ACOUSTICS
part – 3
Sound Engineering Course

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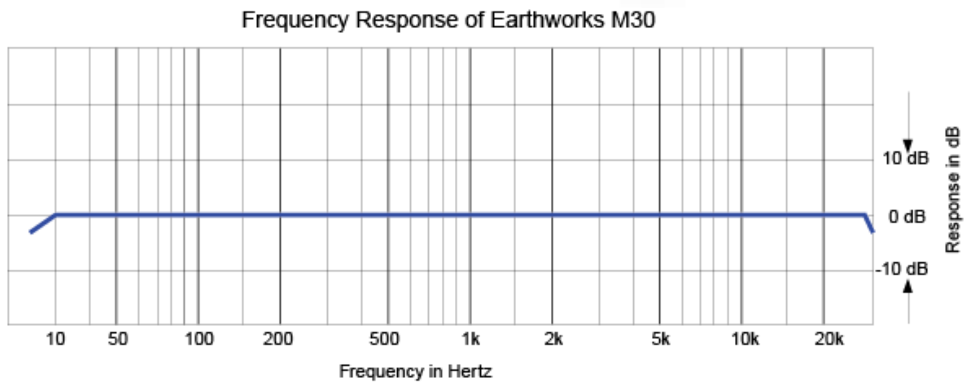
Microphones

Omnidirectional microphones

- ISO3382 recommends the usage of omni mikes of no more than 13mm
- These are the same microphones usually employed on sound level meters:

Sound Level Meter (records a WAV file on the internal SD) (left)

Measurement-grade microphone and preamplifier (to be connected to a sound card) (right)



Spatial analysis by directive microphones

- The initial approach was to use directive microphones for gathering some information about the spatial properties of the sound field “as perceived by the listener”
- Two apparently different approaches emerged: binaural dummy heads and pressure -velocity microphones:

Binaural microphone (left)

and

variable-directivity microphone (right)

