



Annex 51

Acoustic Signatures of Heat Pumps

Final Report – Part 2

1.0 Introduction

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Preface

This project was carried out within the Technology Collaboration Programme on Heat Pumping Technologies (HPT TCP), which is a Technology Collaboration Programme within the International Energy Agency, IEA.

The IEA

The IEA was established in 1974 within the framework of the Organization for Economic Cooperation and Development (OECD) to implement an International Energy Programme. A basic aim of the IEA is to foster cooperation among the IEA participating countries to increase energy security through energy conservation, development of alternative energy sources, new energy technology and research and development (R&D). This is achieved, in part, through a programme of energy technology and R&D collaboration, currently within the framework of nearly 40 Technology Collaboration Programmes.

The Technology Collaboration Programme on Heat Pumping Technologies (HPT TCP)

The Technology Collaboration Programme on Heat Pumping Technologies (HPT TCP) forms the legal basis for the implementing agreement for a programme of research, development, demonstration, and promotion of heat pumping technologies. Signatories of the TCP are either governments or organizations designated by their respective governments to conduct programmes in the field of energy conservation.

Under the TCP, collaborative tasks, or "Annexes", in the field of heat pumps are undertaken. These tasks are conducted on a cost-sharing and/or task-sharing basis by the participating countries. An Annex is in general coordinated by one country which acts as the Operating Agent (manager). Annexes have specific topics and work plans and operate for a specified period, usually several years. The objectives vary from information exchange to the development and implementation of technology. This report presents the results of one Annex.

The Programme is governed by an Executive Committee, which monitors existing projects and identifies new areas where collaborative effort may be beneficial.

Disclaimer

The HPT TCP is part of a network of autonomous collaborative partnerships focused on a wide range of energy technologies known as Technology Collaboration Programmes or TCPs. The TCPs are organized under the auspices of the International Energy Agency (IEA), but the TCPs are functionally and legally autonomous. Views, findings and publications of the HPT TCP do not necessarily represent the views or policies of the IEA Secretariat or its individual member countries.

The Heat Pump Centre

A central role within the HPT TCP is played by the Heat Pump Centre (HPC).

Consistent with the overall objective of the HPT TCP, the HPC seeks to accelerate the implementation of heat pump technologies and thereby optimize the use of energy resources for the benefit of the environment. This is achieved by offering a worldwide information service to support all those who can play a part in the implementation of heat pumping technology including researchers, engineers, manufacturers, installers, equipment users, and energy policy makers in utilities, government offices and other organizations. Activities of the HPC include the production of a Magazine with an additional newsletter 3 times per year, the HPT TCP webpage, the organization of workshops, an inquiry service and a promotion programme. The HPC also publishes selected results from other Annexes, and this publication is one result of this activity.

For further information about the Technology Collaboration Programme on Heat Pumping Technologies (HPT TCP) and for inquiries on heat pump issues in general contact the Heat Pump Centre at the following address:

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Acoustic Signatures of Heat Pumps

IEA HPT

Annex 51

1.0: Introduction



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1. Introduction

As is known, heat pumps are proposed as an interesting alternative to traditional heating methods and have few rivals for cooling environments. Although heat pumps contain more and more advanced technologies, and their energy efficiency has grown over the years, a traditional obstacle for their diffusion is constituted by the noise produced which in many cases is not yet comparable to that of the older and still widespread technologies. The purpose of this chapter is to analyze the problem, to list which are the noise sources and their main problems, to describe the sound phenomenon in general.

2. What is acoustics?

1.1 What is acoustics – Sound definition

Before entering into the details of the problem, it is important to introduce the acoustic phenomenon:

Acoustics is a scientific discipline that deals with the characteristics of sound, that is, studying how the sound effects are produced and propagated, and in the field of psychoacoustics, how the sound is perceived.

The sound is a physical phenomenon that stimulates the sense of hearing: it is caused by the rapid movement (vibration) of any body (a rope, a rubber band, a piece of wood, a column of air, etc.). The sounds are simply pressure fluctuations created by vibrations obtained in different ways that propagate as waves in a medium.

The sound is characterized by some fundamental quantities: amplitude, frequency (or period), wavelength and propagation speed.

In order to propagate, the sound requires a source and an elastic medium that can be air, or any other element. The vibrations produced by the source cause a succession of compressions and rarefactions in the elastic propagation medium.

The propagation speed depends on the propagation medium. Some examples of propagation speed:

- around 340 meters / sec. in air;
- about 1500 meters / sec. in water;
- about 5000 meters / sec. in iron.

The pitch of a sound, a perception quantity, depends on the **frequency**, that is the speed of the vibrations: given a constant time (one second), the more numerous they are, the more acute the sound is. The pitch of the sound is measured in “**Hertz**” (**Hz**). A Hertz corresponds to a complete oscillation of an elastic body in the time of one second. In nature there are sounds ranging from one Hertz to one million Hertz. The human ear can hear sounds between 16 Hz and 20 kHz.



1.2 Main Acoustics quantities

The most important physical quantities that characterize the sound phenomenon are:

- sound pressure;
- particle speed;
- density of sound energy;
- sound intensity;
- sound power.
- spectral distribution

When the sound wave passes through the elastic medium, a sequence of compressions and expansions occurs, which implies a variation of the ambient pressure with respect to the equilibrium value. These compressions and expansions give rise to the acoustic pressure p which depends on the frequency and amplitude of the harmonic motion of the source, the elastic characteristics and the mass of the acoustic means. The link between the particle velocity of the elastic medium v and the acoustic pressure p is:

$$\frac{p}{v} = \rho_0 \cdot c_0 \quad [\text{kg/m}^2\text{s}] \quad \text{Eq. 2-1}$$

Where ρ_0 is the **density** of the elastic medium and the product $\rho_0 c_0$ it is called **acoustic impedance (Z)** of the plane wave ($\text{kg/m}^2\text{s}$). This quantity can be referred to two different systems thus assuming two different values. The two reference systems can be the MKS system and the CGS system. In this case the SI system will be used, based on MKS system.

When the waveform is complex, the definition of the average amplitude of the signal to be analyzed becomes ambiguous, and the use of the maximum instantaneous value is not representative of human perception. The so-called **Average Effective Value** or **Root Mean Square (RMS)** value of the signal is then used:

$$P_{eff} = P_{RMS} = \sqrt{\frac{1}{T} \int_0^T [p(t)]^2 dt} \quad \text{Eq. 2-2}$$

Since the sound is made up of waves, we can think that there is a share of **sound energy E** transferred from the source through these waves. In acoustics, however, it is more convenient to consider the density of sound energy per unit of volume.

In the case of plane waves in a non-viscous elastic medium, the energy per unit of volume or **density of sound energy D** transferred to the medium can be expressed with the relation:

$$D = \rho_0 \cdot v_{eff}^2 \quad [\text{J/m}^3] \quad \text{Eq. 2-3}$$



where $\mathbf{v} = \mathbf{v}(t)$ is the velocity of the surface that generate a plane wave (for example a piston moving in a cylinder) and, for plane waves in a non-viscous medium, also of the particles of the medium.

For plane waves we have: $\mathbf{v} = \mathbf{p}/(\rho_0 c_0)$. Thus, another important expression is:

$$D = \frac{p_{eff}^2}{\rho_0 \cdot c_0^2} \quad [\text{J/m}^3] \quad \text{Eq. 2-4}$$

which correlates a directly measurable quantity, such as the **RMS (effective)** value of the sound pressure, p_{eff} , with the energy transmitted by the source in the medium.

In the case of non-plane waves, or in the presence of stationary waves (which bounce back and forth) the energy is never completely kinetic or potential, and it is necessary to evaluate separately, at each point and in each instant, the two contributions and add them:

$$D = \frac{1}{2} \cdot \left[\rho_0 \cdot v_{eff}^2 + \frac{p_{eff}^2}{\rho_0 \cdot c_0^2} \right] \quad [\text{J/m}^3] \quad \text{Eq. 2-5}$$

Sound Intensity I is the parameter for evaluating the flow of energy that passes through a given surface.

It is defined as the energy that in unit time passes in a normal direction a unitary surface (W/m^2).

Intensity \vec{I} is a vector parameter defined by a module and a direction:

$$\vec{I}(P, t) = p(P, t) \cdot \vec{v}(P, t) \quad \text{Eq. 2-6}$$

where \mathbf{P} indicates the spatial **coordinates of the point**, and \mathbf{t} indicates the **time**.

In the case of plane waves, in a medium in non-viscous stillness, the relationship between density and intensity of sound energy is given as:

$$I = D \cdot c_0 \quad [\text{W/m}^2] \quad \text{Eq. 2-7}$$

The **sound power W** describes the sound emission capacity of a source and is measured in Watts (W). The power cannot be measured directly, but requires special methods for its determination (for a description of the measurement techniques refer to section 1.2).

The sound power is a univocal descriptor of a sound source and it is an objective quantity independent of the environment in which the source is placed.

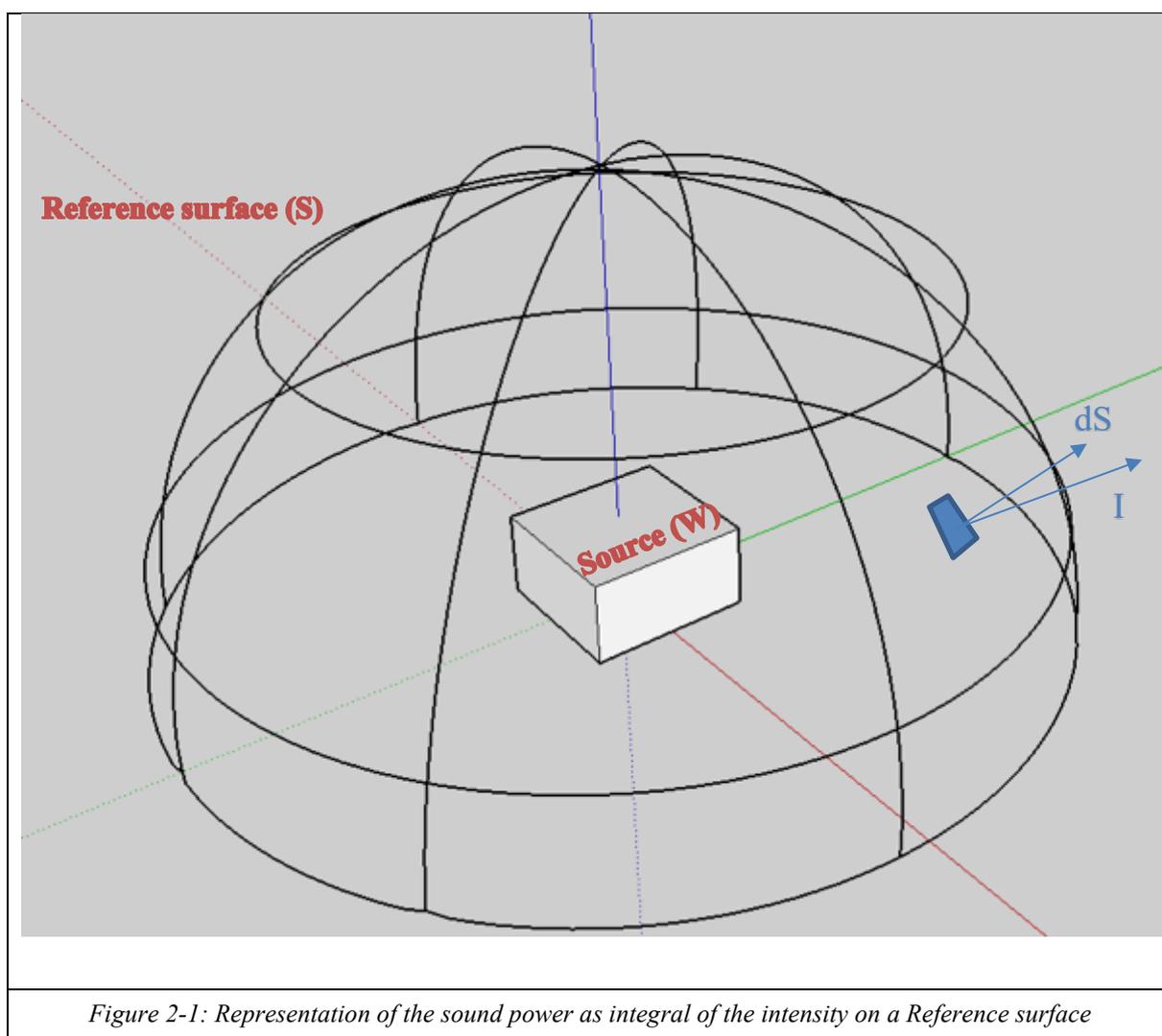
Considering a closed surface S enclosing a sound source, the acoustic power W emitted from the source is given by the integral of the sound intensity I on the surface:



$$W = \int_S \vec{I}(P, t) \cdot n dS \quad \text{Eq. 2-8}$$

In the case in which the closed surface S is decomposable in N elementary S_i surfaces, the expression of the sound power becomes:

$$W = \sum_{i=1}^N I_i \cdot S_i \quad \text{Eq. 2-9}$$





1.3 What are decibels and why they are used

The powers and intensity of sound associated with the phenomena that the human ear can perceive have a wide dynamic.

It ranges from 1 pW/m² (threshold of the audible) to 1 W/m² (pain threshold) for the sound power.

It ranges from 20 μPa (threshold of the audibility) to 20 Pa (pain threshold) for the sound pressure.

For this reason we use a logarithmic scale, in which the logarithm of the ratio between that same value and a predetermined “reference” value is used instead of the physical quantity itself.

The advantage that derives from the use of the decibel scale consists in the evident reduction of the range of variability (with consequent reduction of the dynamics of the quantity to be treated); thus we define the following quantities:

sound pressure level L_p (also referred to as **SPL**) :

$$L_p = 10 \log (p^2/p_{\text{ref}}^2) = 20 \log (p/p_{\text{ref}}) \quad (\text{dB}) \quad \text{Eq. 2-10}$$

where $p_{\text{ref}} = 20 \mu\text{Pa}$ (often indicate as p_0);

sound velocity level L_v :

$$L_v = 10 \log (v^2/v_{\text{ref}}^2) = 20 \log (v/v_{\text{ref}}) \quad (\text{dB}) \quad \text{Eq. 2-11}$$

where $v_{\text{ref}} = 50 \text{ nm/s}$ (often indicate as v_0);

sound intensity level L_I :

$$L_I = 10 \log (I/I_{\text{ref}}) \quad (\text{dB}) \quad \text{Eq. 2-12}$$

where $I_{\text{ref}} = 10^{-12} \text{ W/m}^2$ (often indicate as I_0);

sound density level L_D :

$$L_D = 10 \log (D/D_{\text{ref}}) \quad (\text{dB}) \quad \text{Eq. 2-13}$$

where $D_{\text{ref}} = 3 \cdot 10^{-15} \text{ J/m}^3$ (often indicate as D_0).

In the case of plane waves, in a medium in non-viscous stillness ($\rho_0 c_0 = 400 \text{ kg/m}^2\text{s}$) from the two equations:

$$p/u = \rho_0 \cdot c_0 \quad \text{Eq. 2-14}$$

$$I = p^2 / \rho_0 c_0 = D \cdot c_0 \quad \text{Eq. 2-15}$$



follows that:

$$L_p = L_v = L_I = L_D \quad \text{Eq. 2-16}$$

Finally we can define **sound power level** L_W as:

$$L_W = 10 \log (W/W_{\text{ref}}) \quad (\text{dB}) \quad \text{Eq. 2-17}$$

where $W_{\text{ref}} = 10^{-12}$ W (often indicate as W_0).

But, while the 4 previous “field” levels are identified in a single numerical value, the power level generally takes on a very different value.

Also in the case of a plane and progressive wave (S area piston at the end of a tube/cylinder), the link between power level and intensity level is:

$$L_W = L_I + 10 \log S/S_0 = L_I + 10 \log S \quad (\text{dB}) \quad \text{Eq. 2-18}$$

(true if we consider $S_0 = 1 \text{ m}^2$).

A commonly used quantity is the “**continuous equivalent sound level**” or L_{eq} defined as:

$$L_{\text{eq},T} = 10 \cdot \log \left[\frac{1}{T} \int_0^T \frac{p^2(t)}{p_{\text{ref}}^2} dt \right] (\text{dB}) \quad \text{Eq. 2-19}$$

where T is the **integration time interval**, $p(t)$ is the **instantaneous pressure value** and p_{ref} is the **reference pressure**.

If there are two different sound sources that produce sound emissions that are incoherent with respect to each other (not related to each other), the “inconsistent” sum of the two levels (two different sounds) can be defined in a receptor point:

Starting with two different pressure levels L_{p1} ed L_{p2} :

$$L_{p1} = 10 \log (p_1/p_{\text{ref}})^2 \quad (p_1/p_{\text{ref}})^2 = 10^{L_{p1}/10} \quad \text{Eq. 2-20}$$

$$L_{p2} = 10 \log (p_2/p_{\text{ref}})^2 \quad (p_2/p_{\text{ref}})^2 = 10^{L_{p2}/10} \quad \text{Eq. 2-21}$$

the (energetic) sum of the two previous pressure levels becomes:

$$(p_T/p_{\text{ref}})^2 = (p_1/p_{\text{ref}})^2 + (p_2/p_{\text{ref}})^2 = 10^{L_{p1}/10} + 10^{L_{p2}/10} \quad \text{Eq. 2-22}$$

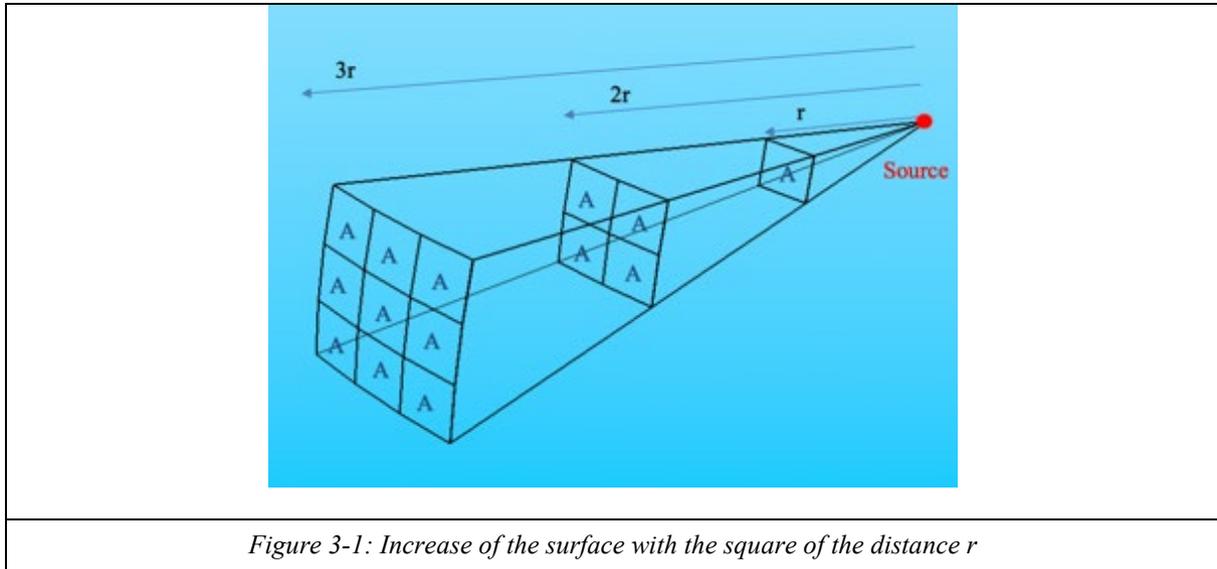
$$L_{pT} = L_{p1} + L_{p2} = 10 \log (p_T/p_{\text{ref}})^2 = 10 \log (10^{L_{p1}/10} + 10^{L_{p2}/10}) \quad \text{Eq. 2-23}$$

The latter is the formula usually used to evaluate the sound pressure level resulting in a given point by the presence of different sound sources.



3. Spherical propagation

Moving away from the source, the sound power that propagates remains in a first approximation constant (no absorption by the air) but the sound intensity decreases because it is distributed over an ever larger surface.



$$I = \frac{W}{S} = \frac{W}{4\pi r^2} \quad \text{Eq. 3-1}$$

The decrease in intensity by doubling the distance and in the case of spherical propagation is always 6dB.

Let's now recall the definitions of sound levels in dB:

$$L_W = 10 \log (W/W_{ref})$$

$$L_I = 10 \log (I/I_{ref})$$

$$L_D = 10 \log (D/D_{ref})$$

$$L_p = 10 \log (p^2/p_{ref}^2)$$

$$L_v = 10 \log (v^2/v_{ref}^2)$$

All levels are expressed in decibels. Only L_W is not homogeneous to the others (the power level depends on the source and consequently L_W remains constant at each point).

Therefore with fixed L_W , we can deduce the level of intensity:

$$L_I = 10 \log \left(\frac{I}{I_0} \right) = 10 \log \left(\frac{W}{4\pi r^2 I_0} \right) = 10 \log \left(\frac{W}{I_0} \frac{W_{ref}}{4\pi r^2 W_{ref}} \right) \quad \text{Eq. 3-2}$$



$$L_I = 10 \log \frac{W}{W_{ref}} + 10 \log \frac{W_{ref}}{I_{ref}} + 10 \log \frac{1}{4\pi} + 10 \log(r^{-2}) \quad \text{Eq. 3-3}$$

Since the reference values W_{ref} and I_{ref} are arbitrary values we can choose them the same in order to simplify the relationship and their actual values are:

$I_{ref} = 10^{-12} \text{ W/m}^2$ (see Eq. 2-12) and $W_{ref} = 10^{-12} \text{ W}$ (see Eq. 2-17).

It is now clearer why the reference values defined in chapter 2 have been chosen.

Now we can calculate the numerical value of the other two terms:

$$10 \log \frac{1}{4\pi} = -11 \quad \text{Eq. 3-4}$$

$$10 \log(r^{-2}) = -20 \log(r) \quad \text{Eq. 3-5}$$

And finally we can derive from Eq. 3-2, Eq. 3-3, Eq. 3-4 and Eq. 3-5 in:

$$L_I = L_W - 20 \log(r) - 11 \quad \text{Eq. 3-6}$$

Obviously, out of the case of a progressive plane wave, the relationships are no longer valid. It has already been verified that in a pipe with rigid termination the p and v values rise and fall alternately (where p is high the speed is canceled, and vice versa). It is not just the case of the tube, but simply in every real room, where the values of I , D , p , v , are slightly different at each point. A quantity is however limited: $L_D \leq L_I$, the first is the only energy that propagates, while the second is the total energy, and therefore the sum of both the energy that propagates, both of the “stationary” energy that oscillates back and forth without propagate.

