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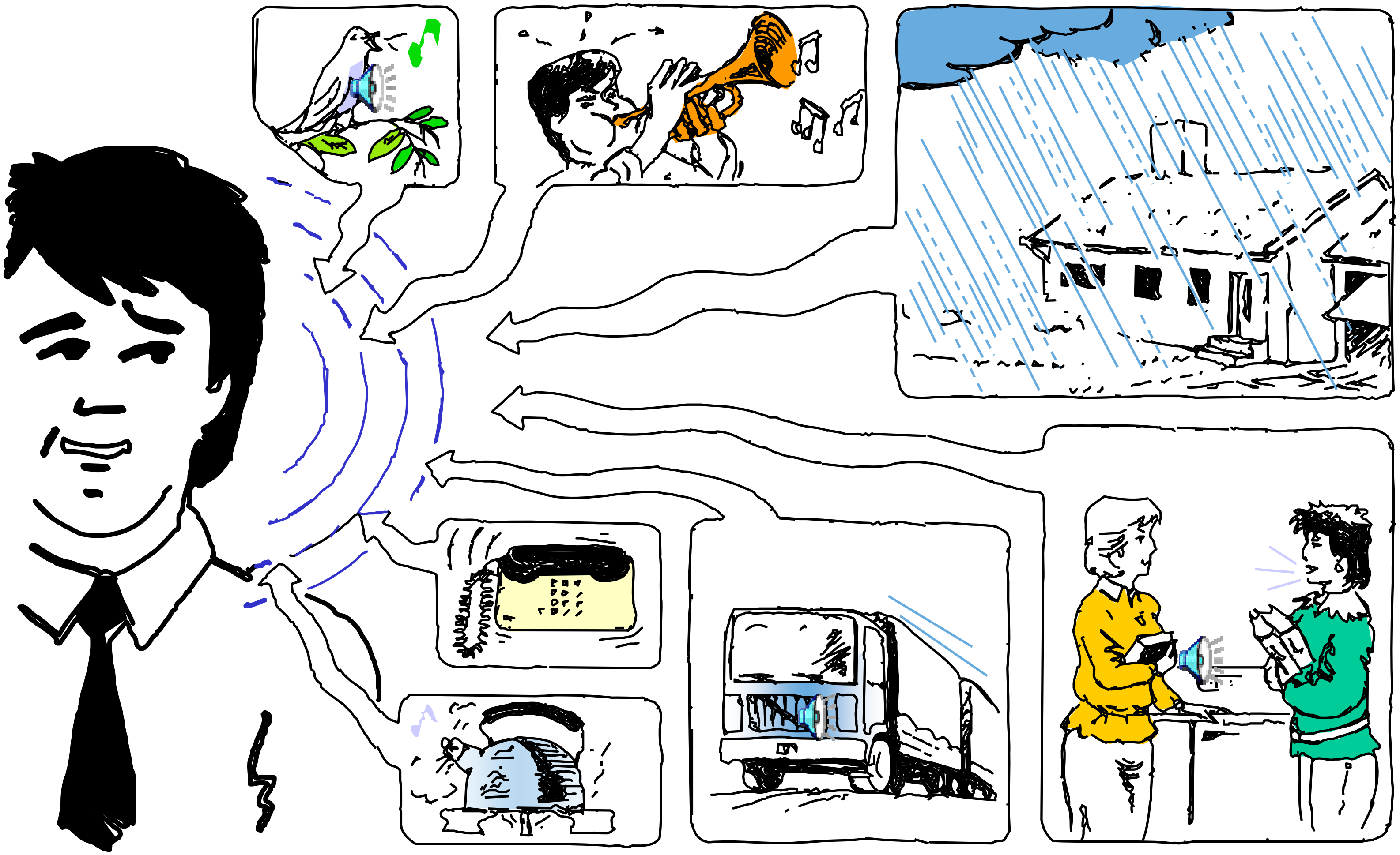
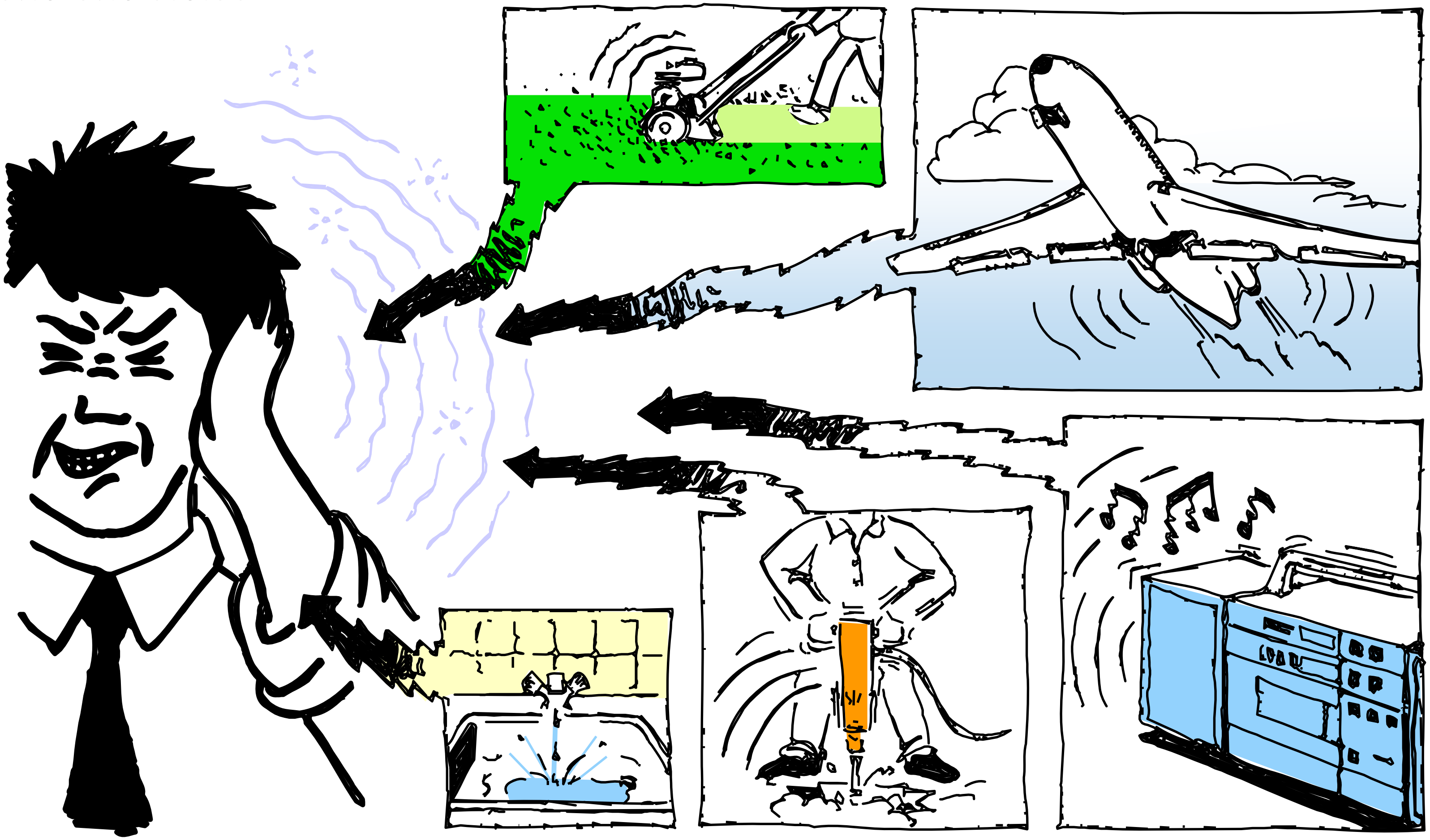
- Sound



Sound is such a common part of everyday life that we rarely appreciate all of its functions.

1.1 Basic Concepts of Sound

- Sound and Noise



Yet, too often in our modern society, sound annoys us. Many sounds are unpleasant or unwanted – these are called noise. However, the level of annoyance depends not only on the quality of the sound, but also our attitude towards it.

- Terminology of Sound

RMS
Peak

Fast
Slow
Impulse

Percentile level

Sound Pressure

dB

Logarithmic scales

Weighting

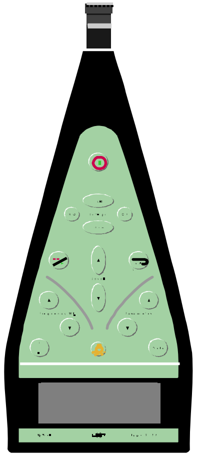
L_{eq}

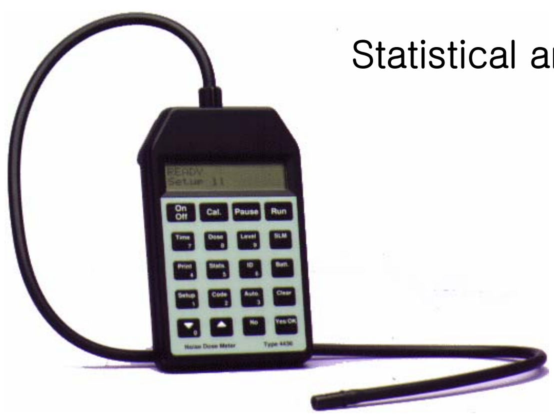
Pascal

L_{10}

RMS

L_{90}





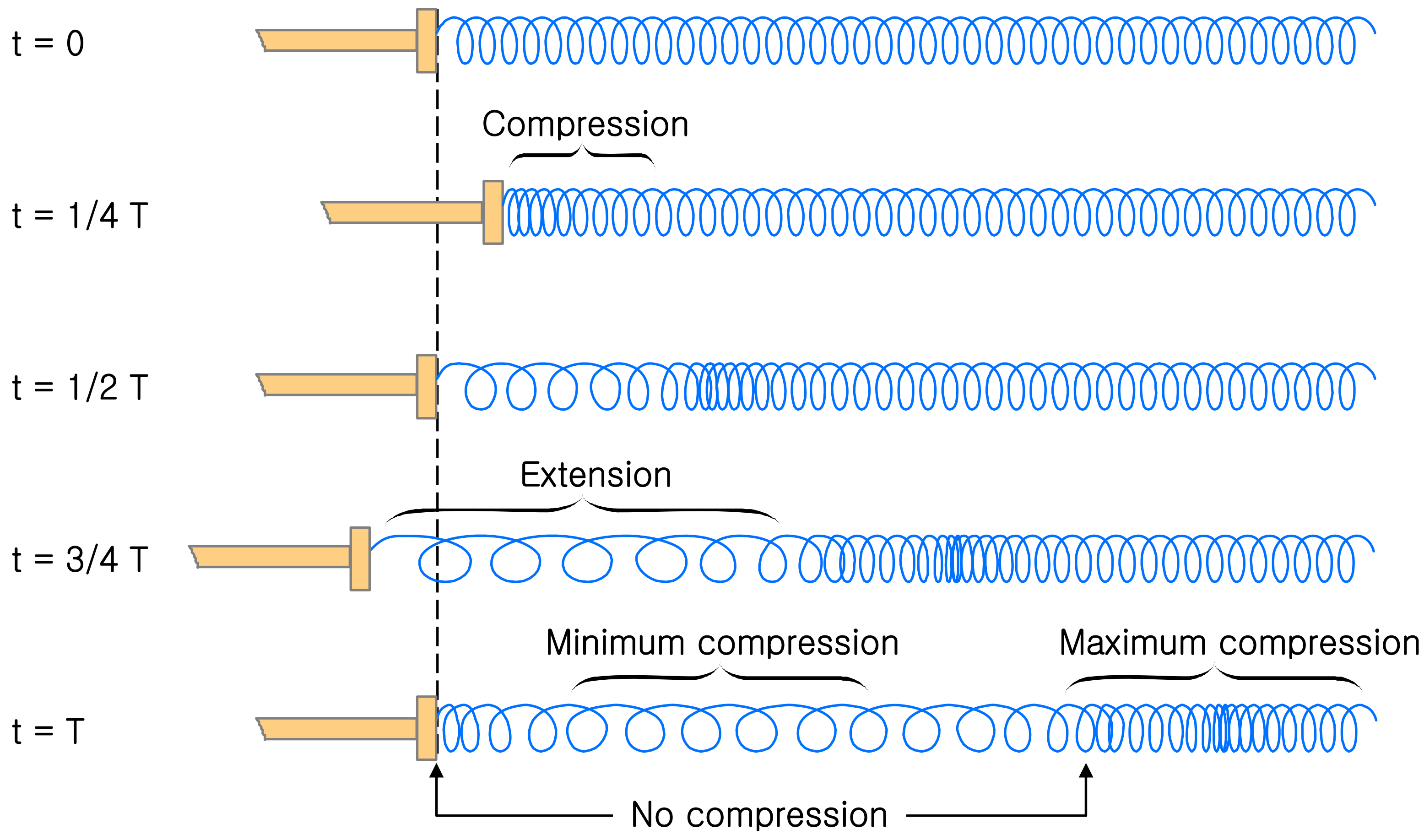
Statistical analysis

1/1 and 1/3 Octave Analysis

Constant percentage bandwidth

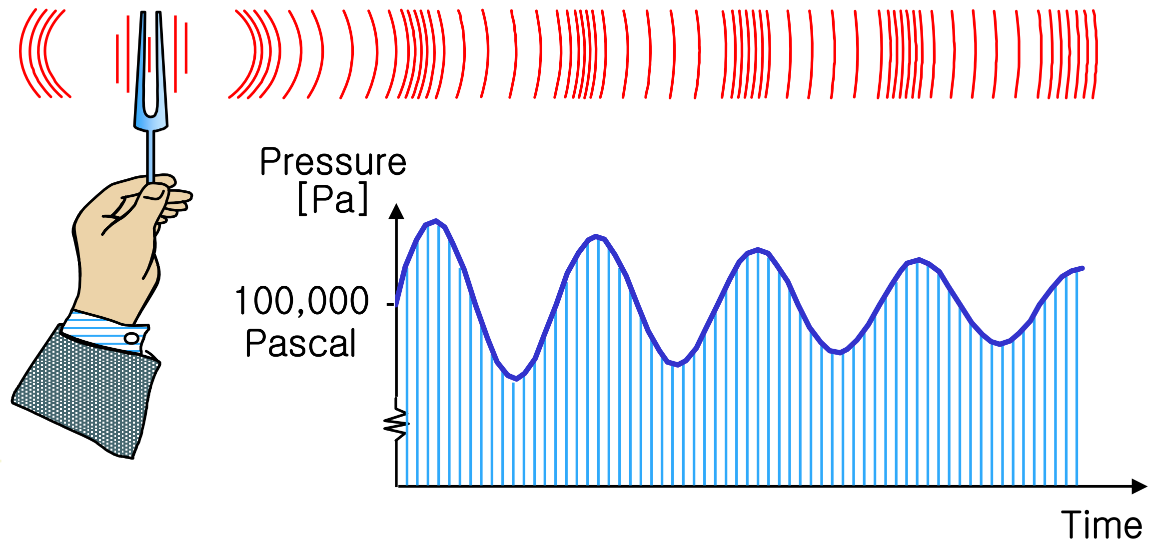
Noise Dose

- Propagation of Sound

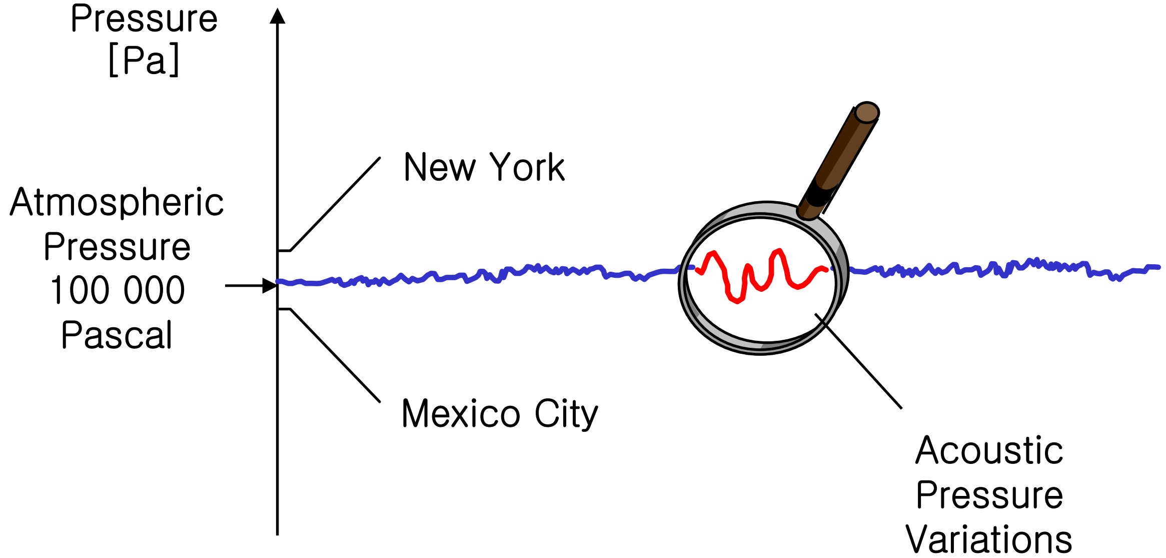


The same happens when air molecules are compressed and extended;
the 'compression' and 'extension' or changes of pressure travel or radiate in air.

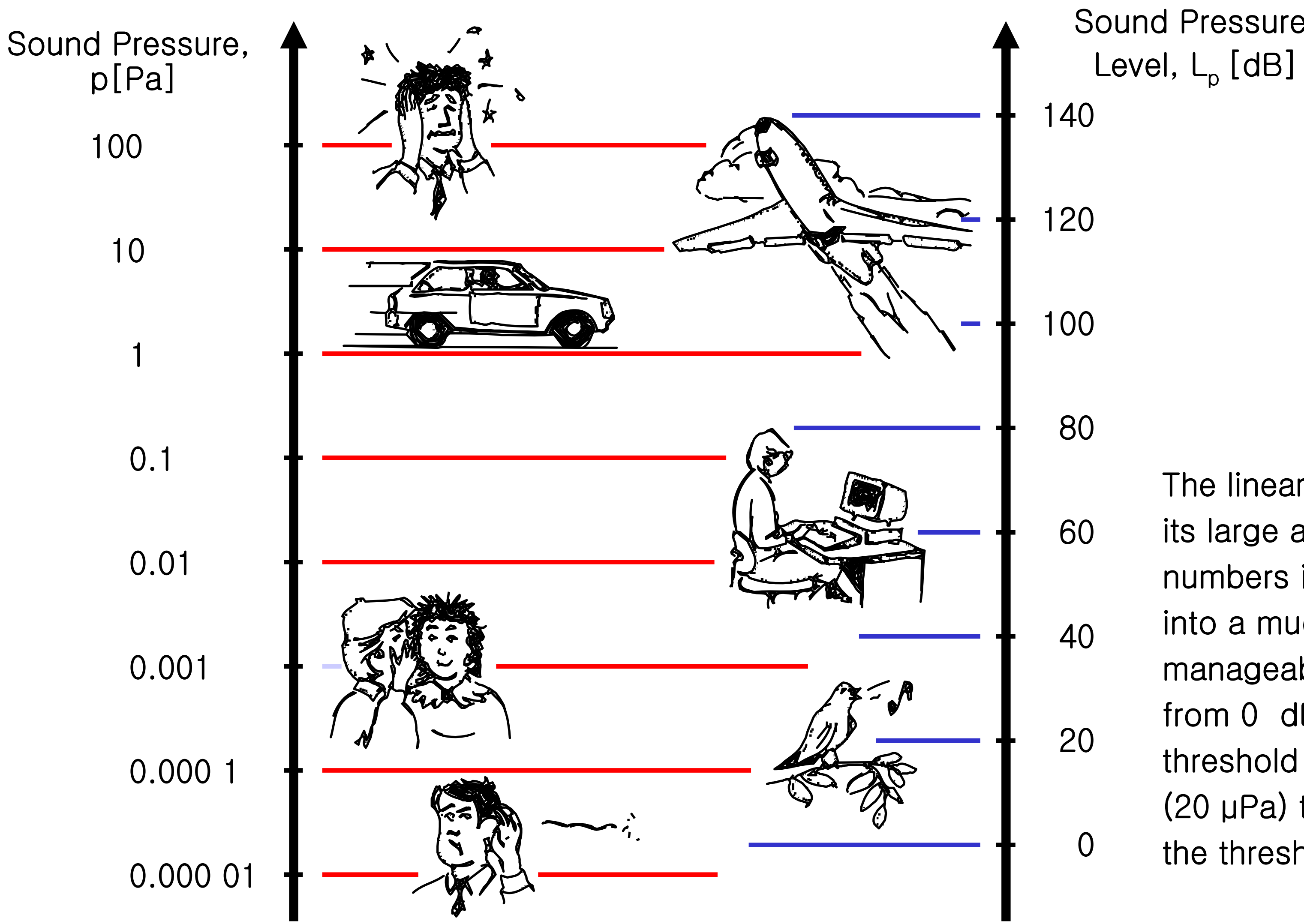
- Sound Pressure



The acoustic pressure vibrations are superimposed on the surrounding static air pressure which has a value of 10^5 Pascal.



- Range of Sound Pressure Level



The linear scale with its large and unwieldy numbers is converted into a much more manageable scale from 0 dB at the threshold of hearing (20 μPa) to 130 dB at the threshold of pain.

- dB – decibel

$$L_p = 20 \log \frac{p}{p_0} \text{ dB}$$

$$(p_0 = 2 \times 10^{-5} \text{ Pa})$$

Ex.1: $p = 1$ Pa

$$L_p = 20 \log \frac{1}{20 \times 10^{-6}}$$

$$= 20 \log 50\,000$$

$$= 94 \text{ dB}$$

Ex.2: $p = 31.7$ Pa

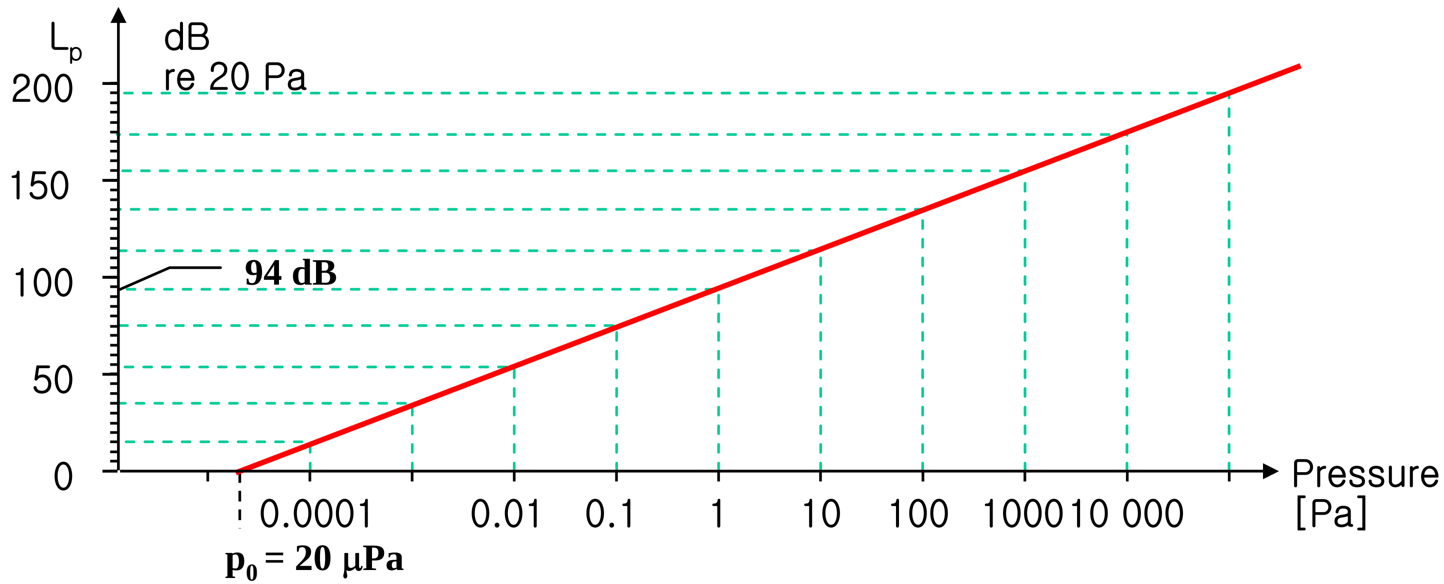
$$L_p = 20 \log \frac{317}{20 \times 10^{-6}}$$

$$= 20 \log 1.58 \times 10^{-6}$$

$$= 124 \text{ dB}$$

- dB – decibel

Change in Sound Level (dB)	Change in Perceived Loudness
3	Just perceptible
5	Noticeable difference
10	Twice (or 1/2) as loud
15	Large change
20	Four times (or 1/4) as loud



- ACOUSTICS

  - Generation
  - Propagation
  - Detection
  - Effects

  } of compressional waves in gases, liquids and solids

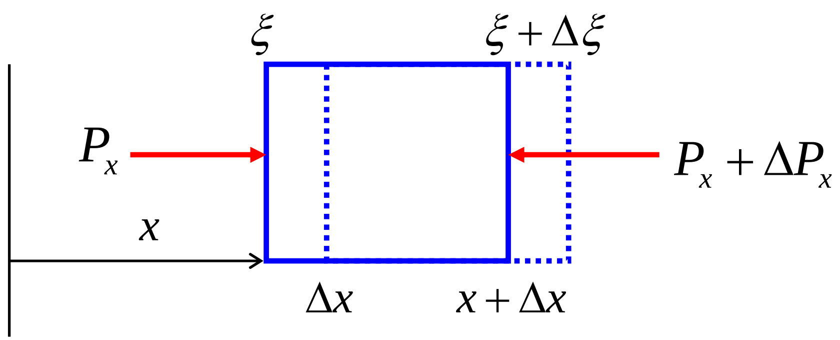
- Sound is a wave and a disturbance which propagates

- Sound is a disturbance in:

  - Pressure
  - Density
  - Velocity (longitudinal)
  - Temperature

- Wave equation (1-D plane wave)

  - Infinitesimal air element with an unit area
  - ξ : Displacement of air particle



By Newton's 2nd law

$$\{P_x - (P_x + \Delta P_x)\}A = \rho A \Delta x \frac{\partial^2 \xi}{\partial t^2} \Rightarrow -\Delta P_x = \rho \Delta x \frac{\partial^2 \xi}{\partial t^2}$$

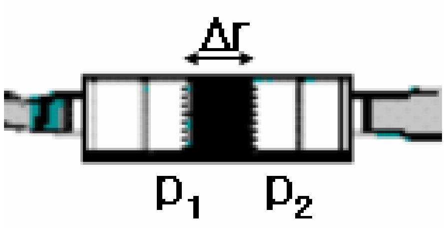
$$\frac{\partial^2 \xi}{\partial t^2} = -\frac{1}{\rho}\frac{\partial p}{\partial x}$$

$$\frac{\partial^2 \xi}{\partial t^2} = -\frac{1}{\rho}\frac{\partial p}{\partial x}$$

$$v = \frac{\partial \xi}{\partial t}$$ (the velocity of particle)

$$\Rightarrow \quad \frac{\partial v}{\partial t} = -\frac{1}{\rho}\frac{\partial p}{\partial x}$$

When measuring a sound intensity, this equation is used.



$$v = -\frac{1}{\rho}\int \frac{\partial p}{\partial x} dt$$

$$v = \frac{1}{\rho}\int \frac{p_1 - p_2}{\Delta r} dt$$

Using Bulk modulus : K

$$p = -K\frac{dV}{V} \qquad \frac{dV}{V} = \frac{\partial \xi}{\partial x} \quad \Rightarrow \quad p = -K\frac{\partial \xi}{\partial x}$$

$$\frac{\partial^2 \xi}{\partial t^2} = -\frac{1}{\rho}\frac{\partial p}{\partial x} \quad \Rightarrow \quad \frac{\partial^2 \xi}{\partial t^2} = \frac{K}{\rho}\frac{\partial^2 \xi}{\partial x^2}$$

: Wave equation

Meanwhile, the velocity of particle is $v = \frac{\partial \xi}{\partial t} = \frac{\partial \xi}{\partial x}\frac{\partial x}{\partial t}$

Wave speed is $c = \frac{\partial x}{\partial t}$ $\Rightarrow$ $v = \frac{\partial \xi}{\partial t} = c\frac{\partial \xi}{\partial x}$

$\Rightarrow$ Differentiate both sides by time variable, t

$$\frac{\partial^2 \xi}{\partial t^2} = c^2\frac{\partial^2 \xi}{\partial x^2}$$

: Wave equation

From two wave equations, the wave speed becomes as the follows.

$$c = \sqrt{\frac{K}{\rho}}$$

The relationship between sound pressure and particle velocity

$$p = -K\frac{\partial \xi}{\partial x} = -K\frac{\partial \xi}{\partial t}\frac{\partial t}{\partial x} = -\frac{K}{c}\frac{\partial \xi}{\partial t}$$

$$K = \rho c^2$$

$$\Rightarrow \quad p = -\frac{K}{c}\frac{\partial \xi}{\partial t} = -\rho c v$$

<u>Sound intensity</u>

$$I = pv = \frac{p^2}{\rho c}$$

$$L_p = 20\log\frac{p}{p_o} = 10\log\frac{p^2/\rho c}{p_o^2/\rho c} = 10\log\frac{I}{I_o}$$

Standard state : 1 atm, 15 ℃

$$\rho c = 415 kg/m^2 s \qquad I_o = \frac{(2\times10^{-5})^2}{415} \approx 1\times10^{-12}\ w/m^2$$

$$L_w = 10\log\frac{w}{w_o} = 10\log\frac{IS}{1\times10^{-12}} = 10\log\frac{I}{1\times10^{-12}} + 10\log S$$

$$L_w = L_p + 10\log S$$

$$\frac{\partial^2 \xi}{\partial t^2} = c^2 \frac{\partial^2 \xi}{\partial x^2} \qquad \text{and} \qquad p = -K \frac{\partial \xi}{\partial x}$$

Differentiate both sides by space variable, x

$$\frac{\partial^2}{\partial t^2}\left(\frac{\partial \xi}{\partial x}\right) = c^2 \frac{\partial^2}{\partial x^2}\left(\frac{\partial \xi}{\partial x}\right) \Rightarrow \frac{\partial^2}{\partial t^2}\left(\frac{\partial \xi}{\partial x}\right) = c^2 \frac{\partial^2}{\partial x^2}\left(\frac{\partial \xi}{\partial x}\right)$$

$$\Rightarrow \frac{\partial^2 p}{\partial t^2} = c^2 \frac{\partial^2 p}{\partial x^2}$$

: Wave equation

Wave Equation $$\frac{1}{c^2}\frac{\partial^2 p}{\partial t^2}=\frac{\partial^2 p}{\partial x^2}$$

Solution $$p(x,t)=f(x-ct)$$

Proof:

$$\frac{\partial p(x,t)}{\partial t}=-cf'(x-ct) \qquad \frac{\partial p(x,t)}{\partial x}=f'(x-ct)$$

$$\frac{\partial^2 p(x,t)}{\partial t^2}=c^2 f''(x-ct) \qquad \frac{\partial^2 p(x,t)}{\partial x^2}=f''(x-ct)$$

so $$\frac{1}{c^2}\frac{\partial^2 p_A}{\partial t^2}=f''(x-ct)=\frac{\partial^2 p_A}{\partial x^2}$$

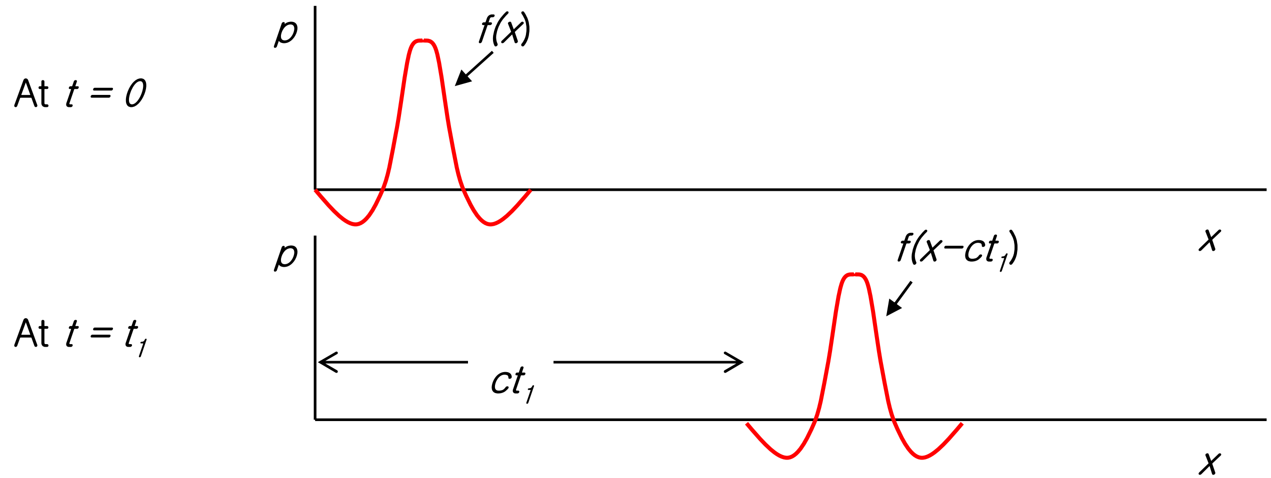
Wave Equation $$\frac{1}{c^2}\frac{\partial^2 p}{\partial t^2}=\frac{\partial^2 p}{\partial x^2}$$

Solution $$p(x,t)=f(x-ct)$$

- f can be any function
- u_{Ax}, ρ_A and T_A also satisfy the wave equation
- $p(x,t)=g(x+ct)$ is also a solution

Why is this a propagating wave?