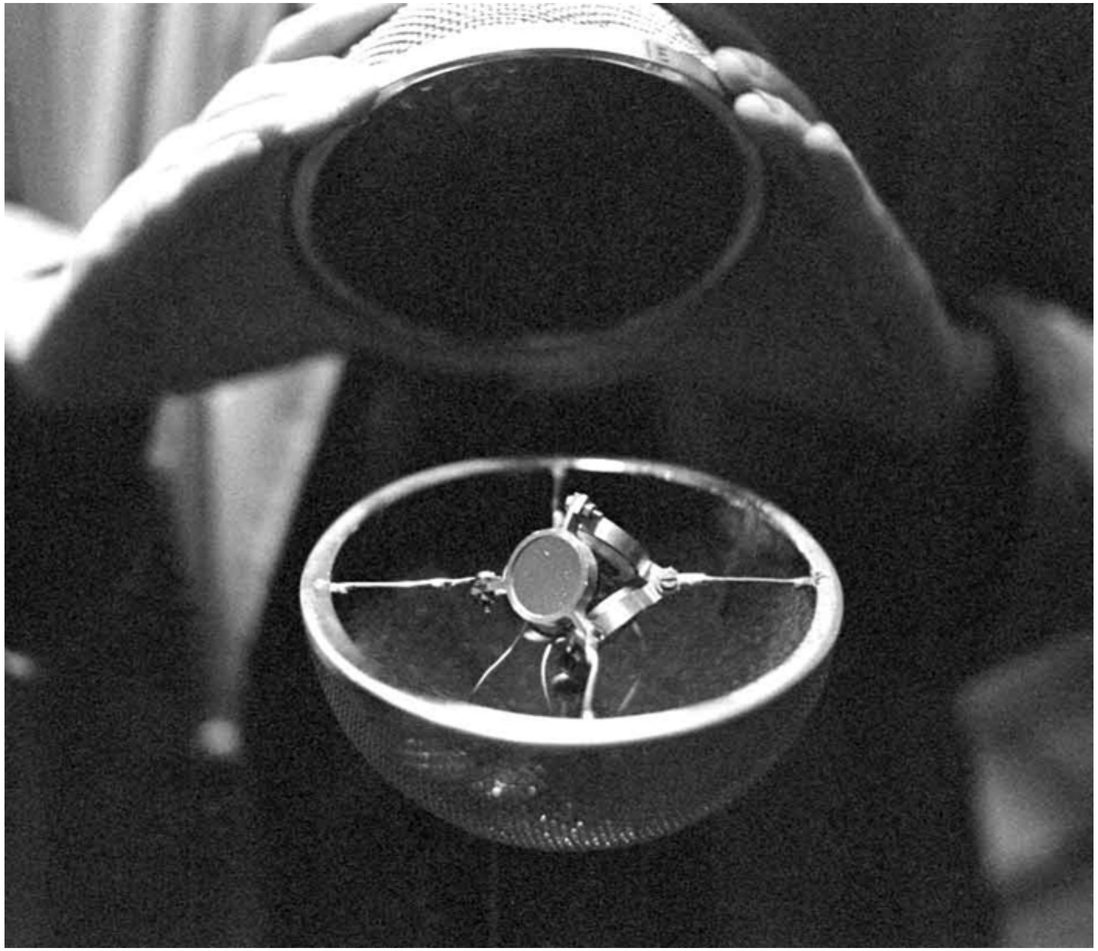
Test with binaural microphones

- Cheap electret mikes in the ear ducts

Capturing Ambisonics signals

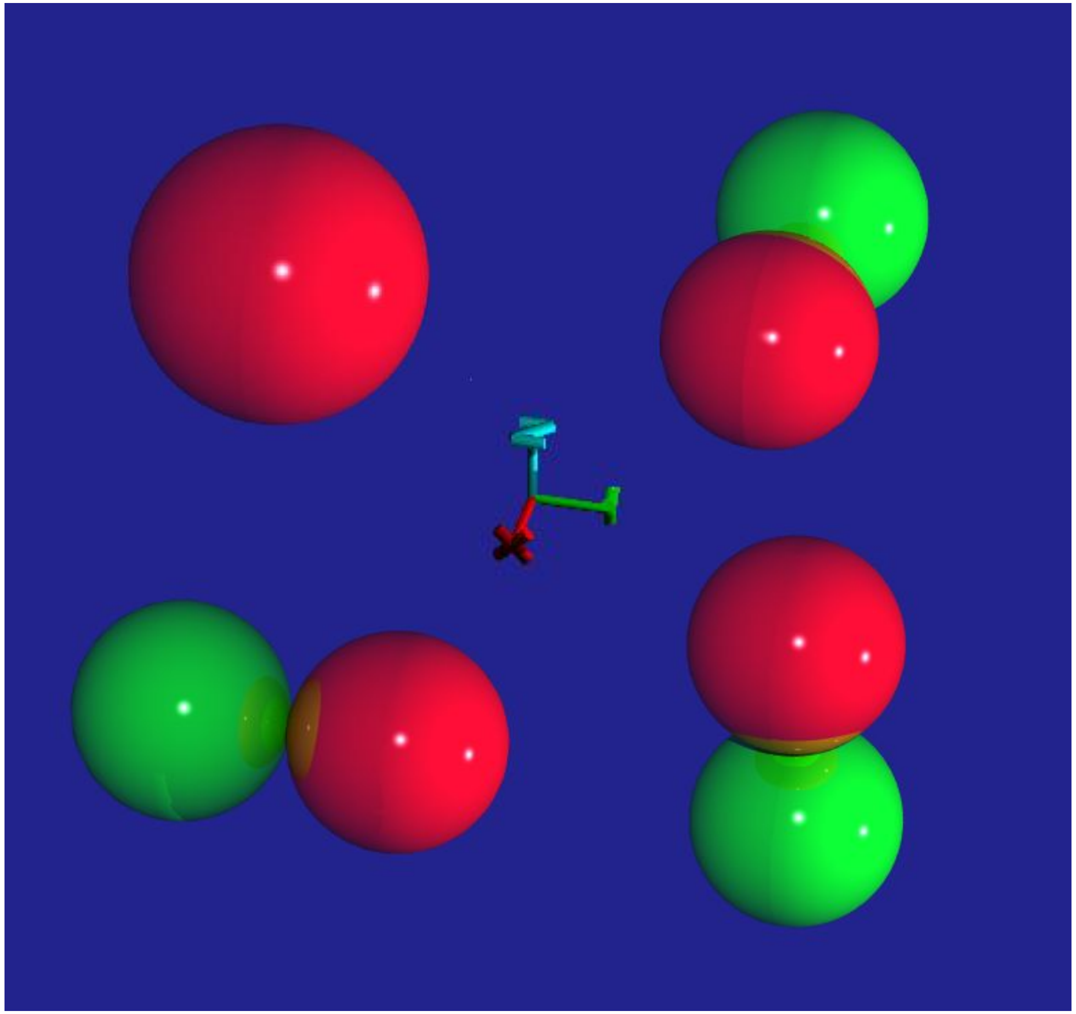
- A tetrahedrical microphone probe was developed by Gerzon and Craven, originating the Soundfield microphone

Soundfield microphones



Soundfield microphones

- The Soundfield microphone allows for simultaneous measurements of the omnidirectional pressure and of the three cartesian components of particle velocity (figure-of-8 patterns)



Directivity of transducers

Soundfield ST-250 microphone

Alternative A-format microphones

- At UNIPR many other 1st-order Ambisonics microphones are employed (Soundfield ^{TM}, DPA-4, Tetramic, Brahma)

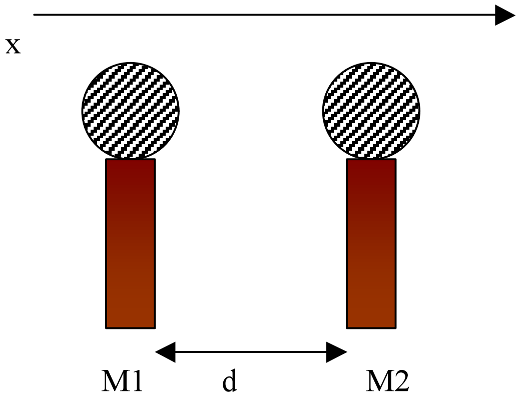
Portable, 4-channels microphone

- A portable digital recorder equipped with tetrahedrical microphone probe: BRAHMA

The Sound Intensity meter

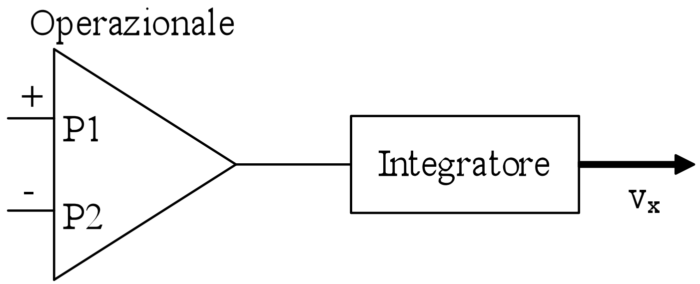
Euler equation: the sound intensity probe

A sound intensity probe is built with two amplitude and phase matched pressure microphones:



The particle velocity signal is obtained integrating the difference between the two signals:

$$grad(p)|_x = \frac{p_2 - p_1}{d}$$



Euler's Equation

Connection between sound pressure and particle velocity; it is derived from the classic Newton's first law (F = m · a):

$$\rho \cdot \frac{\partial v}{\partial \tau} = -grad(p)$$

It allows to calulate particle velocity by time-integrating the pressure gradient:

$$v = -\rho_0 \int grad(p)dt$$

The gradient is (approximately) known from the difference of pressure sampled by means of two microphones spaced a few millimeters (SOund Intensity probe).

Sound Intensity probe (1)

The Sound Intensity is:

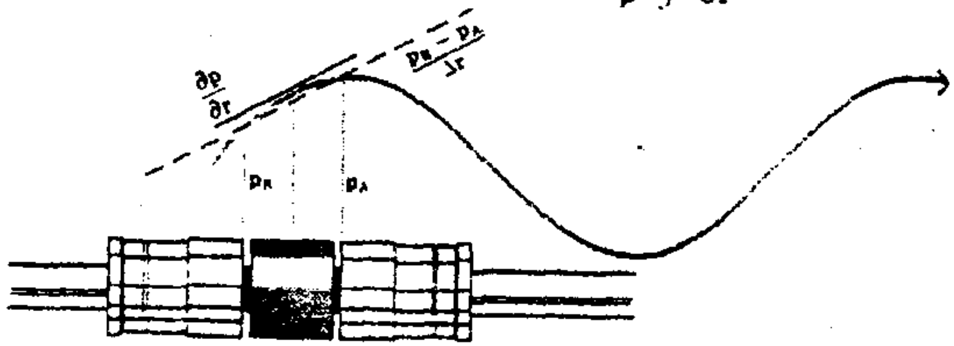
$$\vec{I} = p \cdot \vec{v}$$

Where both ***p*** and ***v*** can be derived by the two pressure signals captured by the two microphones