We can therefore use this **difference** $L_D - L_I$ to estimate the **Reactivity Index** of a sound field.

It is important to recall that the quantities are homogeneous and represent the total energy as well as the part of it that is propagating.

4. Directivity of sound sources

The acoustic field generated by a sound source is, in general, characterized by a different sound energy emission according to the various directions. Therefore, the “**directivity factor**” Q is defined as the ratio between the **sound intensity** I_θ in the **direction** θ and the **sound intensity** I_0 which would have the acoustic field at that point, if the source was **omnidirectional**:

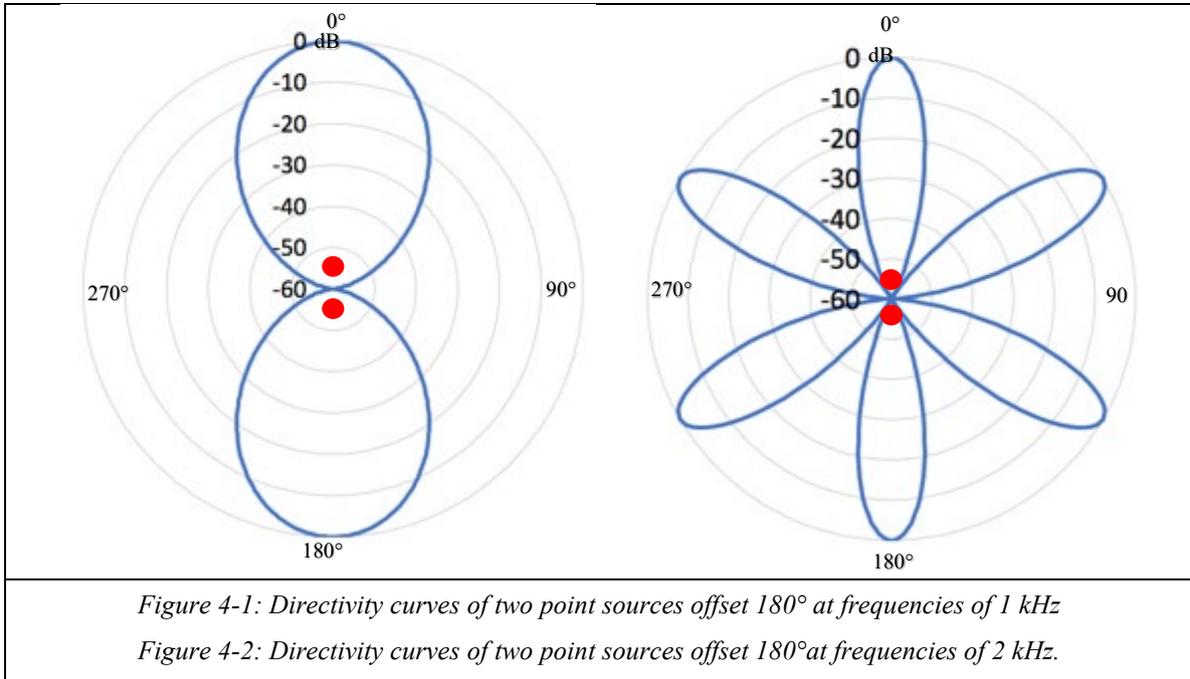
$$Q(\theta) = I_\theta / I_0 \quad \text{Eq. 4-1}$$

In addition to this value, the **directivity index** D is also defined for each direction θ , given as:

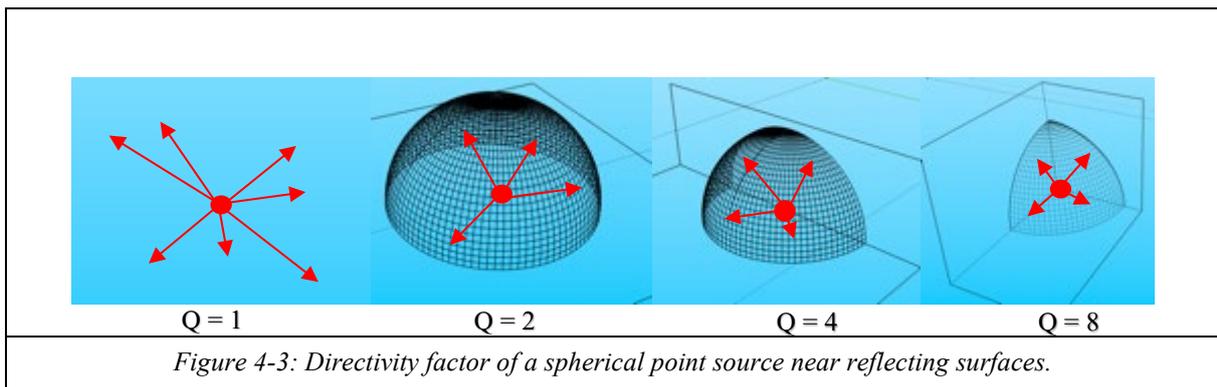
$$D(\theta) = 10 \log Q(\theta) \quad (\text{dB}) \quad \text{Eq. 4-2}$$



In general it is sufficient to know the value of Q (or D), in the vertical and horizontal plane. It should still be remembered that the value of Q depends on the frequency and that it normally increases with it.



As already noted above, the sound field generated by a source can be modified by the presence of obstacles and reflecting surfaces: if, for example, a spherical omnidirectional point source (in this case Q is not a function of θ and $Q = 1$), it is placed on a perfectly reflecting plane, obtains $Q = 2$, as shown in the following figure; if it is placed in a corner, between two reflecting surfaces, we obtain $Q = 4$ and $Q = 8$ for the corner, due to the reflections over hard planes.



Using previous Eq. 3-6 , a new formulation for free field sources that includes also the directivity factor Q can be established:

$$L_1 = L_p = L_w - 20 \log r - 11 + 10 \log Q \quad (\text{dB}) \quad \text{Eq. 4-3}$$



The aforesaid relationship is particularly important as it allows, by measuring the L_p sound pressure levels, to determine the directivity factor of a source and the value of the sound power level. The measurement must be carried out in an anechoic chamber according to the requirements of the **ISO 3745** standard (for a description of the measurement standards and techniques refer to section 1.2).

Other types of sources may exist in addition to the point source (e.g. cylindrical or planar), which we will not consider here: usually heat pumps can be considered point sources, as they are very often positioned at a sufficiently high distance from the receptors that can be disturbed by their sound emissions, although this does not exclude that the emission of a heat pump cloud have a pronounced directivity (in many cases is important to have a description of the sound directivity pattern of this kind of sources).

5. Outdoor propagation

The factors that influence the propagation of sound are linked to environmental phenomena but also to the presence of barriers or surfaces between the source and the receiver. We will now try to understand the changes introduced by these factors and how they can be considered in the design phase.

1.4 Temperature effect

The first factor that influences wave behavior is temperature variation. In fact, the temperature varies as the altitude changes and there are different configurations of variation. We will now examine three cases that may arise:

1.4.1 a) Normal trend

Under normal conditions the temperature decreases with the altitude. The sound beams (in the various figures represented with the field lines orthogonal to the wave front and representing points of equal-intensity sound) are curved upwards. There is a theoretical limit surface tangent to the ground, below which a shadow zone is formed due to the absence of sound waves.

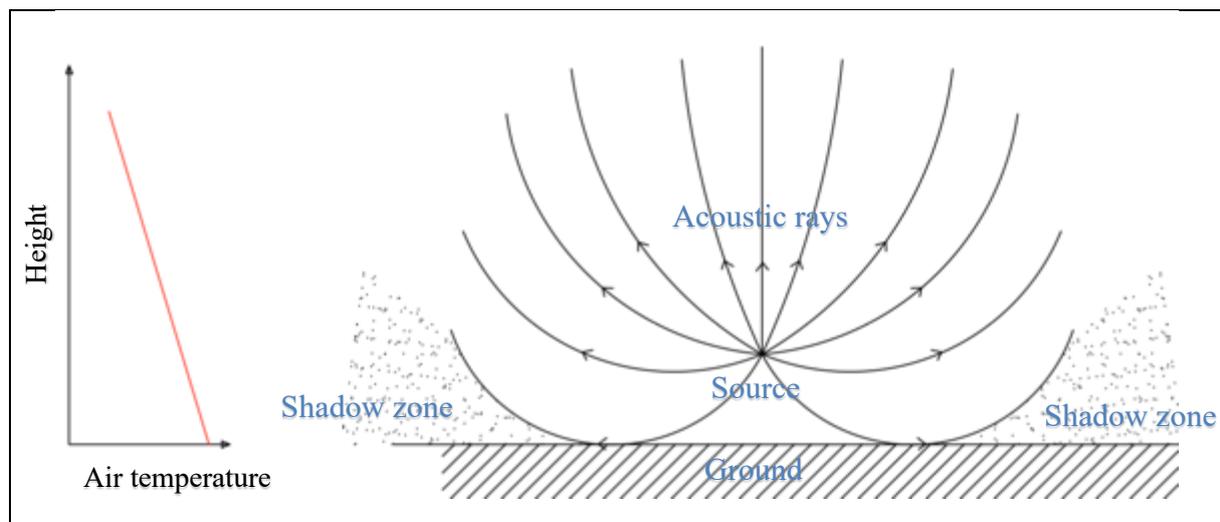
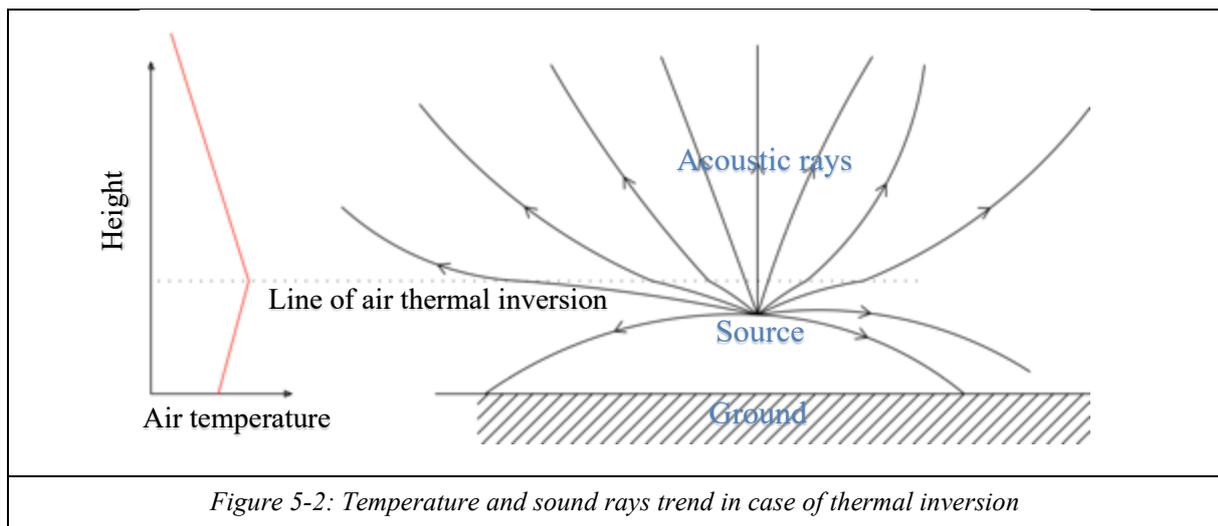


Figure 5-1: Normal behavior of temperature and sound beams



1.4.2 b) Thermal inversion

In this situation the ground is colder than the surrounding air and therefore at low altitudes the ground temperature is lower than the altitude temperature. As the distance from the ground increases, it returns to a normal course. This is one of the climatic conditions typical of areas such as the Po Valley. In these cases the sound beams are curved downwards and this leads to the absence of shaded areas; this can give rise to strange phenomena because the sound can “rain” on areas that would not be reachable if the wave fronts had the usual course.