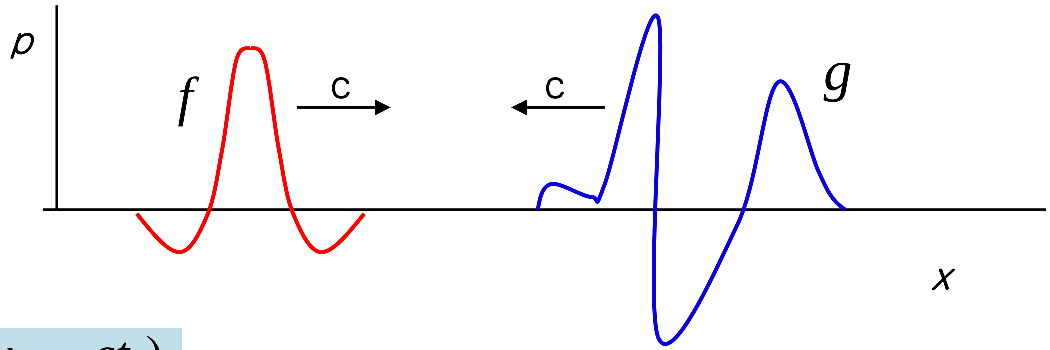
$p(x,t)=f(x-ct)$



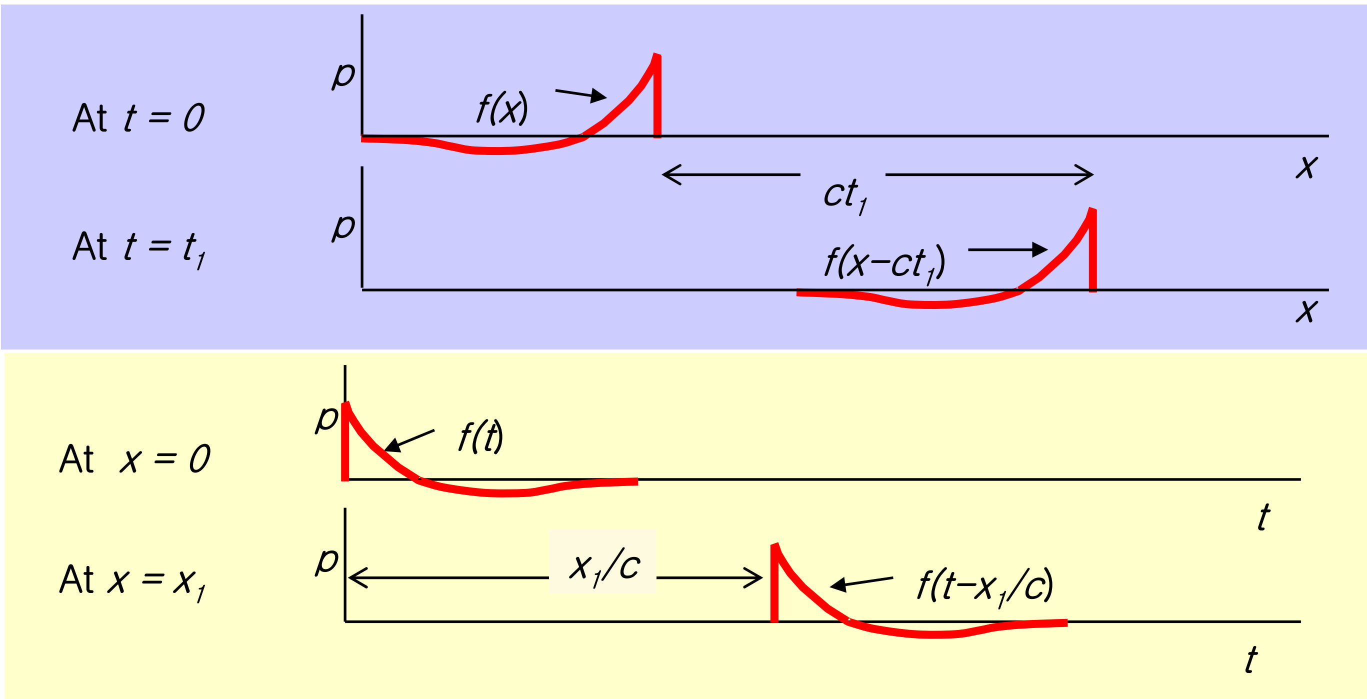
$p(x,t)=g(x+ct)$ propagates in the = $-x$ direction

Most General Solution

$$p(x,t) = f(x-ct) + g(x+ct)$$

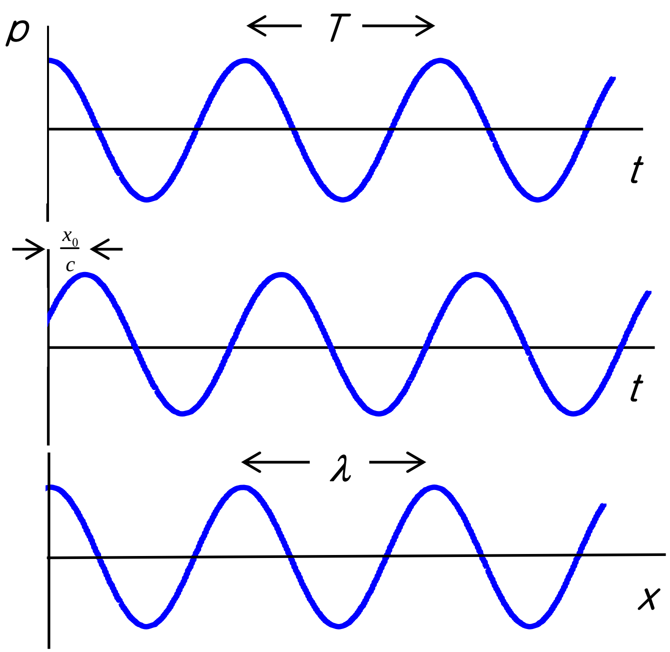


Propagating Wave: $p(x,t) = f(x-ct)$



Sinusoidal Waves

$$e.g.\quad f(\eta) = A\cos\left(\frac{2\pi\eta}{T}\right) \Rightarrow p(x,t) = A\cos\left(\frac{2\pi}{T}\left(t-\frac{x}{c}\right)\right) = A\cos\left(2\pi f\left(t-\frac{x}{c}\right)\right)$$



$x=0$ $$p(0,t) = A\cos\left(\frac{2\pi t}{T}\right)$$

$x=x_0$

T period (s)

$f = 1/T$ frequency (Hz) (# cycles/sec)

$t=0$ $$p(x,0) = A\cos\left(\frac{2\pi x}{cT}\right)$$

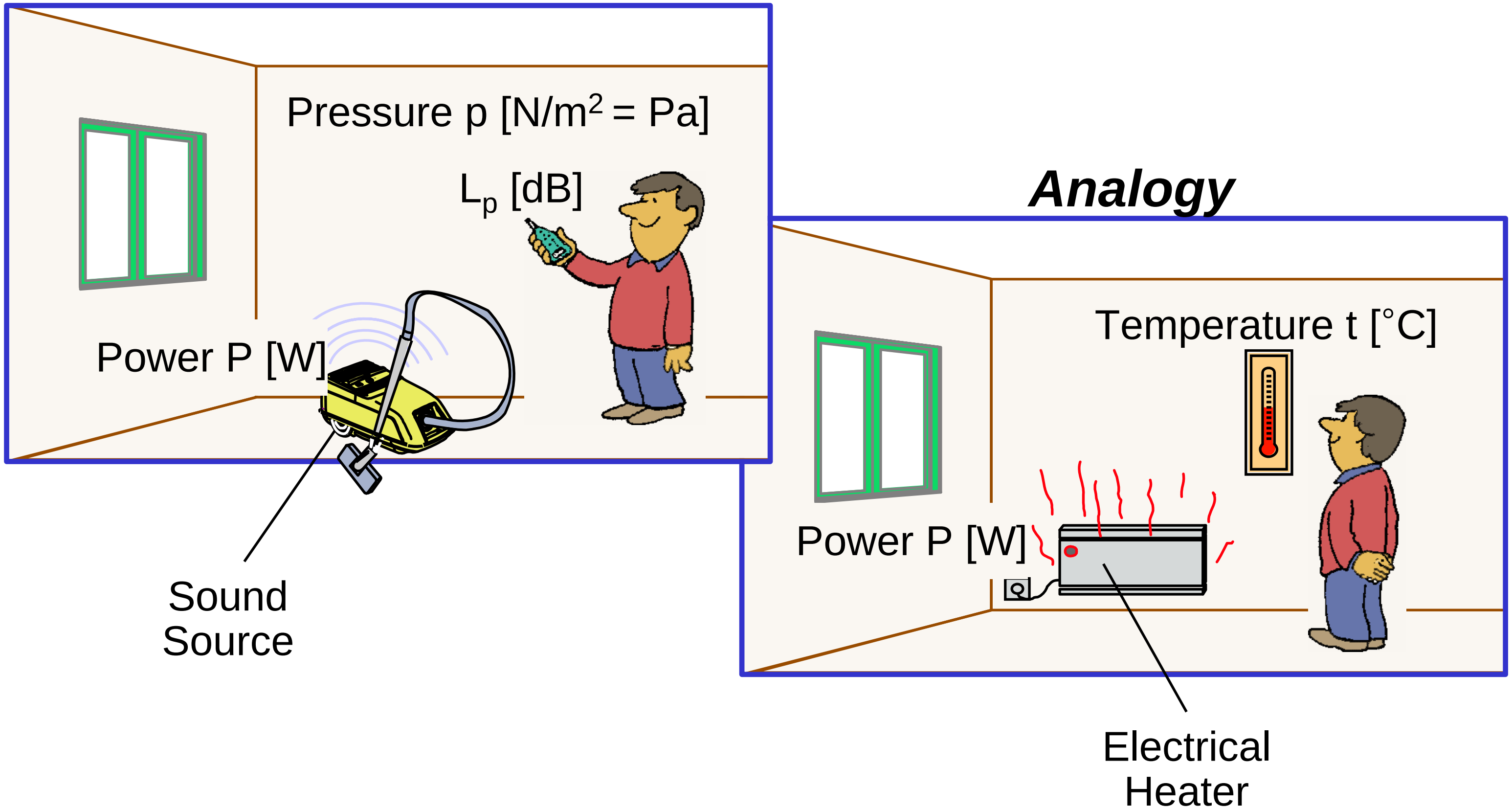
λ is the wavelength (spatial period) $\lambda = cT$ λ distance propagated in one period

$$p_A(x,t) = A\cos(\omega t - kx)$$

$\omega = 2\pi f$ temporal angular frequency

$k = \omega / c$ wavenumber (spatial angular freq)

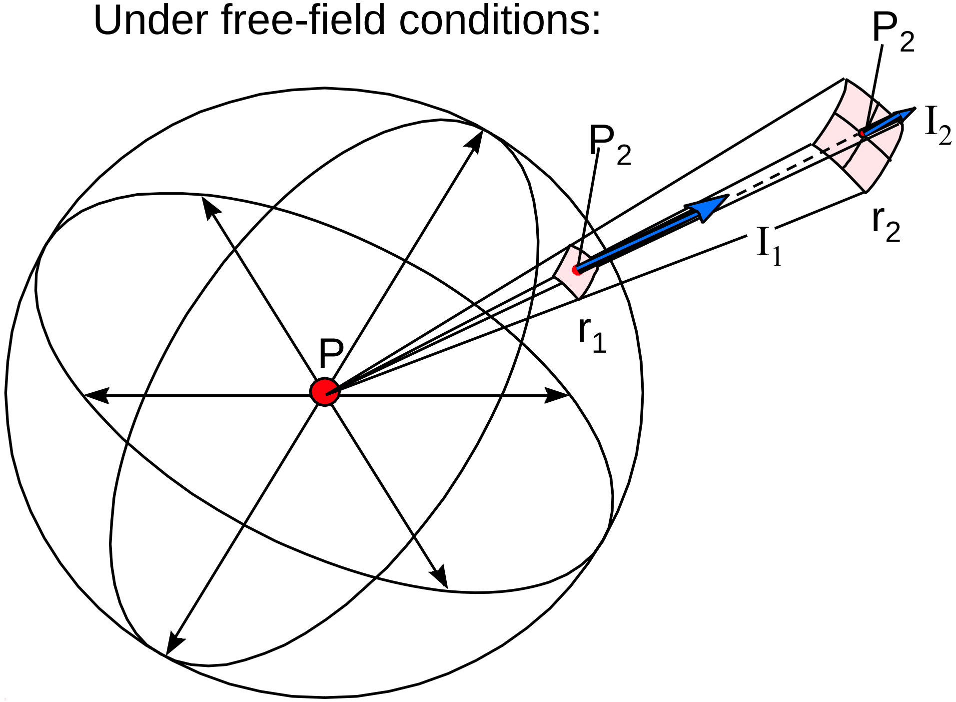
- Pressure vs. Power



A sound source will produce a certain amount of sound energy per unit time [Joule/sec], i.e. it has a certain sound power rating in W [Watt = Joule/sec].
This is a basic measure of how much acoustical energy it can produce, and is independent of its surroundings.

- Basic Parameters of Sound

Under free-field conditions:



The Sound Intensity vector, $\vec{I}$ describes the amount and direction of flow of acoustic energy at a given position

$$I = \frac{P}{4\pi r^2} = \frac{p^2}{\rho c}$$

Power:	P [W]
Intensity:	I [J/s/m^2] = W/m^2
Pressure:	p [Pa = N/m^2]

The energy passing a particular point in the area around the source will give rise to a sound pressure, p, at that point.

r is the density of air, c is the speed of sound.

1.3 Sound Field

- Reference Level

Sound Pressure

$L_P=10\log_{10}p^2/p_0^2$

$P_0=2*10^{-5}N/m^2$

Sound Intensity

$L_1=10\log_{10}I/I_0$

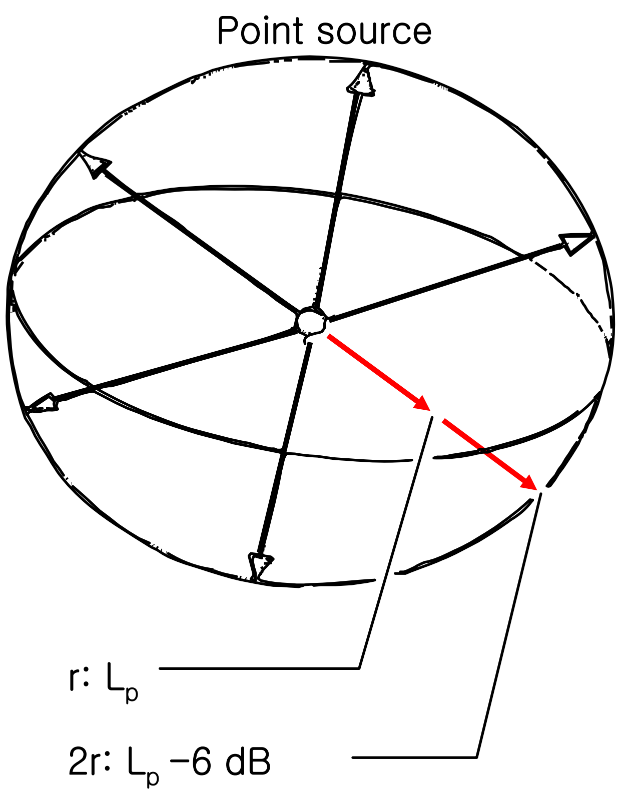
$I_0=1p\ W/m^2$

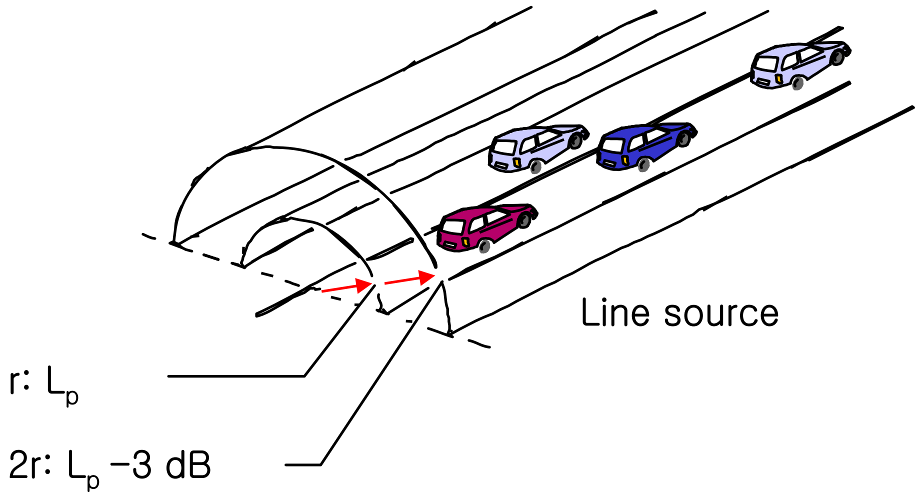
Sound Power

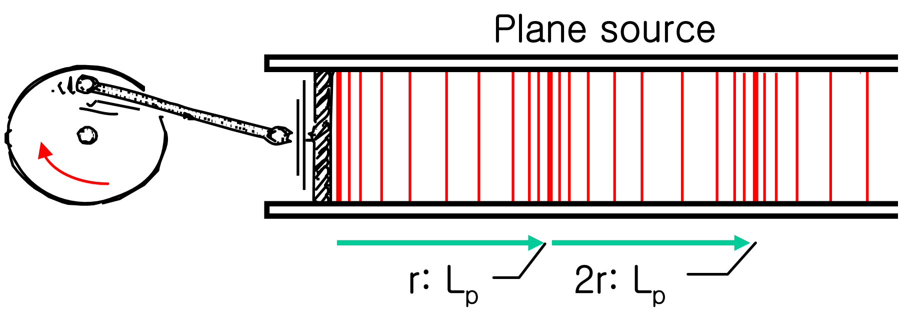
$L_W=10\log_{10}W/W_0$

$W_0=1pW$

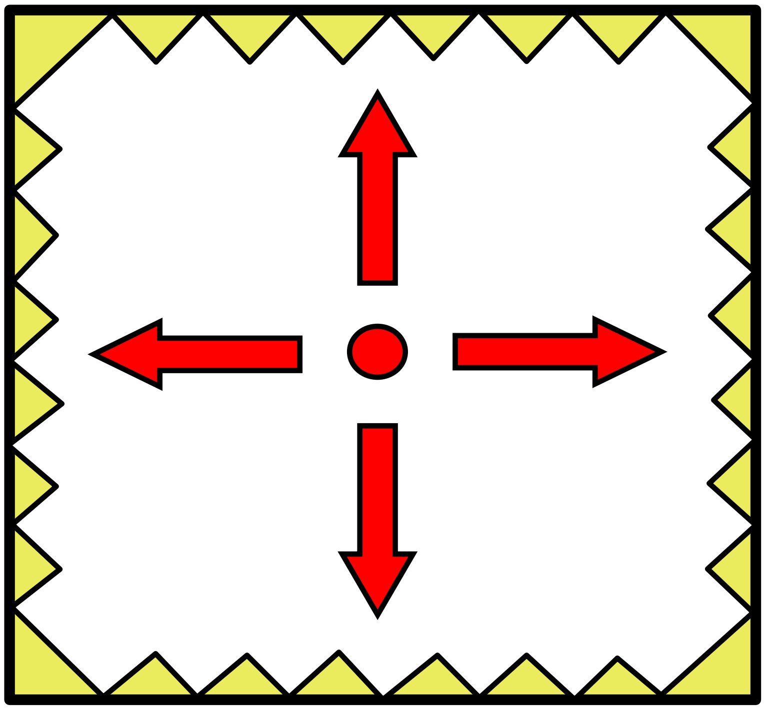
- Types of Sound Sources

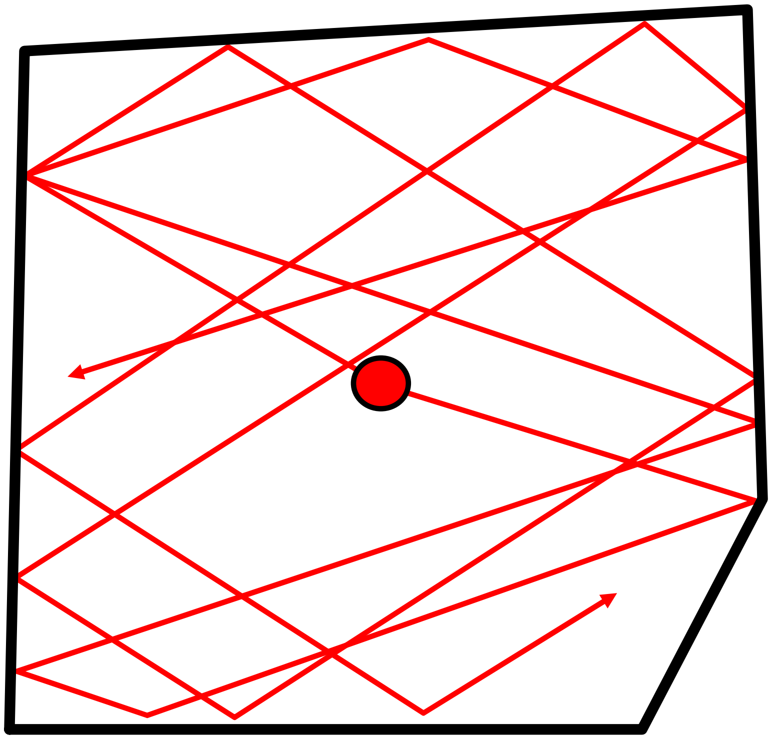






- Anechoic and Reverberant Enclosures

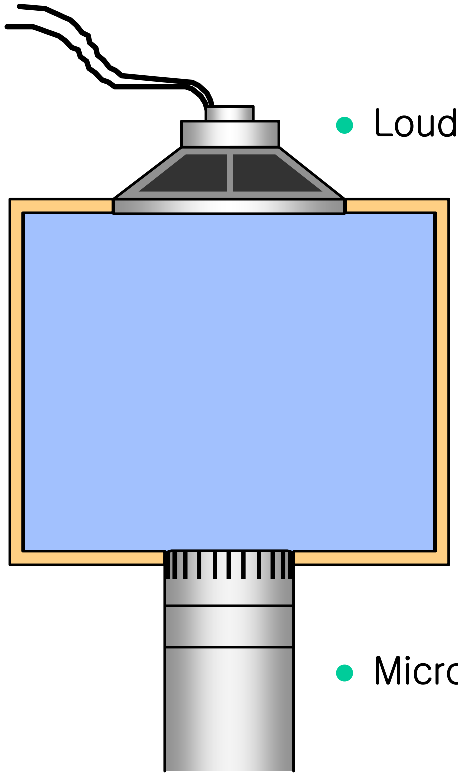




In a room with highly absorbent surfaces all the energy will be absorbed by the surfaces and the noise energy in the room will spread away from the source as if the source was in a free field. Such a room is called an anechoic room.

In a room with hard reflecting surfaces, all the energy will be reflected and a so-called diffuse field with sound energy uniformly distributed throughout the room is set up. Such a room is called a reverberation room.

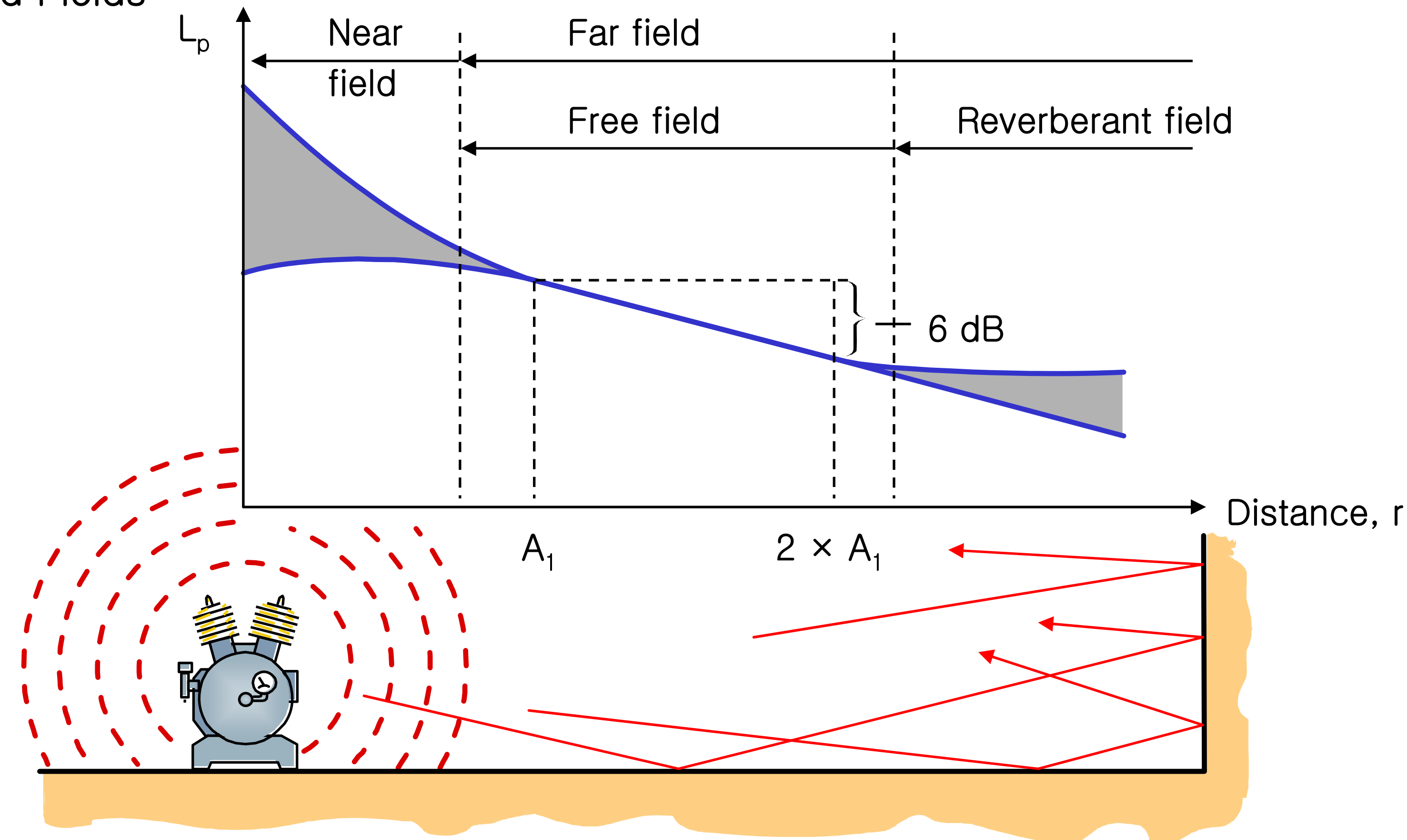
- Pressure Field



In a pressure field where the wavelength is long compared to the dimensions of the enclosure, the pressure is uniform in the enclosure.

This is used in calibrators where an exact sound pressure is applied to an enclosure.

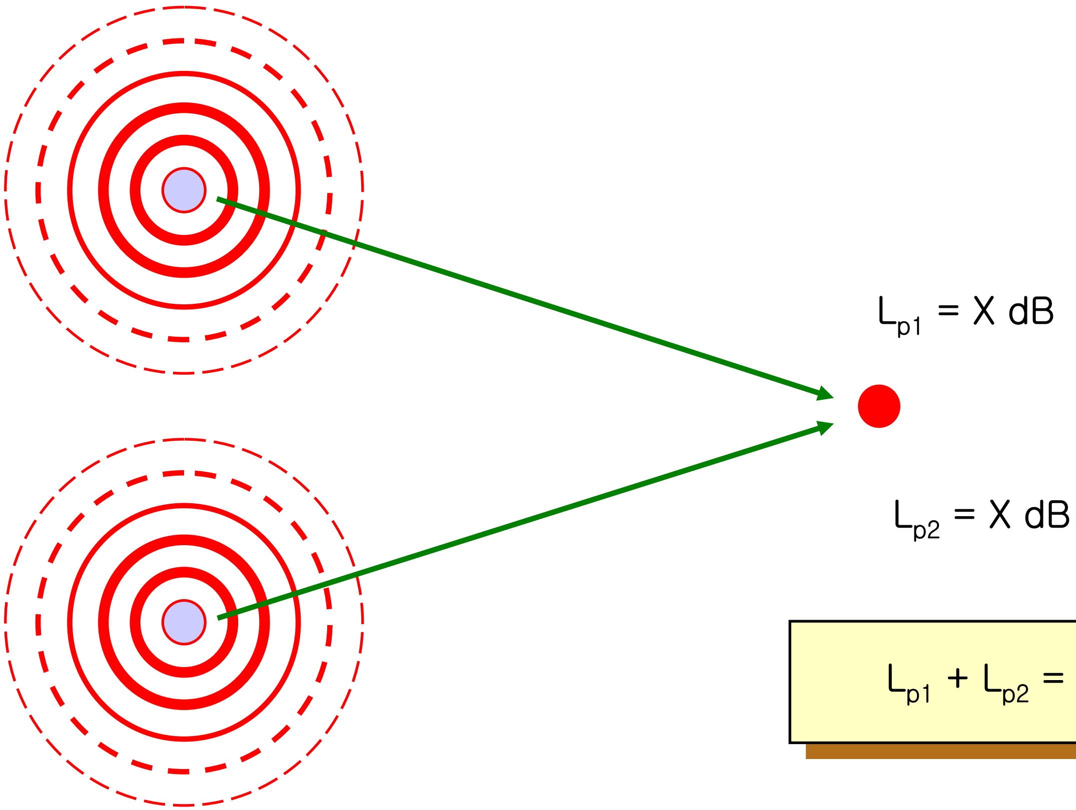
- Sound Fields



The near field is the area very close to the machine where the sound pressure level may vary significantly with a small change in position.

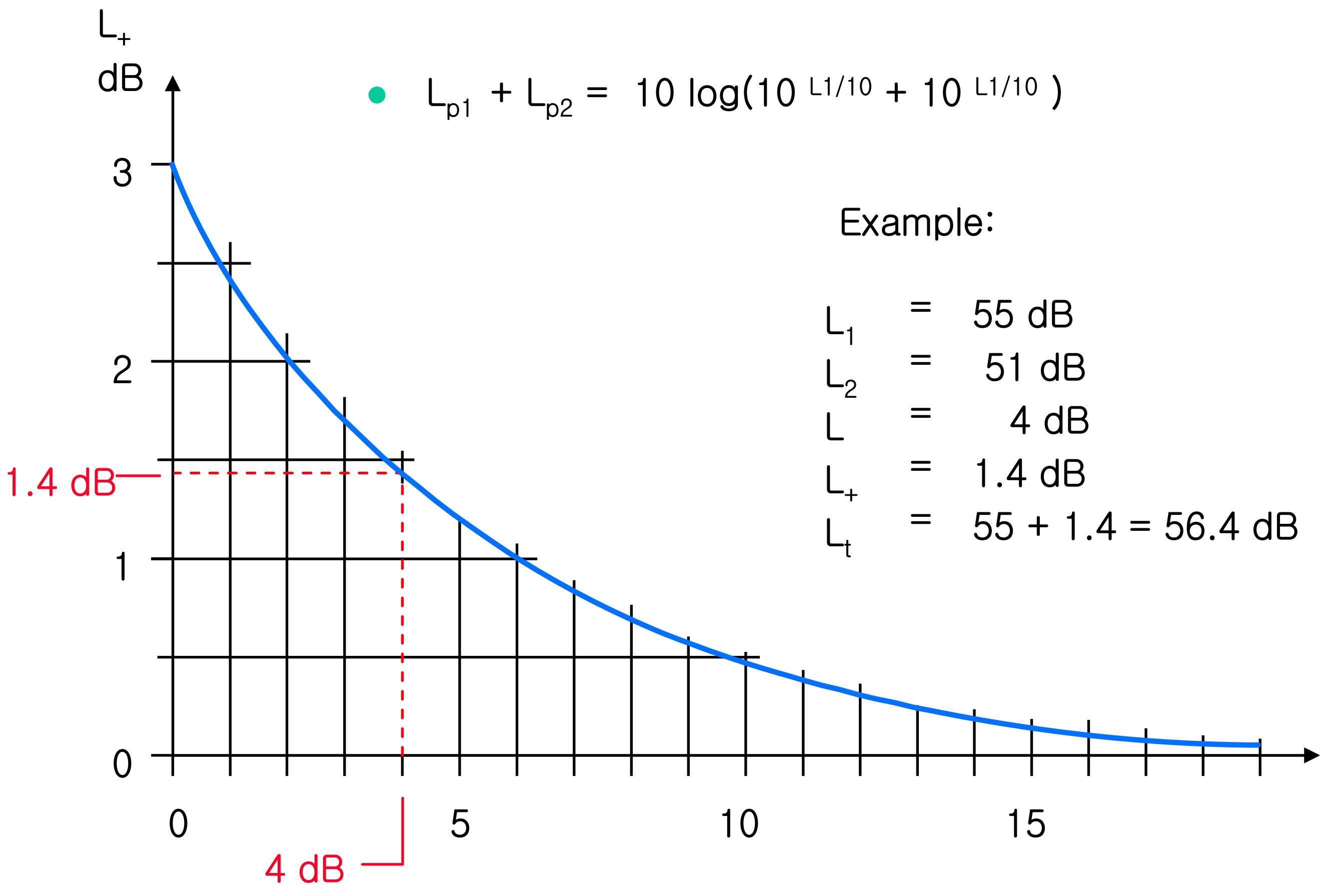
The area extends to a distance less than the wavelength of the lowest frequency emitted from the machine, or at less than twice the greatest dimension of the machine, whichever distance is the greater. Sound pressure measurements in this region should be avoided.

- Two Sound Source



$$L_{p1} + L_{p2} = X + 3\ \text{dB}$$

- Addition of dB Levels

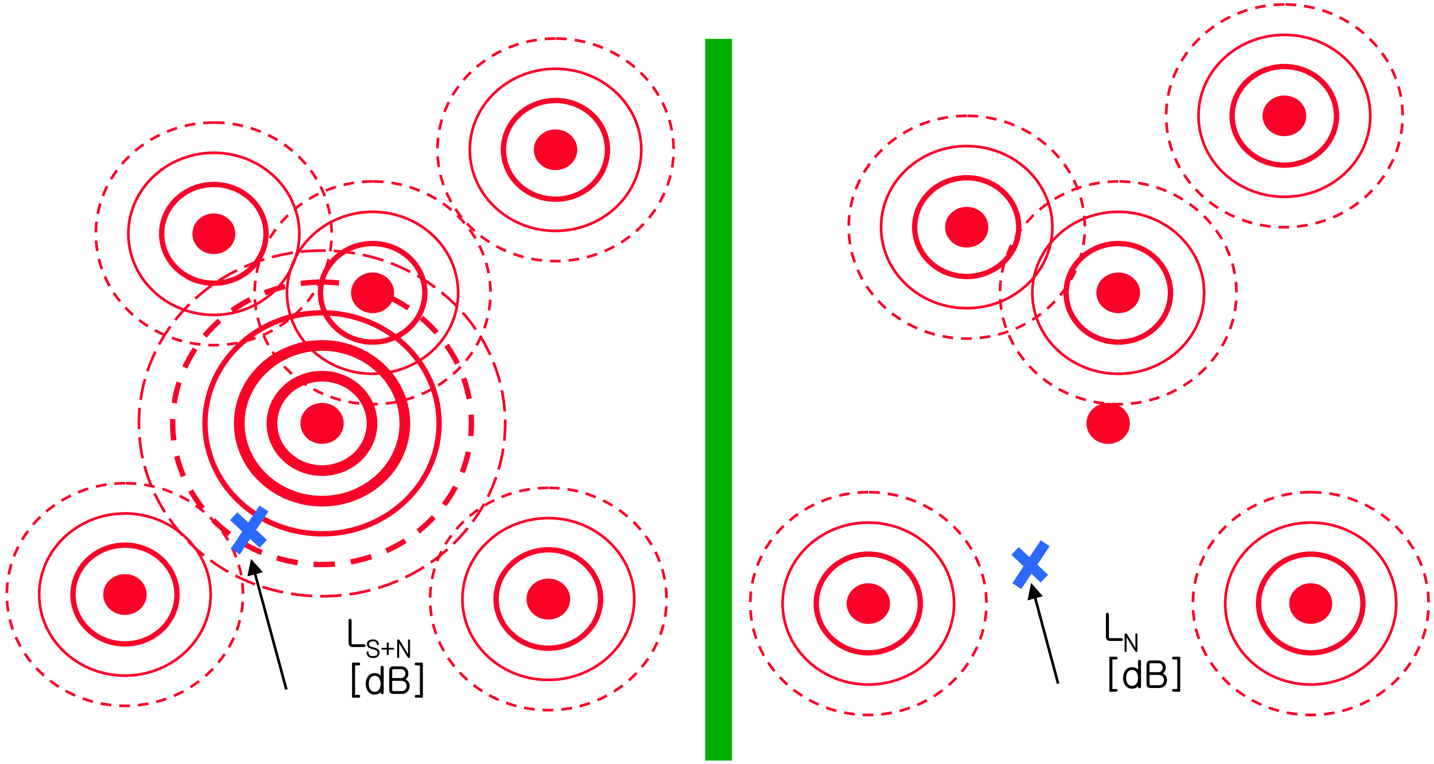


- $L_{p1} + L_{p2} = 10 \log(10^{L1/10} + 10^{L1/10})$

Example:

L_1 = 55 dB
L_2 = 51 dB
L = 4 dB
L_+ = 1.4 dB
L_t = 55 + 1.4 = 56.4 dB

- Subtraction of Noise Levels



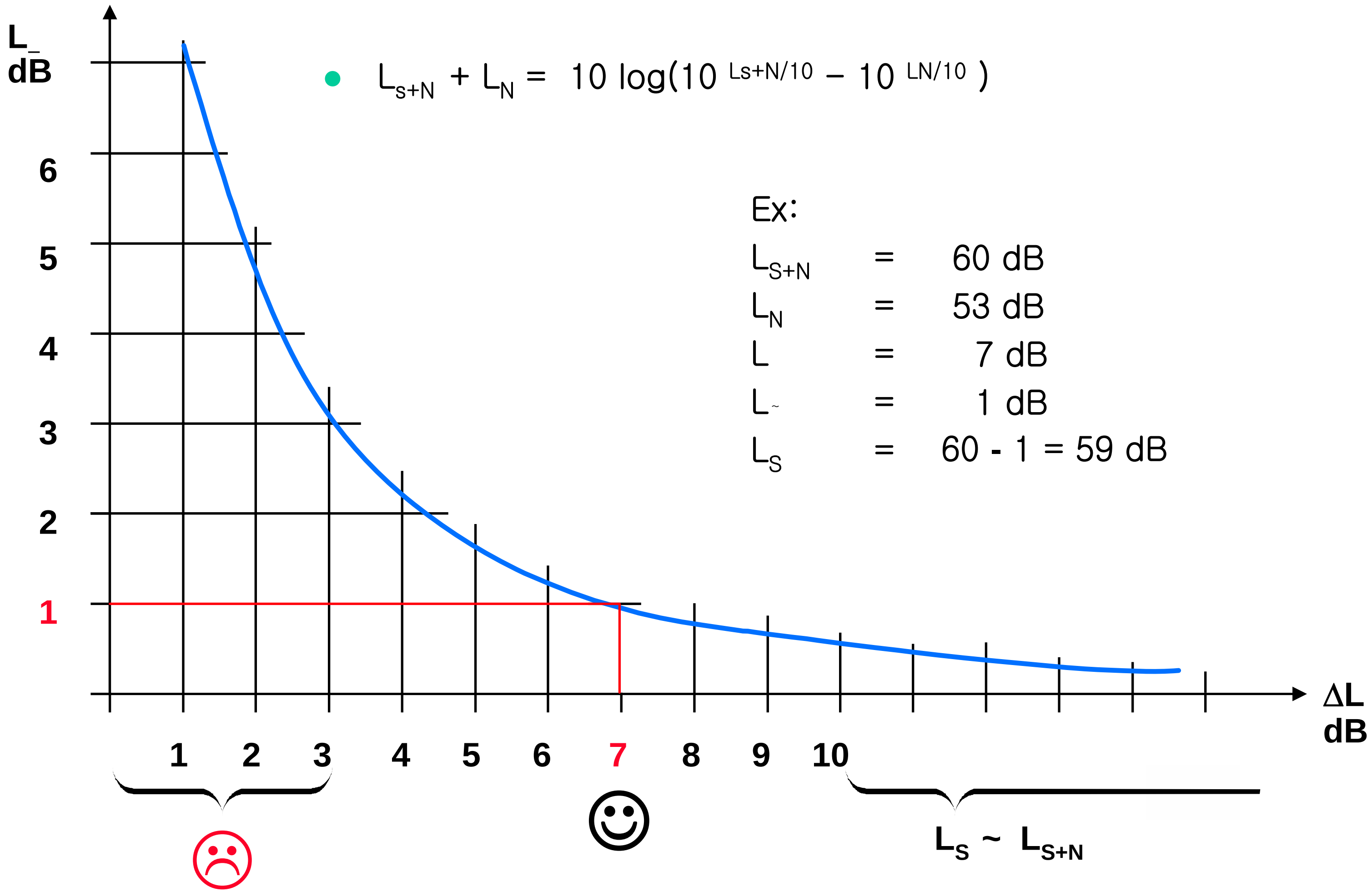
1. Measure the combined effect of machine noise and background noise, L_{S+N}.
2. Switch off the machine and measure the background noise, L_N. In most cases it is possible to switch off the machine under test, whereas the background noise normally cannot be switched off.
3. Finally calculate the difference, $L = L_{S+N} - L_N$ and use the following simple curve to find the correct noise level caused by the machine.