Formulazione in dominio del tempo

Secondo Eulero

$$u = -\frac{1}{\rho}\int \frac{\partial p}{\partial r}\,dt$$



L'approssimazione per differenza finita

$$u = -\frac{1}{\rho}\int \frac{p_B - p_A}{\Delta r}\,dt$$

Pressione media

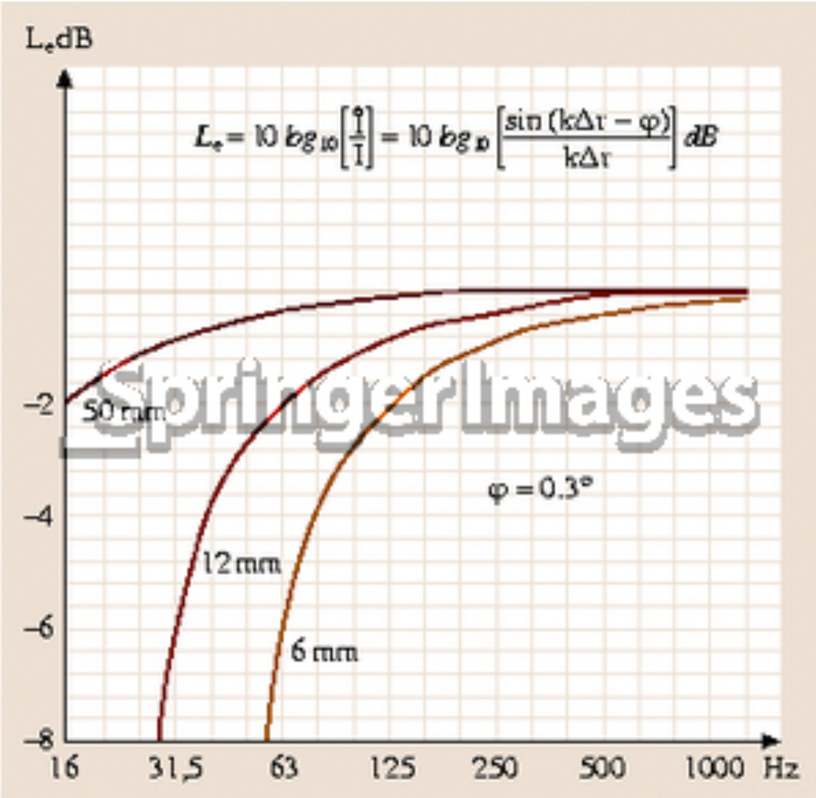
$$p = \frac{p_A + p_B}{2}$$

$$I = \overline{p \cdot u}$$

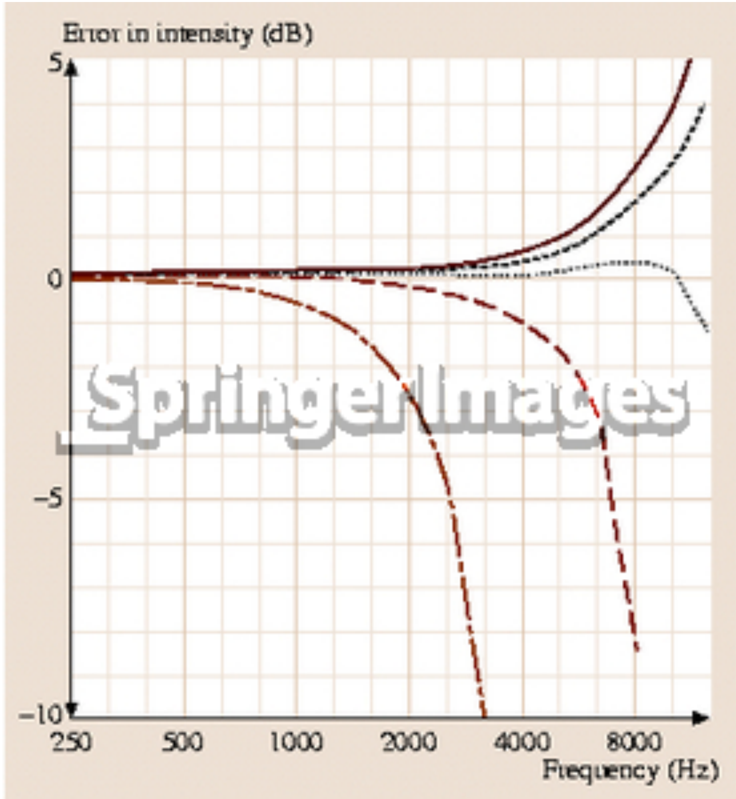
$$I = -\frac{p_A + p_B}{2\rho\Delta r}\int (p_B - p_A)\,dt$$

Sound Intensity probe (2)

Sound Intensity Probe: errors



Phase Mismatch Error due to a phase error of 0.3° in a plane propagating wave



Finite-Differences Error of a sound intensity probe with 1/2 inch microphones in the face-to-face configuration in a plane wave of axial incidence for different spacer lengths: 5 mm (*solid line*); 8.5 mm (*dashed line*); 12 mm (*dotted line*); 20 mm (*long dashes*); 50 mm (*dash-dotted line*)

Outdoors propagation

The D' Alambert equation

The equation comes from the combination of the continuty equation for fluid motion and of the 1st Newton equation (f=m·a).

In practive we get the Euler's equation:

$$grad(p) = \rho \cdot \frac{\partial v}{\partial \tau}$$

now we define the potential Φ of the acoustic field, which is the "common basis" of sound pressure ***p*** and particle velocity ***v***:

$$\vec{v} = grad(\Phi) \qquad p = -\rho_o \cdot \frac{\partial \Phi}{\partial \tau}$$

Subsituting it in Euler's equation we get::

$$\frac{\partial^2 \Phi}{\partial \tau^2} = c^2 \cdot \nabla^2 \Phi$$

D'Alambert equation

Once the equation is solved and Φ is known, one can compute p and v.

Free field propagation: the spherical wave

Let’s consider the sound field being radiated by a pulsating sphere of radius R:

$$v(R) = v_{max}\, e^{i\omega\tau}$$

k = ω/c
wave number

$e^{i\omega\tau} = cos(\omega\tau) + i\, sin(\omega\tau)$

Solving D’Alambert equation for r > R, we get:

$$v(r,\tau) = -v_{\max} \cdot \frac{R^2}{r^2} \cdot \frac{[1+ikr]}{[1+ikR]} \cdot e^{-ik(r-R)} \cdot e^{i\omega\tau}$$

Finally, thanks to Euler’s formula, we get back pressure:

$$p(r,\tau) = \frac{1}{r} \cdot \frac{R^2 \cdot i\omega\rho_0 v_{\max}}{[1+ikR]} \cdot e^{-ik(r-R)} \cdot e^{i\omega\tau}$$

Free field: proximity effect

From previous formulas, we see that in the far field (***r***>>***λ***) we have:

$$p \propto \frac{1}{r} \qquad v \propto \frac{1}{r}$$

But this is not true anymore coming close to the source.

When r approaches 0 (or ***r*** is smaller than ***λ***), ***p*** and ***v*** tend to:

$$p \propto \frac{1}{r} \qquad v \propto \frac{1}{r^2}$$

This means that close to the source the particle velocity becomes much larger than the sound pressure.