This condition can be important in the case of installation of heat pumps on the roof of buildings as sound sources can then be audible even at long distances.

1.4.3 c) Sound channel

It is the most "strange" and rare phenomenon. A sound channel is formed when there is a layer of air that is warmer (or colder) than the surrounding layers. In this case the sound waves are “trapped” in the different temperature layer and can only exit when the temperature changes again; they can therefore travel several kilometers before falling back and this can give rise to so-called "sound mirages". A similar situation can occur in the presence of fog: in fact, the blanket of fog on the ground forms an area where the temperature is lower than that of the ground, while above the layer of fog the sun's rays make the temperature higher. This temperature variation creates a channel in which sound waves can be trapped.

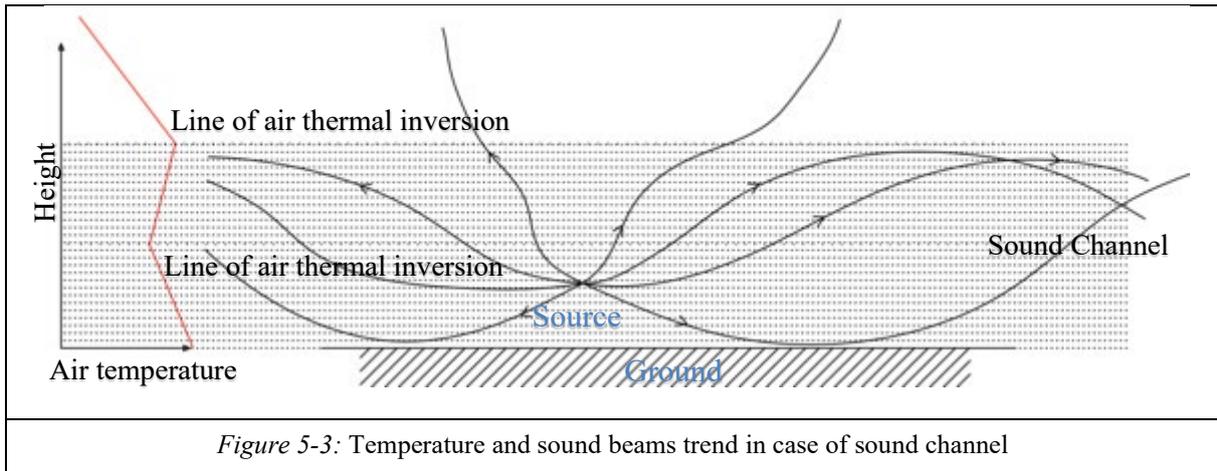


Figure 5-3: Temperature and sound beams trend in case of sound channel

When designing the systems, these thermal phenomena must be taken into account. For example, the Italian legislation states that when calculating the effect of noise propagations it is necessary to consider the average unfavorable case, that normally is the case of thermal inversion.

1.5 Wind effect

The wind can greatly influence the performance of the sound beams. In the presence of wind, the speed of sound and wind add up as a vector composition. In reality, the wind can only carry sound when the wind speed is comparable to that of sound (and this is quite rare).

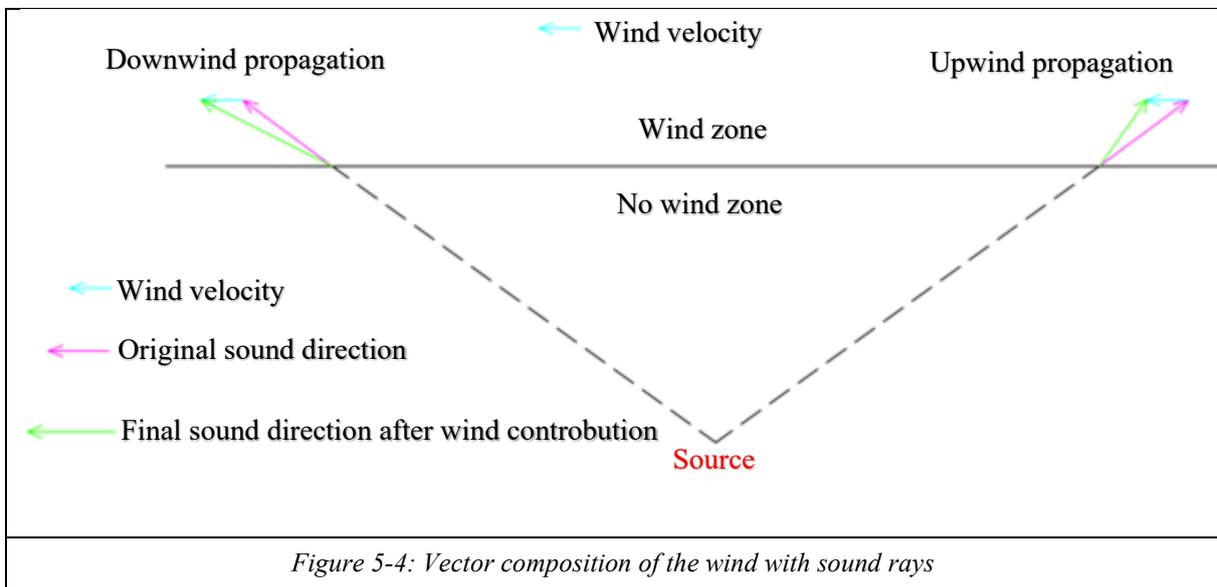
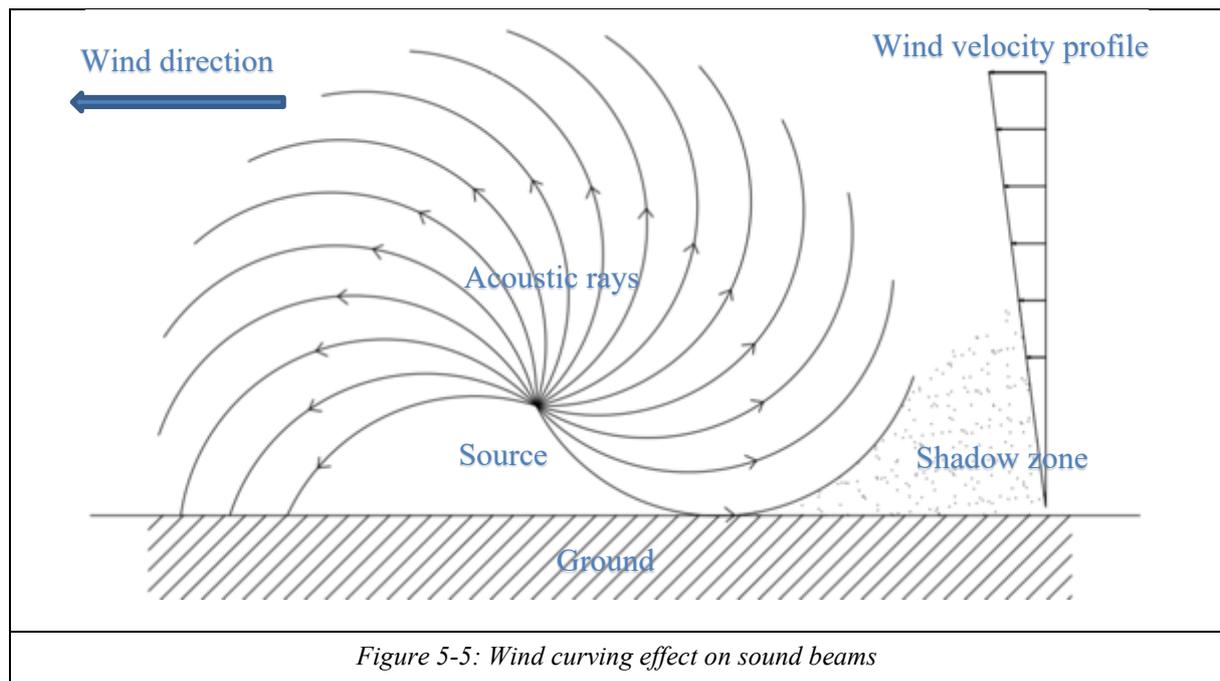


Figure 5-4: Vector composition of the wind with sound rays

Similar to the temperature gradients, wind velocity gradient (i.e. the wind speed changes with the altitude) the sound rays also curve downwind.



This curvature given by the wind leads to the formation of a zone of shadow over wind and an area in which the sound “rains” downwind.

In order to take into account these phenomena, the **ISO 9613-2** standard describes the appropriate calculation methods. However, this standard is extremely complex, so as to be practically inapplicable without the use of computers.

To simplify, the calculation must always consider the condition of leeward and so it must imagine that the sound beams always bent downwards. Generally a radius of curvature of 2000 or 3000 meters is used. In any case, the curvature of the sound beam is appreciated only when the propagation distance is comparable with the selected radius of curvature.

1.6 Air absorption

Air is not a perfectly elastic medium, and consequently there is a weak dissipation of acoustic energy into heat. The phenomenon grows with the frequency squared, and depends in a very complex way on the physical parameters temperature and humidity. The **ISO 9613-1** standard contains the complex formulations necessary for the analytical calculation of the air absorption, which however are not reported here.

At the end of the standard instead, extensive and detailed tables are reported, which provide the attenuation of the air expressed in dB/km, at various frequencies, and for all relative temperatures and humidity. Below we report only a very brief excerpt of the tabulated data, please refer to the text of the standard for complete data.