- Subtraction of dB Levels

- $L_{s+N} + L_N = 10 \log(10^{Ls+N/10} - 10^{LN/10})$

Ex:

L_{S+N}	=	60 dB
L_N	=	53 dB
L	=	7 dB
$L_{\sim}$	=	1 dB
L_S	=	60 - 1 = 59 dB



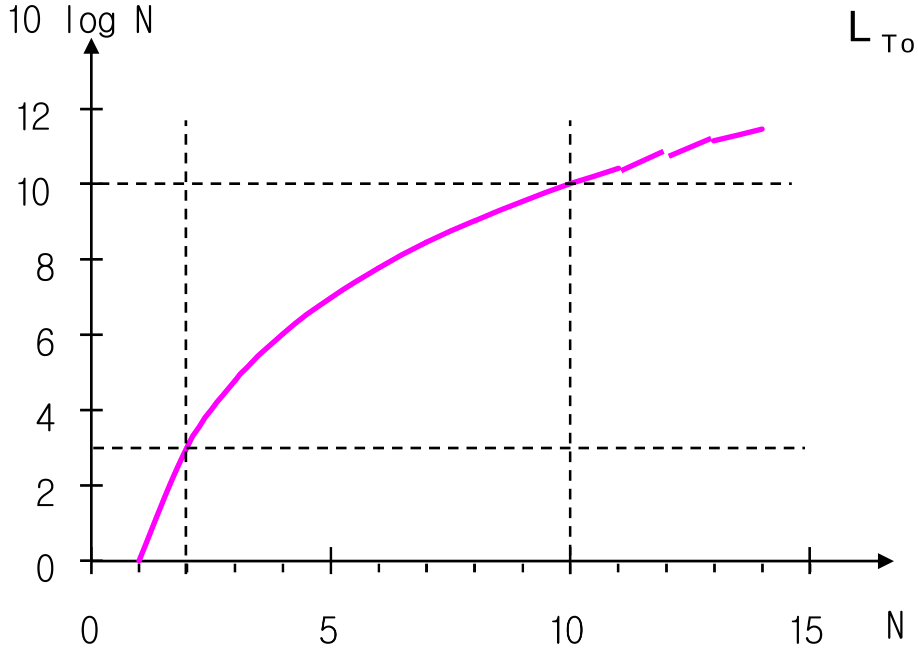
- Subtraction of dB Levels

  - $L_{Total} = 10 \log (10^{0.1 L1} + 10^{0.1 L2} + 10^{0.1L3} \ldots\ldots + 10^{0.1Ln})$

Addition of sound levels : $L_1 + L_2 \ldots\ldots + L_N = ?$

For $L_1 = L_2 = L_3 \ldots\ldots = L_N$

$$L_{Total} = L_1 + 10 \log N$$

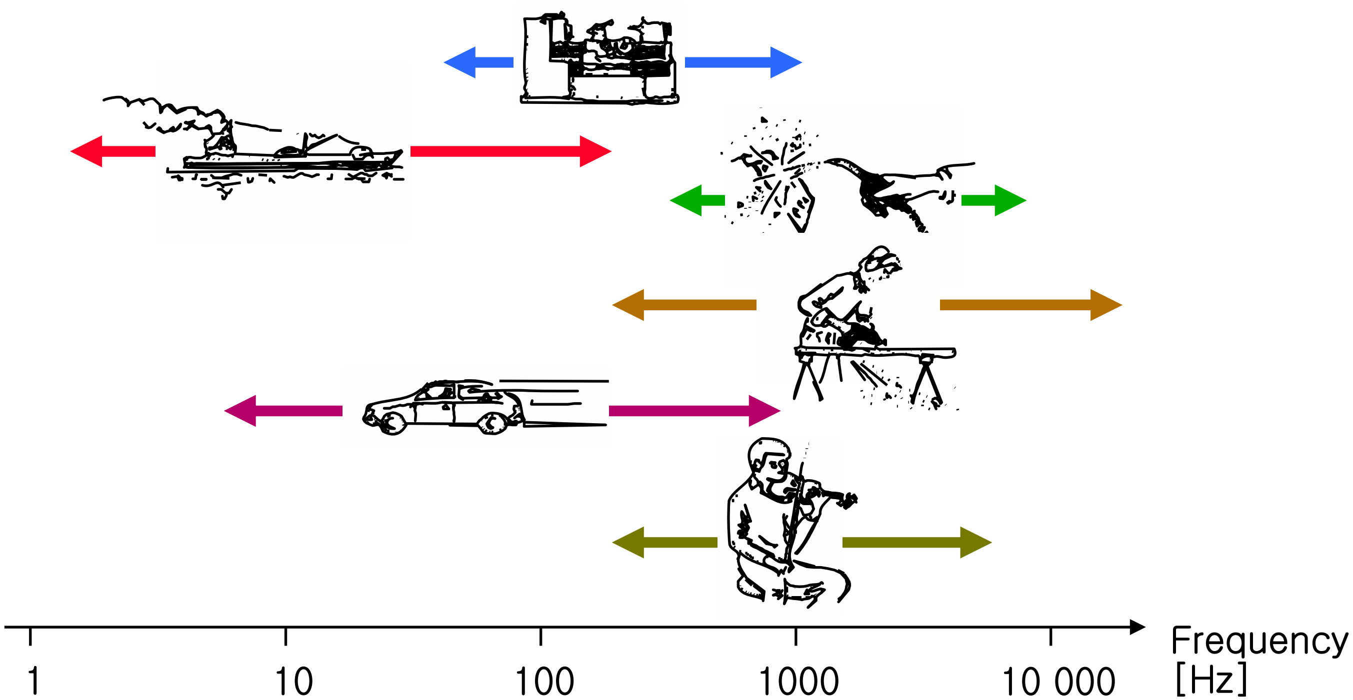


Examples:

N = 2: $L_{Total} = L_1 + 3$ dB

N = 10: $L_{Total} = L_1 + 10$ dB

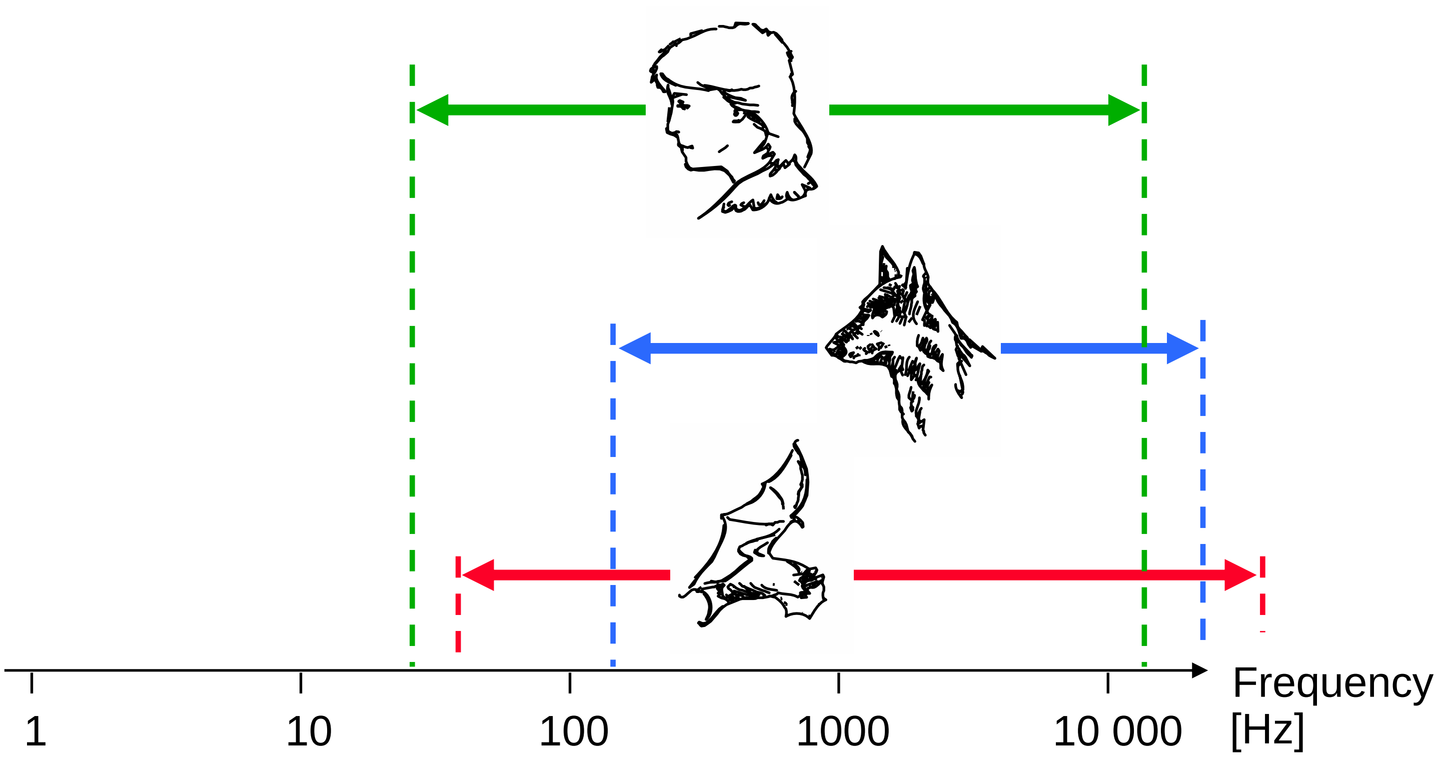
- Frequency Range of Different Sound Sources



The frequency span of the sounds that typically surround human beings vary considerably. Normally, young human beings can detect sounds ranging from 20 to 20000 Hz.

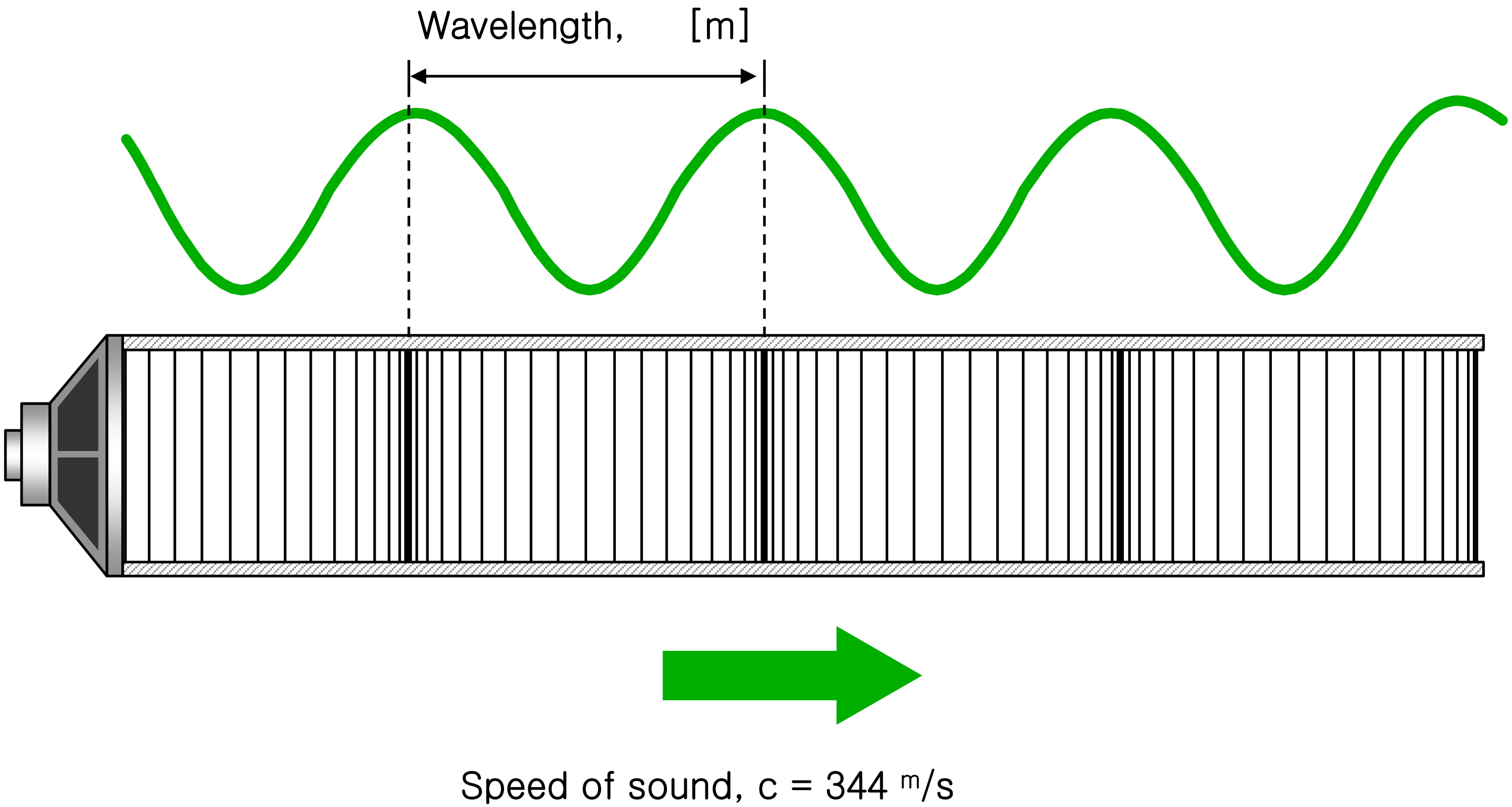
However, infrasounds in the range from 1 to 20 Hz and ultrasounds between 20000 to 40000 Hz can affect other human senses and cause discomfort.

- Audible Range



As can be seen, the range of perception of sound for humans – at a maximum for young people – goes from 20 to 20000 Hz.
With age, the human perception of the highest frequencies decreases gradually.

- Wavelength

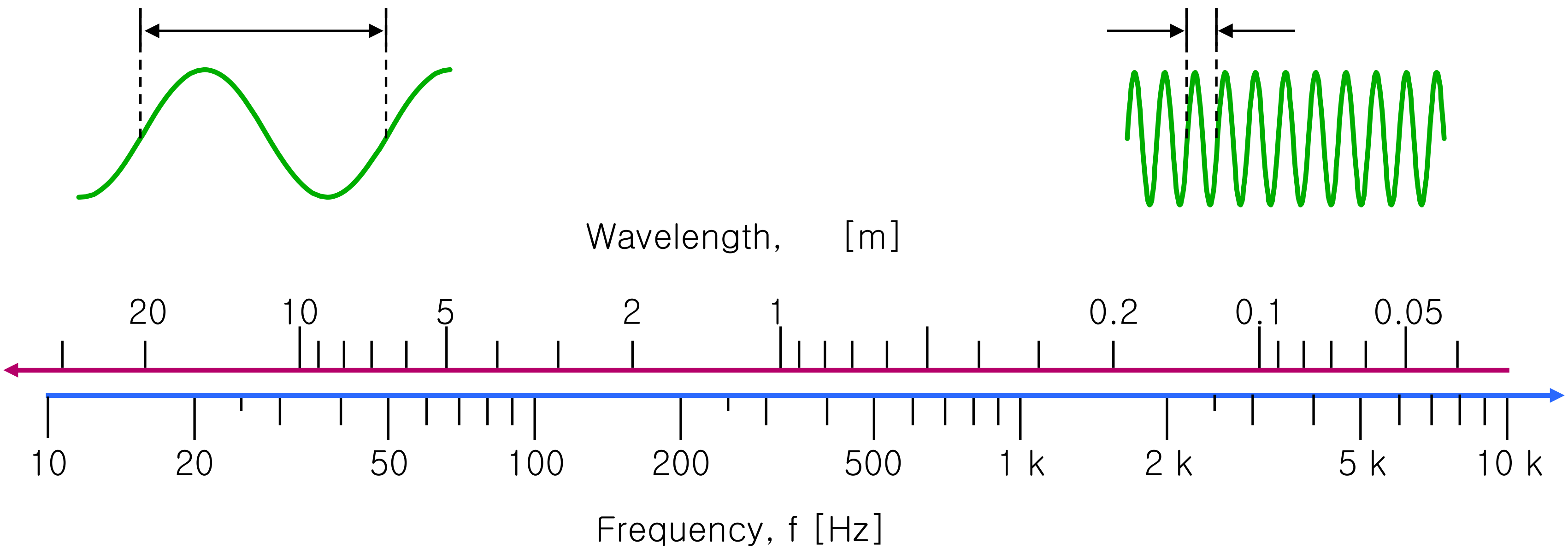


A sound signal from a loudspeaker mounted at one end of a tube will produce a sound wave that propagates forward at a speed of 344 m/s.

1.3 Sound Field

- Wavelength and Frequency

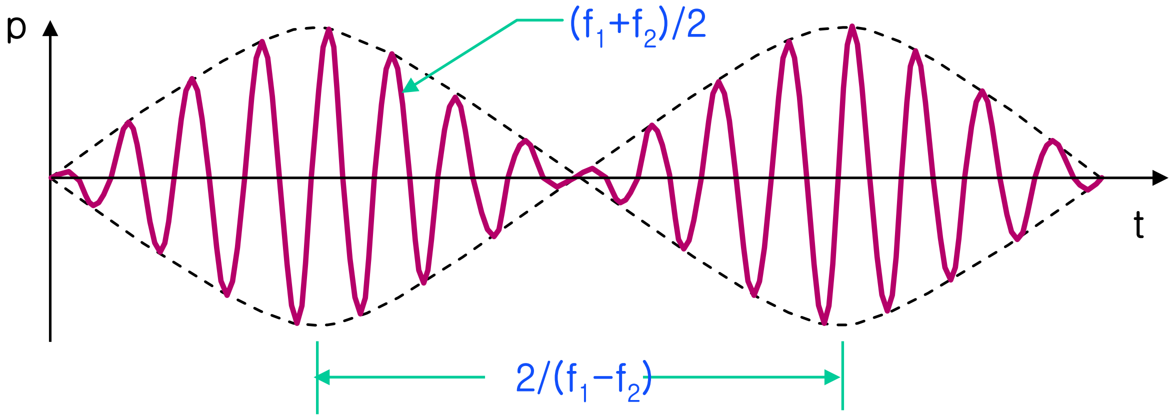
$$\lambda = \frac{c}{f}$$



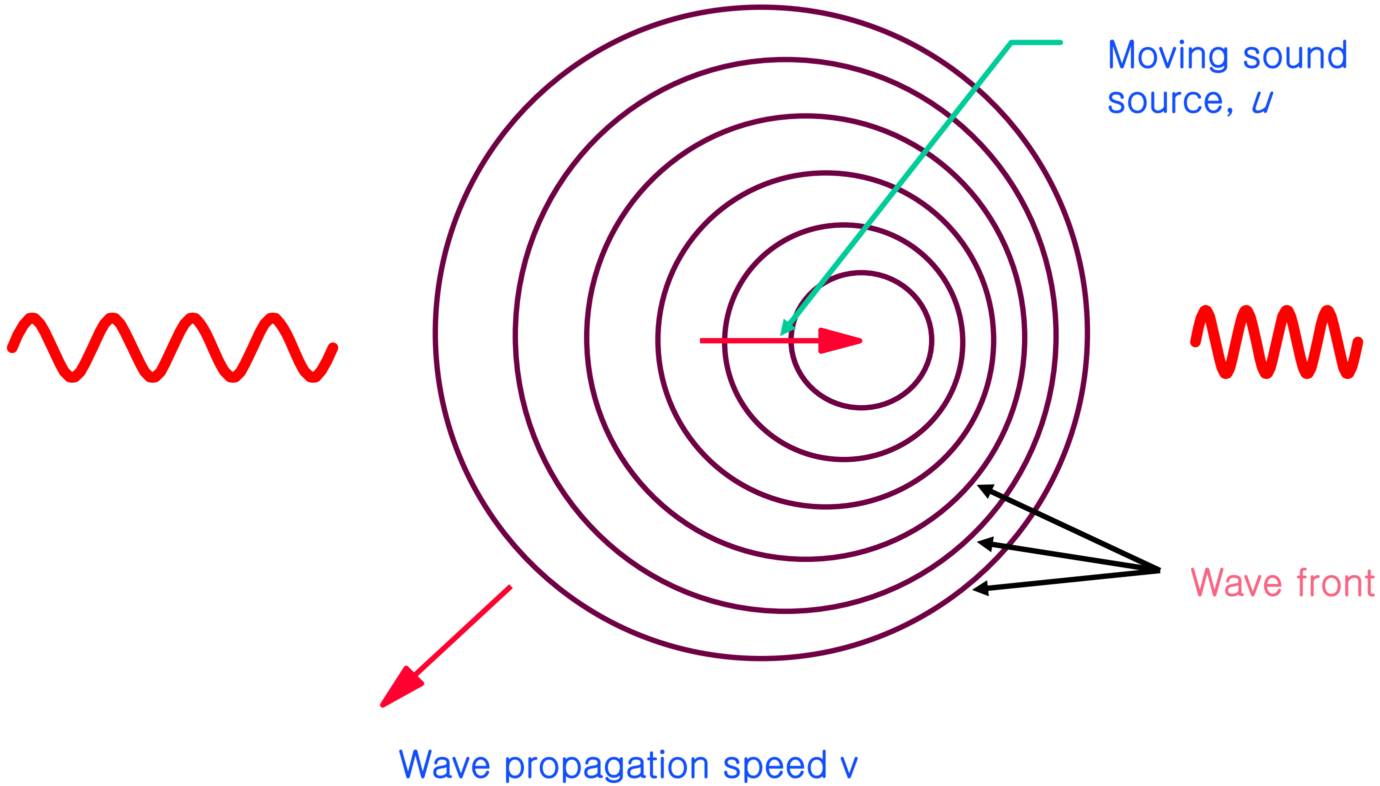
At 1 kHz the wavelength is close to 34 cm or one foot.

At 20 Hz it is close to 17 m, and only 1.7 cm at 20 kHz.

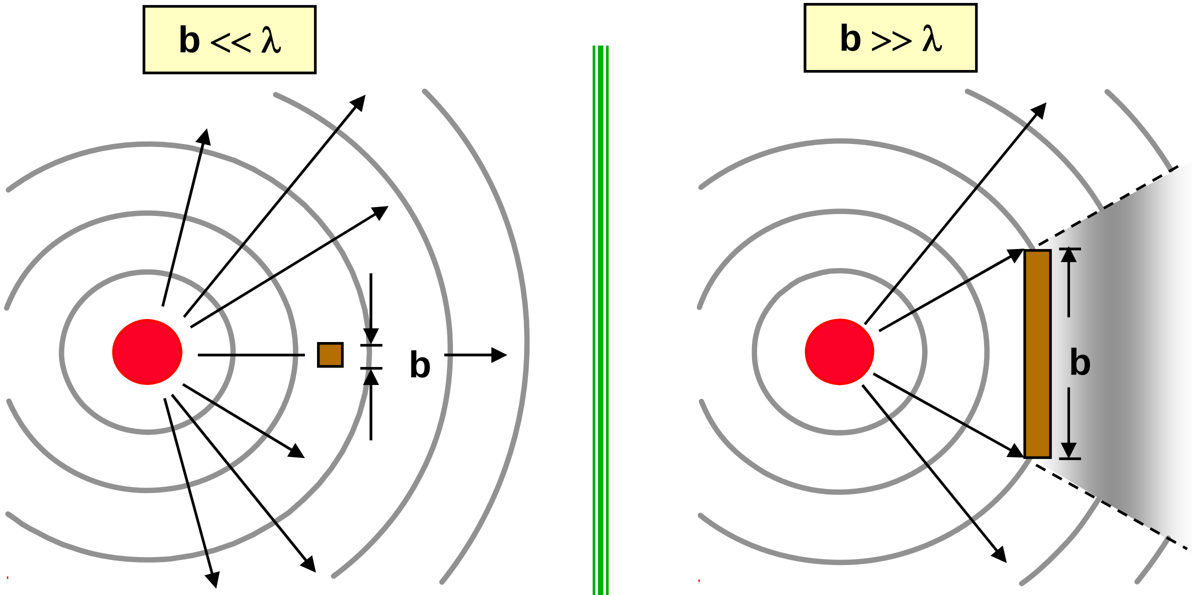
- Beating Phenomena



- Doppler Effect



- Diffraction of Sound

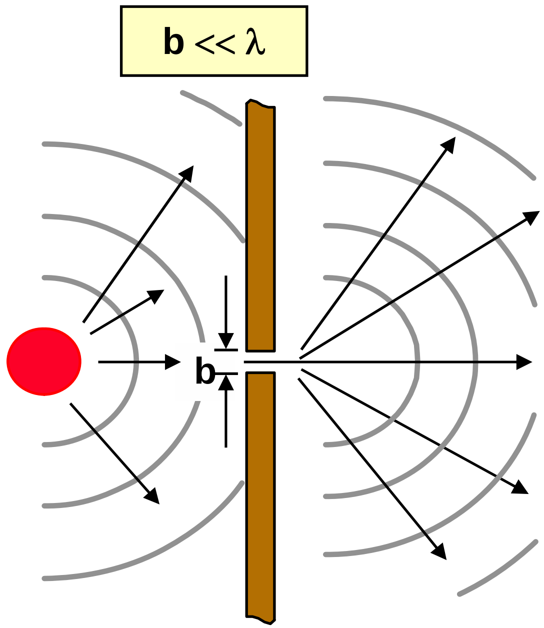


Example :
b = 0.1 m
λ = 0.344 m (≈ f = 1 kHz)

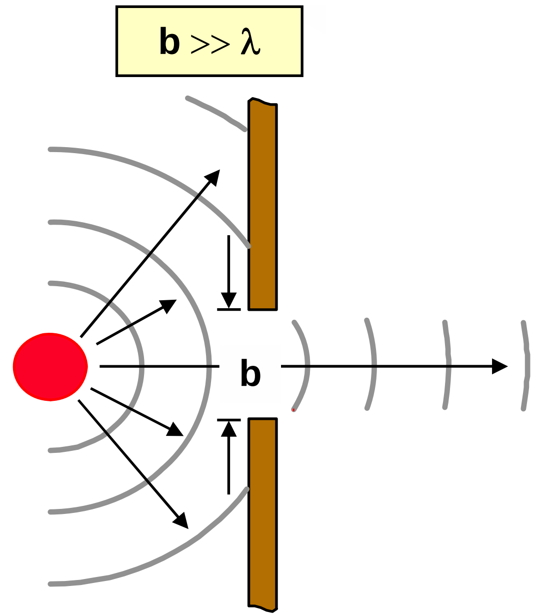
Example:
b = 1 m
λ = 0.344 m (≈ f = 1 kHz)

If the obstruction is larger than the wavelength, the effect is noticeable as a shadowing effect.

- Diffusion of Sound



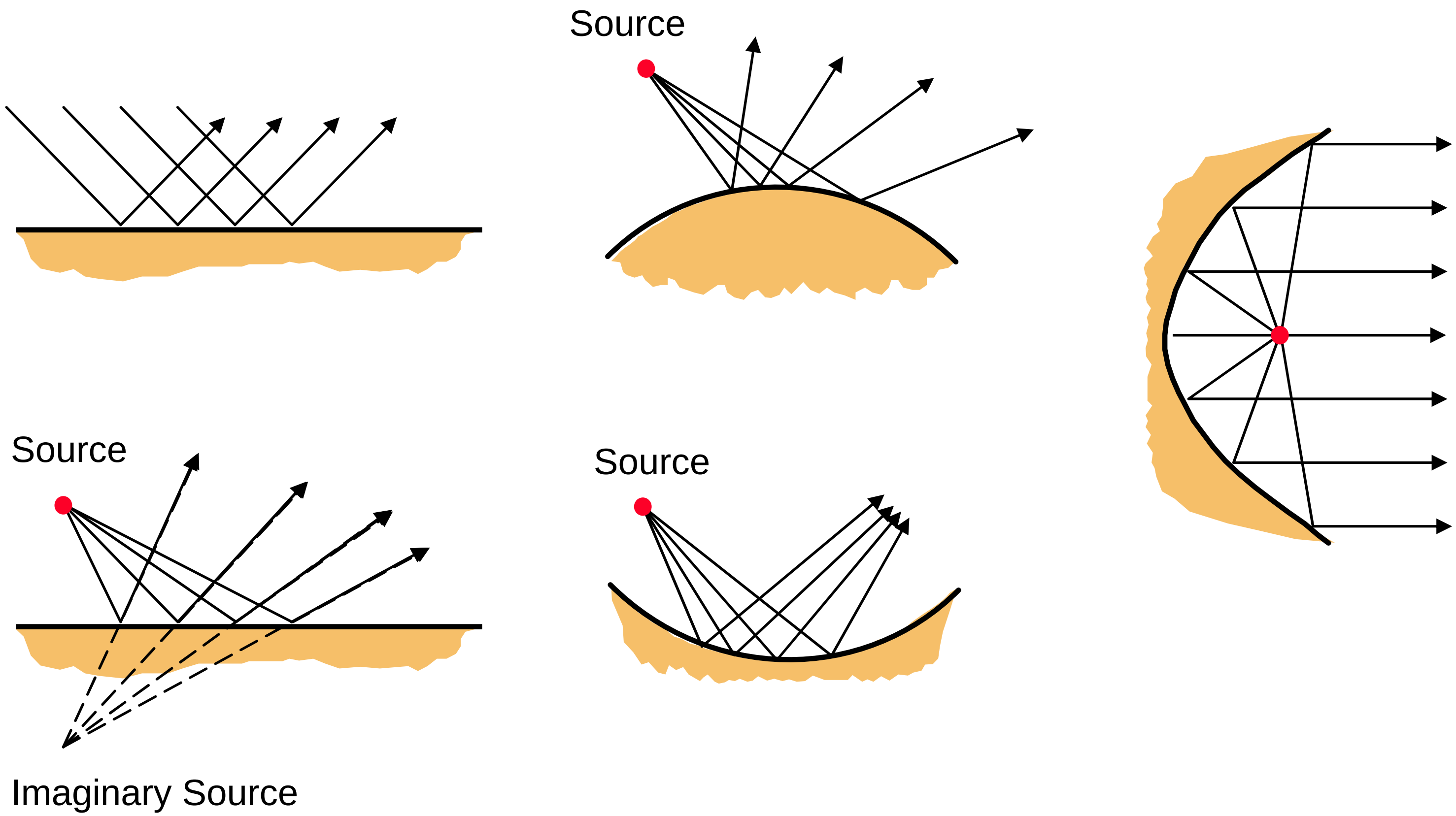
Example :
b = 0.1 m
λ = 0.344 m (≈ f = 1 kHz)



Example:
b = 0.5 m
λ = 0.344 m (≈ f = 1 kHz)

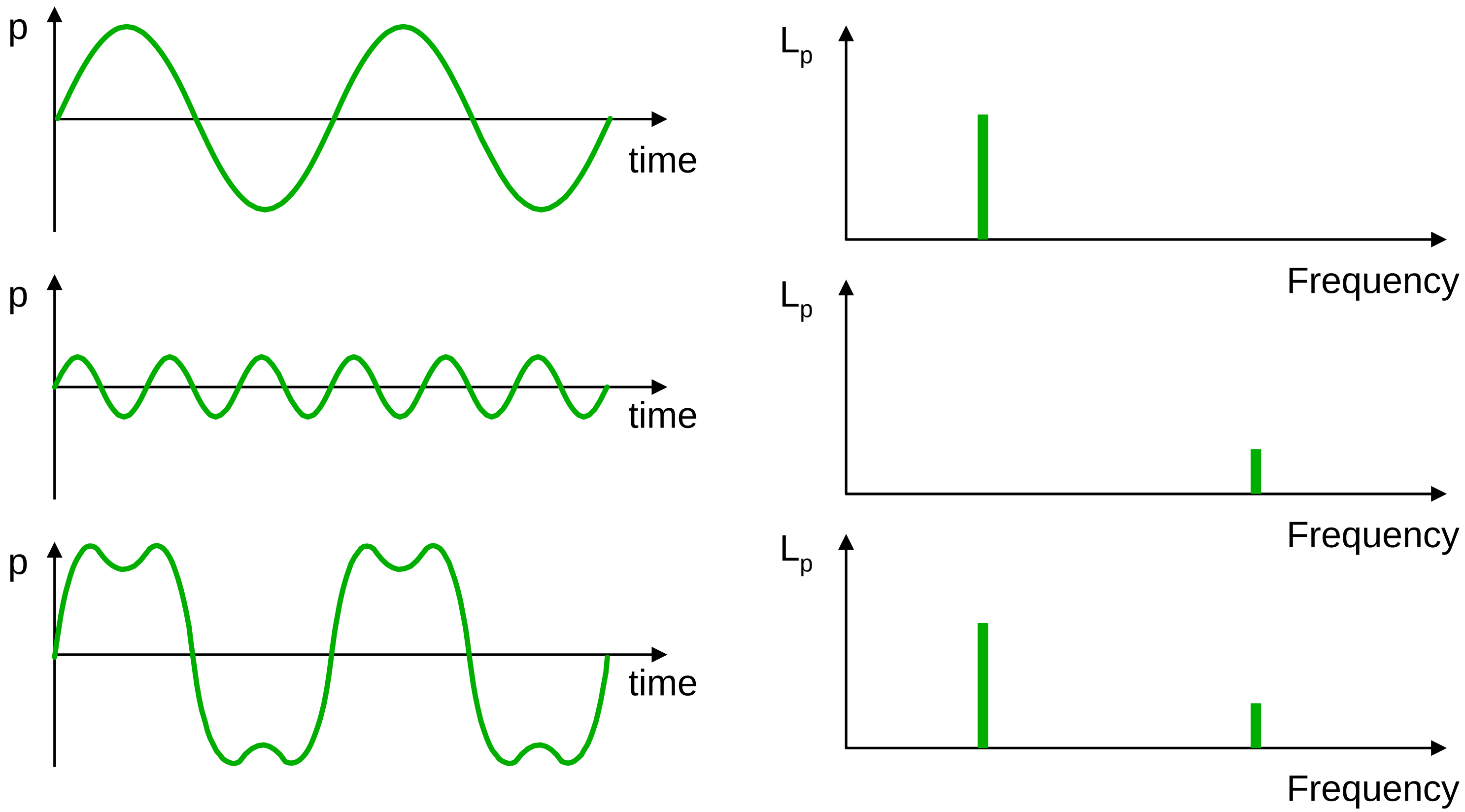
If the holes are small compared to the wavelength of the sound, the sound passing will re-radiate in an omnidirectional pattern similar to the original sound source.
When the hole has larger dimensions than the wavelength of the sound, the sound will pass through with negligible disturbance.