Free field: proximity effect

The more a microphone is directive (cardioid, hypercardioid) the more it will be sensitive to the partcile velocty (whilst an omnidirectional microphone only senses the sound pressure).

So, at low frequency, where it is easy to place the microphone "close" to the source (with reference to λ), the signal will be boosted. The singer "eating" the microphone is not just "posing" for the video, he is boosting the low end of the spectrum...

Free field: Impedance

If we compute the impedance of the spherical field (z=p/v) we get:

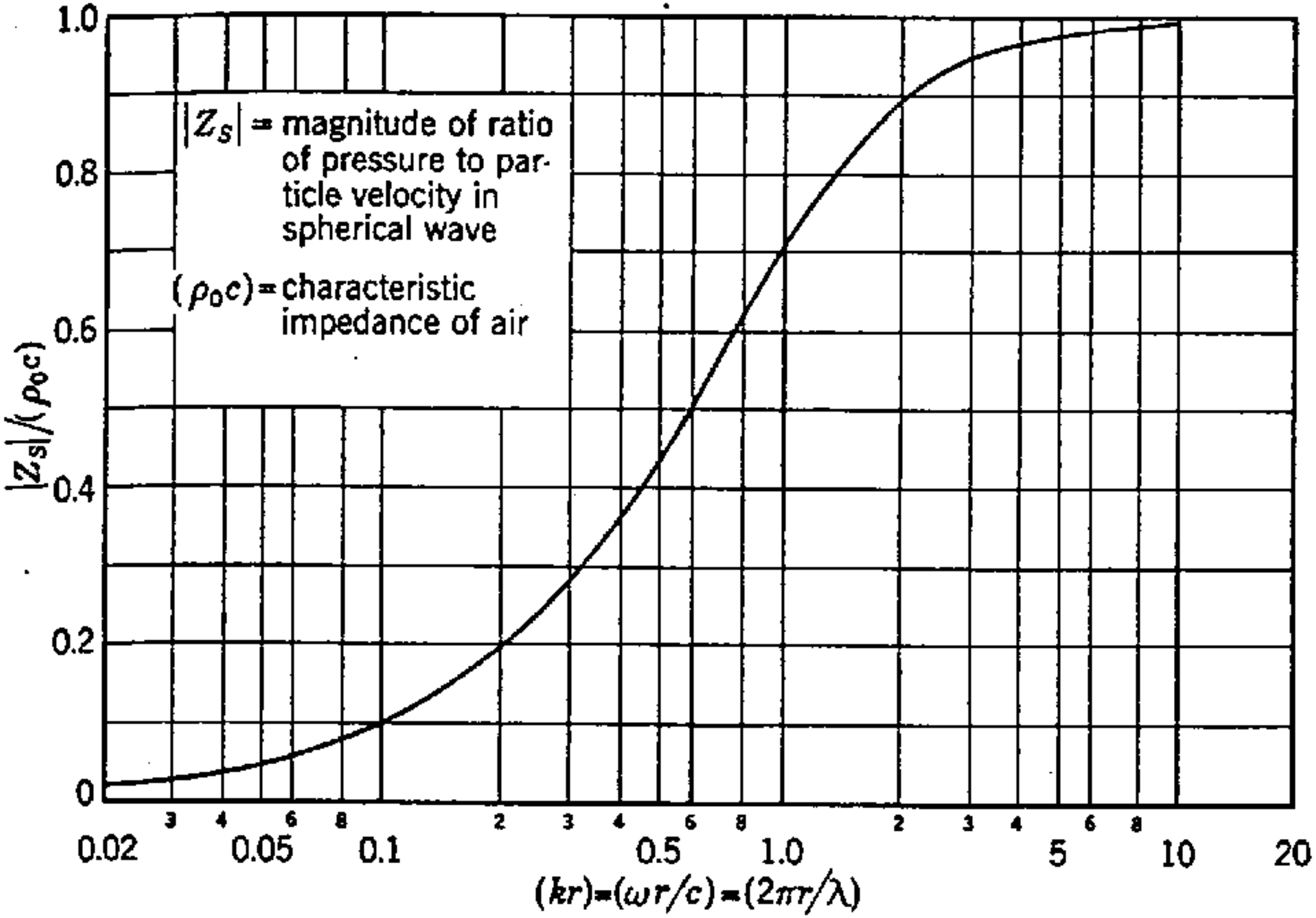
$$Z(r) = \frac{i\omega\rho_0 r}{1 + ikr} \quad (r > R)$$

When $\boldsymbol{r}$ is large, this becomes the same impedance as the plane wave ($\boldsymbol{\rho} \cdot \boldsymbol{c}$).

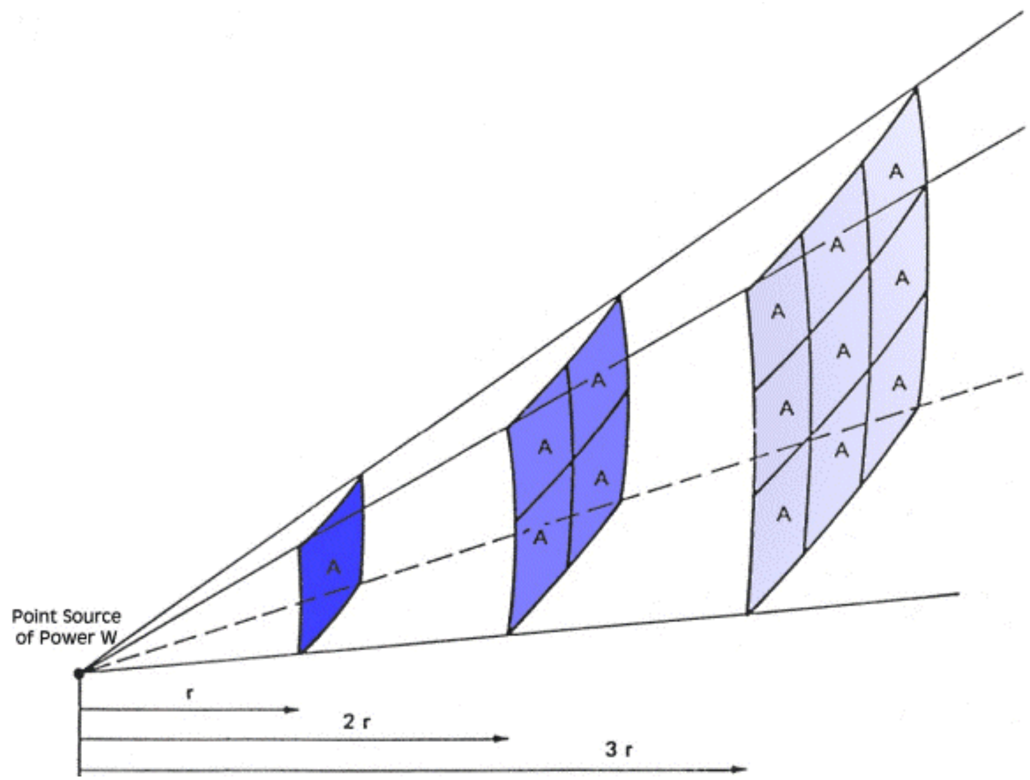
Instead, close to the source ($\boldsymbol{r} < \boldsymbol{\lambda}$), the impedance modulus tends to zero, and pressure and velocity go to quadrature (90° phase shift).