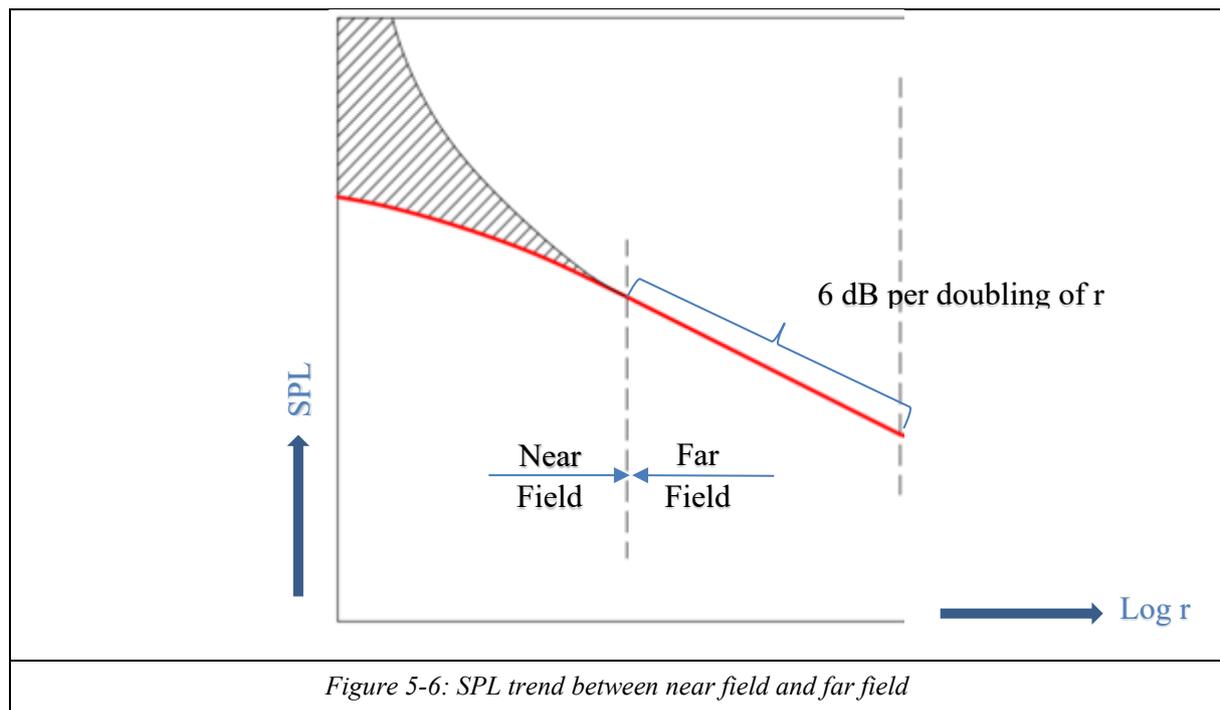


		Central frequencies of octave band							
T(°C)	U,R,(%)	63	125	250	500	1000	2000	4000	8000
10	70	0,12	0,41	1,04	1,93	3,66	9,66	32,8	117,0
15	20	0,27	0,65	1,22	2,70	8,17	28,2	88,8	202,0
15	50	0,14	0,48	1,22	2,24	4,16	10,8	36,2	129,0
15	80	0,09	0,34	1,07	2,40	4,15	8,31	23,7	82,8
20	70	0,09	0,34	1,13	2,80	4,98	9,02	22,9	76,6
30	70	0,07	0,26	0,96	3,14	7,41	12,7	23,1	59,3

Table 5-1: Sound absorption coefficients in dB / km (from ISO 9613-1) for certain combinations of temperature and relative air humidity

1.7 Outdoor and indoor propagation

For the propagation of sound two aspects must be taken into consideration. The first involves the mutual distance between source and receiver (distance usually compared to the maximum size of the sound source). The second is based on a near field and a far field. The equations so far introduced are valid for the hypotheses of waves that can be considered plane. The most common sources, especially those that are taken into account in this study, are point-like, so the propagation is always spherical. At a sufficiently high distance these waves can however be approximated as plane waves. In this hypothesis we are in the case of far field and the equations relative to the spherical divergence are valid. The Figure 5-6 shows the trend of the sound pressure level measured for a point source that emits a constant sound: in the far field there is a constant decrease of 6 dB for each doubling of the distance between source and receiver (value valid for one point source placed in free field). Typically this value is valid for distances between source and receiver that are greater than 10 times the maximum size of the source.

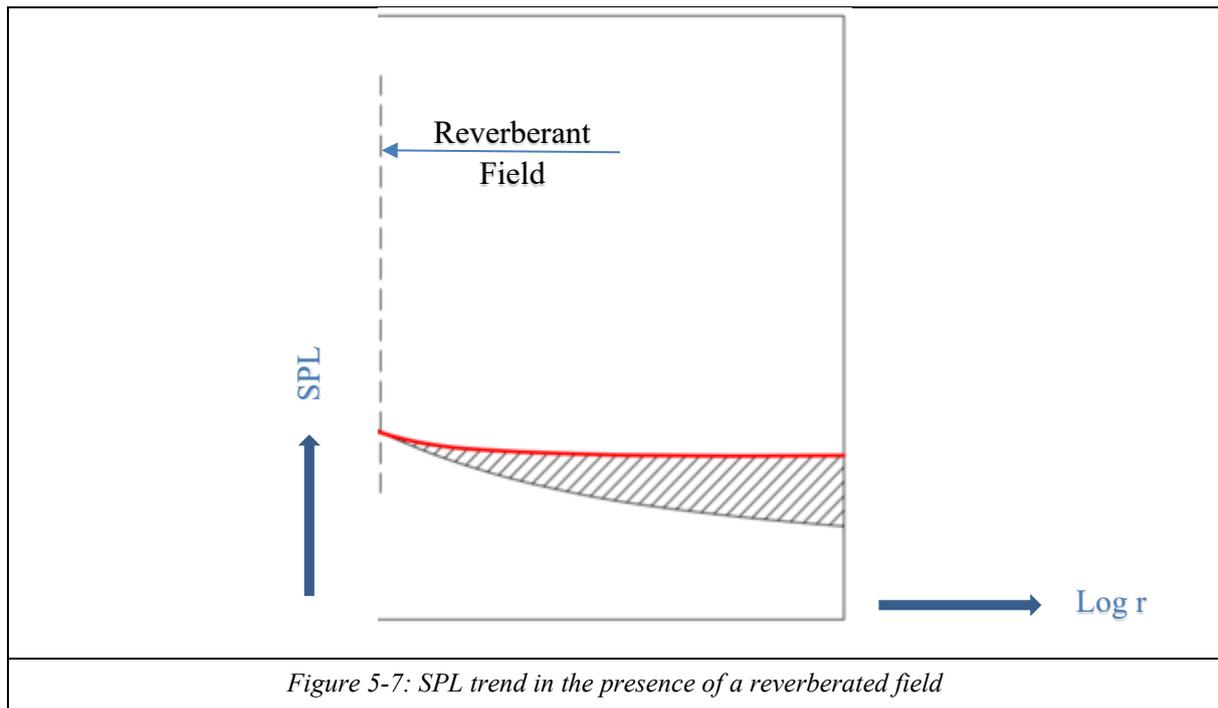


Approaching the source, at each halving of the distance the **SPL** value increases by 6 dB, up to a boundary line between near field and far field. This is because the energy is spread over an area that increases with the radius squared. From that point on, the increase is less than 6dB. For positions very close to the source, the **SPL** value tends to stabilize itself and present itself almost constant.

The propagation of sound is also affected by the type of environment in which the source and receiver are placed. The environments can typically be divided into three: free field, reverberant field, mixed field.

The free field is made up of an environment free of other objects or obstacles, that can reflect the sound and raise the sound pressure level at the receiver. A good approximation of free field can be made up of particular test environments such as anechoic chambers. Sometimes it is convenient to consider a reflecting surface on which the source is placed: it is the typical case of the heat pumps, whose units are often placed on reflecting surfaces, usually the floor, while sometimes the smaller units are placed on the wall. A good approximate free field with a reflecting plane can be created by a semi-anechoic chamber or by an open-air square (a parking lot, very far from the nearest building or from the nearest object). In free field, and in far field, the law of the decrease of 6dB is valid with the doubling of the distance previously described (normative references for the determination of the sound power in free field: **ISO 3744**, **ISO 3745**, **ISO 9614**. For more details on these standards see section 1.2).

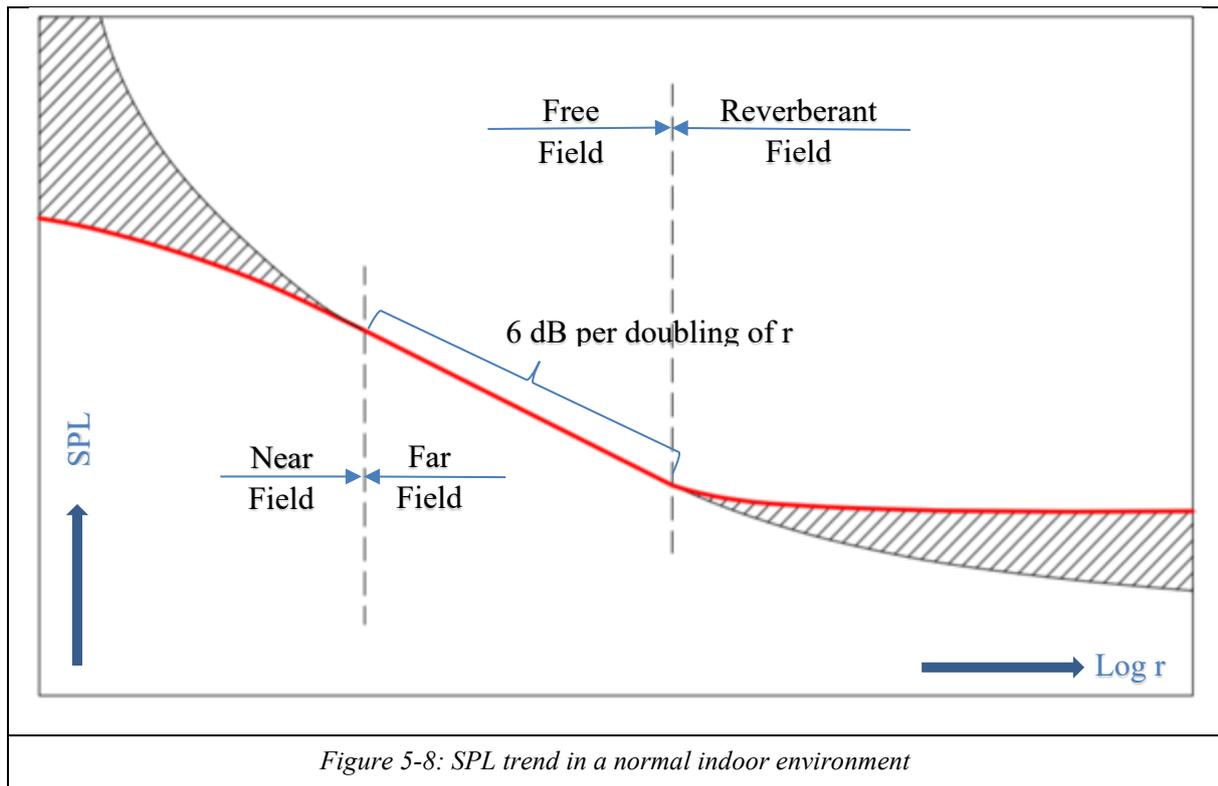
The second type of sound field consists of a perfectly reverberating field. By definition, a field is called reverberant if, out of the near field, the **SPL** value is constant in all points and thus does not vary any more as the mutual distance between source and receiver changes. What happens in this case is what is reported in the Figure 5-7:



The dashed area represents the divergence between the free field case and the reverberated field case: if we were in the free field case there would be the usual 6 dB decrease for each doubling of the mutual distance between source and receiver. In the case of a reverberated field, the sound pressure level remains constant.

A good reverberation field approximation is obtained in test environments called reverberation chambers.