- Reflection of Sound

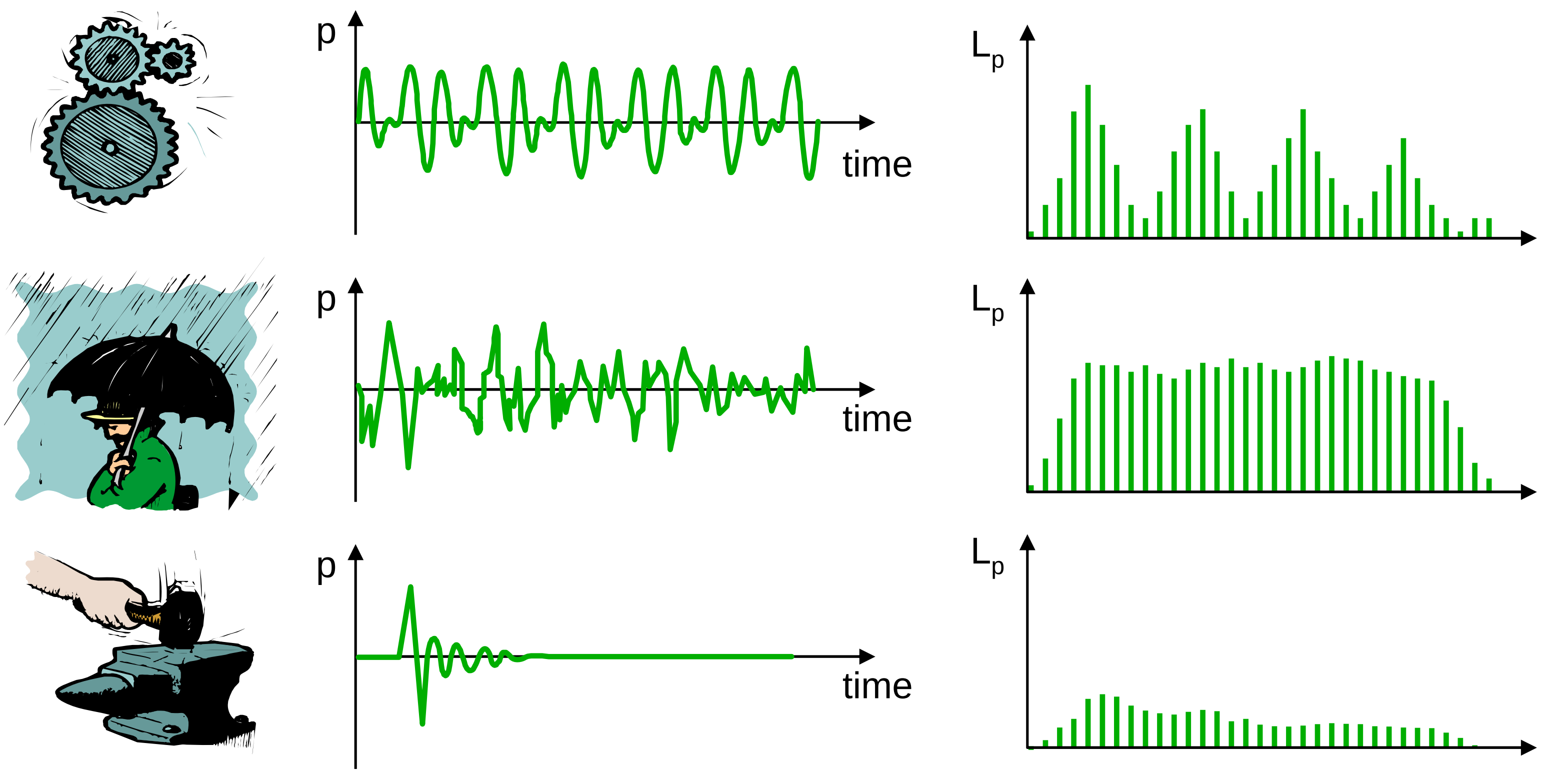


When sound hits obstructions large in size compared to it's wavelength, reflections take place. If the obstruction has very little absorption, all the reflected sound will have equal energy compared to the incoming sound.

• Waveforms and Frequencies

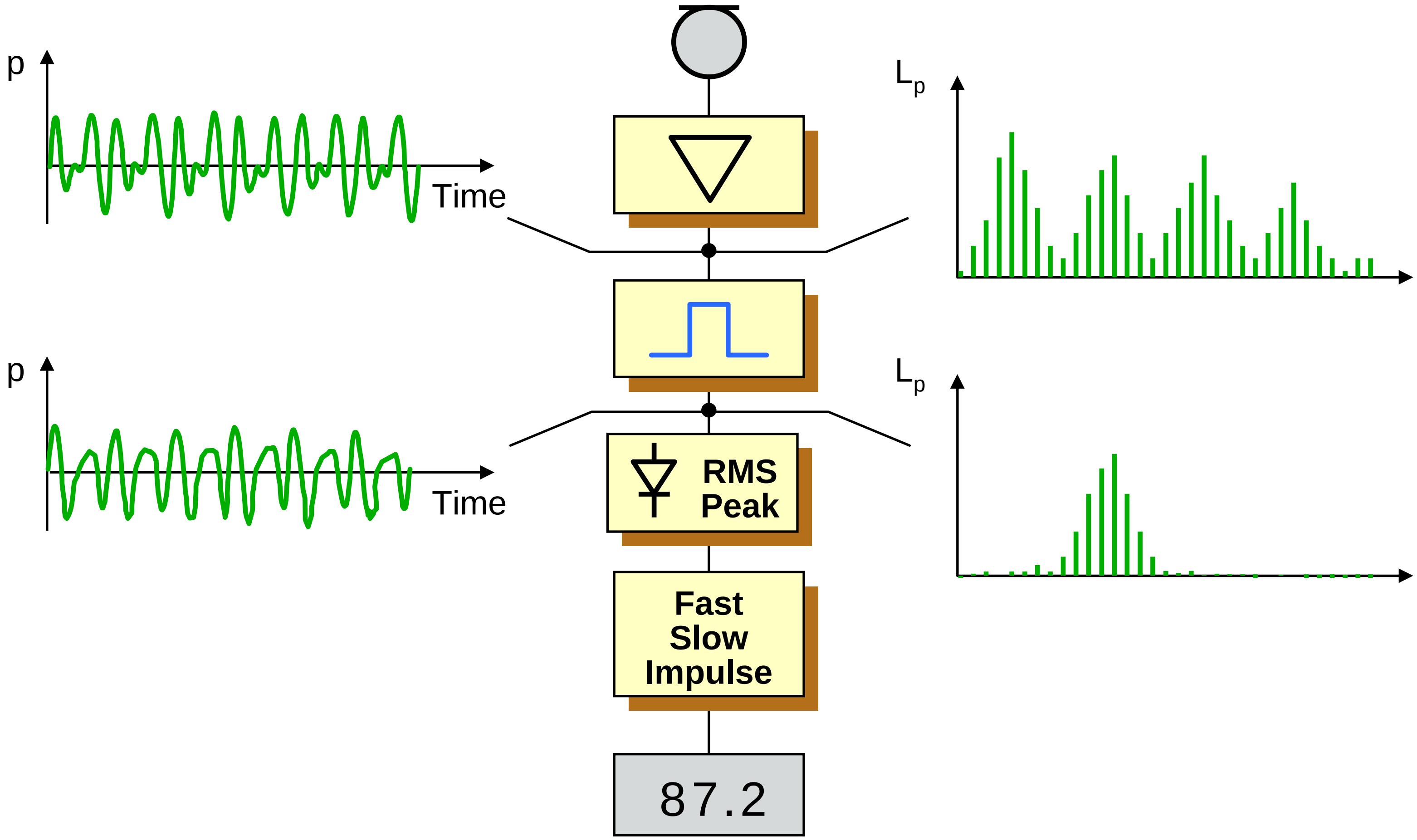


- Typical Sound and Noise Signals

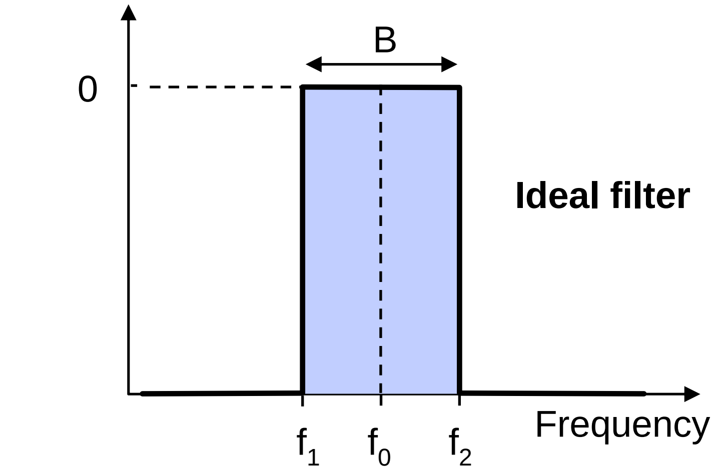


The primary result of a frequency analysis is to show that the signal is composed of a number of discrete frequencies at individual levels present simultaneously.

- Filters

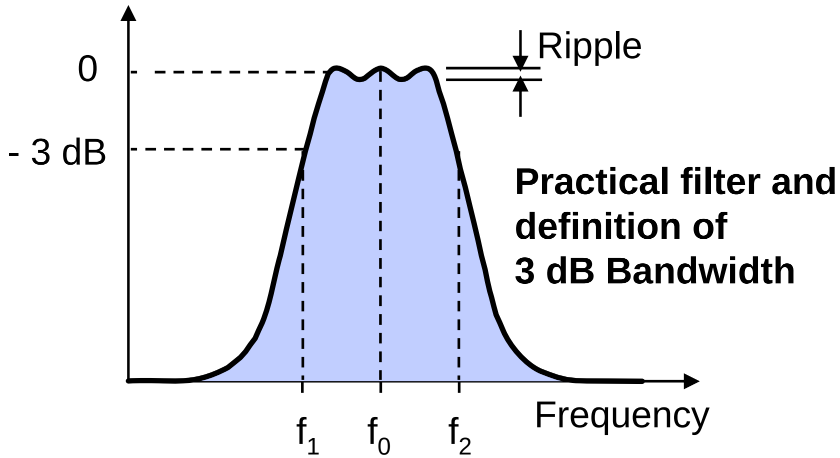


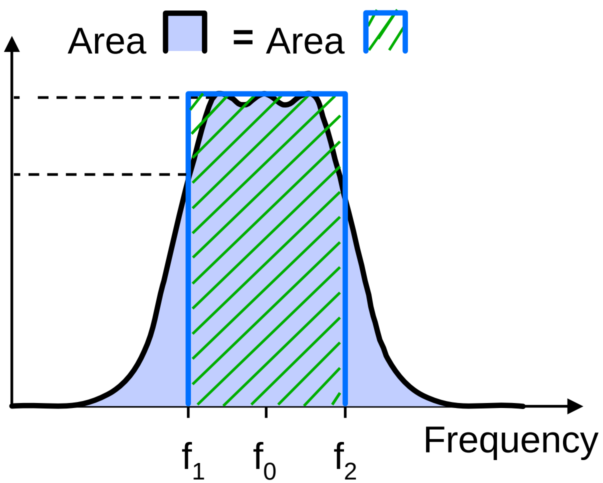
- Bandpass Filters and Bandwidth



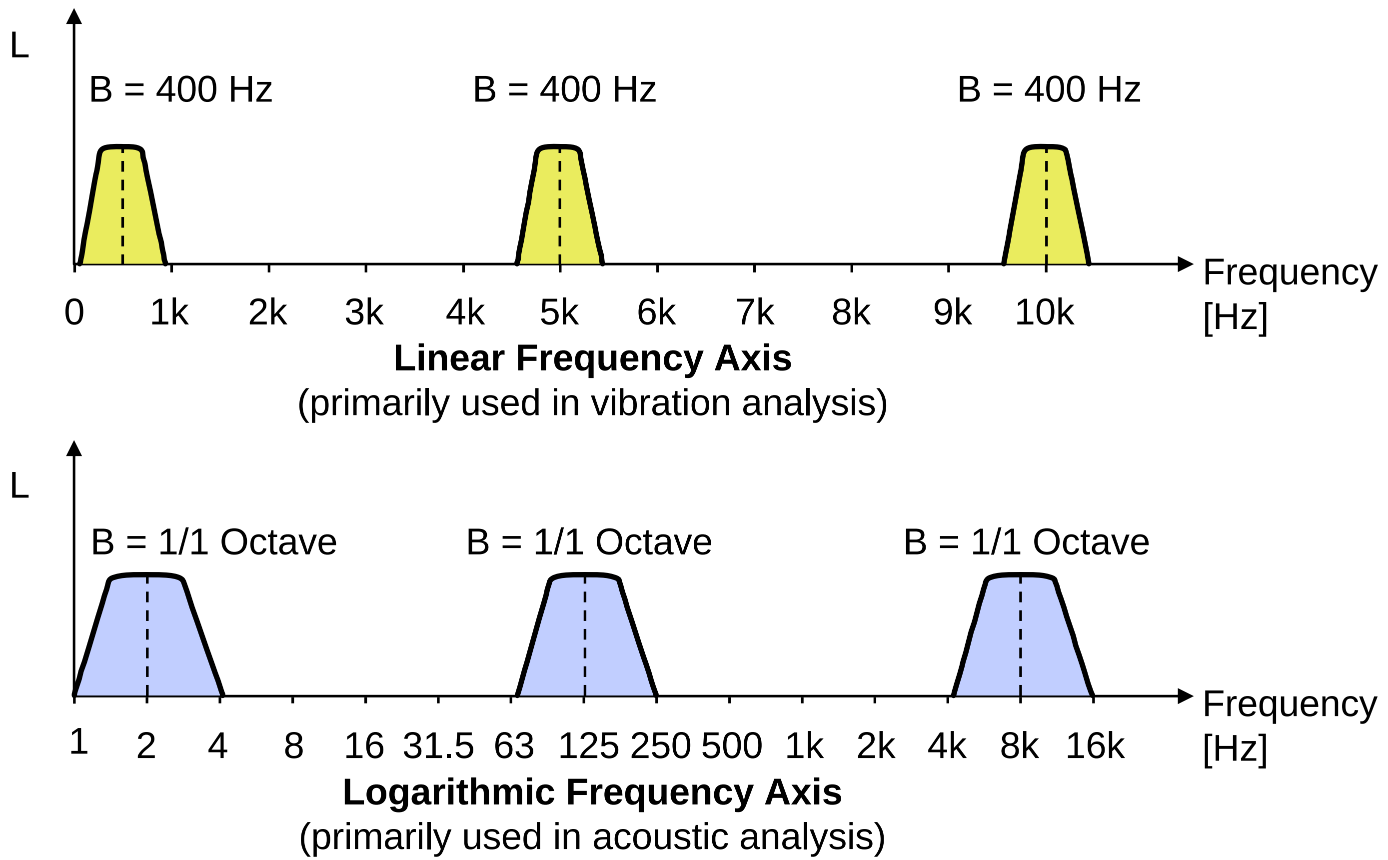
Bandwidth = $f_2 - f_1$

Centre Frequency = f_0





- Typical Sound and Noise Signals

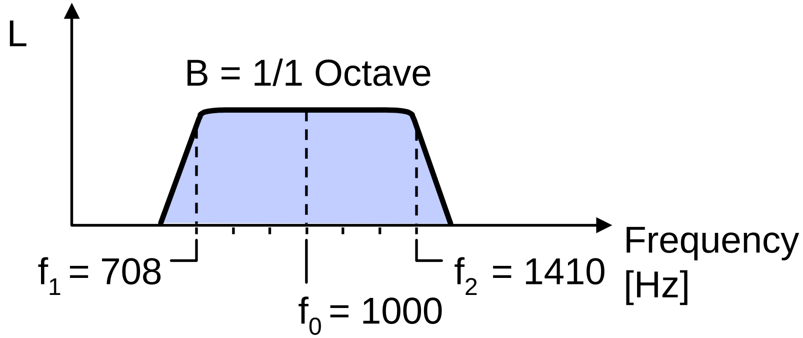


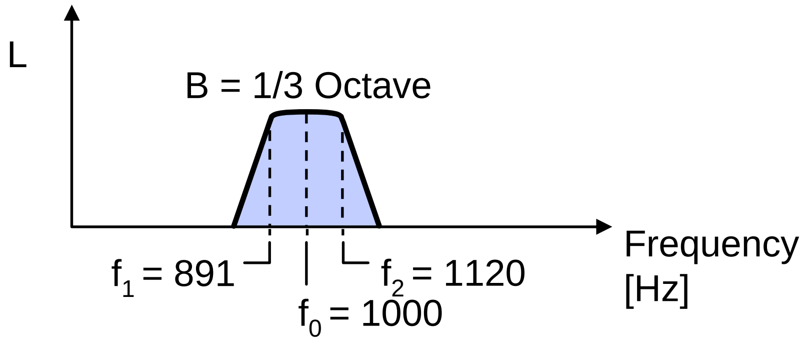
Constant bandwidth filters are mainly used in connection with analysis of vibration signals.

Analysis with CPB filters (and logarithmic scales) is almost always used in connection with acoustic measurements, because it gives a fairly close approximation to how the human ear responds.

1.3 Sound Field

- 1/1 and 1/3 Octave Filters





1/1 Octave

$f_2 = 2 \times f_1$

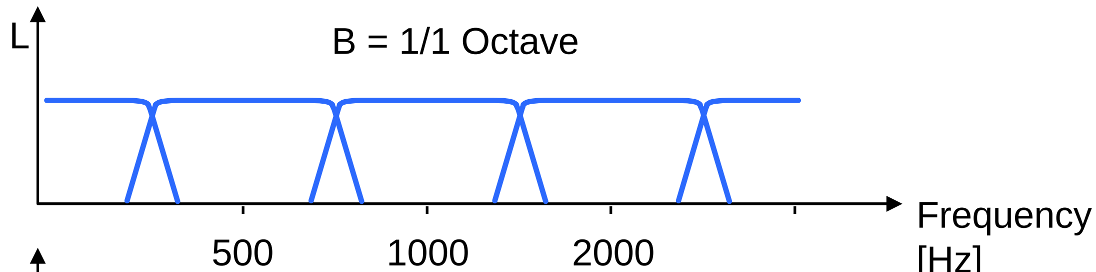
$B = 0.7 \times f_0 \approx 70\%$

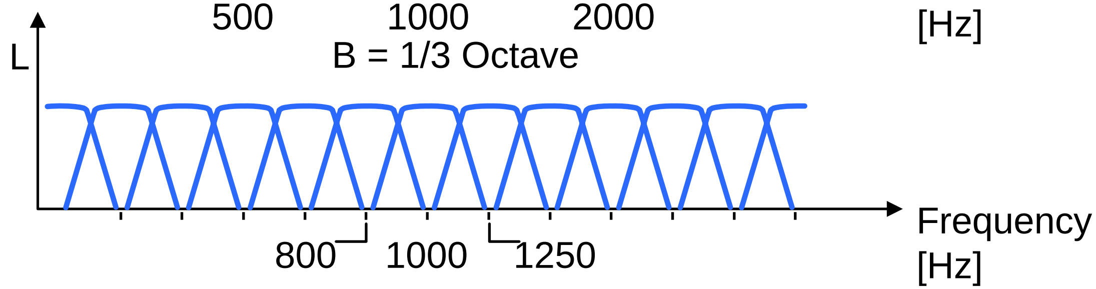
1/3 Octave

$f_2 = \sqrt[3]{2} \times f_1 = 1.25 \times f_1$

$B = 0.23 \times f_0 \approx 23\%$

- 3 × 1/3 Oct. = 1/1 Oct.

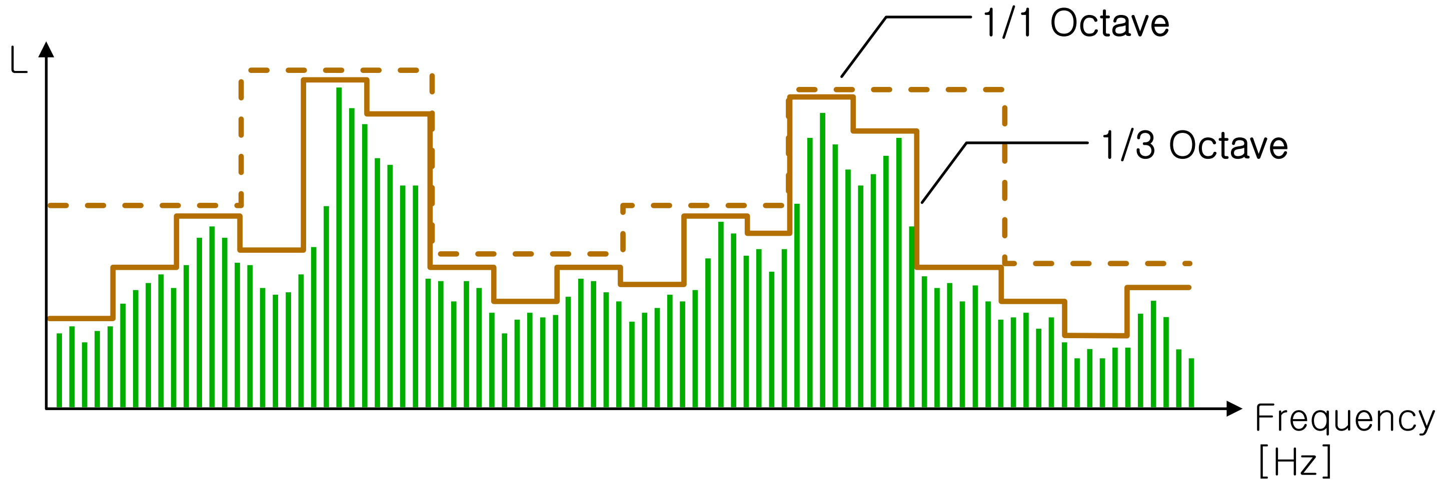




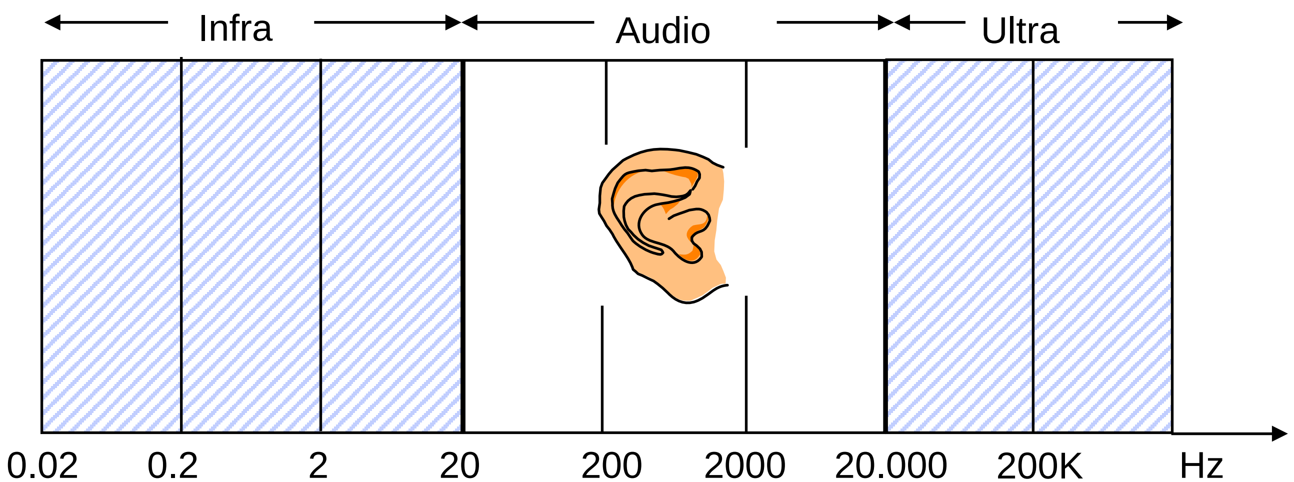
- Third-octave and Octave Passband

Band No.	Nominal Centre Frequency Hz	Third-octave Passband Hz	Octave Passband Hz
1	1.25	1.12 – 1.41	
2	1.6	1.41 – 1.78	
3	2	1.78 – 2.24	1.41 – 2.82
4	2.5	2.24 – 2.82	
5	3.15	2.82 – 3.55	
6	4	3.55 – 4.47	2.82 – 5.62
27	500	447 – 562	355 – 708
28	630	562 – 708	
29	800	708 – 891	
30	1000	891 – 1120	780 – 1410
31	1250	1120 – 1410	
32	1600	1410 – 1780	
40	10 K	8910 – 11200	
41	1.25 K	11.2 – 14.1	
42	16 K	14.1 – 17.8 K	11.2 – 22.4 K
43	20 K	17.8 – 22.4 K	

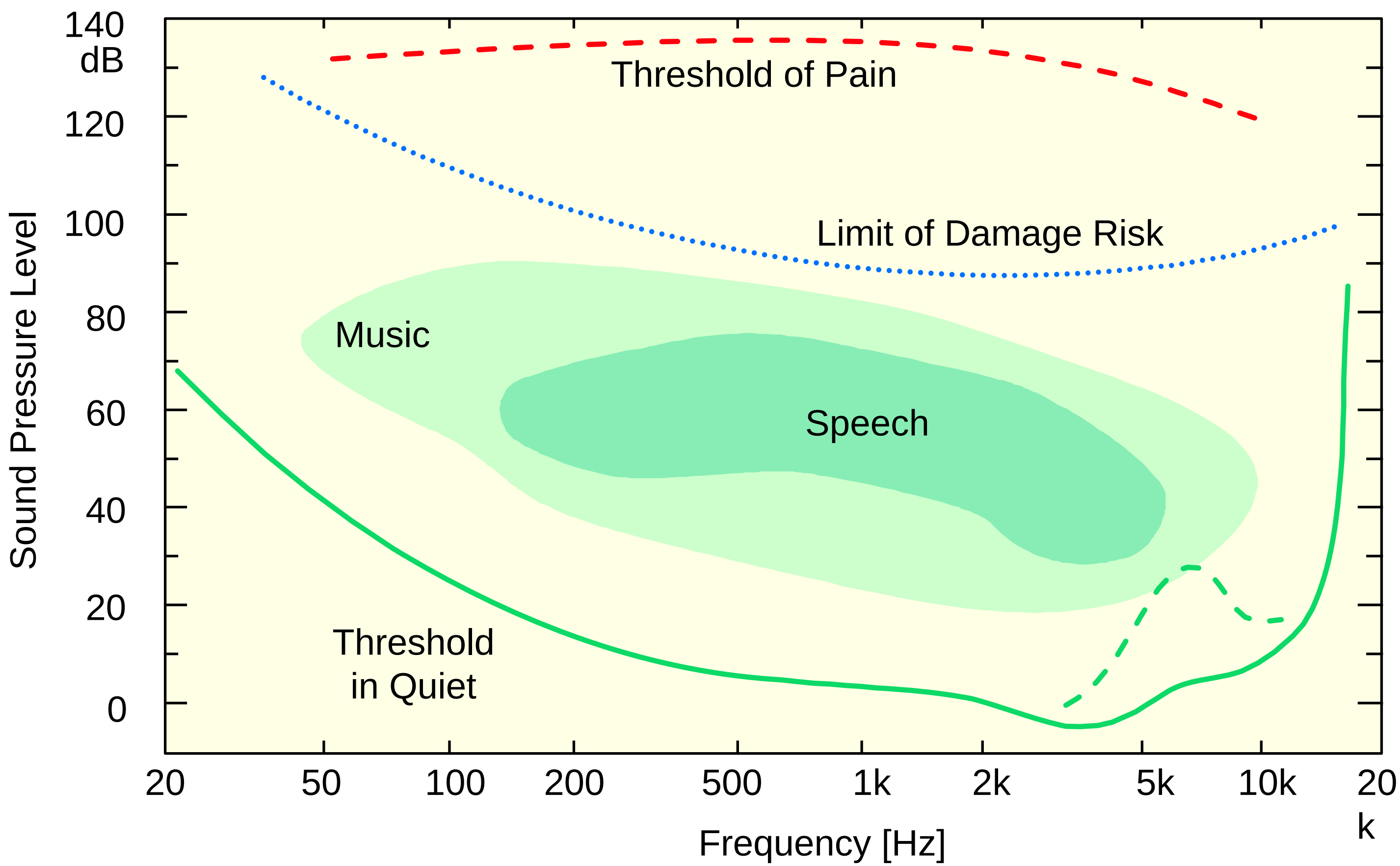
- The Spectrogram



- Sound Frequencies

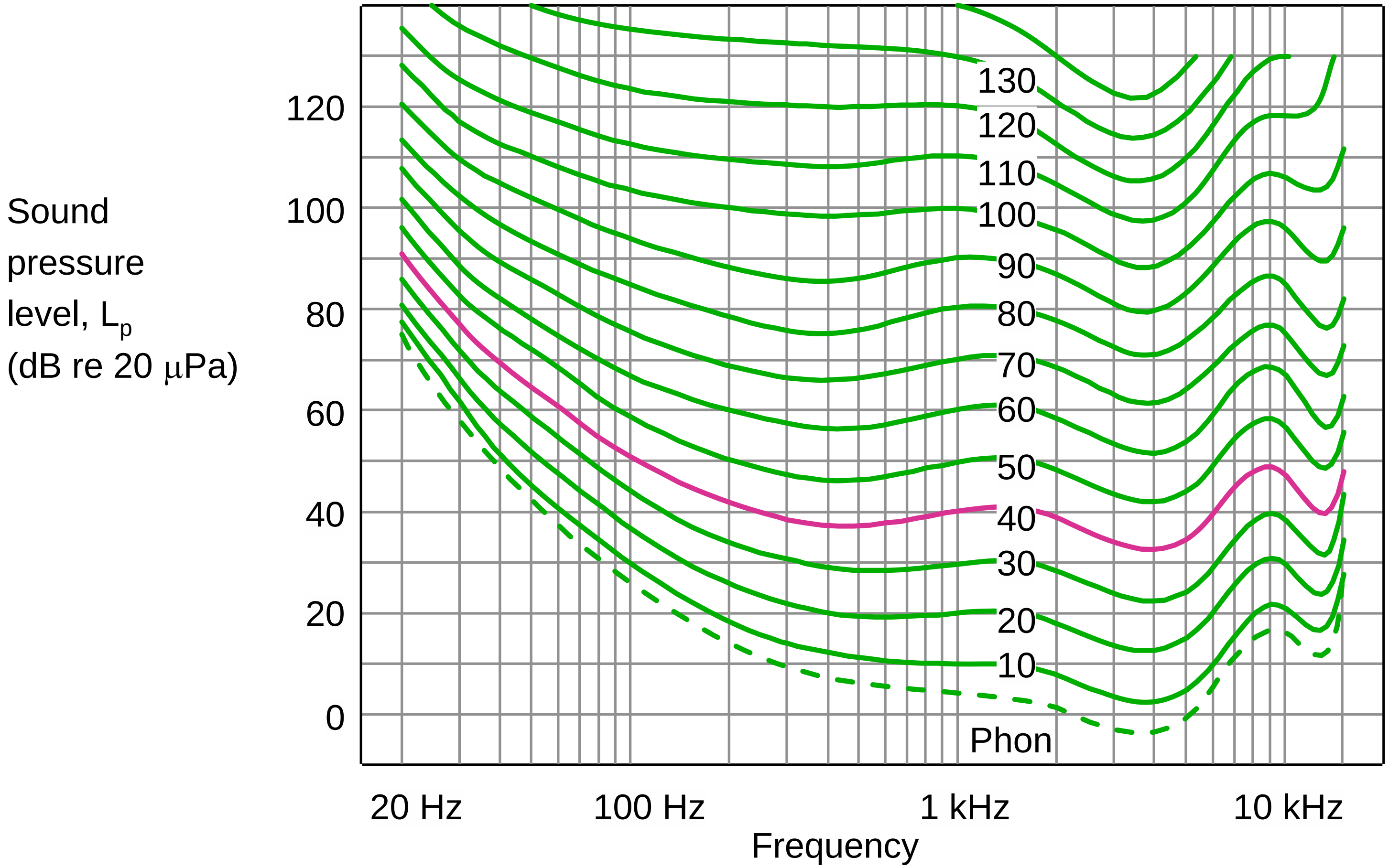


- Auditory Field



This display of the auditory field illustrates the limits of the human auditory system.
The solid line denotes, as a lower limit, the threshold in quiet for a pure tone to be just audible.

- Equal Loudness Contours for Pure Tones

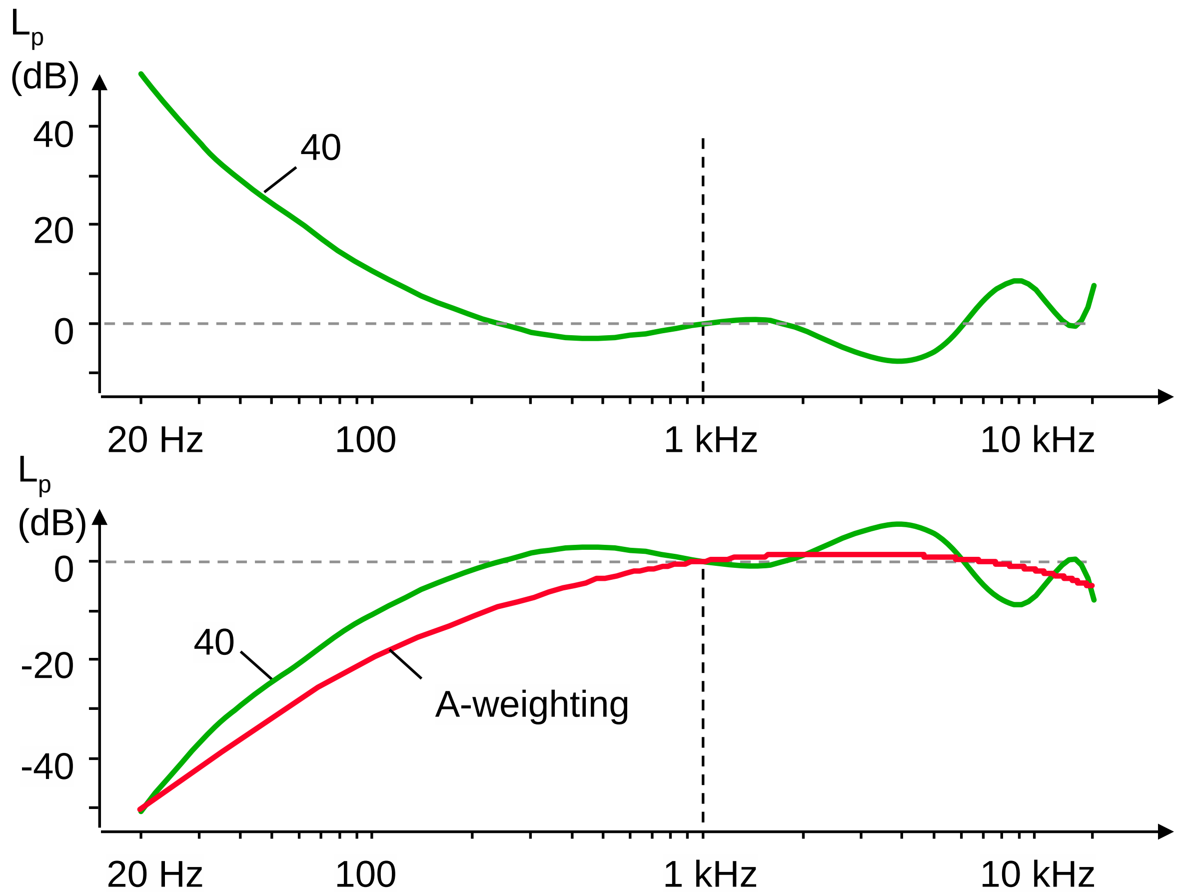


Here are shown the normal equal loudness contours for pure tones. The dashed curve indicates the normal binaural minimum audible field.
Note the very non-linear characteristics of the human perception. Almost 80 dB more SPL is needed at 20 Hz to give the same perceived loudness as at 3-4 kHz.