Of consequence, it becomes difficult for a sphere smaller than the wavelength $\boldsymbol{\lambda}$ to radiate a significant amount of energy.

Free Field: Impedance



Free field: energetic analysis, geometrical divergence



The area over which the power is dispersed increases with the square of the distance.

Free field: sound intensity

If the source radiates a known power W, we get:

$$I = \frac{W}{S} = \frac{W}{4\pi r^2}$$

Hence, going to dB scale:

$$L_I = 10\log\frac{I}{I_0} = 10\log\left(\frac{\frac{W}{4\pi r^2}}{I_0}\right) = 10\log\left(\frac{\frac{W}{4\pi r^2}}{I_0}\frac{W_0}{W_0}\right) = 10\log\frac{W}{W_0} + 10\log\frac{W_0}{I_0} + 10\log\frac{1}{4\pi} + 10\log r^{-2}$$

$$L_I = L_W - 11 - 20\log r$$

A spherical wave is propagating in **free field** conditions if there are no obstacles or surfacecs causing reflections.

Free field conditions can be obtained in a lab, inside an anechoic chamber.

For a point source at the distance r, the free field law is:

- $L_p = \mathbf{L_I} = L_W - 20 \log r - 11 + 10 \log Q \qquad \text{(dB)}$

where L_W the power level of the source and Q is the *directivity factor*.

When the distance ***r*** is doubled, the value of ***Lp*** decreases by 6 dB.

Free field: directivity (1)

Many soudn sources radiate with different intensity on different directions. Hence we define a direction-dependent **“directivity factor” Q** as:

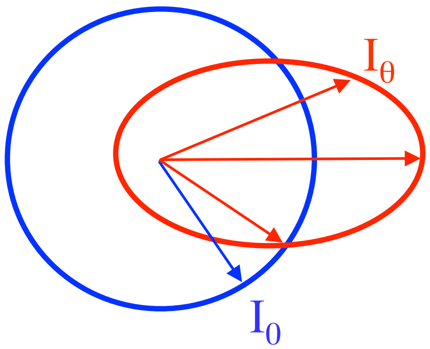
- $Q = I_\theta / I_0$

where I_θ è is **sound intensity** in direction **θ,** and I_0 is the **average sound intensity** consedering to average over the whole sphere.

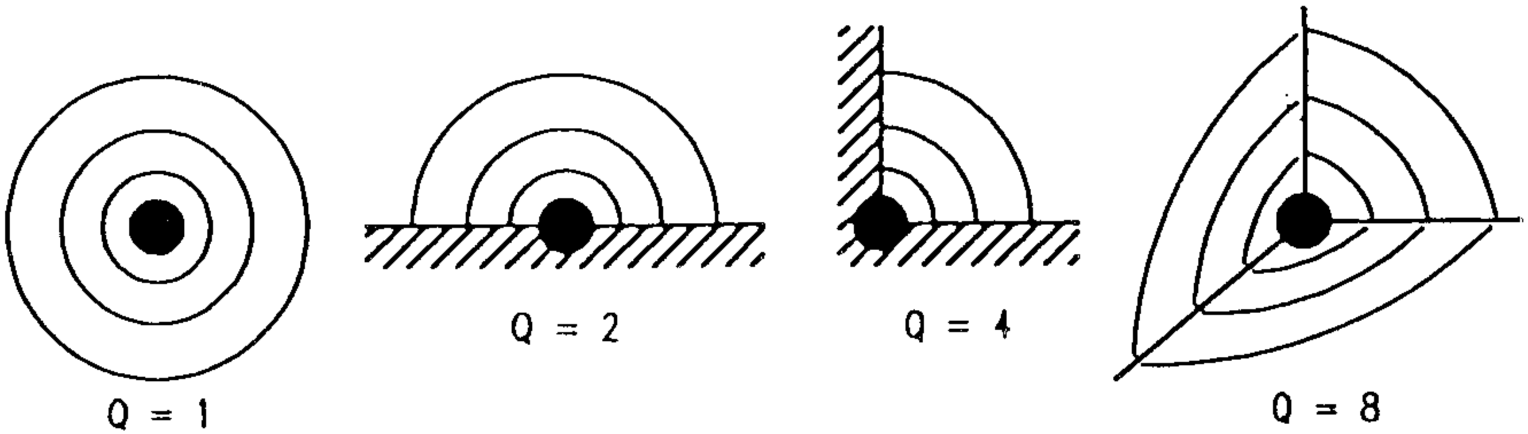
From Q we can derive the **direcivity index** *DI*, given by:

- DI = 10 log Q (dB)

Q usually depends on frequency, and often increases dramatically with it.



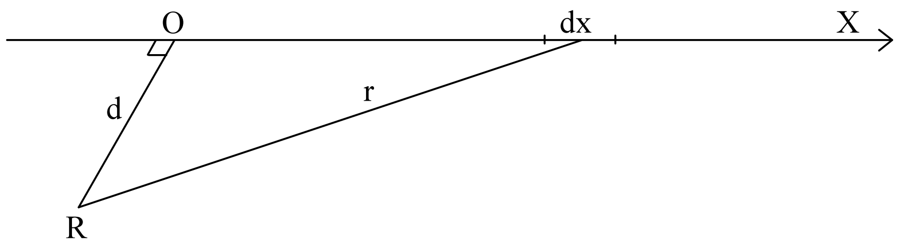
Free Field: directivity (2)



- Q = 1 ⇒ Omnidirectional point source
- Q = 2 ⇒ Point source over a reflecting plane
- Q = 4 ⇒ Point source in a corner
- Q = 8 ⇒ Point source in a vertex

Outdoor propagation – cylindrical field

Many noise sources found outdoors can be considered line sources: roads, railways, airtracks, etc.