The real environments, especially the confined ones, that is the indoor spaces, apart from the special test chambers mentioned above, constitute the third case, the most frequent one, of the real indoor environments. Their behavior involves a mixed course and is arranged by the Figure 5-8 obtained by combining the two figures: Figure 5-6 and Figure 5-7.



In real environments, very close to the source the sound pressure level does not grow so rapidly (near field behavior) with the decreasing of distance, as expected due to the 6dB increment per halving the distance r , and at a sufficient distance from the source the sound energy reflected by surrounding objects and walls begins to get into action increasing the sound pressure level (reverberant field behavior) expected due to a 6dB decrement per doubling the distance from the source. In real indoor environments the “6dB law” is valid only in a restricted interval (this interval depends on the characteristics of the environment).

1.8 Absorption, reflection and transmission of sound

Leaving aside a complete discussion, when a sound wave hits a surface it can be absorbed, reflected or partially absorbed and reflected. Usually a part of the incident sound energy is absorbed, a part is reflected, and a part is transmitted through the surface. One speaks therefore of coefficients of:

reflection: r

absorption: a

transmission: t

Each of these can vary from 0 to 1 and in the generic case are linked by the following relation (easily obtainable from a balance starting from the noise powers incident on the surface considered):

$$a + r + t = 1$$

Eq. 5-1



moreover, everything that is not reflected is called **apparent absorption** and is indicated with the symbol α (quantity usually considered in place of absorption a).

$$\alpha = 1 - r = a + t$$

Eq. 5-2

Each type of material or surface treatment in particular has a certain acoustic absorption coefficient (which varies according to the frequency). Some examples are reported in Table 5-2.

Material	Octave band frequency in Hz						
	125	250	500	1k	2k	4k	8k
Walls, hard surfaces average (brick walls, plaster, hard floors, etc.)	0.02	0.02	0.03	0.03	0.04	0.05	0.05
Walls, rendered brickwork	0.01	0.02	0.02	0.03	0.03	0.04	0.04
Rough concrete	0.02	0.03	0.03	0.03	0.04	0.07	0.07
Smooth unpainted concrete	0.01	0.01	0.02	0.02	0.02	0.05	0.05
Rough lime wash	0.02	0.03	0.04	0.05	0.04	0.03	0.02
Smooth brickwork with flush pointing, painted	0.01	0.01	0.02	0.02	0.02	0.02	0.02
Smooth brickwork, 10 mm deep pointing, pit sand mortar	0.08	0.09	0.12	0.16	0.22	0.24	0.24
Brick wall, stuccoed with a rough finish	0.03	0.03	0.03	0.04	0.05	0.07	0.07
Ceramic tiles with a smooth surface	0.01	0.01	0.01	0.02	0.02	0.02	0.02
Limestone walls	0.02	0.02	0.03	0.04	0.05	0.05	0.05
Reverberation chamber walls	0.01	0.01	0.01	0.02	0.02	0.04	0.04
Concrete floor	0.01	0.03	0.05	0.02	0.02	0.02	0.02
Marble floor	0.01	0.01	0.01	0.02	0.02	0.02	0.02
Material	Octave band frequency in Hz						
	125	250	500	1k	2k	4k	8k
Single pane of glass, 3 mm	0.08	0.04	0.03	0.03	0.02	0.02	0.02
Glass window, 0.68 kg/m ²	0.10	0.05	0.04	0.03	0.03	0.03	0.03
Lead glazing	0.30	0.20	0.14	0.10	0.05	0.05	–
Double glazing, 2–3 mm glass, > 30 mm gap	0.15	0.05	0.03	0.03	0.02	0.02	0.02
Double glazing, 2–3 mm glass, 10 mm gap	0.10	0.07	0.05	0.03	0.02	0.02	0.02
Double glazing, lead on the inside	0.15	0.30	0.18	0.10	0.05	0.05	–

Table 5-2: Examples of alfa coefficient for some materials



In confined environments, the value of α of all surfaces is low and the environment will behave like a reverberant field. Actually, for reverberant test chambers (reference: **ISO 3741**) the discourse is much more complex and must take into account many other aspects that are not treated here for convenience (modal analysis, echo flutter, external acoustic insulation, decoupling from vibrations, etc. ...).

1.9 Heat pumps case:

NOTE: in the case of technical documentation of heat pumps, the catalog values are often inaccurate and not very comparable with each other. The most accurate data (necessary for some certification systems) is that of the sound power level (a value that does not depend on the measurement position, since the sound power is an intrinsic property of the source or of its mode of operation). The first problem is that often the values show “something” expressed in dB, but it is not known whether it is a sound power level, or a sound pressure level. In the second case sometimes it is not indicated at what distance the measurement was performed. The second problem is that often a sound pressure level (measured or calculated) is often indicated at one meter. Sometimes a measured value (or calculated) is indicated at 10 meters or still at 30 meters. It should be noted that the correct measurement of SPL should still be performed in the free field and not in the near field. From what has been said above it is important to remember that in order to know the sound pressure level of a heat pump at a distance of one meter it is usually not enough to perform a generic measurement by simply placing the microphone one meter away from it, since the microphone could be placed in the near field, producing an underestimation of the measure. In addition, the source may not have been measured in a correct environment (a free field) and the reflected sound energy counter may produce an overestimation of the source. This is the reason why the simple sound pressure measurements are carried out at adequate distances (7-10-30 meters) from the source, in free field preferably, and in different positions (to take into account the directivity of the source). The single value of sound pressure level should refer either to an average of values taken at different angles or to the value taken in the direction of maximum sound emission.

6. Psychoacoustics

The physical phenomenon, as it can be defined by the quantities introduced up to now and measured by microphones and instruments, does not exactly correspond to that perceived by human ears: the human ear response is not linear in frequency and this non-linearity also varies with “loudness” (perceived volume) of the phenomenon.

1.10 Human auditory system: how it is made and how it works

The human auditory system can be seen composed of three parts: the outer ear, the middle ear and the inner ear. Inside the auricle there is a conduit called the ear canal which ends on a tympanic membrane. The tympanic membrane is a thin, elastic diaphragm, impermeable to water and air that separates the outer ear from the middle ear. Figure 6-1 shows an example of human ear.

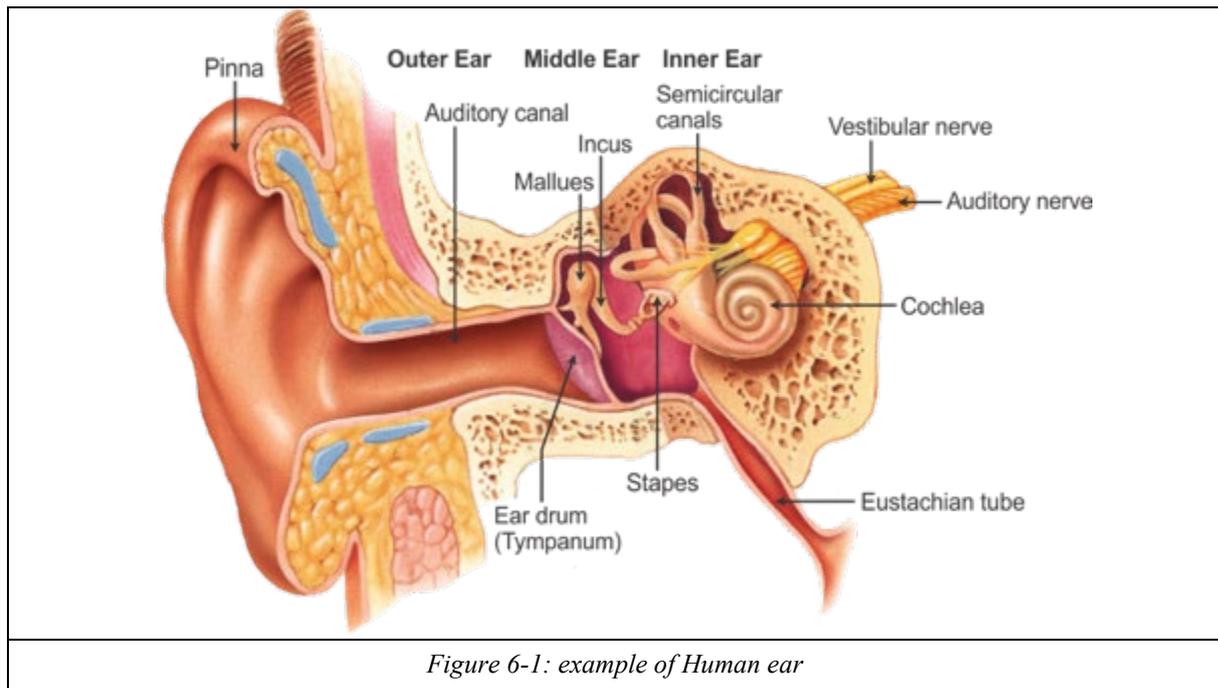


Figure 6-1: example of Human ear

The middle ear consists of an internal cavity of the skull bone, which also contains air, and a complex chain of bones (called Ossicles and shown in Figure 6-2) for transmitting the vibration of the tympanic membrane to the hearing organ properly called, the cochlea, which it is located in the inner ear. Also in the inner ear is a second organ that is unrelated to the auditory system, the labyrinth (or semicircular canals). It contains the center of equilibrium.

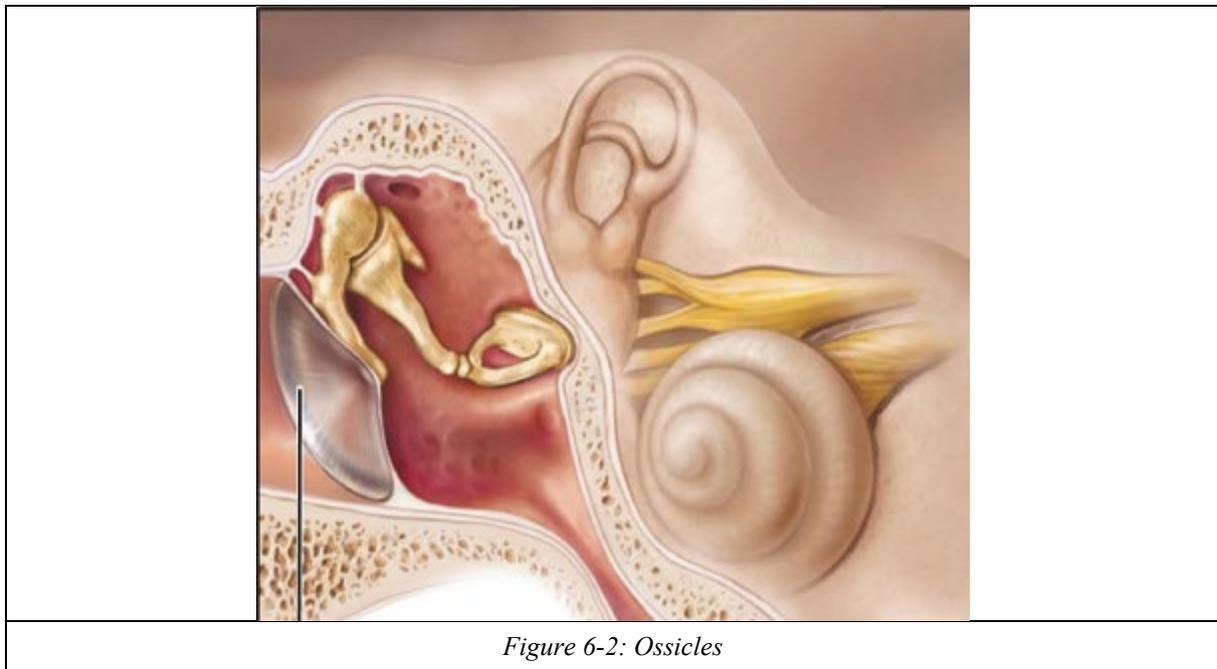


Figure 6-2: Ossicles

The three ossicles (very small bones) are located inside the middle: “malleus”, “incus” and “stapes” are hinged to each other. The first one is in contact with the tympanic membrane, while