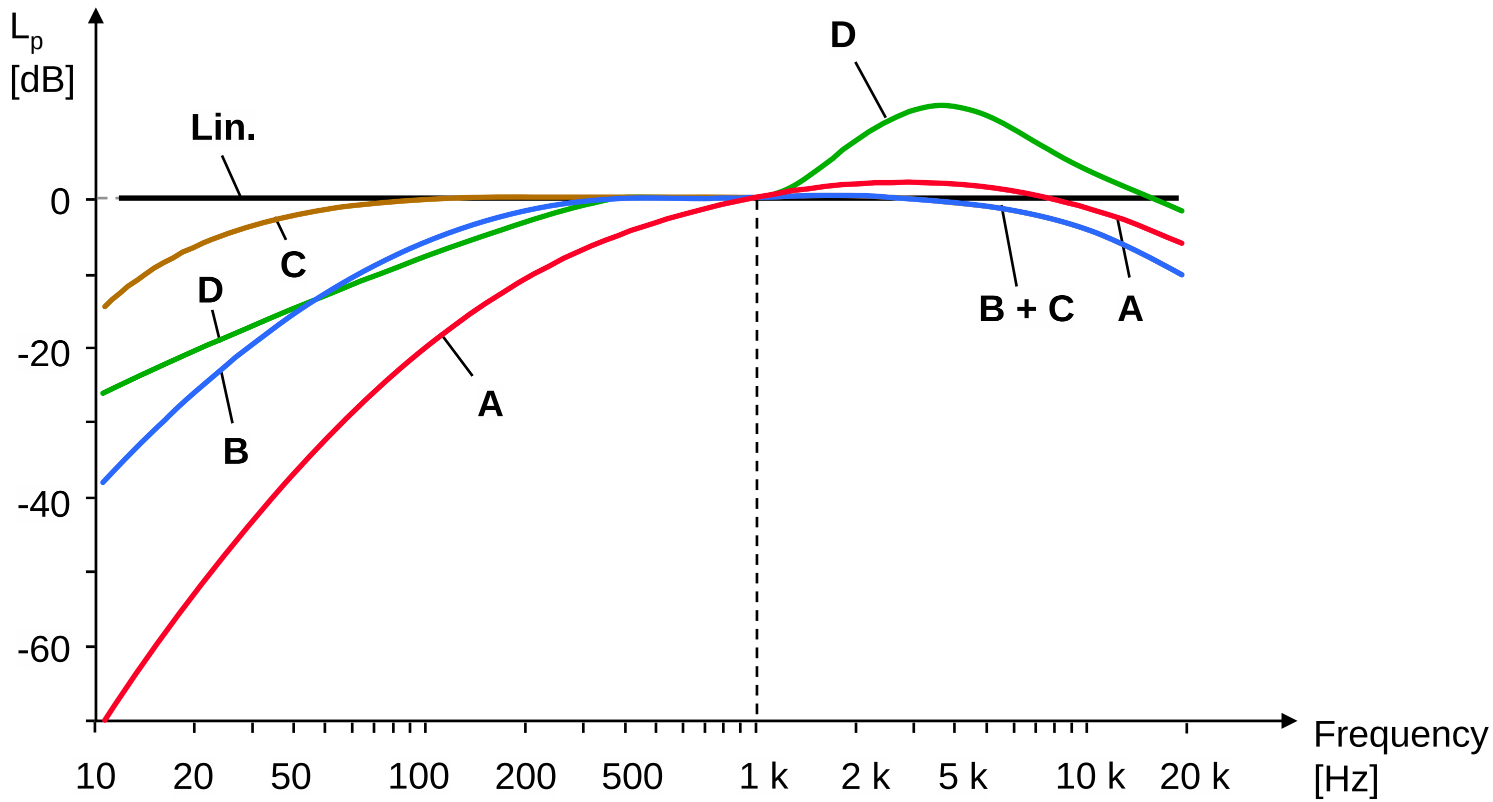
- Equal Loudness Contours for Pure Tones

- 40 dB Equal Loudness Contour normalized to 0 dB at 1kHz

- 40 dB Equal Loudness Contour inverted and compared with A–weighting

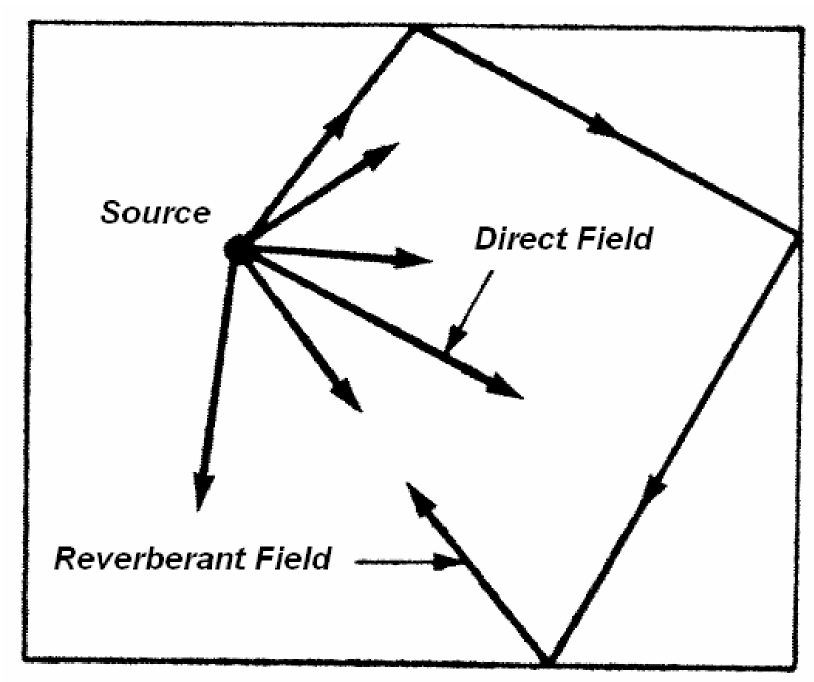


- Frequency Weighting Curves



The A-weighting, B-weighting and C-weighting curves follow approximately the 40, 70 and 100 dB equal loudness curves respectively.
D-weighting follows a special curve which gives extra emphasis to the frequencies in the range 1 kHz to 10 kHz. This is normally used for aircraft noise measurements.

- Room Acoustics



- Direct field :

Sound Intensity $I \propto \dfrac{1}{r^2}$

- Reverberant field :

Sound Intensity $I = I_r$: constant

  – Steady-state, multiple reflection sounds

  – It depends on the size of room and the reflectivity of surface.

- Acoustic energy density

$$\delta = \frac{I}{c} = \frac{p^2}{\rho c^2} \quad [joule/m^3]$$

Acoustic energy passing thorough unit area during △t

$$\Delta E = I \cdot \Delta t$$

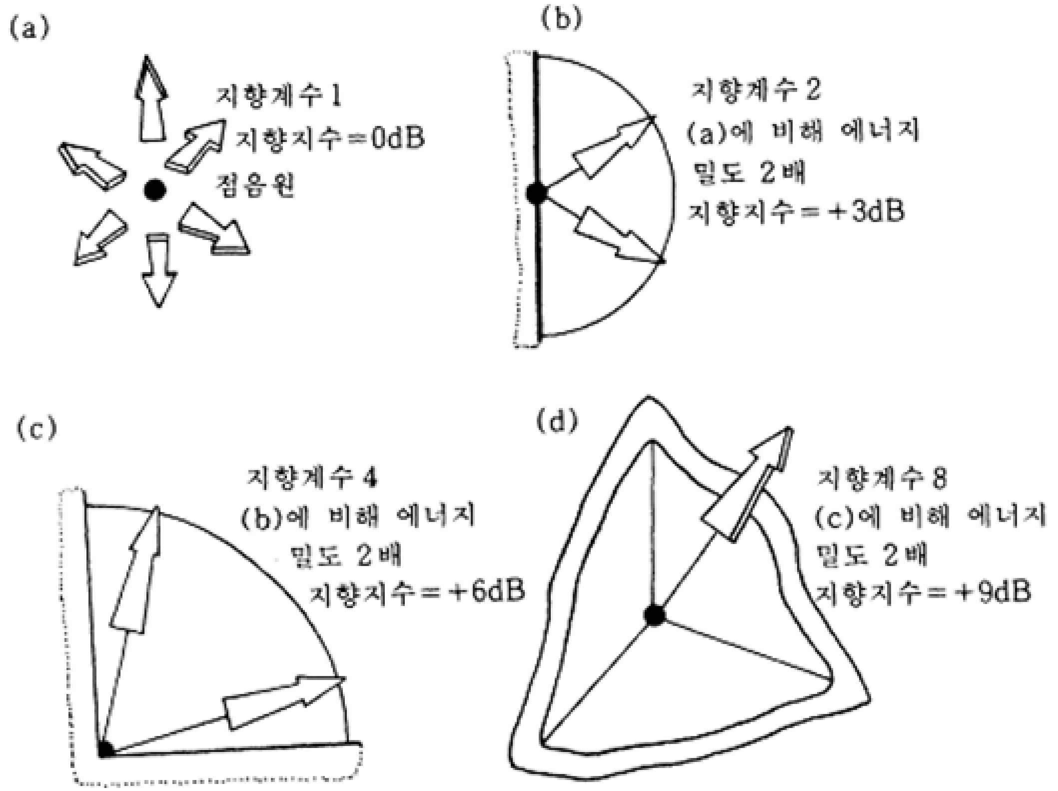
Volume made during △t

$$\Delta V = 1 \cdot c\Delta t$$

- Energy density in direct sound field

  - Spherical wave

$$I_o = \frac{W}{4\pi r^2} = \frac{p^2}{\rho c}$$



  - If the directivity index is considered,

$$I_o = \frac{QW}{4\pi r^2}$$

  - Acoustic energy density

$$\delta_o = \frac{QW}{4\pi r^2 c}$$

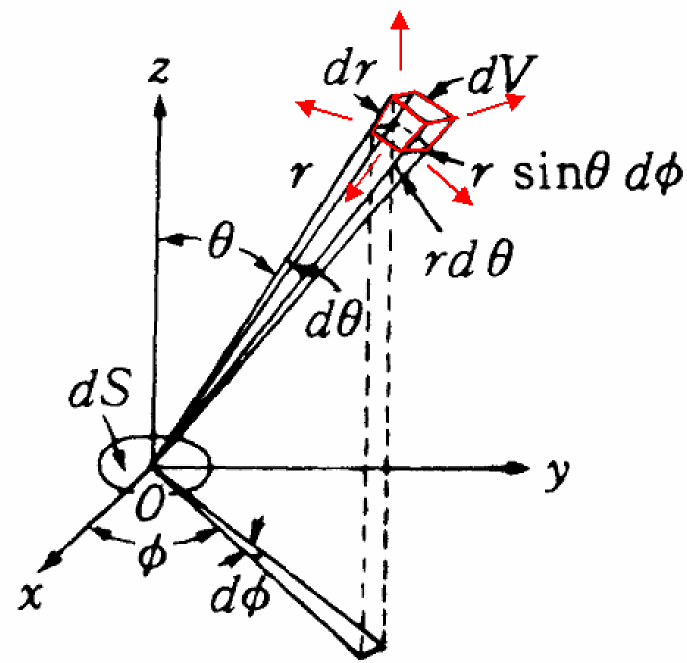
- Energy density in reverberant sound field

$$\text{Let energy density} = \delta' \quad [J/m^3]$$

- Acoustic energy of dV
  having a distance r from the infinitesimal wall dS

$$\delta' \Delta V \quad [J]$$

: It propagates sound energy in all direction
as a point source.



- Incident energy on dS

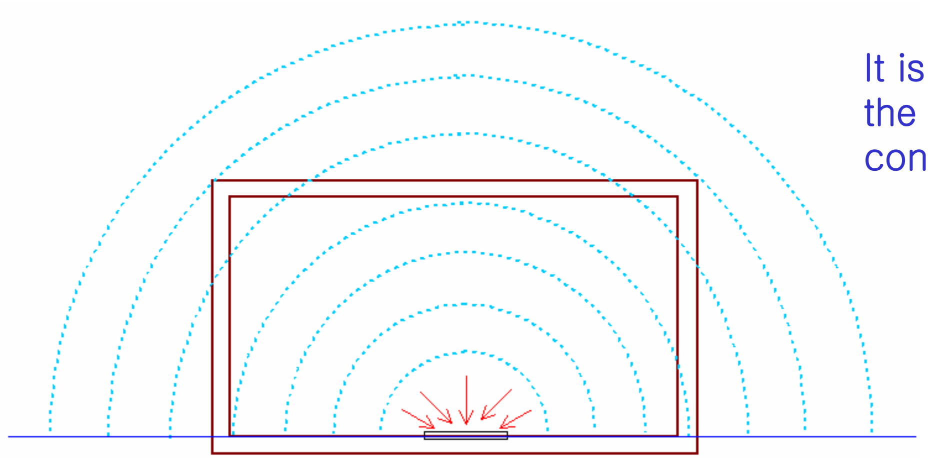
$$Let \quad \Delta I' dS \qquad \Delta I' dS = \frac{\delta' \Delta V}{4\pi r^2} dS \cos\theta$$

$\Delta I'$: energy per unit area
which is normally incident on dS

$\frac{\delta' \Delta V}{4\pi r^2}$: energy per unit area on the surface of sphere
which has a distance r from the source

- Total energy during 1 sec. (incident on dS)

It can be obtained by integrating the above volume
in the hemisphere with radius c., which is the incident range on dS during 1 sec.



It is assumed that
the reverberant sound energy is
constant and uniform in a room.

$$\Delta V = r^2 \sin\theta \, dr d\theta d\phi \qquad \Delta I' dS = \frac{\delta' \Delta V}{4\pi r^2} dS \cos\theta = \frac{\delta' dS}{4\pi} \cos\theta \sin\theta d\theta d\phi dr$$

$$I' dS = \frac{\delta' dS}{4\pi} \int_0^{\frac{\pi}{2}} \cos\theta \sin\theta d\theta \int_0^{2\pi} d\phi \int_0^c dr$$

$$\int_0^{\frac{\pi}{2}} \cos\theta \sin\theta d\theta = \frac{1}{2}\int_0^{\frac{\pi}{2}} \sin 2\theta d\theta = \frac{1}{2}$$

$$= \frac{\delta' c}{4} dS$$

- Energy balance method

  - Total energy during 1 sec. (incident on unit area of surface)

$$I' = \frac{\delta' c}{4}$$

  - The energy balance in the perfect reverberant room (during 1 sec.)

$$W(1-\bar{\alpha}) = I' S\bar{\alpha}$$

The acoustic power reflected one time from the surface
= The acoustic power absorbed into the surface

$$W(1-\bar{\alpha}) = \frac{\delta' c}{4} S\bar{\alpha} \quad \Rightarrow \quad \therefore \ \delta' = \frac{4W}{c} \frac{(1-\bar{\alpha})}{S\bar{\alpha}} = \frac{4W}{cR}$$

$$\bar{\alpha} = \frac{\sum S_i \alpha_i}{\sum S}$$

: the average Sabine absorptivity

$$R = \frac{S\bar{\alpha}}{(1-\bar{\alpha})}$$

: Room constant

- Sound pressure level in room
  - Acoustic energy density at the distance r from the sound source

$$\delta = \frac{QW}{4\pi r^2 c} + \frac{4W}{cR}$$

$$\delta = \frac{p^2}{\rho c^2} \quad \Rightarrow \quad p^2 = W\rho c\left[\frac{Q}{4\pi r^2} + \frac{4}{R}\right]$$

$$SPL = 10\log\left(\frac{p}{p_o}\right)^2 = 10\log W + 10\log \rho c + 10\log\left[\frac{Q}{4\pi r^2} + \frac{4}{R}\right]$$

$$= 10\log\left(\frac{W}{W_o}\right) + 10\log\left[\frac{Q}{4\pi r^2} + \frac{4}{R}\right] + 10\log W_o - 10\log p_o + 10\log \rho c$$

In normal condition : $\rho c \cong 407 \quad [kg/m^2 s] \quad \Rightarrow \quad 10\log W_o - 10\log p_o + 10\log \rho c \approx 0$

$$SPL = PWL + 10\log\left[\frac{Q}{4\pi r^2} + \frac{4}{R}\right]$$

1.3 Sound Field

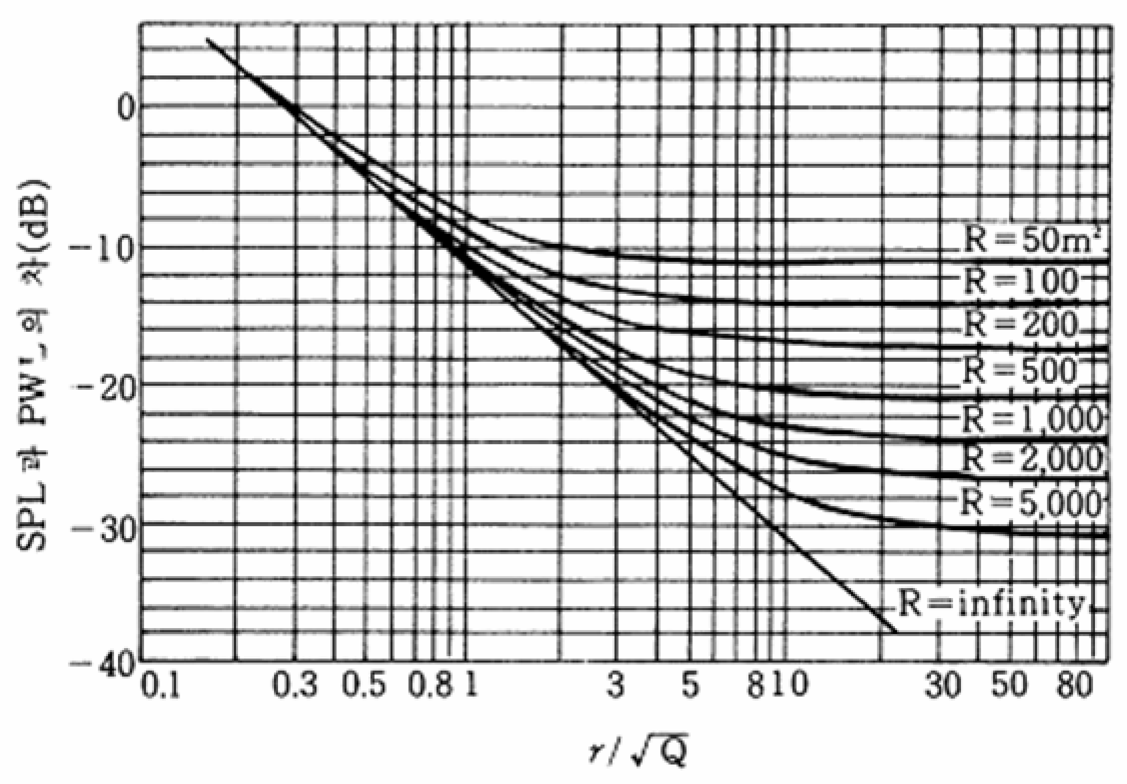
- Class of room according to average absorptivity

$\overline{\alpha} = 0.99$: anechoic room

$\overline{\alpha} = 0.5$: dead room

$\overline{\alpha} = 0.1$: quasi-reverberant room

$$SPL = PWL + 10\log\left[\frac{Q}{4\pi r^2} + \frac{4}{R}\right]$$



- Sound state equation in room

Acoustic energy density $\Rightarrow$ Turn on Acoustic power W

$$\delta \quad \underset{dt}{\Rightarrow} \quad \delta + d\delta$$

From the energy balance,

$$\delta V + Wdt - \frac{c\delta}{4}\cdot\bar{\alpha}Sdt = V(\delta + d\delta) \quad \Rightarrow \quad Wdt = Vd\delta + \frac{c\delta}{4}\cdot\bar{\alpha}Sdt$$

If the above equation is divided by Vdt,

$$\frac{W}{V} = \frac{d\delta}{dt} + \frac{c\bar{\alpha}S}{4V}\delta$$

$$A = \frac{d\delta}{dt} + B\delta \qquad A = \frac{W}{V}, \quad B = \frac{c\bar{\alpha}S}{4V}$$

$$\frac{d\delta}{A - B\delta} = dt \quad \Rightarrow \quad -\frac{1}{B}\ln(A - B\delta) = t + C$$

$$\ln(A - B\delta) = -Bt - BC \quad \Rightarrow \quad A - B\delta = e^{-Bt}\cdot e^{-BC}$$

$$Let \quad C' = e^{-BC} \qquad \delta = \frac{1}{B}(A - C'e^{-Bt})$$

- Sound state equation

$$\delta = \frac{4V}{c\bar{\alpha}S}\left(\frac{W}{V} - C'e^{-\frac{c\bar{\alpha}S}{4V}t}\right)$$

- Sound state equation in room

$$\delta = \frac{4V}{c\bar{\alpha}S}\left(\frac{W}{V} - C' e^{-\frac{c\bar{\alpha}S}{4V}t}\right)$$

1) When the source is turned on : $t = 0, \quad \delta = 0$ ➡ $C' = \frac{4W}{c\bar{\alpha}S}$

$$\delta = \frac{4V}{c\bar{\alpha}S}\left(1 - e^{-\frac{c\bar{\alpha}S}{4V}t}\right)$$: Growing sound equation

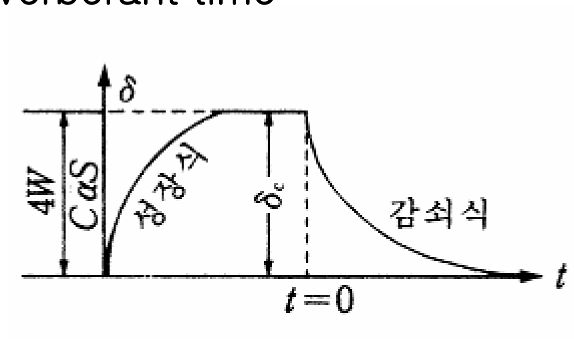
2) Steady-state condition : $t = \infty, \quad \delta = \delta_c$

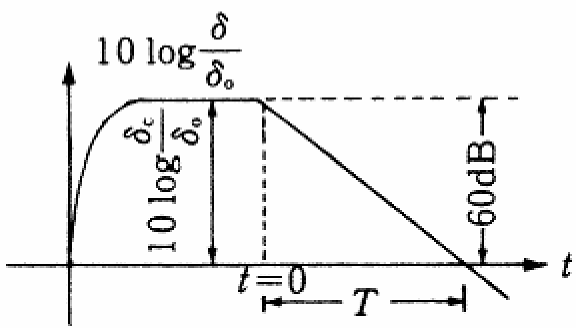
$$\delta_c = \frac{4V}{c\bar{\alpha}S}$$: Steady-state equation

3) When the source is turned off in steady-state : $t = 0, \quad W = 0 \quad and \quad \delta = \delta_c$

$$\delta = \delta_c e^{-\frac{c\bar{\alpha}S}{4V}t}$$: Decreasing sound equation ➡ $C' = -\delta_c$

- Reverberant time





- <u>Sabine's Equation</u>

$$\delta = \delta_c e^{-\frac{c\overline{\alpha}S}{4V}t} \qquad 10\log\left(\frac{\delta_c}{\delta}\right) = (10\log e)\frac{c\overline{\alpha}S}{4V}t$$

Reverberant time T : defined as the time required for the energy density or level of the sound to drop by 10^{-6} or 60 dB.

$$60 = 10\log\left(\frac{\delta_c}{\delta}\right) = (10\log e)\frac{c\overline{\alpha}S}{4V}T \quad \Rightarrow \quad \therefore \; T = \frac{4V}{c\overline{\alpha}S}\frac{60}{(10\log e)} = \frac{0.161\cdot V}{\overline{\alpha}S}$$

$$T = \frac{0.161\cdot V}{A} \quad [\text{sec.}] \qquad A = \overline{\alpha}S \quad [m^2, \textit{metric sabine}]$$

: the total sound absorption of the room

- Eyring's Equation

  - Sabine's equation agrees with experimental result when the average absorptivity is low.
    But it does not so when the average absorptivity is high. $Ex) \quad \bar{\alpha} = 1 \quad \Rightarrow \quad T \neq 0$

  - Eyring's equation is based on the mean free path between reflection.

    The mean distance traveled between successive reflections from the walls of rectangular enclosure is $y = \dfrac{4V}{S}$

    so that the number of reflections per second is $N = \dfrac{cS}{4V}$

Per second

$N = \dfrac{c}{y}$: the number of reflections

$E = \delta V \dfrac{c}{y}$: the total acoustic energy supplied to the total surface

$E = \dfrac{\delta c S}{4}$: the total acoustic energy incident on the total surface

$\Rightarrow \quad \delta V \dfrac{c}{y} = \dfrac{\delta c S}{4} \quad \Rightarrow \quad y = \dfrac{4V}{S}$

- Eyring's Equation

The number of reflections for t seconds

$$N = \frac{cS}{4V} t$$

Acoustic energy density $\Rightarrow$ Turn off Acoustic power W

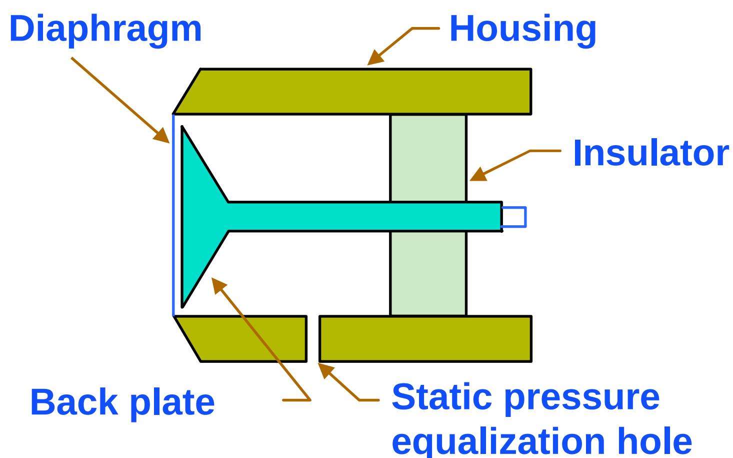
$$\delta_c \underset{t}{\Rightarrow} \delta = \delta_c (1-\overline{\alpha})(1-\overline{\alpha}) \cdots = \delta_c (1-\overline{\alpha})^{\frac{cS}{4V}t}$$

$$\frac{\delta_c}{\delta} = (1-\overline{\alpha})^{\frac{cS}{4V}T} \quad \Rightarrow \quad 60 = 10\log\left(\frac{\delta_c}{\delta}\right) = -\frac{cS}{4V} T \log(1-\overline{\alpha})$$

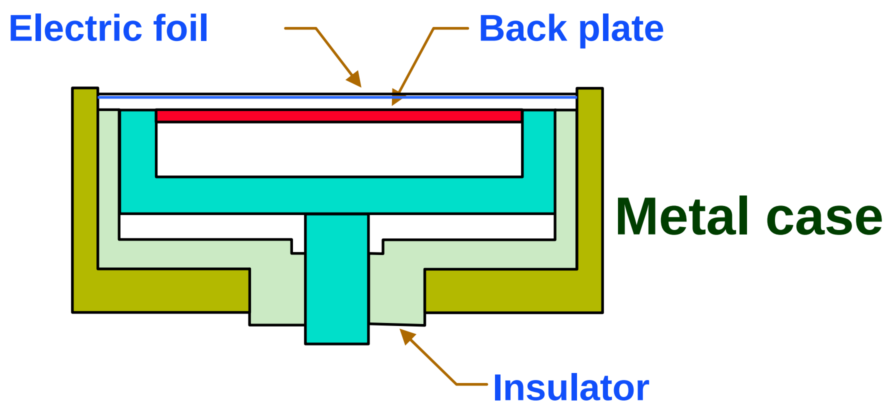
$$\therefore \quad T = \frac{24V}{-cS\log(1-\overline{\alpha})} = \frac{0.071V}{-S\log(1-\overline{\alpha})} = \frac{0.071V\log_e 10}{-S\log_e(1-\overline{\alpha})} = \frac{0.163V}{-S\log_e(1-\overline{\alpha})}$$

$$\therefore \quad T = \frac{0.161V}{-S\log_e(1-\overline{\alpha})} = \frac{0.161V}{-S\{2.30\log(1-\overline{\alpha})\}} \quad [\text{sec.}]$$

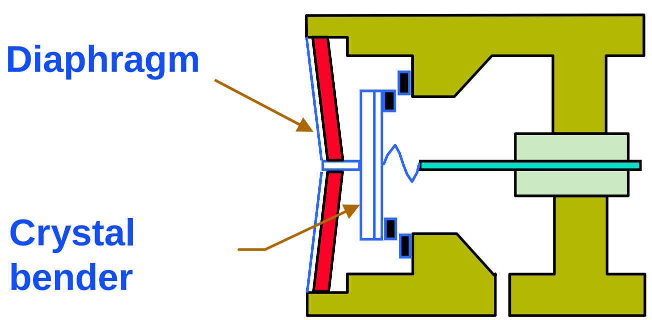
- Types of Microphones



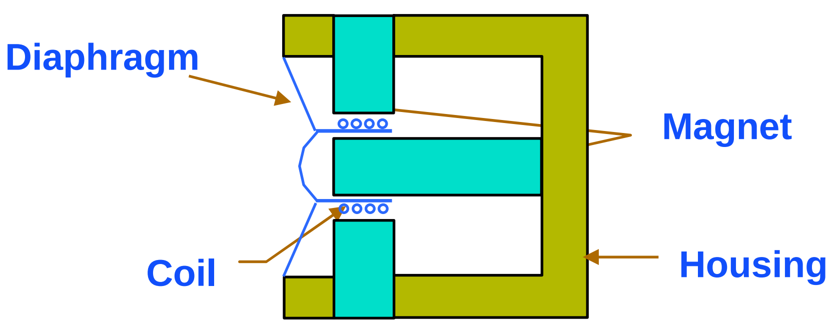
Condenser Microphone



Electric Microphone

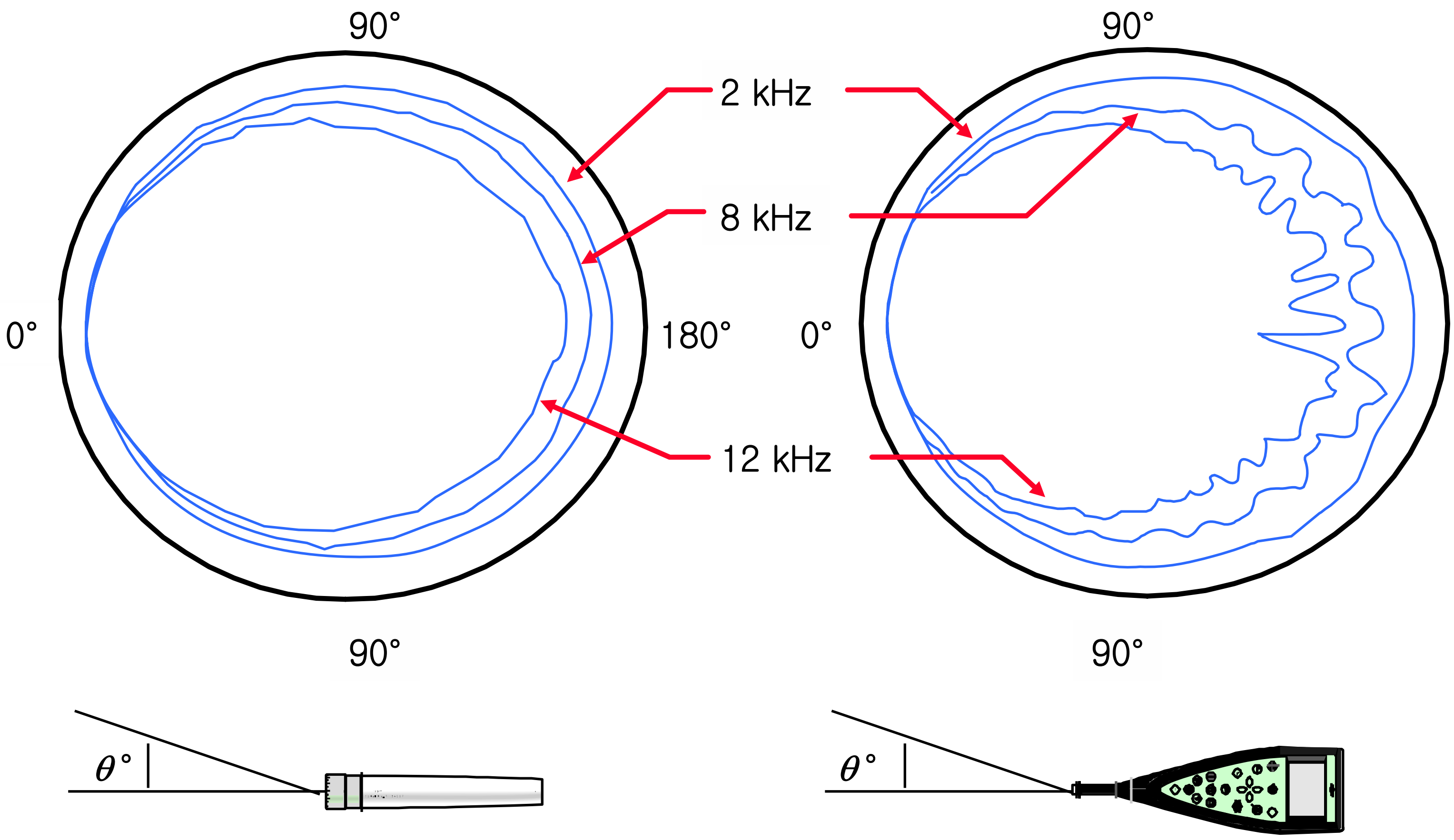


Piezoelectric Microphone

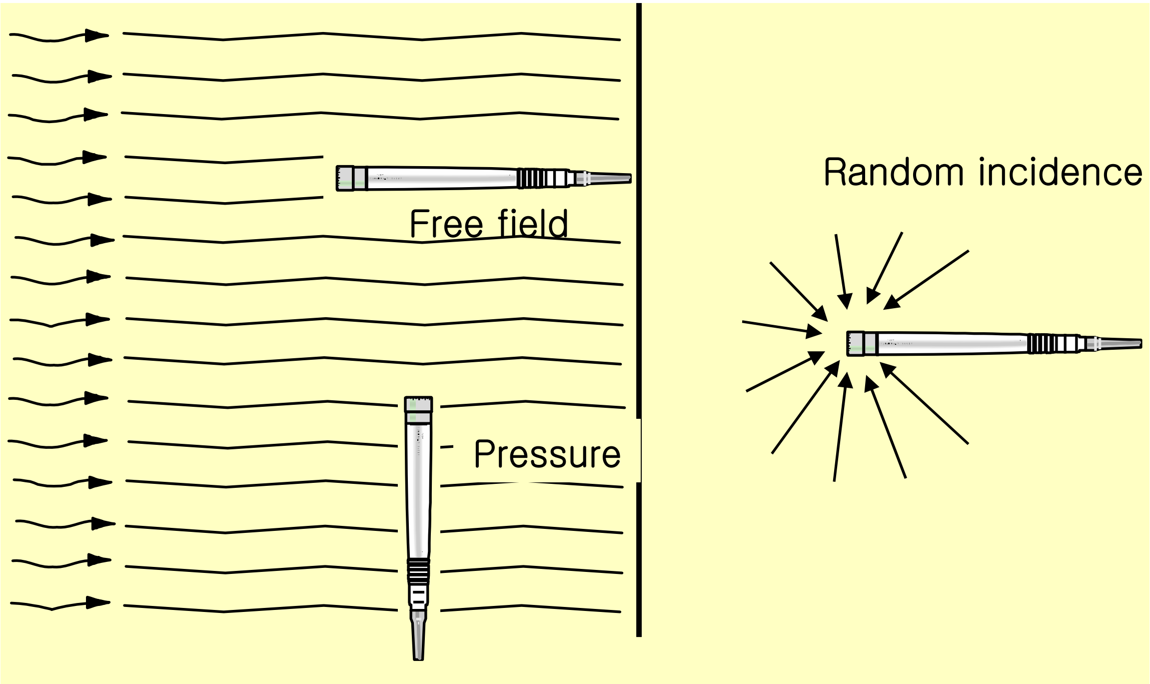


Dynamic Microphone

- Directional Characteristics



- Types of Microphones

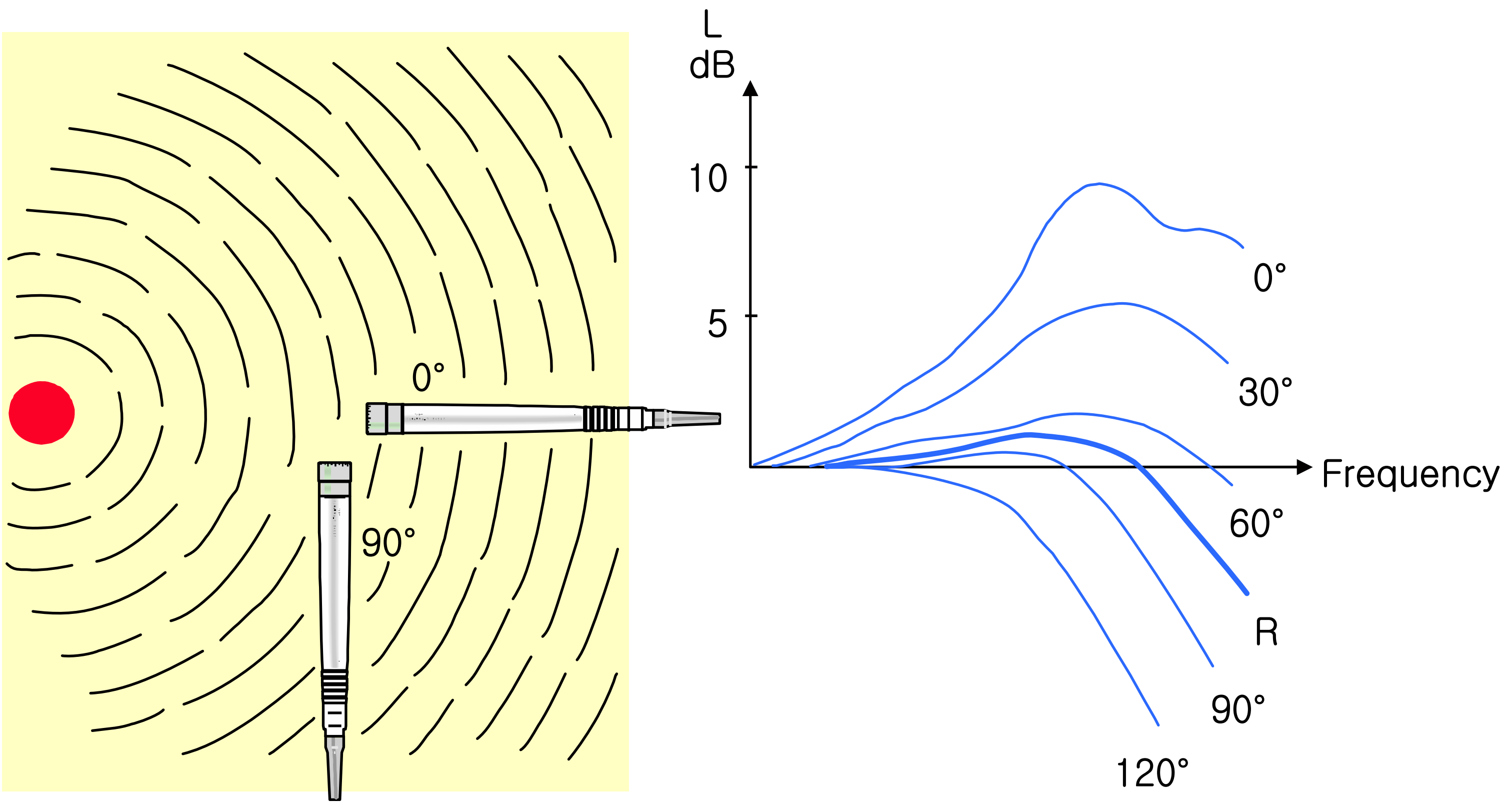


Free field microphones have uniform frequency response for the sound pressure that existed before the microphone was introduced into the sound field. It is of importance to note that any microphone will disturb the sound field, but the free field microphone is designed to compensate for its own disturbing presence.