Geometry for propagation from a line source to a receiver

- in this case the total power is dispersed over a cylindrical surface:

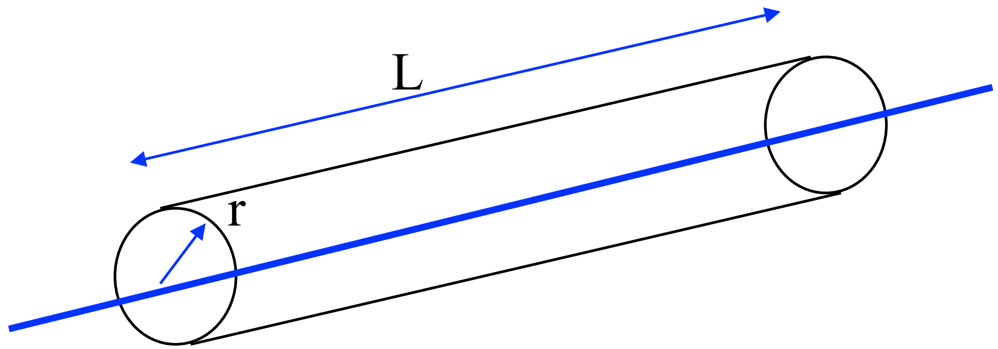
$$L_p = L_{W'} - 10\log d - 6 \qquad (incoherent \quad emission)$$

$$L_p = L_{W'} - 10\log d - 8 \qquad (coherent \quad emission)$$

In which Lw' is the sound power level **per meter of line source**

Coherent cylindrical field

- The power is dispersed over an infinitely long cylinder:



$$I = \frac{W}{S} = \frac{W}{2\cdot\pi\cdot r\cdot L}$$

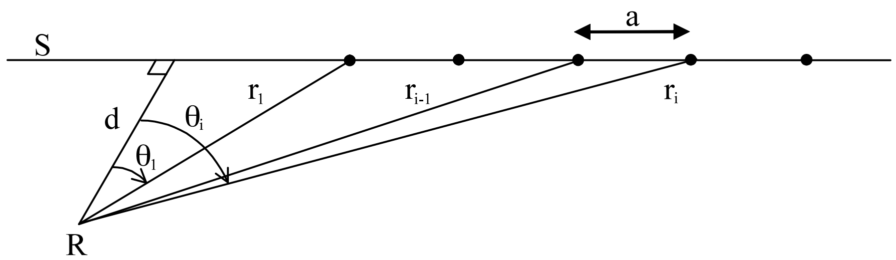
$$L_I = 10\cdot \lg\left[\frac{I}{I_o}\right] = 10\cdot \lg\left[\frac{\frac{W}{2\cdot\pi\cdot r\cdot L}}{I_o}\right] = 10\cdot \lg\left[\frac{\frac{W}{2\cdot\pi\cdot r\cdot L}}{I_o}\cdot\frac{W_o}{W_o}\right] = 10\cdot \lg\left[\frac{W}{L\cdot W_o}\right] - 10\cdot \lg[2\cdot\pi] - 10\cdot \lg[r]$$

$$L_I = L_W' - 8 - 10\cdot \lg[r]$$

In which Lw’ is the sound power level **per meter of line source**

“discrete” (and incoherent) linear source

Another common case is when a number of point sources are located along a line, each emitting sound mutually incoherent with the others:



Geomtery of propagation for a discrete line source and a receiver

- We could compute the SPL at teh received as the energetical (incoherent) summation of many sphericla wavefronts. But at the end the result shows that SPL decays with the same cylndrical law as with a coherent source:

$$L_p = L_{Wp} - 10\log(a) - 10\log(d) - 6 \quad [dB]$$

The SPL reduces by 3 dB for each doubling of distance d.

Note that the incoherent SPL is 2 dB **louder** than the coherent one!

Outdoors propagation – excess attenuation

Other factors causing **additional attenuation** during outdoors progation are:

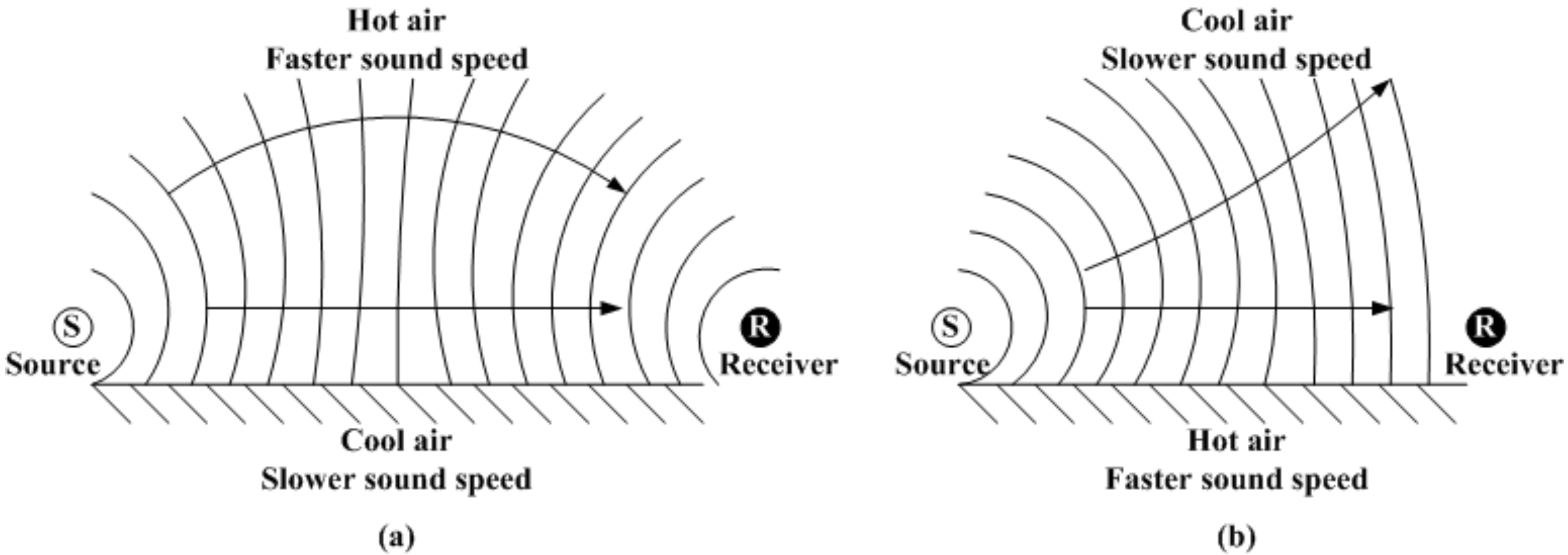
- air absorption
- absorption due to presence of vegetation, foliage, etc.
- metereological conditions (temperature gradients, wind speed gradients, rain, snow, fog, etc.)
- obstacles (hills, buildings, noise barriers, etc.)