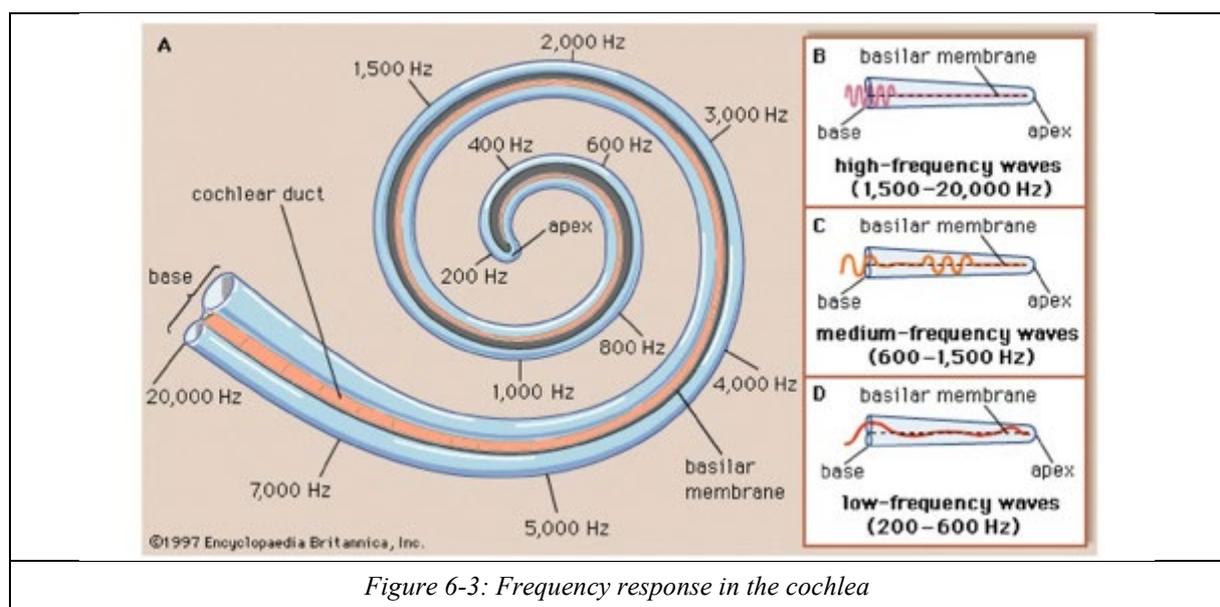
the third one rests on a further diaphragm, the oval window, which connects it to the cochlea. The cochlea is filled with a liquid with an impedance similar to that of water, and therefore a thousand times higher than that of air.

Now, the tympanic membrane has an acoustic impedance only slightly higher than the atmospheric value and this is compensated by the shape of the ear canal of the external ear which performs a horn loading to allow a perfect coupling of impedance between the conduit and the eardrum. The result is an excellent transducer of the acoustic field that performs with the maximum energy transfer. However, it should be noted that the conduit is small so it works better at high frequencies (3000 - 5000 Hz), while at low frequencies the impedance is mismatched and the response is less (we will see however that this is not a problem).

The transfer of energy from the tympanic membrane to the oval window is instead more problematic, since the liquid inside the cochlea has, as mentioned above, an impedance thousand times higher than that of the air and also inside it the sound travels at a speed 4-5 times higher: the result is that the impedance of the tympanic membrane is about three thousand times lower than that of the oval window and therefore, if they were put in direct contact, the transfer of energy would be very scarce.

The ossicles therefore function as levers, transforming the large movements associated with small forces of the tympanic membrane into small movements and great forces of the stirrup. A mechanism of this type is called a mechanical impedance transformer, and although it increases the ratio between the internal pressure and the sound pressure received from the outside it is not a true amplifier, because this process is performed by decreasing the speed of the wave: in other words, one energy increases at the expense of another.

The signal thus transformed reaches the cochlea (also called a snail for its shape. The snail is composed of two 30mm long channels (or scales) placed in contact with each other through the basement membrane: the vestibular canal, which carries the sound towards the center of the snail, and the tympanic canal, which guides the signal outwards towards the round window. Figure 6-3 shows a section and the frequency response of the cochlea.





The basement membrane is subjected to a deflection while the sound travels through it, due to the differences in pressure that are created in the two ducts placed in contact, and these efforts are recorded by the ciliated cells that entrust the new signal to a neural network. Due to this system, our brain receives extremely selective information about how the sound is distributed at various frequencies. The response times are not instantaneous, but vary from 25 to 150 milliseconds depending on the frequency of the signal: it can be said that the high frequency sounds are heard earlier.

In fact at the entrance of the snail the basal membrane is thin and taut like a violin string and the impedance of the liquid inside the cochlea is greater (since the conduit is wider): the resonant frequency in this zone is therefore high. Proceeding towards the center, the membrane becomes thicker and less taut, to be like a double bass string: this mechanism causes the components of the low frequency sound to find their resonance zone only after having traveled the 30 mm length of the vestibular canal. Therefore they are heard with some delay and attenuate.

Being the only transmission channel, the low volume sound components are rendered inaudible by those at high volume and frequency close to them. This phenomenon is called “spectral masking”, and is exploited to compact audio information as in the case of minidisc and MP3 files, in which the contributions that the auditory system would not perceive are eliminated.

1.11 Sound sensation

We therefore see that the response of our auditory system is not the same for all frequencies: we can say that two sounds at different frequencies can have the same intensity but give a different level of sensation. The Figure 6-4, obtained experimentally, indicates the threshold of audibility, this is the minimum intensity that a sound must have to be heard at different frequencies, and the pain threshold, beyond which the sound has harmful effects even for short exposure. Between these two lines extends the area of sounds audible to humans.

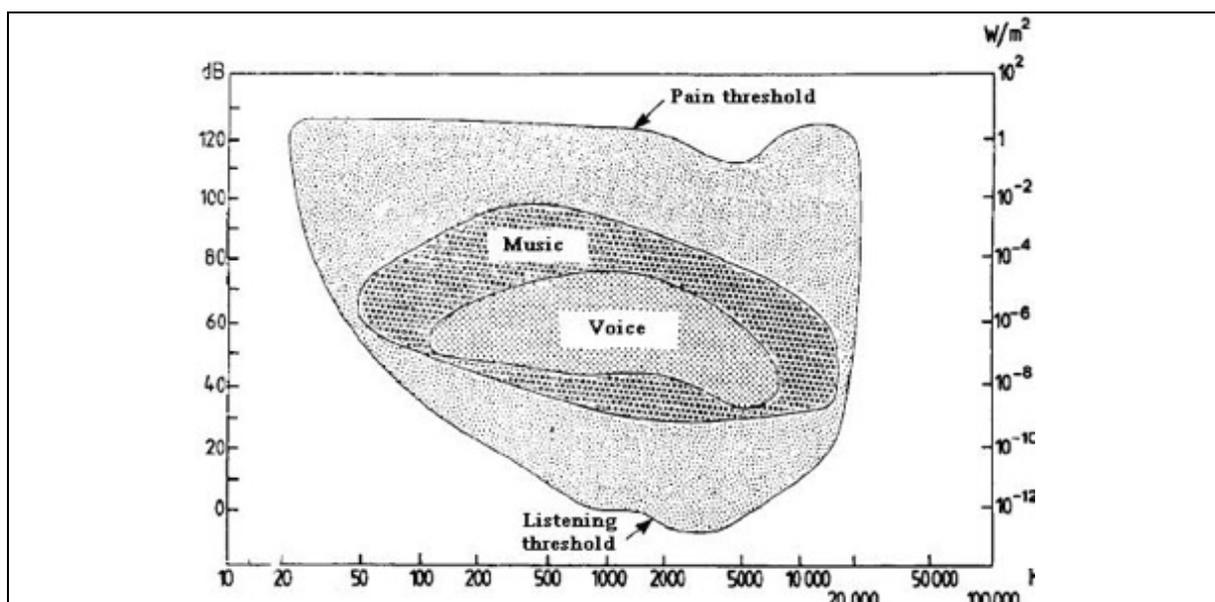


Figure 6-4: Area of auditory sensation



In the 1930s Fletcher and Munson, derived isophonic curves by examining a large number of subjects. These curves represent the sound pressure level that a sound must have to give the same intensity or loudness sensation at various frequencies. The observers are subjected alternately to a pure tone of a certain frequency and to another tone at the reference frequency (1000 Hz). The intensity of the latter is adjusted to give the same sensation as the first sound, and in this way it is established to which curve the first pair of values belongs (frequency - intensity). Figure 6-5 shows the isophonic curves obtained from the comparison work previously described.