The pressure microphone is designed to have a uniform frequency response to the actual sound level present. When the pressure microphone is used for measurement in a free sound field, it should be oriented at a 90° angle to the direction of the sound propagation, so that the sound grazes the front of the microphone.

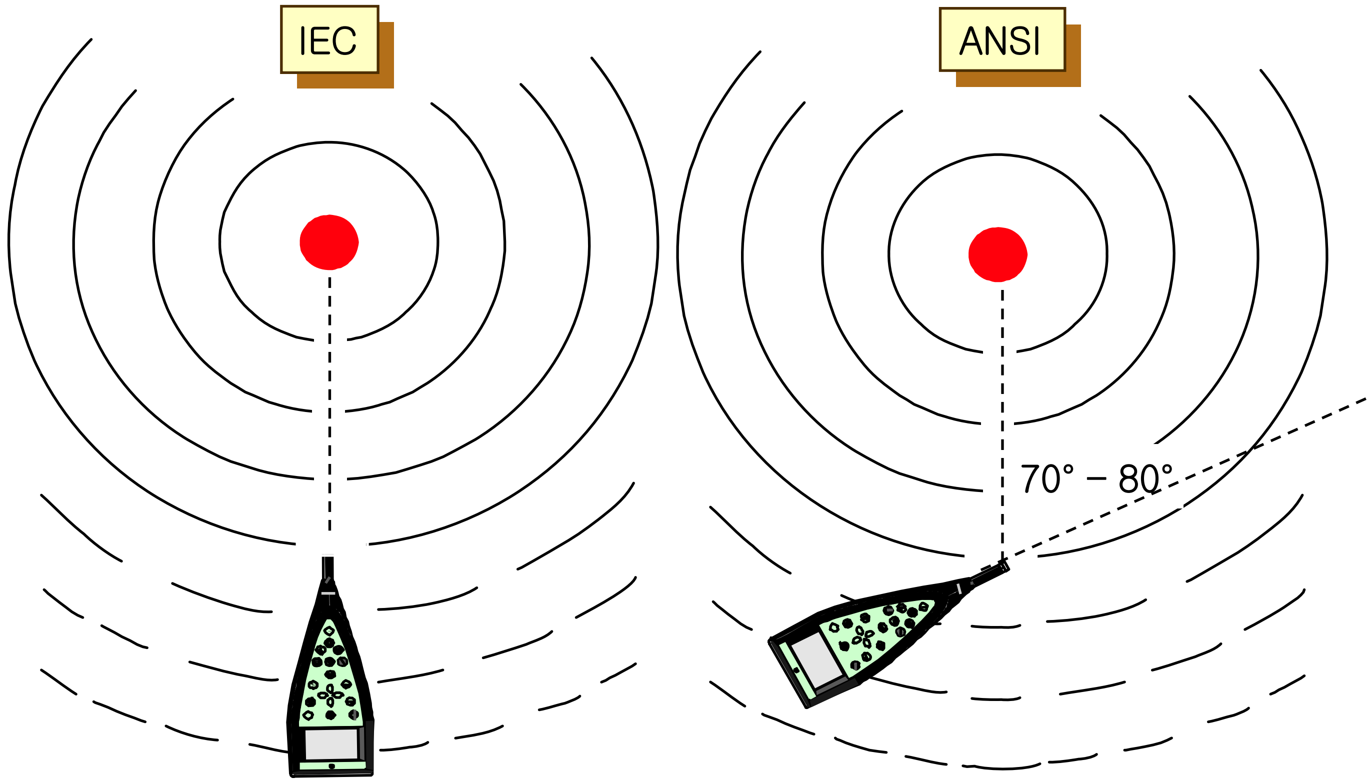
The random incidence microphone is designed to respond uniformly to signals arriving simultaneously from all angles. When used in a free field it should be oriented at an angle of 70° – 80° to the direction of propagation.

- Free Field Correction



The biggest increase in "sensitivity" is obtained when the sound wave comes from a direction perpendicular to the diaphragm (defined as 0° incidence). At all other angles the increase will be less pronounced as shown here. The curve labeled R, which stands for random incidence, is a calculated average response to sound arriving with equal probability from all directions.

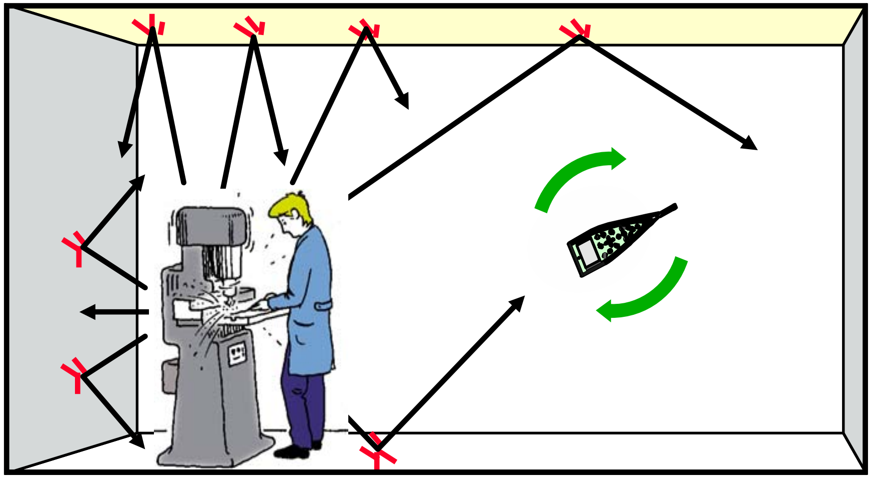
- Measuring in Accordance with Standards



This means that when sound level measurements are made in accordance with IEC a free field microphone should be used, and the sound level meter pointed towards the source (0° incidence). When measurements are made in accordance with ANSI a random incidence microphone should be used, and the sound level meter held at an angle of 70° – 80° to the direction of incidence.

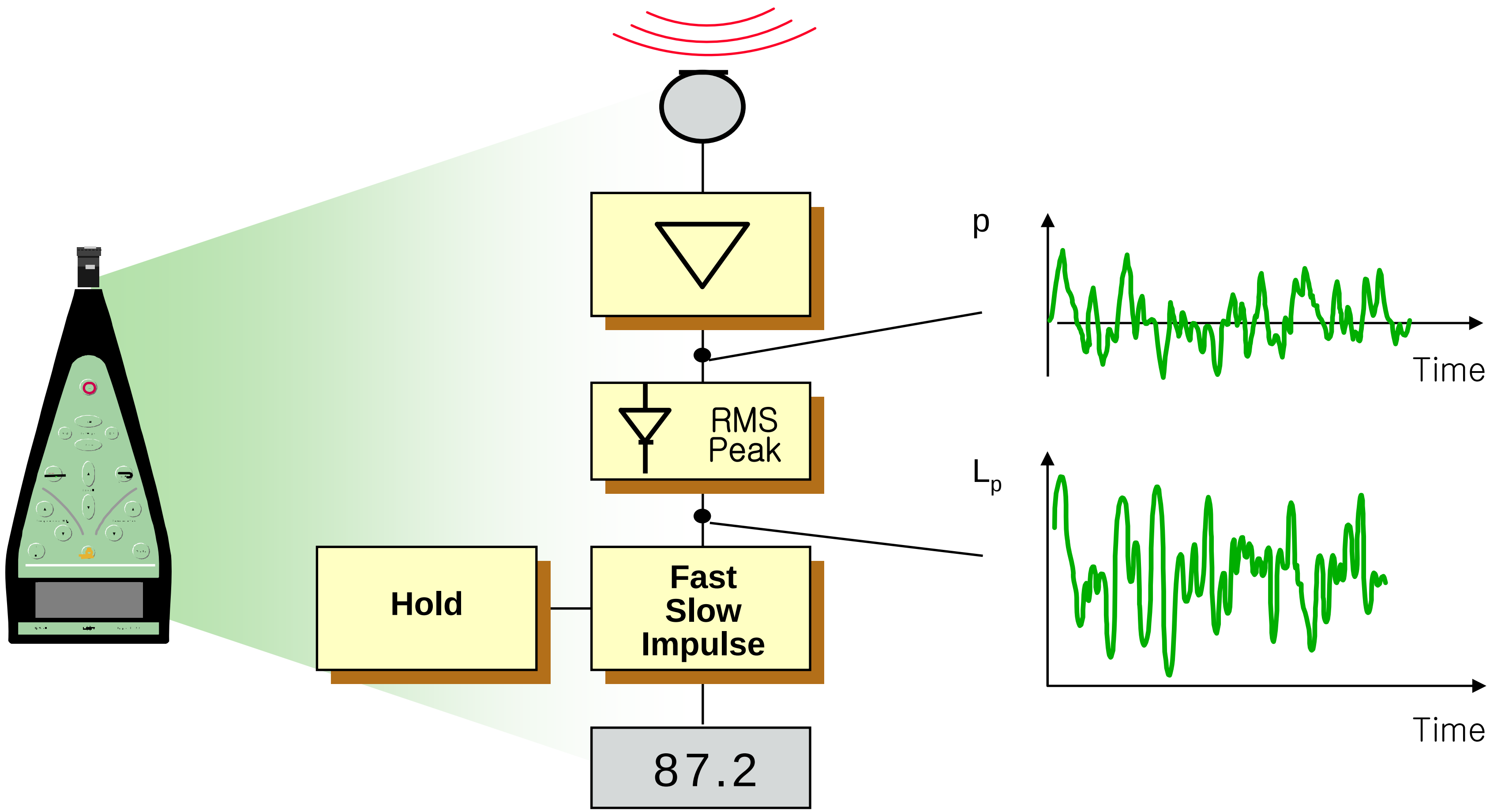
- Use of Microphones

The free field microphone is used in all applications where the sound mainly comes from one direction.

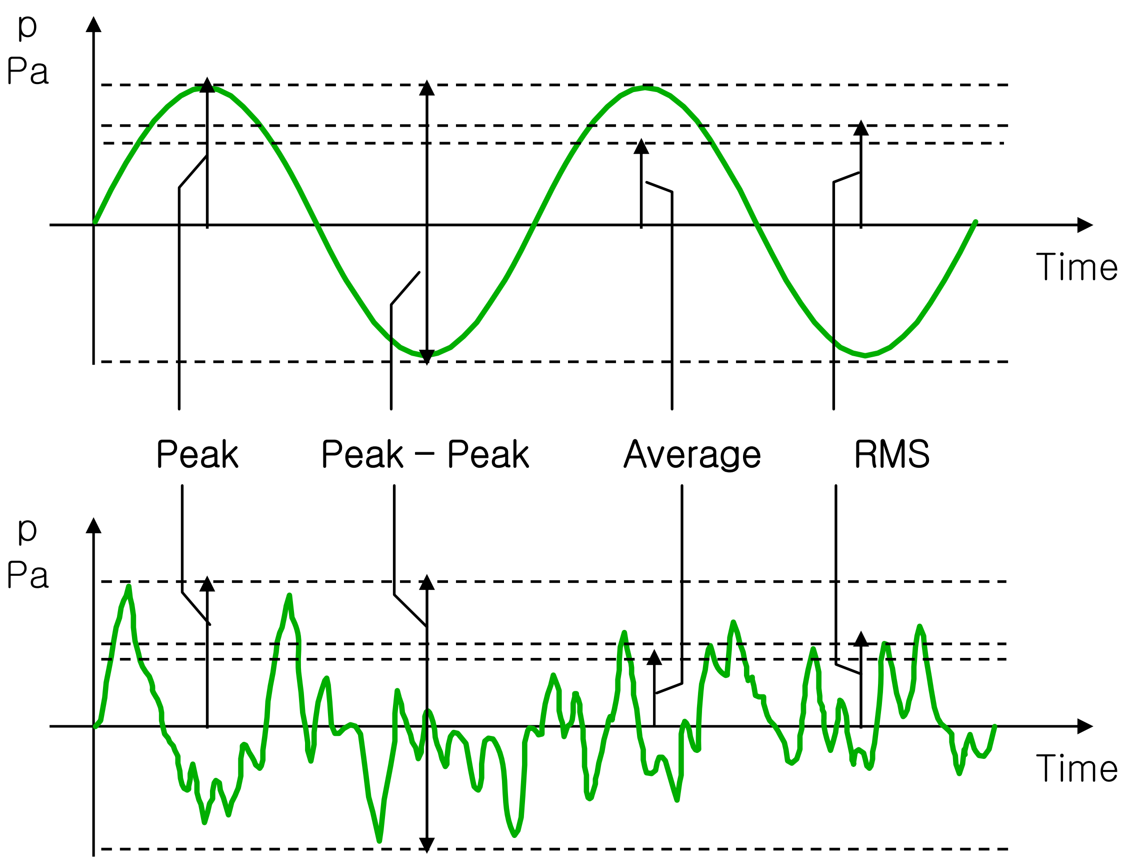


The random incidence microphone should not only be used for measurement in reverberation chambers, but in all situations where the sound field is a diffuse sound field e.g. in many indoor situations where the sound is being reflected by walls, ceilings, and objects in the room.

- The Sound Level Meter



- Sound Level Parameters

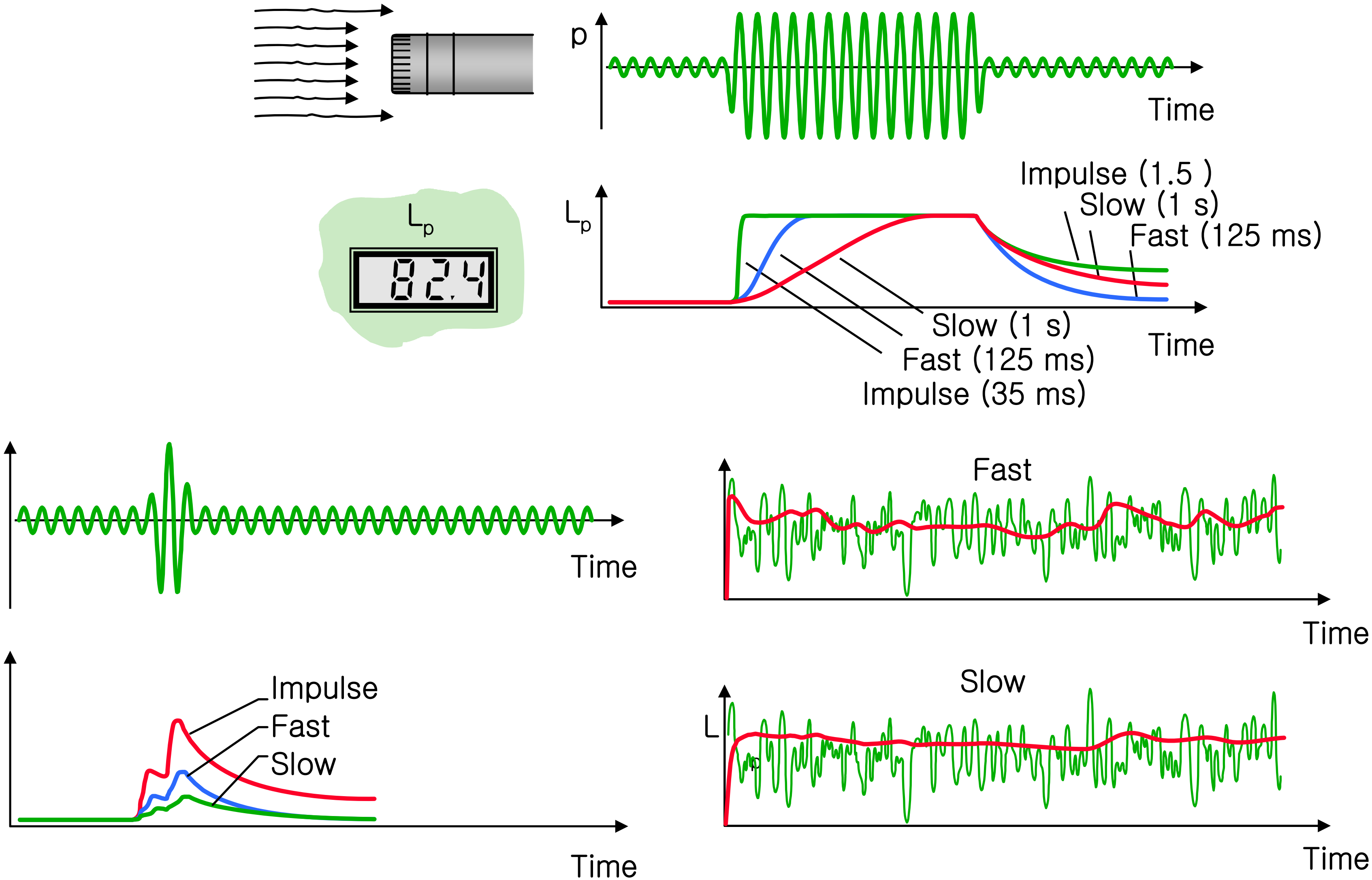


$$\text{RMS} = \sqrt{\frac{1}{T}\int_0^T x^2(t)dt}$$

$$\text{Average} = \frac{1}{T}\int_0^T |x|dt$$

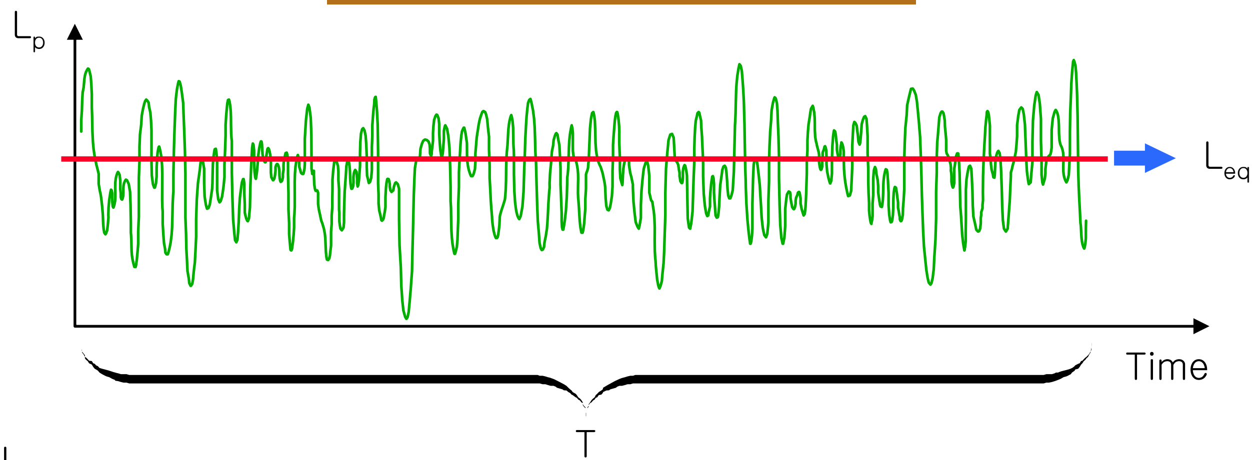
$$\text{Crest factor} = \frac{\text{Peak}}{\text{RMS}}$$

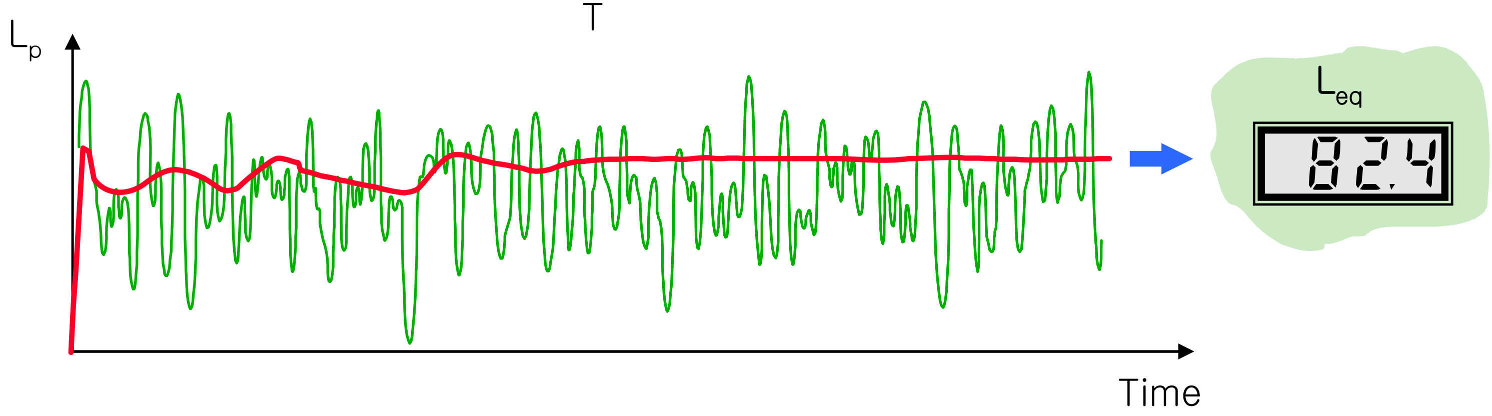
- Time weighting



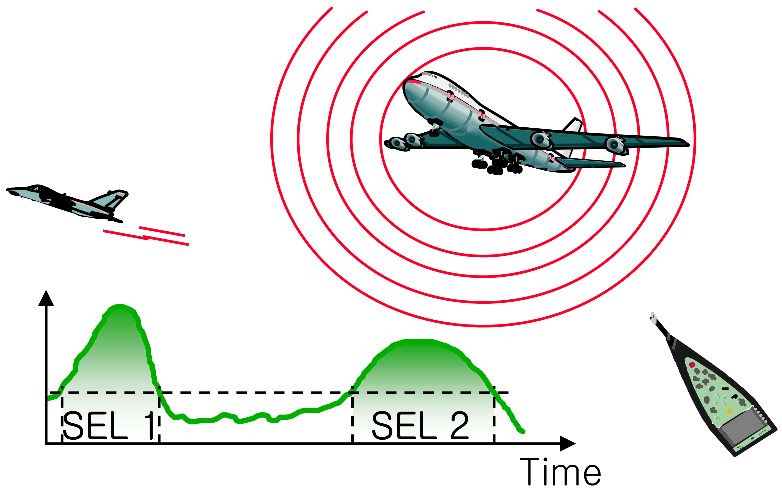
- Equivalent Level, L_{eq}

$$L_{eq} = 10\log_{10}\frac{1}{T}\int_0^T\left(\frac{p(t)}{p_0}\right)^2 dt$$

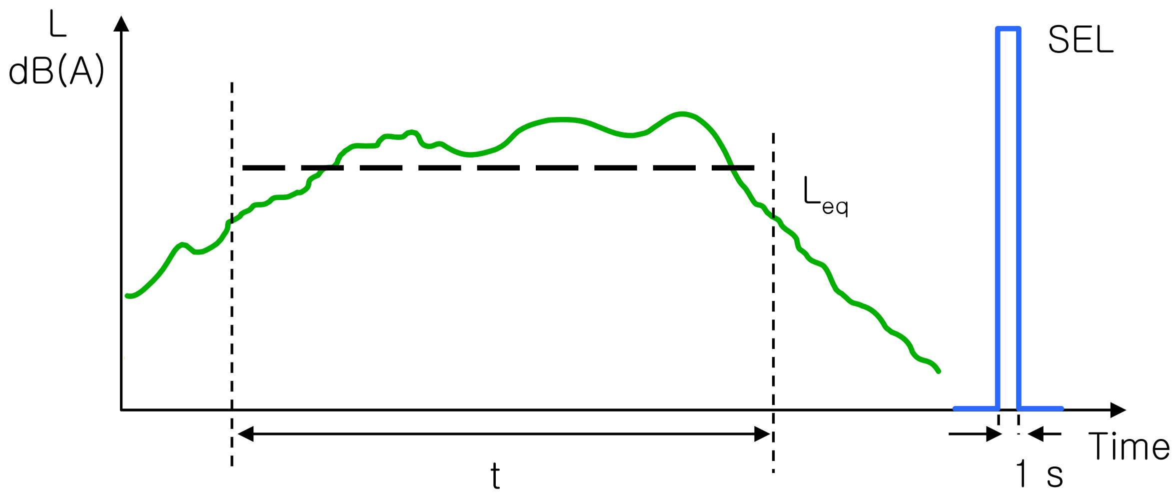




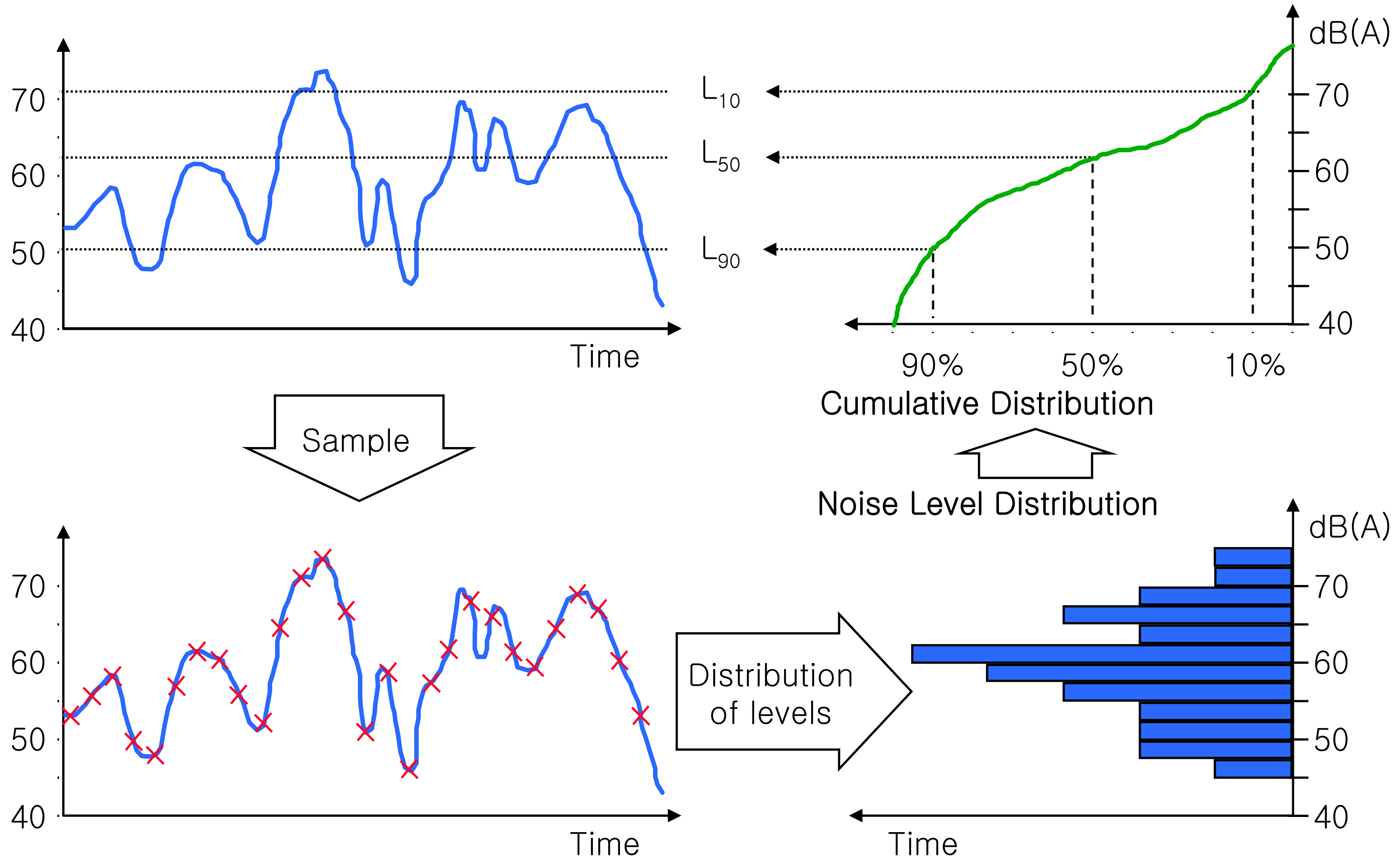
- Sound Exposure Level and its Origin



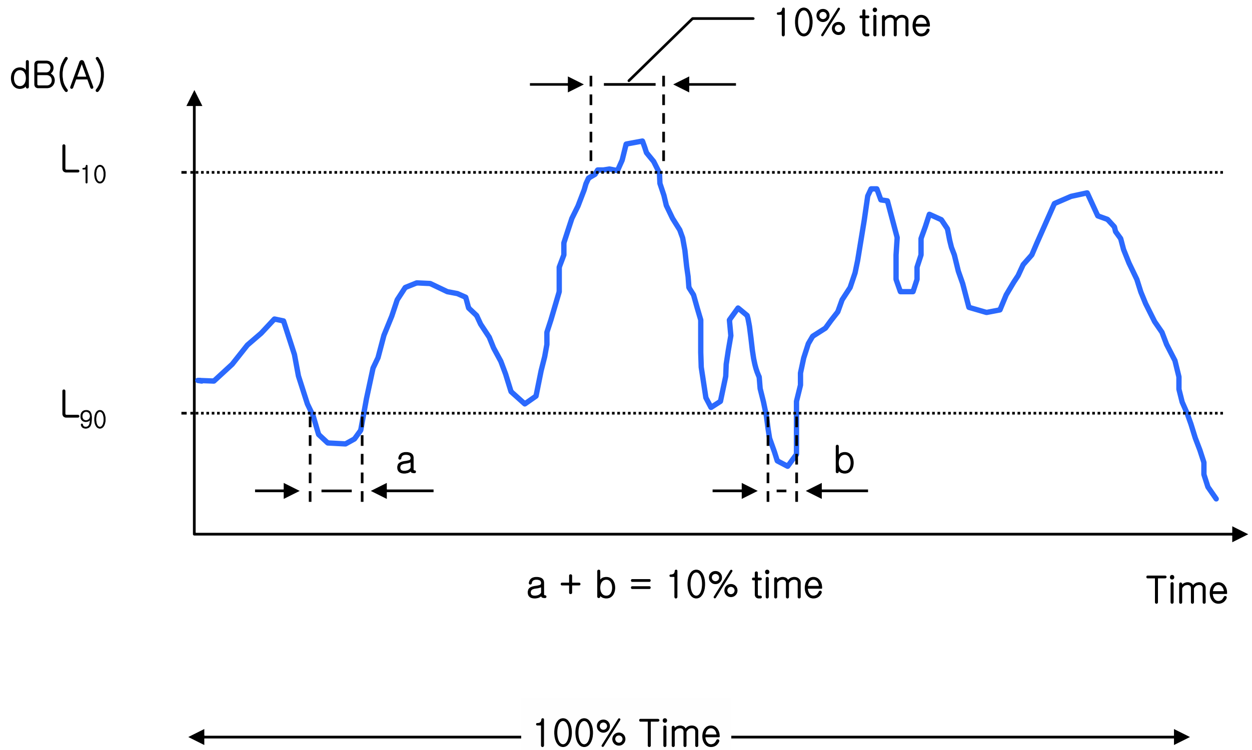
$$SEL = L_{eq} + 10 \log\left(\frac{t}{1s}\right)$$



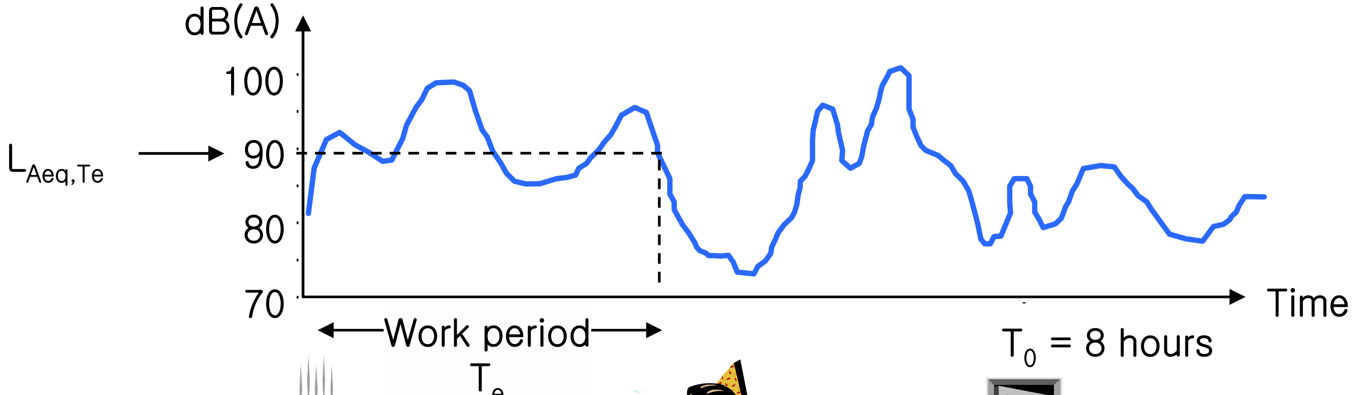
- Statistical Analysis of Noise Levels



- Percentile Levels

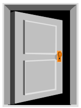


- Daily Personal Noise Exposure, $L_{EP,d}$



Morning

Lunch



Go Home

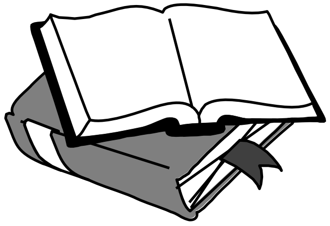
$$L_{EP,d} = L_{Aeq,Te} + 10\log_{10}\frac{T_e}{T_0}$$

Example:

$L_{Aeq,Te}$ = 89.2 dB and T_e = 4 hours

$$L_{EP,d} = 89.2 + 10\log_{10}\frac{4}{8} = 89.2 - 3 = \underline{86.2\ dB}$$

- Daily Personal Noise Exposure, $L_{EP,d}$



- ISO 9612
  Guidelines for the measurement and assessment of exposure to noise in a working environment
  International

- ISO 1999
  Determination of occupational noise exposure and estimation of noise-induced hearing impairment
  International

- IEEC Directive EEC/86/188
  The protection of workers from the risk related to the exposure to noise at work
  The European Union

- OSHA
  Occupational Safety and Health Act
  USA and others

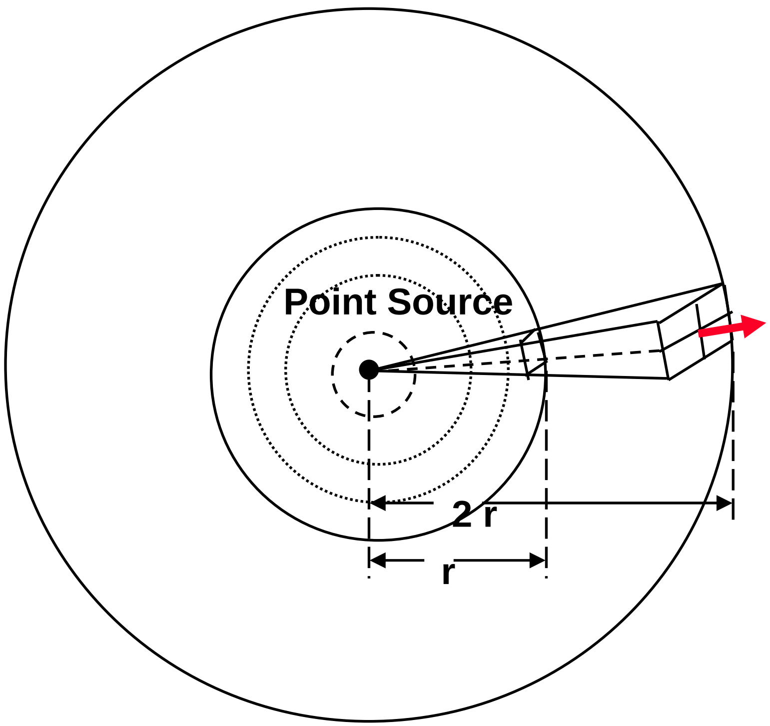
1.5 Sound Intensity

- Agenda
  - Sound Intensity
    - Theory, Limitations
  - Sound Intensity Instrumentation
    - 2144, 2260
  - Sound Power measurements
    - ISO 9614-2, 2260 benefits
  - Noise Source Location