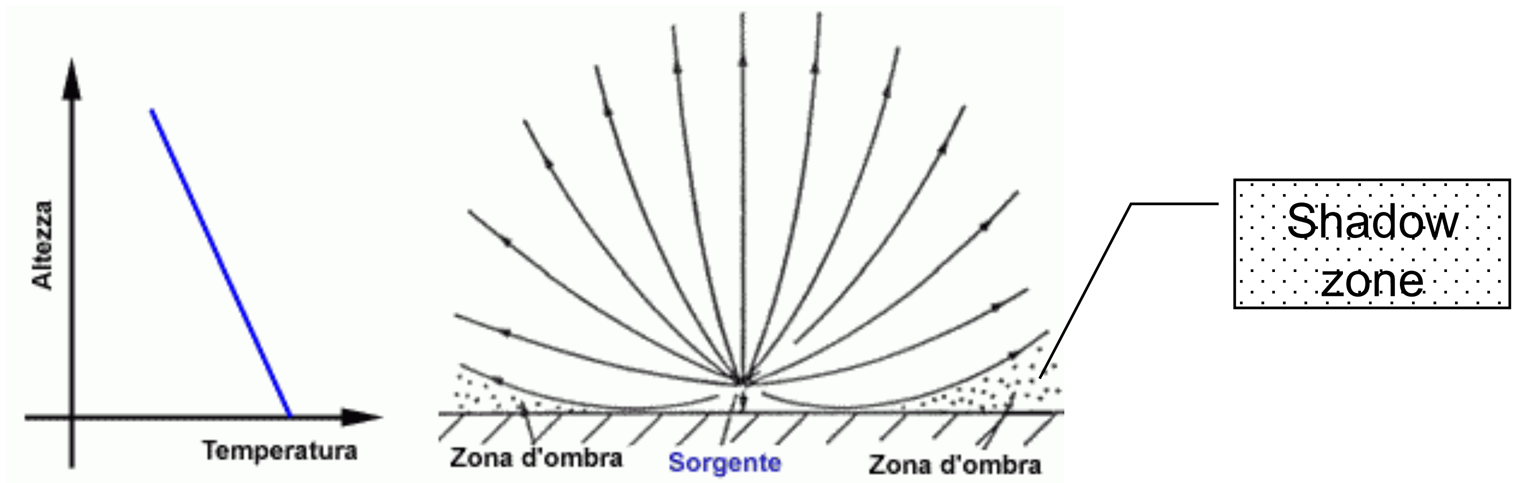
All these effects are combined into an additional term ΔL, in dB, which is appended to the free field formula:

$$L_I = L_p = L_W - 20 \log r - 11 + 10 \log Q - \mathbf{\Delta L} \qquad \text{(dB)}$$

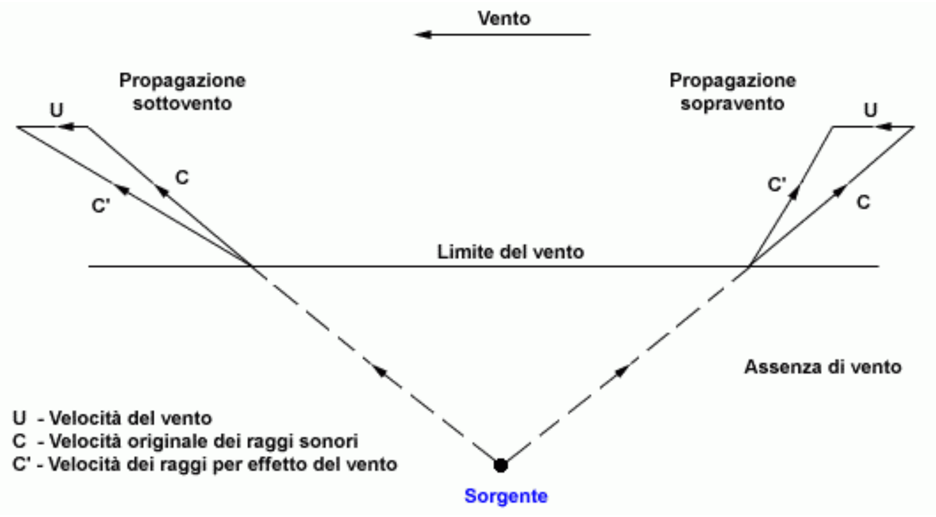
Most of these effects are relevant only at large distance form the source. The exception is shielding (screen effect), which instead is maximum when the receiver is very close to the screen

Excess attenuation: temperature gradient

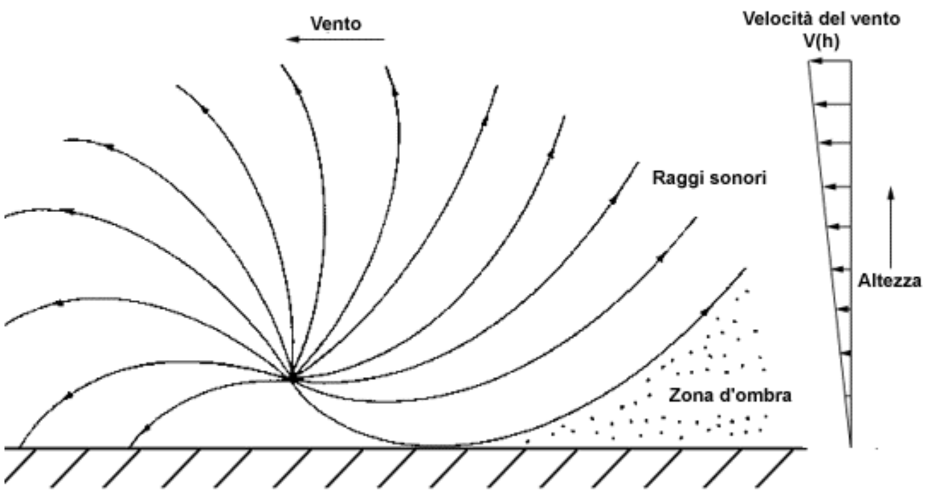




Excess attenuation: wind speed gradient



Vectorial composition of wind speed and sound speed



Effect: curvature of sound "rays"

Excess attenuation: air absorption

Air absorption coefficients in dB/km (from ISO 9613-1 standard) for different combinations of frequency, temperature and humidity:

		Frequency (octave bands)							
T (°C)	***RH (%)***	***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

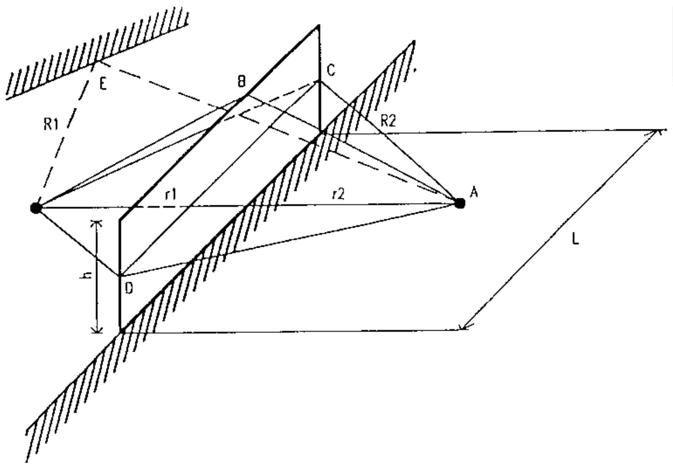
Excess attenuation – barriers

Noise screens (1)

A noise screen causes an **insertion loss** *ΔL*:

- $\Delta L = (L_0) - (L_b)$ (dB)

where L_b and L_0 are the values of the SPL with and without the screen.