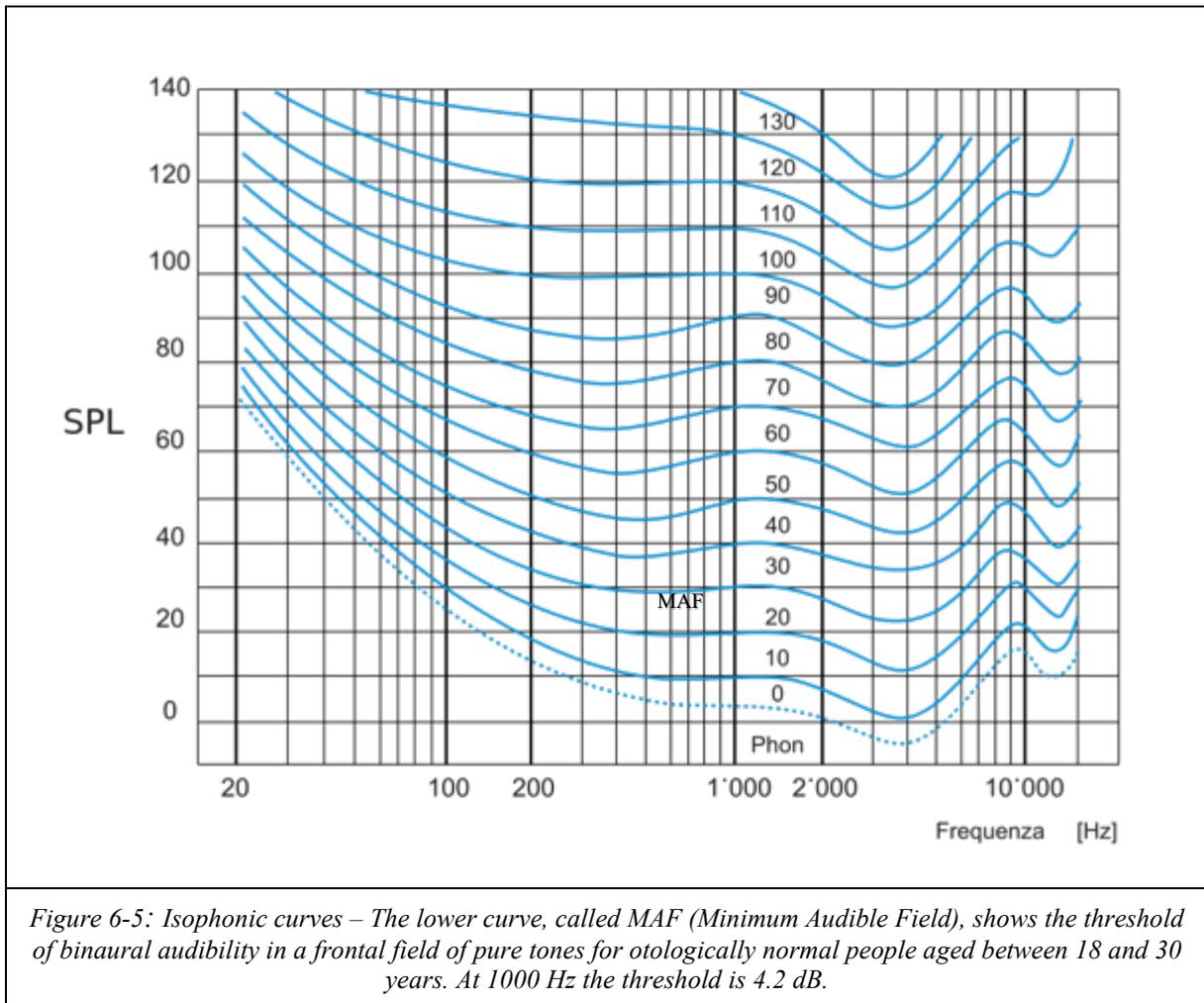


Figure 6-5: Isophonic curves – The lower curve, called MAF (Minimum Audible Field), shows the threshold of binaural audibility in a frontal field of pure tones for otologically normal people aged between 18 and 30 years. At 1000 Hz the threshold is 4.2 dB.

1.12 The dB scale

All the isophonic curves have a very similar shape, with a peak of audibility around 4000 Hz. However, it can be noted that as the intensity increases the response of the auditory system flattens out. Nevertheless it is possible to obtain the doubling unit, this is the factor for which it is necessary to multiply the sound intensity to have a doubling sensation. This value was established by Graham Bell in 3.16, that is $\sqrt{10}$. We reiterate that it is only a mediated value, since the response to a variation in sound pressure is different depending on the frequency and amplitude.



Taking an arbitrary scale, at the various pressures we could have the following results:

Sound pressure P	Sensation ($Sens$)
0,01 Pa	1
0,0316	2
0,1	3
0,316	4
1	5

Table 6-1: Equivalence between sound pressure and sensation

where an increase of $Sens$ of a unit is equivalent to a doubling sensation. Bell defined the **sound sensation** as:

$$Sens = \log \frac{p^2}{p_{ref}^2} \quad [B] \quad Eq. 6-1$$

where the unit of measure between square brackets is the **Bel**, while P_0 is the reference pressure, established in $2 \cdot 10^{-5}$ Pa, corresponding to the weakest sound audible to humans at 1.000 Hz. Note that now it is no longer considered as such, as the MAF (Minimum audible field) curve shows, however it continues to be taken as a reference pressure.

This scale turned out to be too coarse, and today the most commonly used unit of measurement is the **decibel (dB)**, or **the tenth of Bel**. To avoid confusion the value in dB is called **level (L)** and not sensation, so we will write:

$$L = 10 \cdot \log \frac{p^2}{p_{ref}^2} \quad [dB] \quad Eq. 6-2$$

Some observations: a 0 dB sound, according to Bell, corresponded to the weakest audible sound at 1.000 Hz (in fact, because the logarithm is zero, its argument must be 1, or P must be equal to P_0). The fact that the pressure terms are elevated to the square suggests that our auditory system has a proportional response to their average effective value, and therefore to the energy content (which we know is proportional to the square of the pressure).

Ultimately, the characteristics with which we can construct a tool more similar to the human ear works at **RMS** pressure levels with fast time constant (125 ms). In formula:

$$Sens = \log \frac{P_{RMS}^2}{P_{ref}^2} \quad Eq. 6-3$$

However for acousticians a more interesting quantity than Sens, is the Sone, unit widely used for loudness measurement in the place of phon, or together with it. For definition 1 Sone is



equivalent to the loudness of 40 phons. After that, doubling the perceived loudness doubles the value of sone. For that the sone scale is not a logarithmic scale, but increasing the loudness value of 10 phons, the loudness value expressed in sone is doubled as shown in Table 6-2.

Phon	10	20	30	40	50	60	70	80	90	100	110	120	130
Sone	0,125	0,25	0,5	1	2	4	8	16	32	64	128	256	512

Table 6-2: Sone scale and phon scale comparison

1.13 The dB(A) scale

In order to achieve a good approximation of the human response, it is also necessary to compensate instrumentally due to the fact that the ear hears higher frequencies better than the low ones. This operation, called weighting, is performed by means of the Fletcher Munson diagram, this means going to see to which **isophonic curve** a certain frequency-level pair belongs. To facilitate the operation, it is sufficient to have available a graph of Fletcher Munson overturned, which allows us to establish what value we must add to the sound levels obtained at the various frequencies to obtain the actual human sensation.

As already mentioned, the **isophonic curves** are similar to each other, but still vary as the level increases, so we need more curves to be used in the various cases. In this regard, there is **curve A** (for levels below 60 dB), **curve B** (between 60 and 80 dB), **curve C** (over 80) and **curve D** (for very loud noises, such as those of aircraft) and the measurements are defined in **dB (A)**, **dB (C)** etc. depending on the weighting curve used. To avoid confusion, the unweighted measurements can be indicated in **dB (LIN)**. Instead of **dB (LIN)** there is now a Z-weighting indicated as **dB(Z)**. The difference is that **dB (Z)** has a defined frequency range from 10 Hz to 20 kHz whereas **dB (LIN)** is to a certain degree manufacturer-specific due to different frequency cut-offs.

Nevertheless, for our purposes it will be useful to have only the weighting curve A available, which also includes the tabulated values. In fact, the curve **B** and the curve **D** usually are not taken into account by international laws, while the **C** concerns only very loud noises.

1.14 For which reason we use only the curve A

The weighting curve **A** was found to be on average better correlated with the subjective human response to broad band generic noises; this fact, combined with the ease of a phonometric measurement in dB(A), has led to the adoption of the **A** curve in many national and international laws and regulations. On the other hand, it is well known that this way of proceeding lends itself to many criticisms:

- there are many other scales of evaluation of the sound sensation, generally much more refined than the **A** curve;
- the isophonic curves have been built working with pure tones, while the curve **A** is generally used to evaluate wideband noises;
- moreover, it is now widely demonstrated that the **A** curve does not give an adequate evaluation when the noise has strong tonal components or is of an impulsive type;



- the low frequency noise disturbance is certainly underestimated using a single number in dB (A).

For these and other reasons, today it is believed that the A curve no longer has the meaning originally intended to be attributed to it. Nevertheless, the A curve remains for its simplicity a common reference for a first approximate evaluation of broadband noise. In reality, the strongest motivation for maintaining the A curve seems to be its omnipresence in the sector regulations. At this point its meaning is purely conventional, which is why the electroacoustic regulations, that define the characteristics of sound level meters, prefer the use of a precise mathematical expression defined by the A (and C) curve instead of trying to define or use more complicated psychoacoustic quantities.

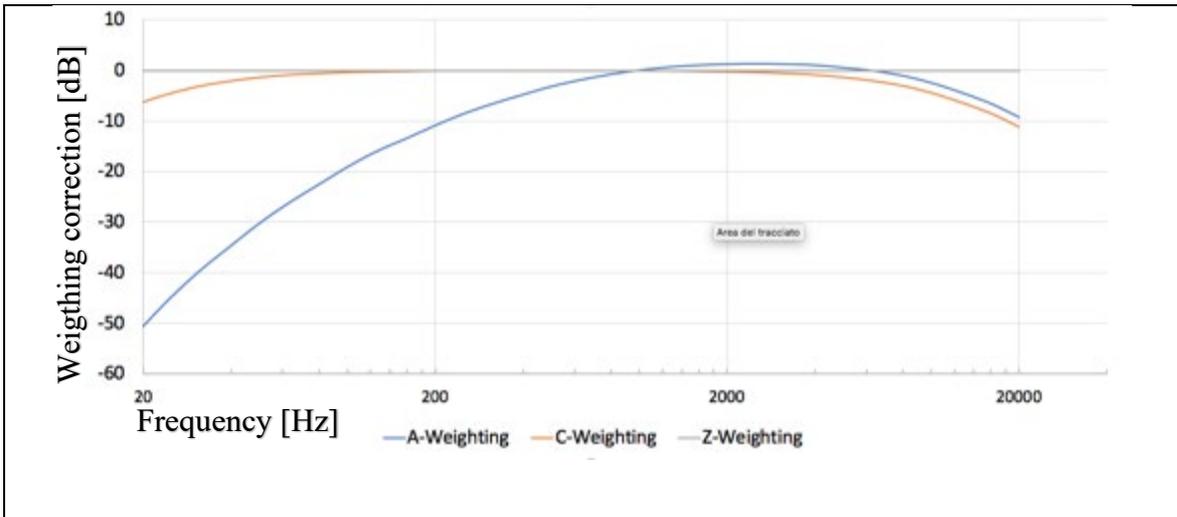


Figure 6-6: A, C and Z weighting curves

Frequency [Hz]	A-Weighting	C-Weighting	Z-Weighting
20	-50,5	-6,2	0
25	-44,7	-4,4	0
31,5	-39,4	-3	0
40	-34,6	-2	0
50	-30,2	-1,3	0
63	-26,2	-0,8	0
80	-22,5	-0,5	0
100	-19,1	-0,3	0
125	-16,1	-0,2	0
160	-13,4	-0,1	0
200	-10,9	0	0
250	-8,6	0	0
315	-6,6	0	0
400	-4,8	0	0
500	-3,2	0	0
630	-1,9	0	0
800	-0,8	0	0
1000	0	0	0
1250	0,6	0	0
1600	1	-0,1	0
2000	1,2	-0,2	0
2500	1,3	-0,3	0
3150	1,2	-0,5	0
4000	1	-0,8	0
5000	0,5	-1,3	0
6300	-0,1	-2	0
8000	-1,1	-3	0
10000	-2,5	-4,4	0
12500	-4,3	-6,2	0
16000	-6,6	-8,5	0
20000	-9,3	-11,2	0

Table 6-3: A, C and Z weighting values



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Figure 8-16: Area of auditory sensation - from https://en.wikibooks.org/wiki/Acoustics/Fundamentals_of_Psychoacoustics

Figure 8-17: Isophonic curves – The lower curve, called MAF (Minimum Audible Field), shows the threshold of binaural audibility in a frontal field of pure tones for otologically normal people aged between 18 and 30 years. At 1000 Hz the threshold is 4.2 dB – from https://en.wikibooks.org/wiki/Acoustics/Fundamentals_of_Psychoacoustics

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