- Definition of Sound Intensity:
  - Sound Intensity is the time-averaged rate of energy flow per unit area
  - The sound intensity vector equals the time-averaged product of the instantaneous pressure and the corresponding instantaneous particle velocity at the same position:

$$\vec{I} = \overline{p(t) \cdot \vec{v}(t)} \quad \left[W / m^2\right]$$

- Sound Power from Sound Intensity

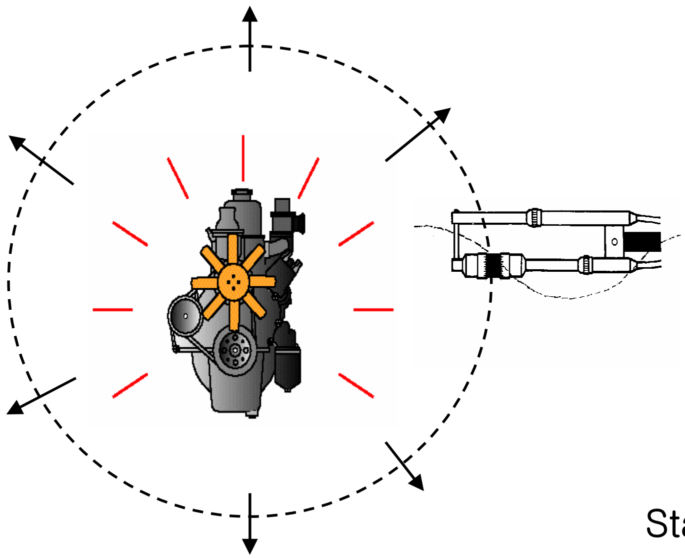


$$P = \int_s \vec{I} \cdot \vec{ds}$$

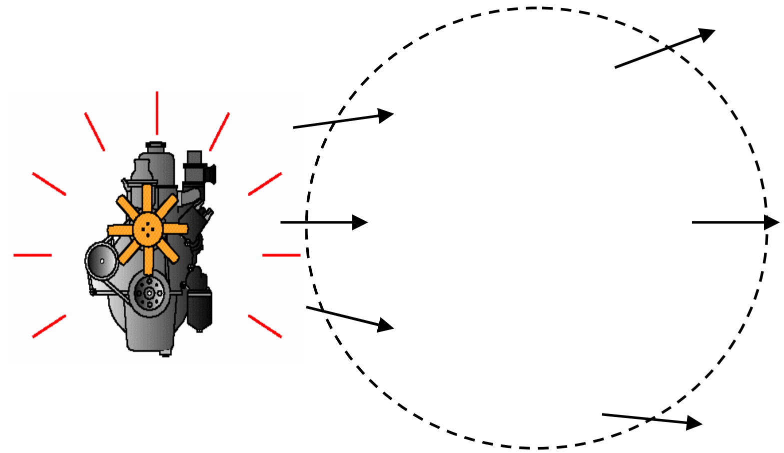
$$I = \frac{P}{4 \pi r^2}$$

- Effects of External Sources

$$\int_S \vec{\mathbf{I}} \cdot d\vec{\mathbf{S}} = W$$



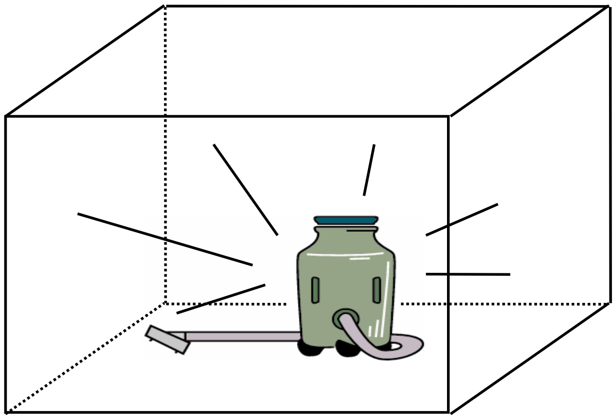
$$\int_S \vec{\mathbf{I}} \cdot d\vec{\mathbf{S}} = 0$$



Stationary Sources

A stationary bacground noise can be removed

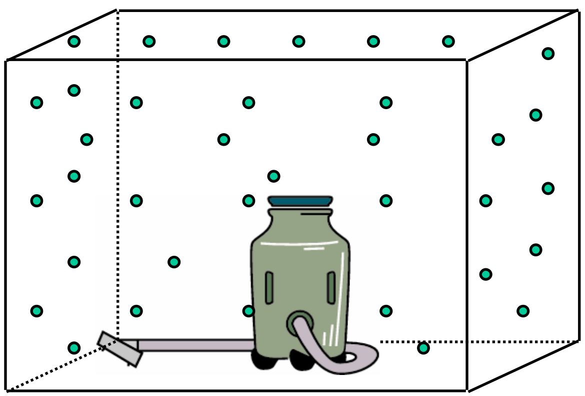
- Effects of External Sources



$$L_W = L_I + 10 \cdot \log \ S/S_0$$

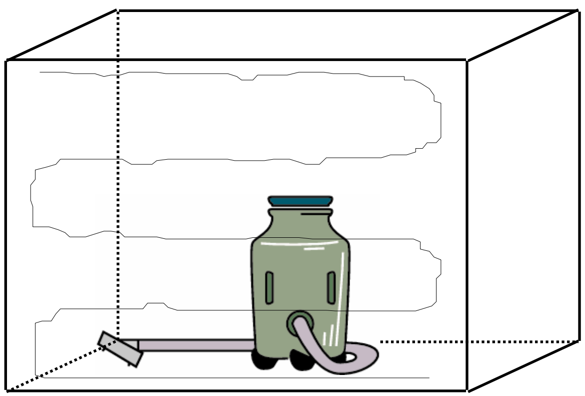
$S_0 = 1 \ m^2$

Point Measurements



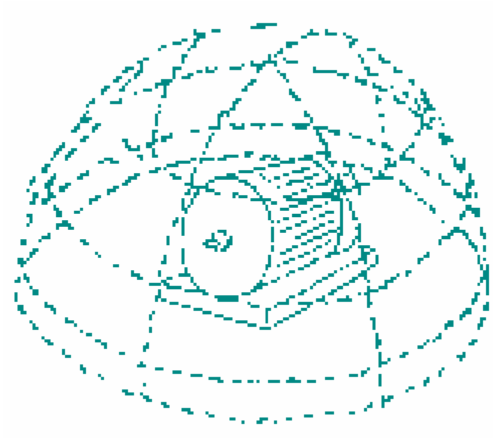
ISO 9614 Part 1

Sweeps

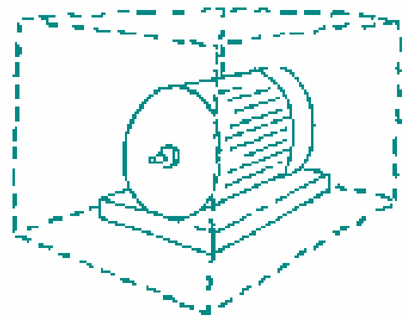


ISO 9614 Part 2

- Why Use Sound Intensity ?
  - Less sensitive to background noise
  - Gives directional information (This is useful when locating sources.)
  - Enables segmentation of source
  - Isolates the object under investigation so that other sources can remain in operation
  - Quality control of measurement (such as extraneous noise and repeatability)



Hemisphere



Box

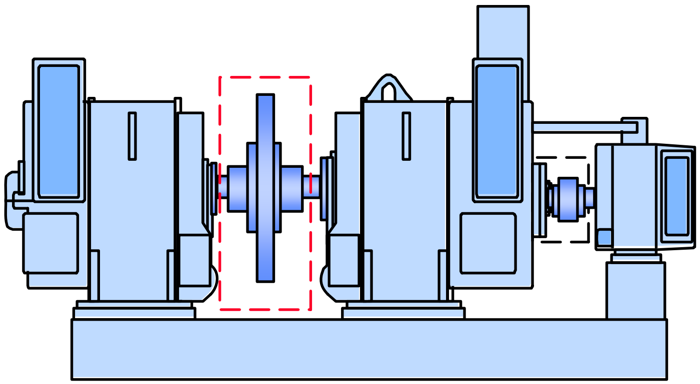
- Pros and Cons of Sound Intensity

Pro

- Less demands to measurement site
  ⇒ save site preparation costs
- Identifies causes of problem
  ⇒ saves time when solving problems

Contra

- More complex instrumentation
  ⇒ higher instrument and training costs
- More measurements
  ⇒ may increase measurement time



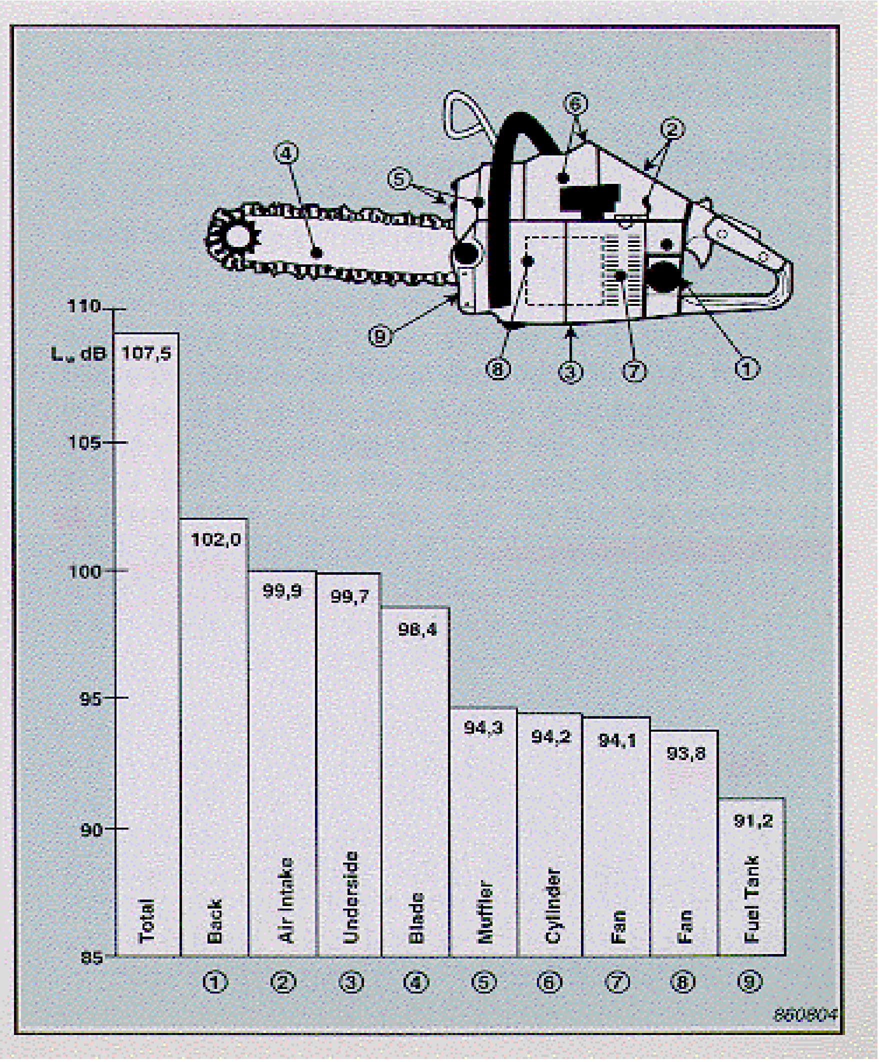
1.5 Sound Intensity

- <u>What is Sound Intensity used for?</u>

- Sound Power Determination
  - Non-standard survey (indication)
  - Measurement according to standard
  - Data gathering for modelling
- Noise Source Location
  - Source location
  - Source ranking
  - Noise mapping

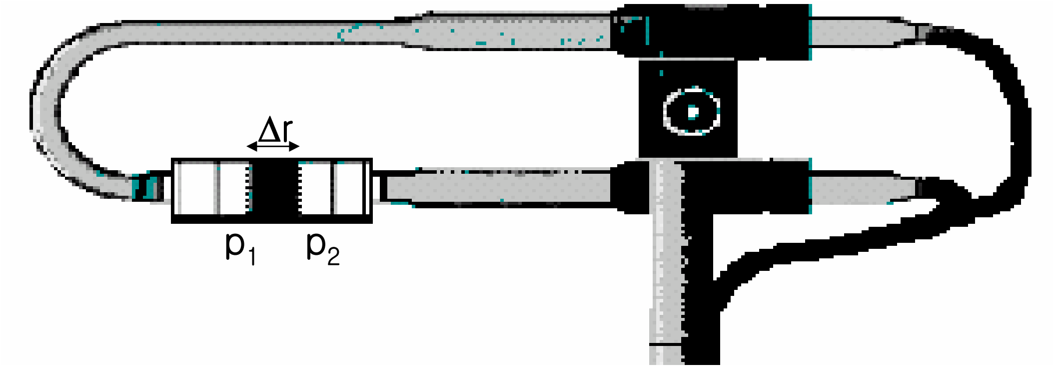
- Building Acoustics
  - Sound Reduction Index
  - Leakage detection
  - Sound absorption
- Sound Field Investigation

1.5 Sound Intensity



• How is Sound Intensity Measured

$$I_r = \overline{p(t) \cdot v_r(t)}$$



Sound Intensity in one direction "r" is measured with two closely spaced microphones

The average pressure is measured as: $$p = \frac{p_1 + p_2}{2}$$

The average particle velocity is measured as: $$v = -\frac{1}{\rho} \int \frac{p_2 - p_1}{\Delta r} dt$$

The Sound Intensity is: $$I = -\frac{p_1 + p_2}{2\rho\Delta r} \int (p_2 - p_1) dt$$

- Sound Intensity in Frequency Domain

Frequency Domain Formulation

$$I = -\frac{1}{\rho\omega\Delta r}\,\mathrm{Im}G_{AB}$$

ω is the angular frequency
$\mathrm{Im}[G_{AB}]$ is the imaginary part of the cross spectrum

- Proof

Cross Correlation function between the pressure and particle velocity is defined as

$$R_{pu}(\tau)=\lim_{T\to\infty}\frac{1}{T}\int_0^T p(t)u(t+\tau)dt$$

Hence, the mean intensity component in direction $\vec{n}$ is given by

$$I_n=\lim_{T\to\infty}\frac{1}{T}\int_0^T p(t)u(t)dt=R_{pu}(0)$$

Cross Spectral Density:

$$S_{pu}(f)=\int_{-\infty}^{\infty}R_{pu}(\tau)e^{-2\pi f\tau}d\tau \qquad R_{pu}(f)=\int_{-\infty}^{\infty}S_{pu}(f)e^{2\pi f\tau}df$$

and $$I_n=R_{pu}(0)=\int_{-\infty}^{\infty}S_{pu}(f)df$$

The Frequency distribution of the mean intensity component

$$I_n(f)=S_{pu}(f)+S_{pu}(-f)$$
$$=2\,\mathrm{Re}\{S_{pu}(f)\}=\mathrm{Re}\{G_{pu}(f)\}$$

$$\mathrm{Re}\{S_{pu}(f)\}=\mathrm{Re}\{S_{pu}(-f)\}$$
$$\mathrm{Im}\{S_{pu}(f)\}=-\mathrm{Im}\{S_{pu}(-f)\}$$

$$G_{pu}(f)=2S_{pu}(f) \quad f>0$$
$$G_{pu}(f)=S_{pu}(f) \quad f=0 \quad (\because \text{one value})$$
$$G_{pu}(f)=0 \quad f<0$$

- Proof

Finite different approximation

$$\frac{\left(P_1(f)-P_2(f)\right)}{d} \approx j\omega\rho_o U_n(f) \qquad P(f) \approx \frac{1}{2}\left[P_1(f)+P_2(f)\right]$$

Hence

$$G_{pu}(f) = -\frac{j}{2\rho_o\omega d}\lim_{T\to\infty}\frac{2}{T}\left[P_1^*(f)+P_2^*(f)\right]\left[P_1(f)-P_2(f)\right]$$

$$= -\frac{j}{2\rho_o\omega d}\left[G_{p1p1}(f)-G_{p2p2}(f)-G_{p1p2}(f)+G_{p2p1}(f)\right]$$

$$G_{p1p2}(f) = A + jB$$

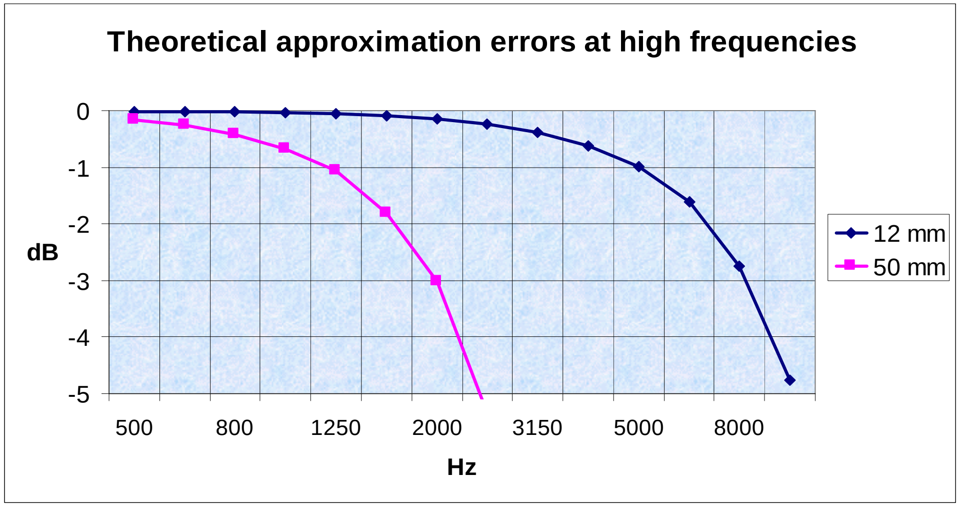
$$G_{p2p1}(f) = A - jB$$

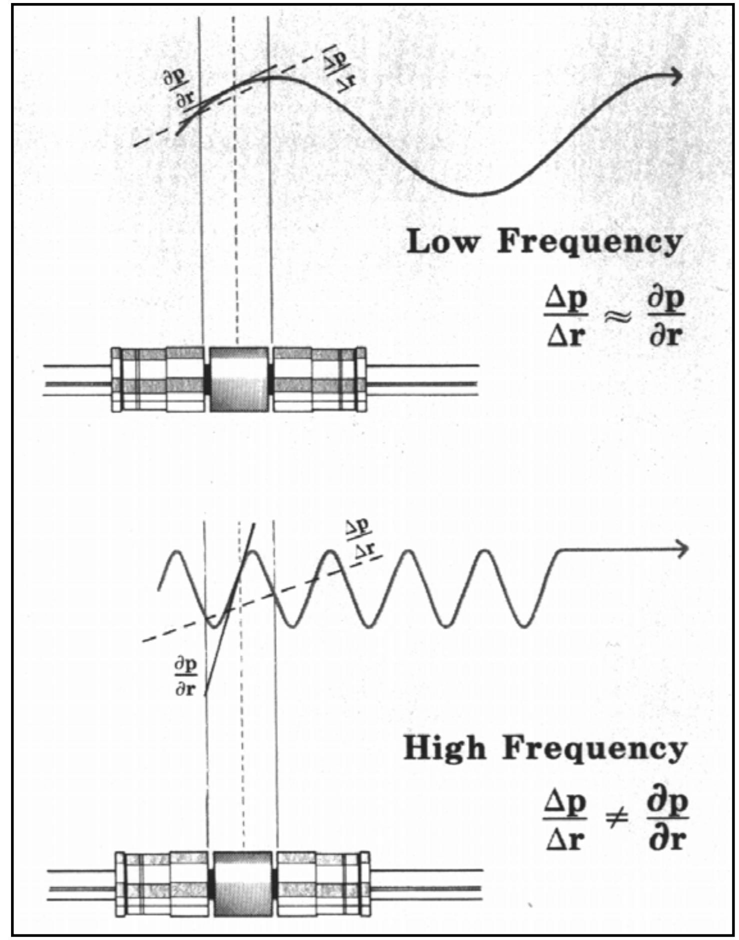
$$-G_{p1p2}(f)+G_{p2p1}(f) = -2jB$$

$$I_n(f) = \mathrm{Re}\left\{G_{pu}(f)\right\} \Rightarrow \boxed{I_n(f) = -\frac{1}{\rho_o\omega d}\,\mathrm{Im}\left[G_{p1p2}(f)\right]}$$

• Theoretical Limitations at High Frequencies

• Finite Difference Approximation Errors





Accuracy within 1dB:

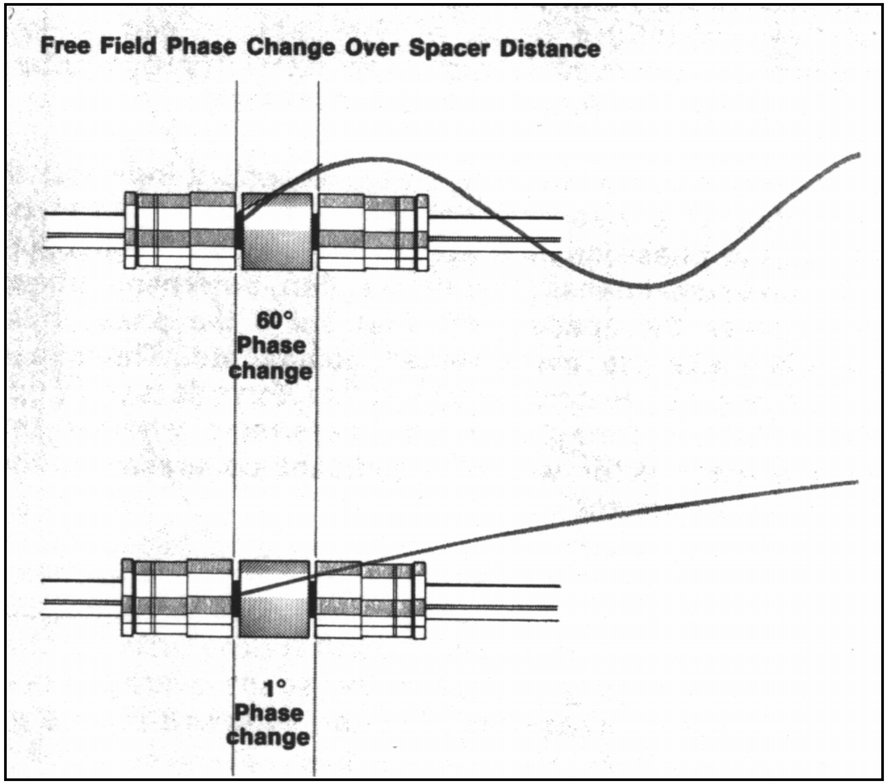
Spacer	Limit
50 mm:	up to 1.25 kHz
12 mm:	up to 5 kHz*)

*) without correction for resonance's at high frequency

• Limitations at Low Frequencies

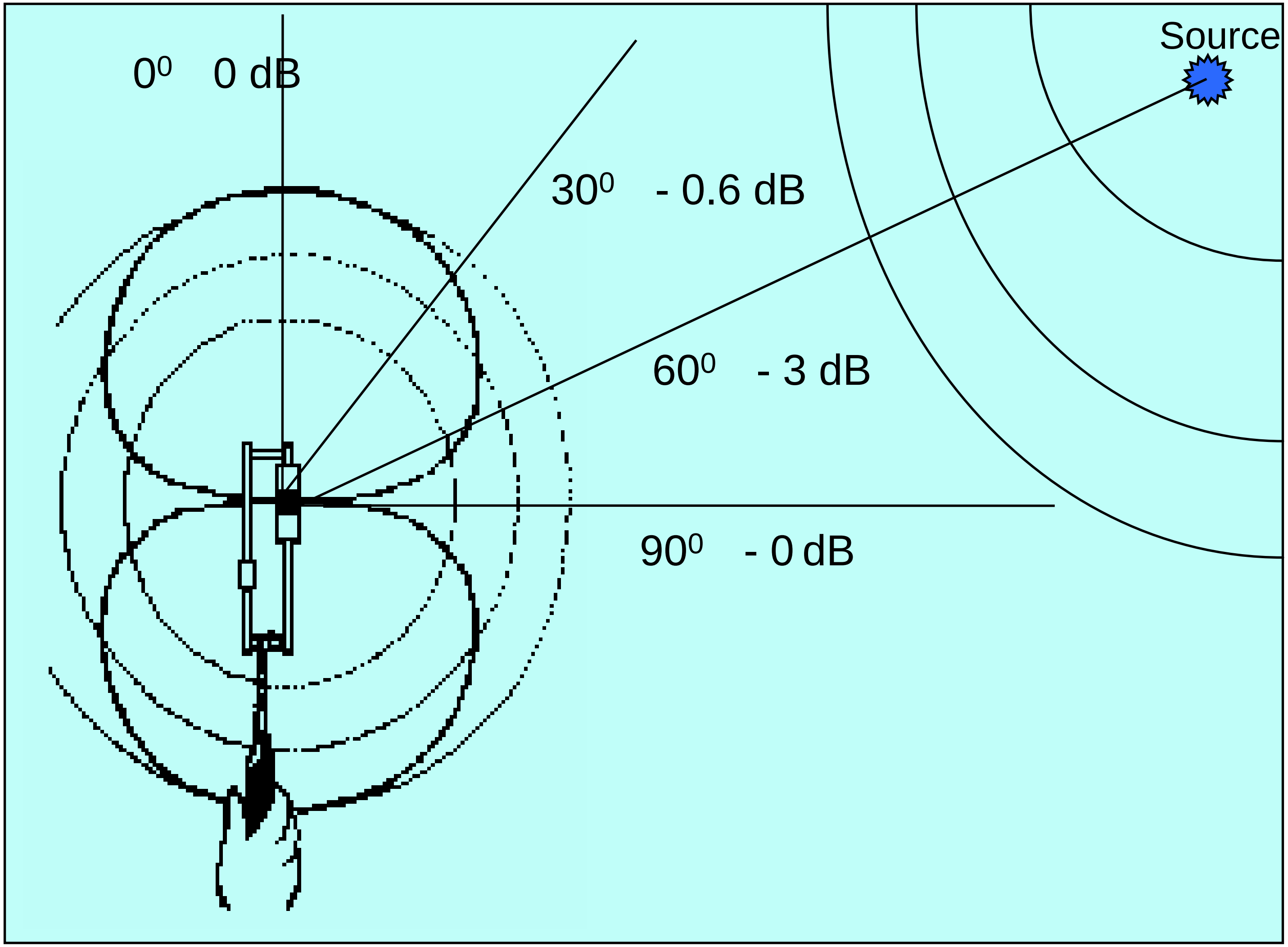
- The Sound Intensity is proportional to the phase change over the spacer
- A phase mismatch of the analyzing system introduces an error to the calculated sound intensity

The phase mismatch error is most severe at low frequencies



If the phase change is small, it is difficult to estimation the intensity.

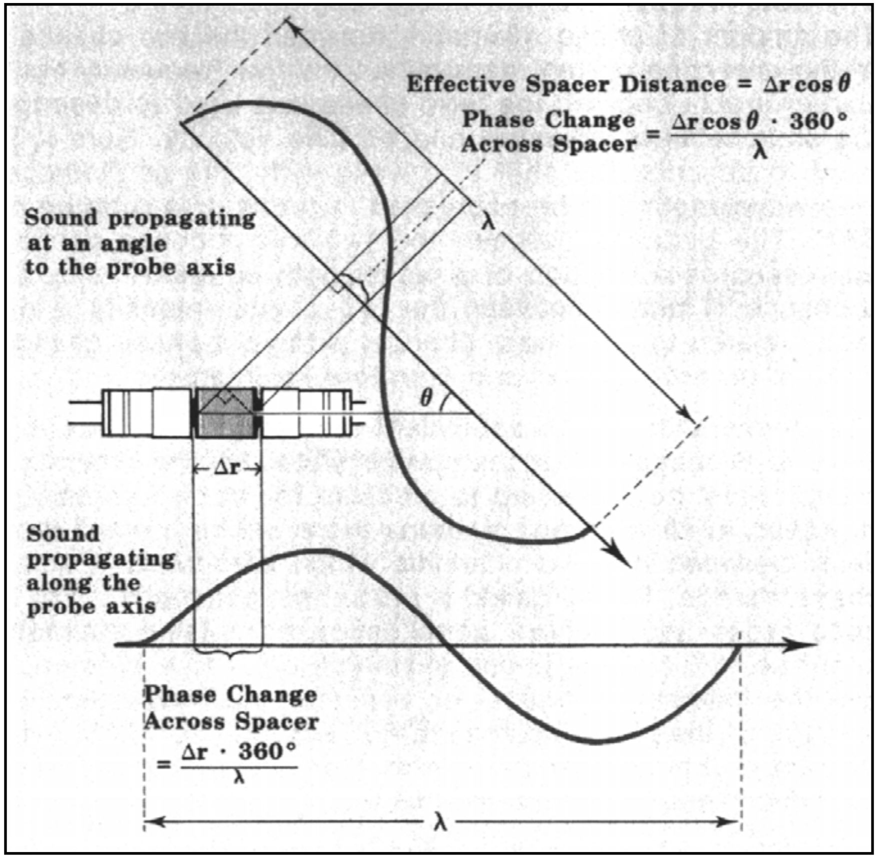
- Intensity Probe Directivity (Intensity)



- Pressure-Intensity Index

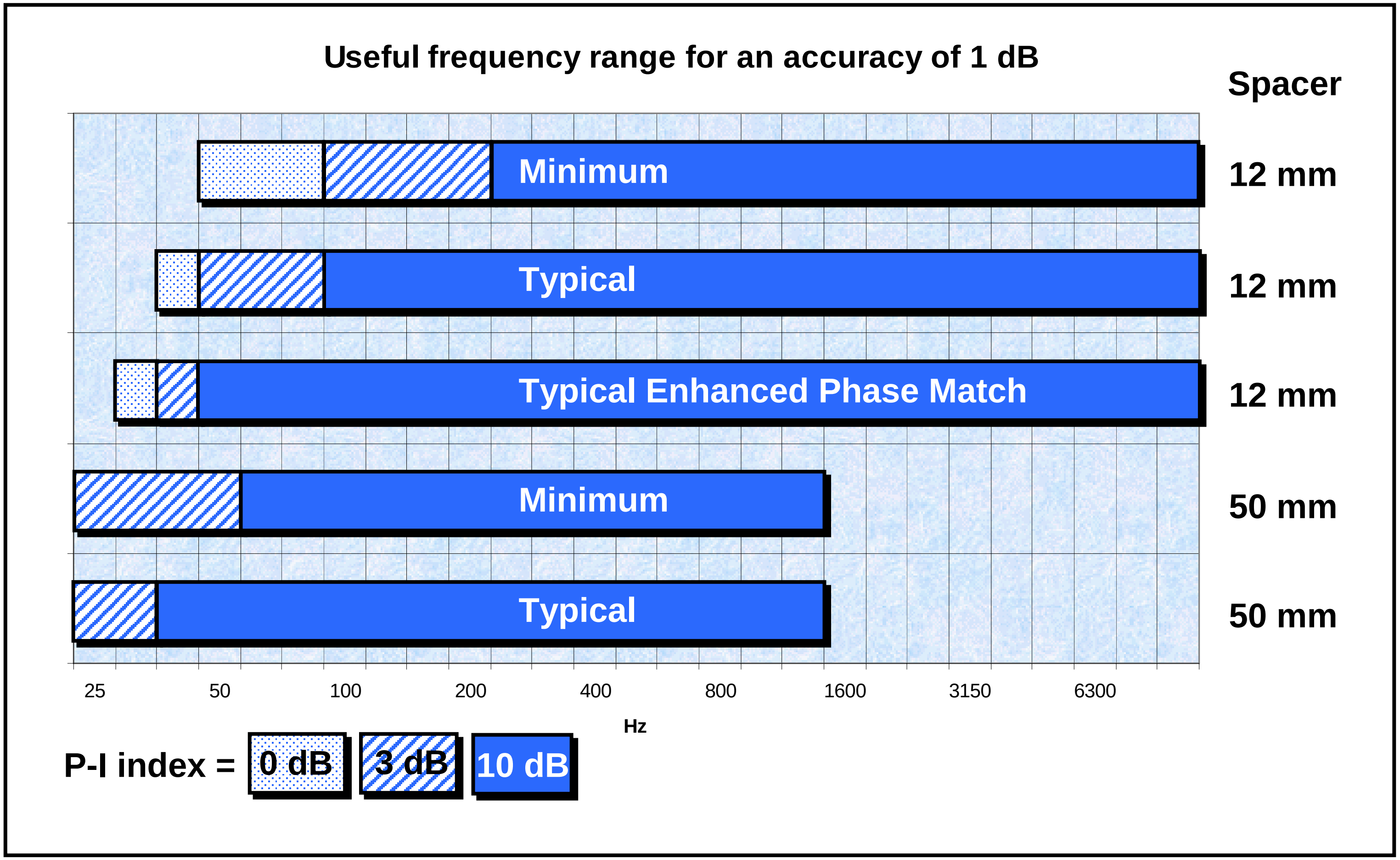
- Sound Intensity = Sound Pressure:
  - for sound propagating along the probe axis in one direction only (free field)
- Sound Intensity < Sound Pressure:
  - for sound propagating at an angle to the probe axis
  - for diffuse sound fields

