)
% This function allows you to filter an
audio signal as described in ANSI
% standard, Class 0. It also transforms the
information to dB units. This
% needs a line-vector, mono audio.
%
% I.E.   Y = filtanddb (X)
%     Filters X signal (1x#) and makes Y
signal (1x30).
%% FILTERING
d = fdesign.octave(3,'Class 0', 'N,F0', 6,
1000, 48000);    %Creates filter object in
order to develop third octave filter by ANSI
standard class 0
F0=validfrequencies(d);    %Third    octave
central frequencies
for freq = 1:length(F0) %Creating each
filter.
    d.F0 = F0(freq);
         F(freq)  =  design  (d,  'butter');
%Determines Butterwoth
                      filtered(:,freq)     =
filter(F(freq),signal);     %Applies  each
filter and writes into a matrix
end
%% dB
audiotorms  =  filtered.^2;       %Squared
pressure
audiosum = zeros(1,30);   %Reset to for loop

for k=1:30  %Sums every row in each column
    audiosum (k) = sum (audiotorms(:,k));
end

audio2    =    audiosum/length(audiotorms);
%Divides by 'T'
audiodb = audio2/((20e-6)^2); %To dB
filteredindb = 10*log10(audiodb);
end
```