In the most general case, there arey many paths for the sound to reach the receiver whne the barrier is installed:

- diffraction at upper and side edges of the screen (B,C,D),

- passing through the screen (SA),

- reflection over other surfaces present in proximity (building, etc. - SEA).

Noise screens (2): the MAEKAWA formulas

If we only consider the enrgy diffracted by the upper edge of an infinitely long barrier we can estimate the insertion loss as:

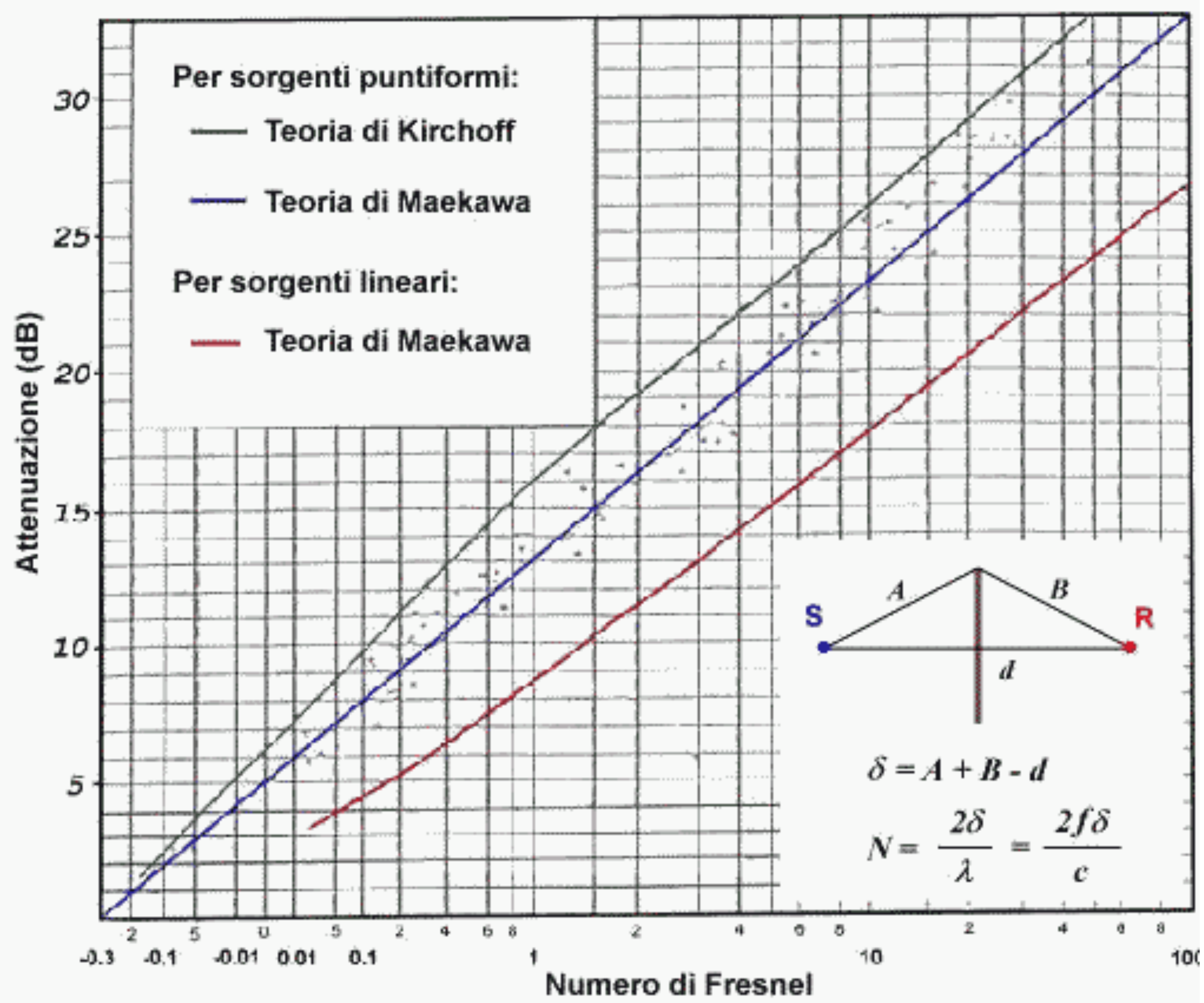
- **ΔL** = 10 log (3+20 N) for N>0 (point source)
- **ΔL** = 10 log (2+5.5 N) for N>0 (linear source)

where N is *Fresnel number* defined by:

- $N = 2\ \delta/\lambda = 2\ (SB + BA - SA)/\lambda$

in which λ is the *wavelength* and δ is the *path difference* among the diffracted and the direct sound.

Maekawa chart



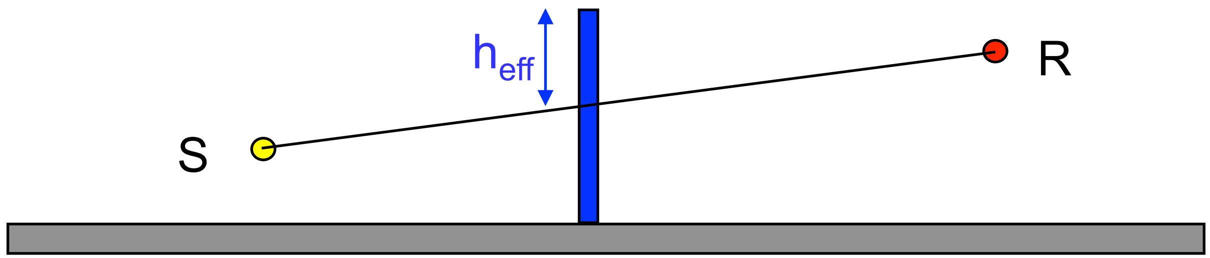
Noise screens (3): finite length

If the barrier is not infinte, we need also to soncider its lateral edges, each withir Fresnel numbers (N_1, N_2), and we have:

- $\mathbf{\Delta L} = \Delta L_d - 10 \log (1 + N/N_1 + N/N_2)$ (dB)

Valid for values of N, N_1, $N_2 > 1$.

The lateral diffraction is only sensible when the side edge is closer to the source-receiver path than 5 times the “effective height”.



Analysis:

The insertion loss value depends strongly from frequency:

- low frequency ⇒ small sound attenuation.

The spectrum of the sound source must be known for assessing the insertion loss value at each frequency, and then recombining the values at all the frequencies for recomputing the A-weighted SPL.