Pressure-Intensity Index : Lp – LI



1.5 Sound Intensity

- Useful frequency range

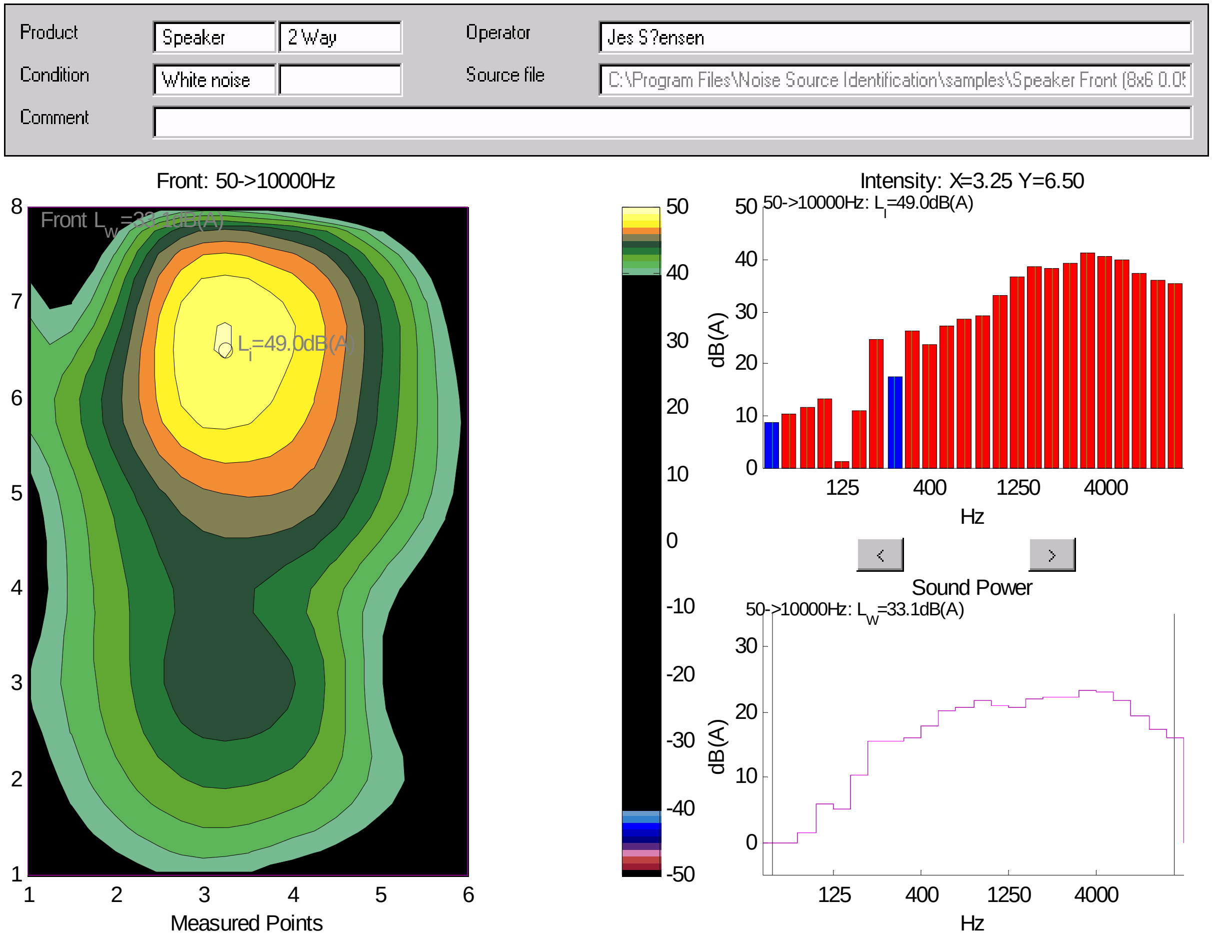


- Sound Power Measurements

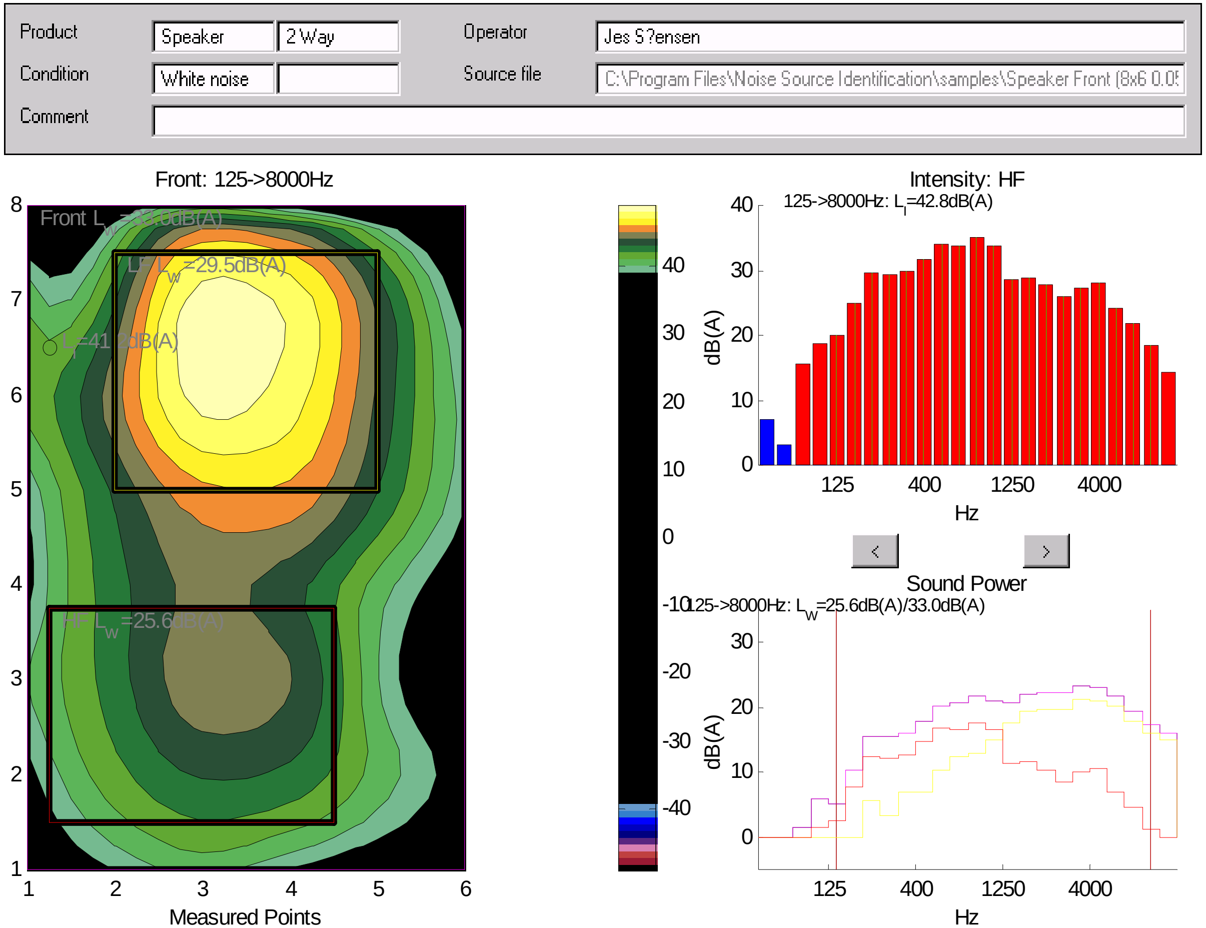
  - Determine Sound Power using standards:
    - ISO 9614-1 (not directly supported by BZ 7205)
    - ISO 9614-2
    - ECMA-160
    - ANSI s12.12

- Benefits of using Sound Intensity:
  - Results are determined within a given precision
  - Results are trustworthy, repeatable and comparable

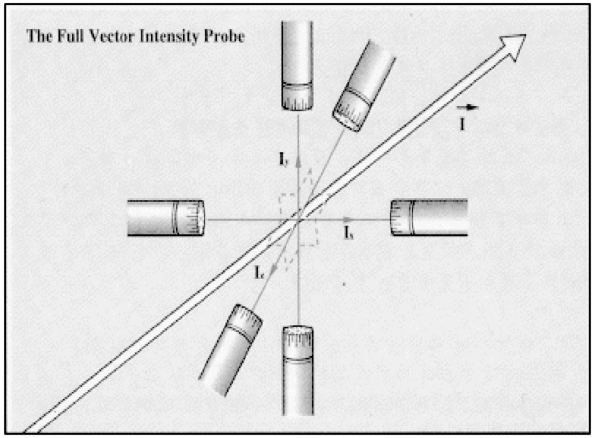
- Contour Plot I



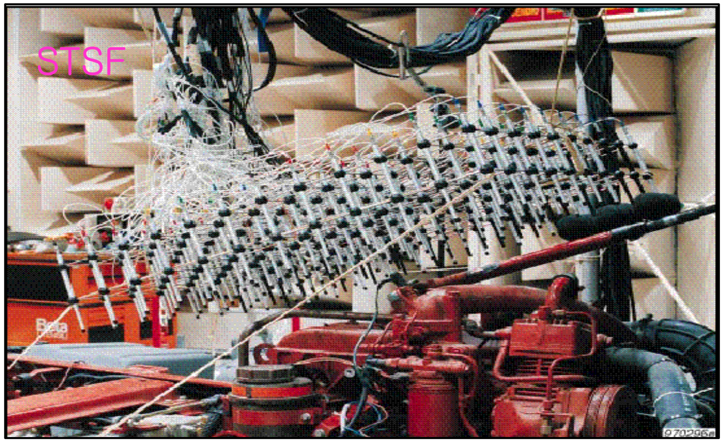
- Contour Plot II



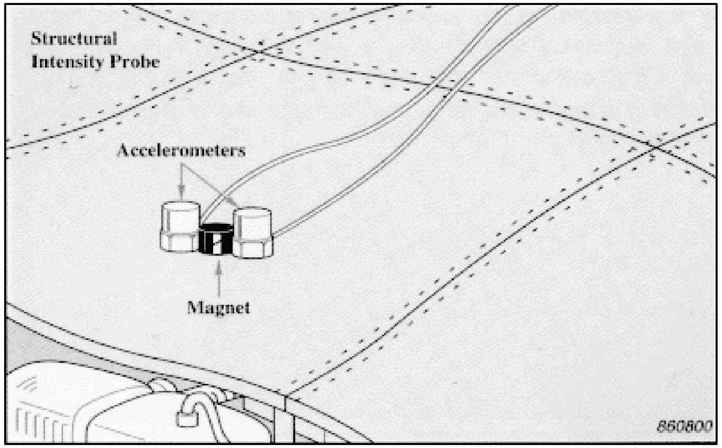
- Intensity Application – I



3 –Dim Vector Intensity

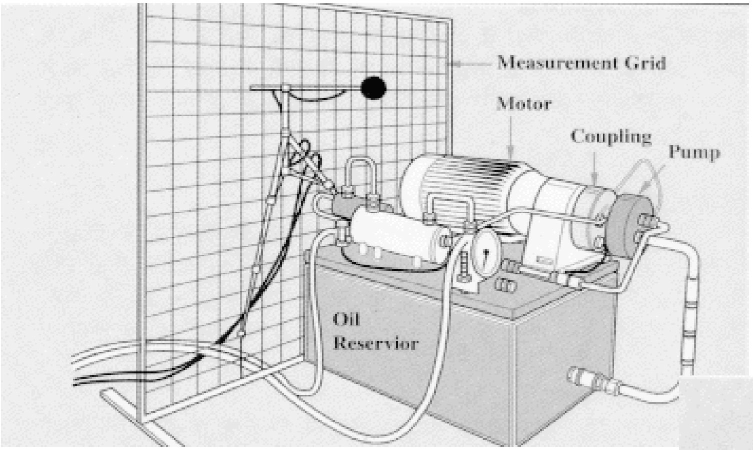


Structural Intensity



STSF : Spatial Transformation of Sound Field

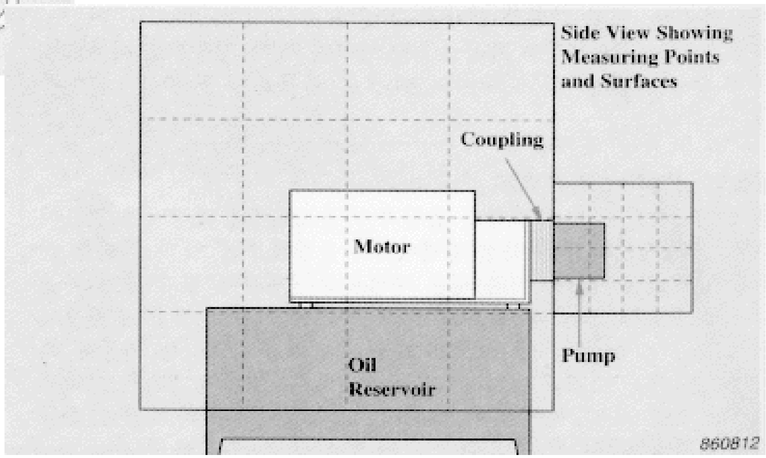
- Intensity Application – II



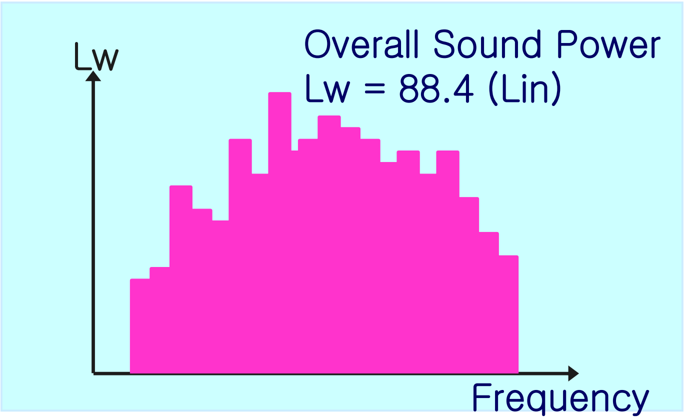
Problem

Pump + Motor
(Working state)

Make Grid



- Intensity Application – III

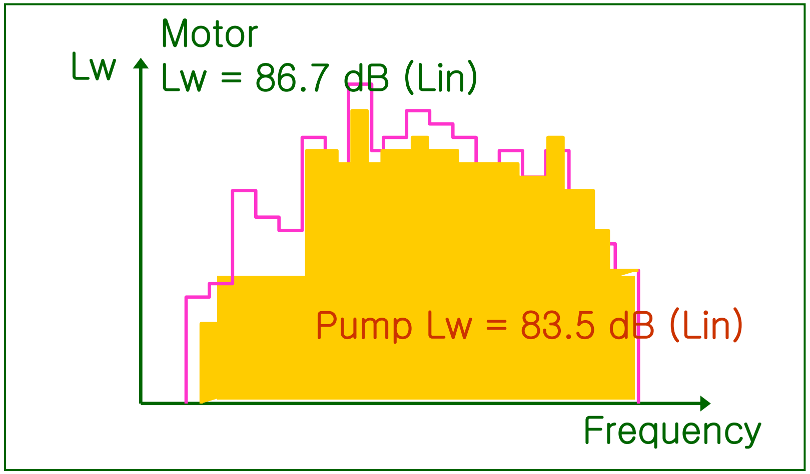


Motor : 86.7 dB

Pump : 83.5 dB

Transmit vibration from coupling

Remove Oil Reservoir



Result

Overall Sound Power :
88.4 dB(Lin)

- Mechanism of noise generation

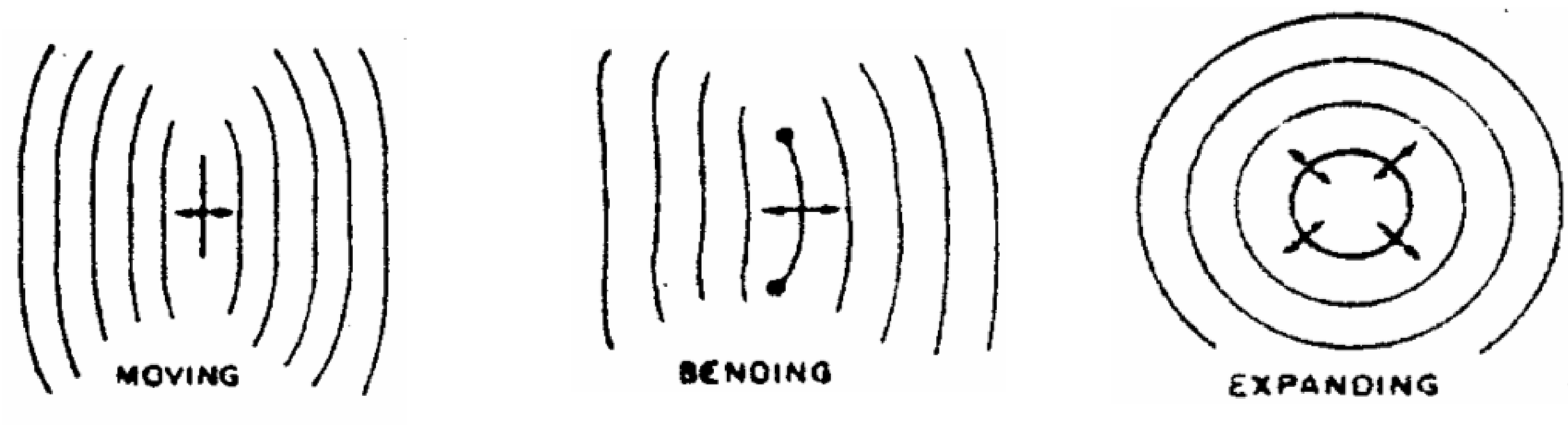
  - Airborne noise:

    This refers to noise which is fundamentally transmitted by way of the air and can be attenuated by the use of barriers and walls placed physically between the noise and receiver.

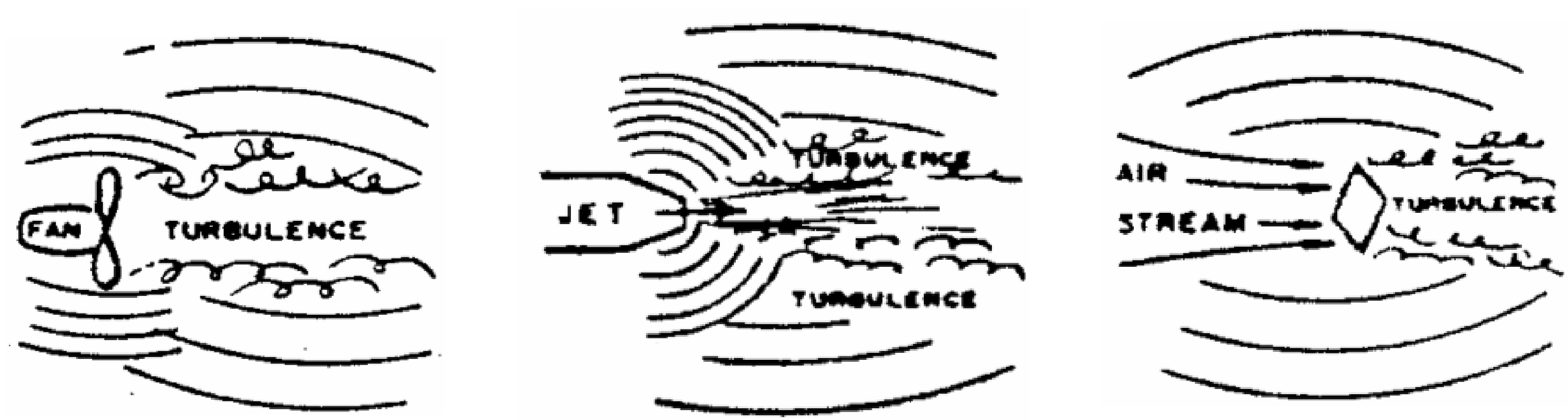
  - Structure-borne noise:

    This refers to noise which is generated by vibrations induced in the ground and/or structure and transmitted by way of the structure. These vibrations excite walls and slabs in buildings and cause them to radiate noise.
    This type of noise can not be attenuated by barriers or walls but requires the interposition of a resilient (neoprene, springs etc.) break between the source and the receiver.

- Noise sources

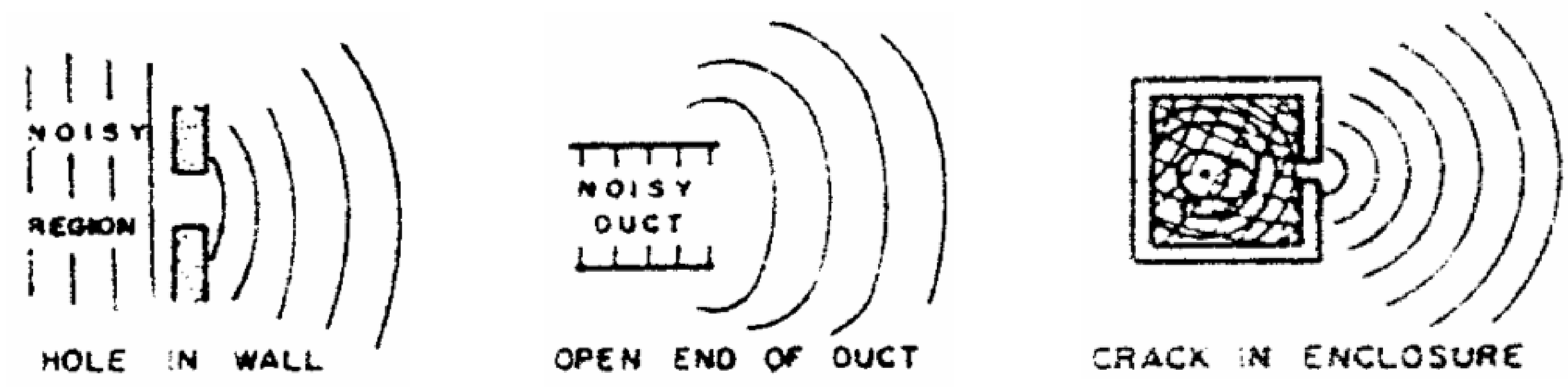


Solid sources



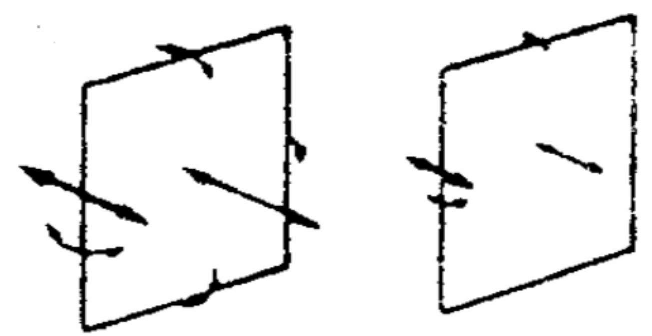
Air sources

- Noise sources

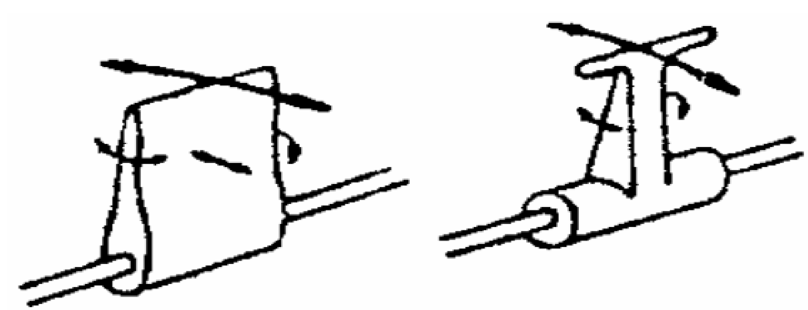


Secondary sources

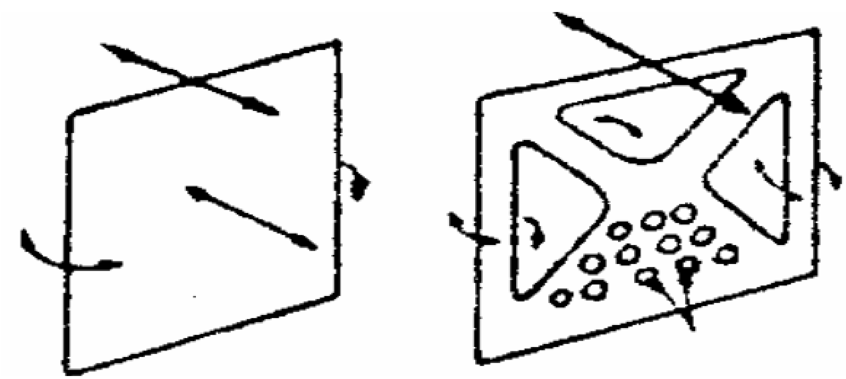
- Airborne noise reduction from solid structure



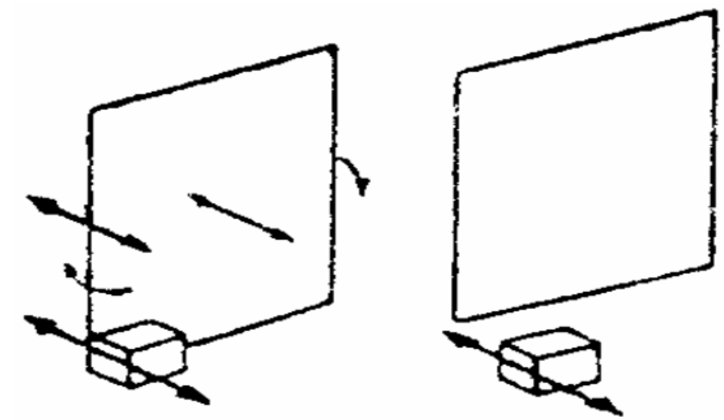
Reduce motion



Reduce radiation area

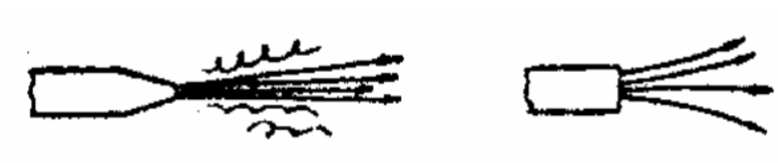


Provide air leak

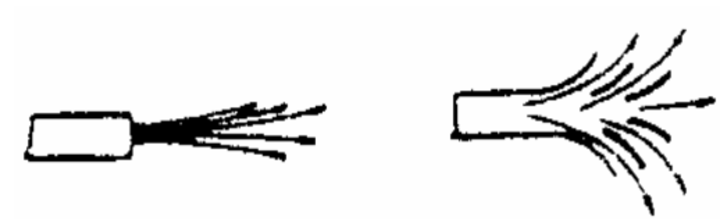


Disconnect large radiating areas from vibrating part

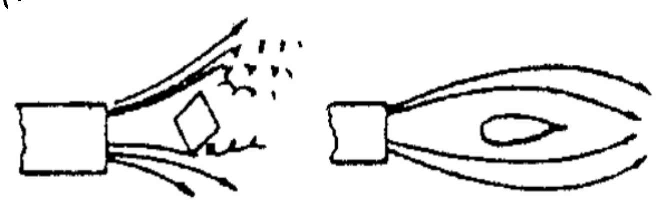
- <u>Airborne noise reduction from air sources</u>



Reduce air speed

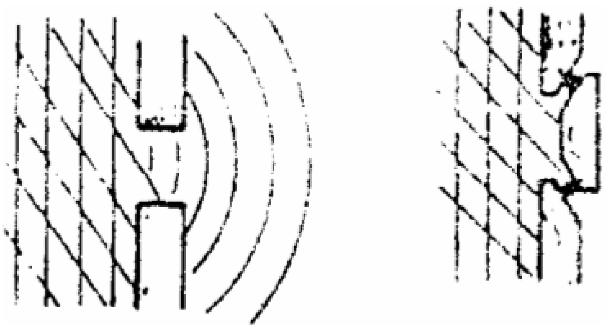


Add diffusing section

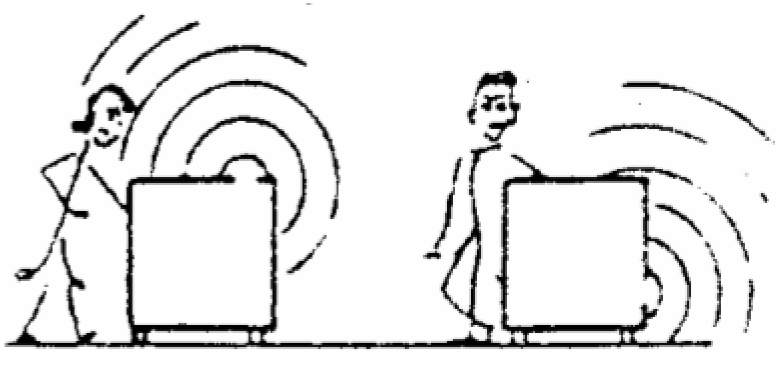


Remove obstacles or make them stream line

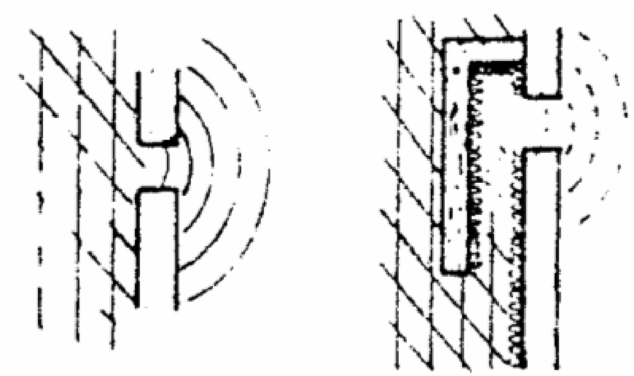
- Airborne noise reduction from secondary sources



Provide airtight closure

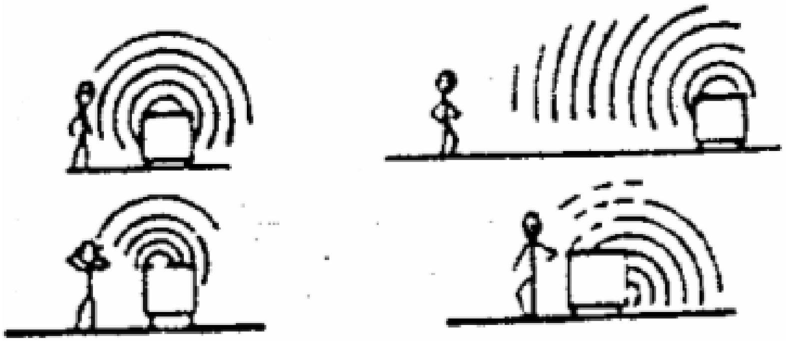


Redirect opening

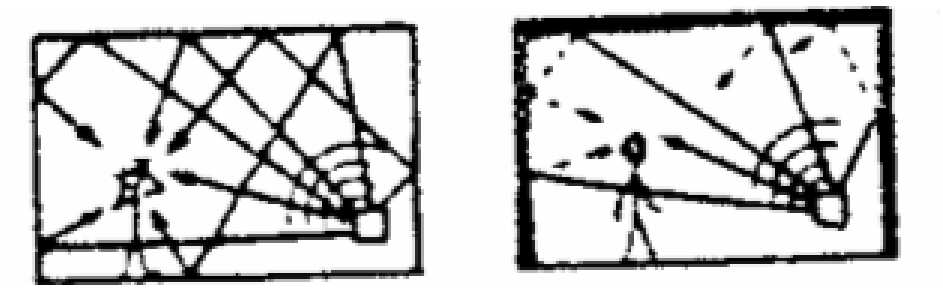


Add muffler or baffle ahead of opening

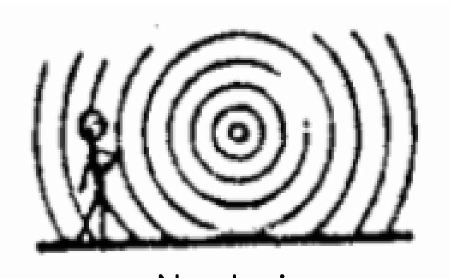
- <u>Airborne sound isolation</u>



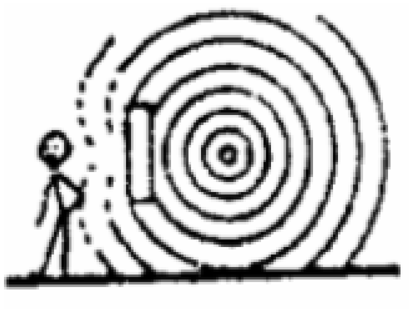
Increase distance



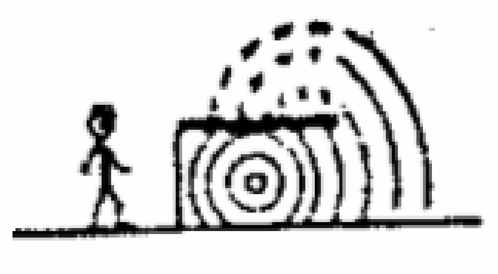
Acoustical absorption on confining walls



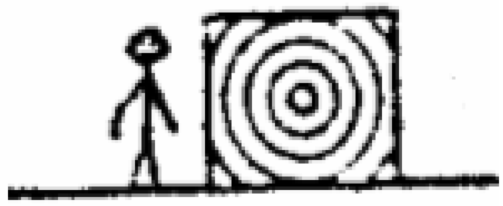
No device



Baffles give little sound reduction



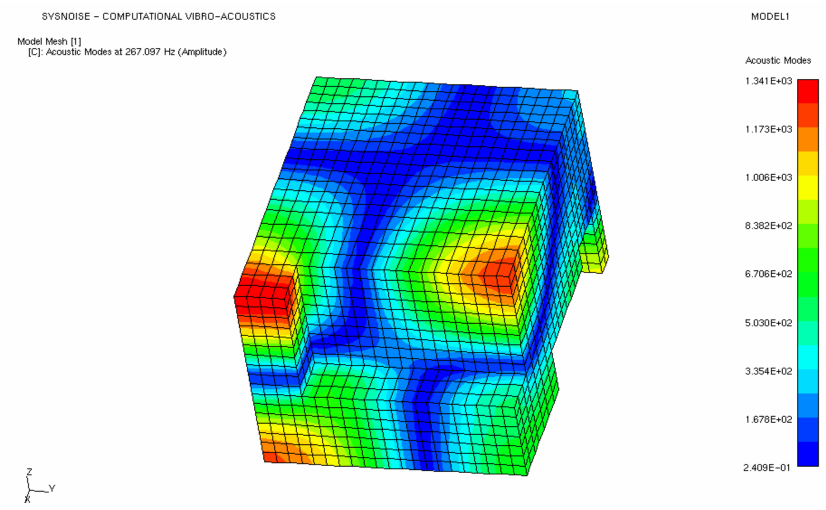
Partial enclosures are better



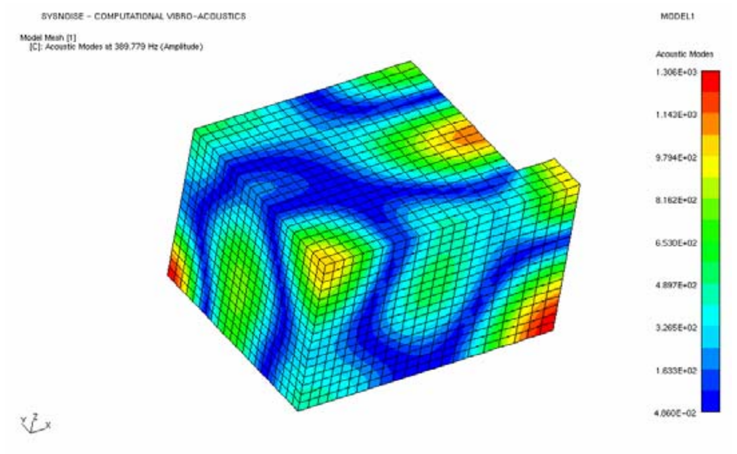
Airtight enclosures are best

- <u>Acoustical modal analysis</u>

267Hz



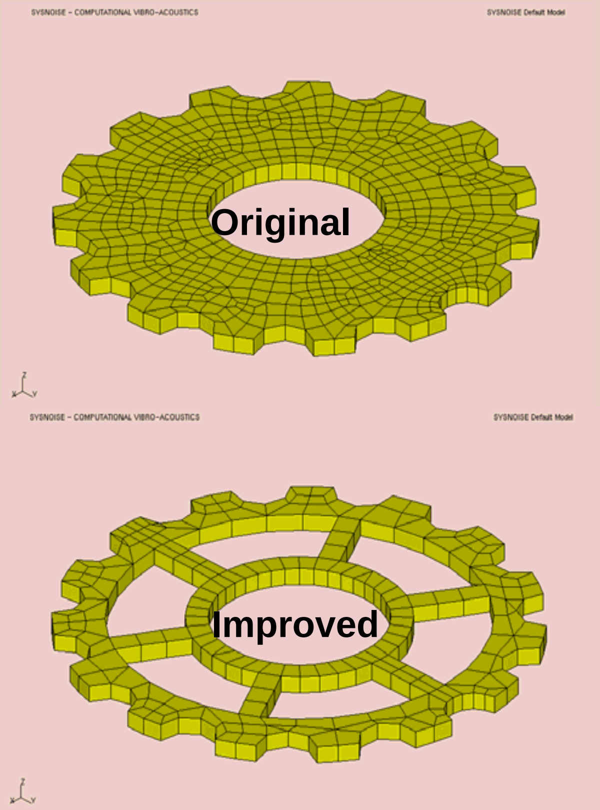
400Hz

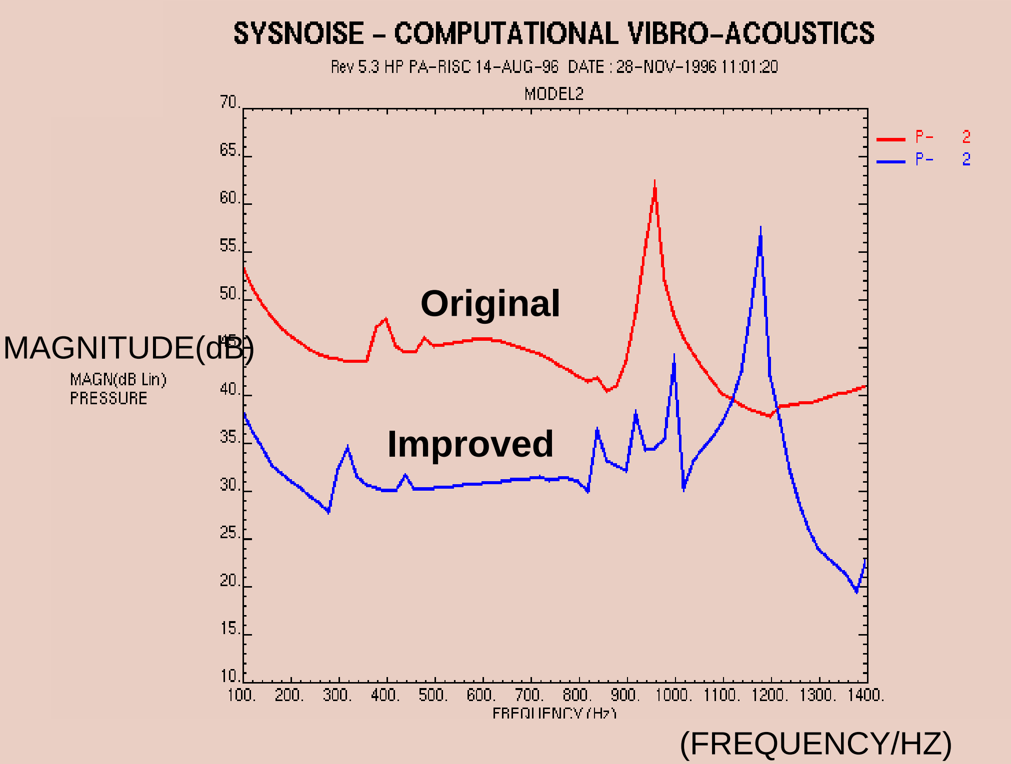


Cavity modes of control room

When implementing an absorbing material at the corner of room, it will be effective to reduce the corresponding noise.

- Acoustic field analysis





■ **Comparison of radiated noise levels**

– Impact noise of gear → emitted noise power ∝ area × vibrating velocity^2

– The noise level reduces more than 10dB.