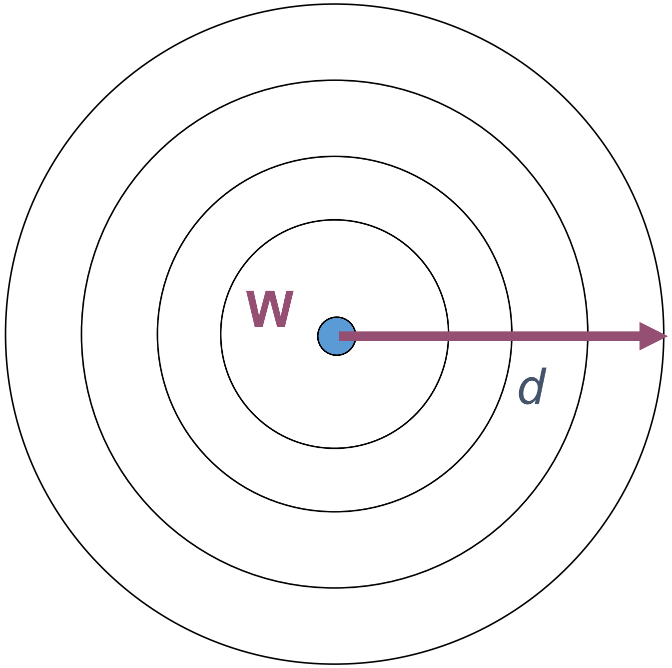
Sound propagation



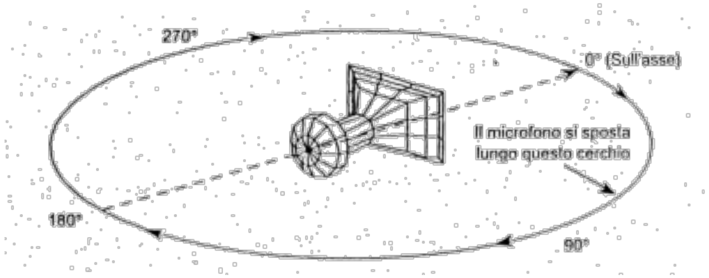
$$L_W = 10\log\frac{W}{W_{RIF}}$$

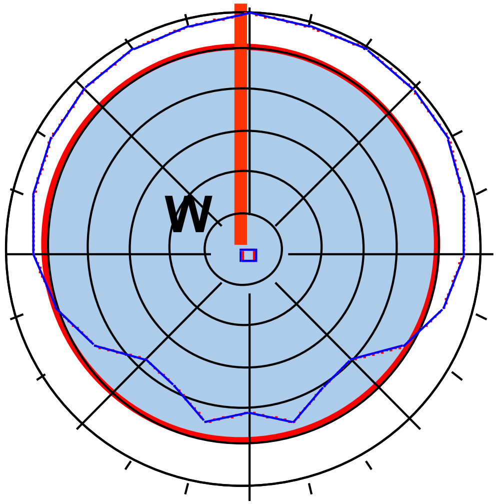
$$I = \frac{W}{4\pi d^2} \quad (Wm^{-2})$$

$$L_I = L_W - 10\log(4\pi) - 10\log d^2$$

$$L_p = L_W - 20\log d - 11$$

Sound propagation





$$Q = \frac{I}{I_S} = \frac{p_{eff}^2}{p_{effS}^2}$$

$$DI = 10 \log Q$$

$$L_p = L_W + DI - 20 \log d - 11$$

Sound propagation

$$L = L_W + DI - 20\log r - 11 - \sum_i Att_i$$

- Attenuations are due to:
  - Atmospheric absorption
  - Ground interference/absorption
  - Barriers and other obstacles
  - Weather-dependent phenomena (wind, temperature)

Sound propagation

$$L = L_W + DI - 20\log r - 11 - \sum_i Att_i$$

- Attenuations are due to:
  - Atmospheric absorption
  - Ground interference/absorption
  - Barriers and other obstacles
  - Weather-dependent phenomena (wind, temperature)

Sound propagation

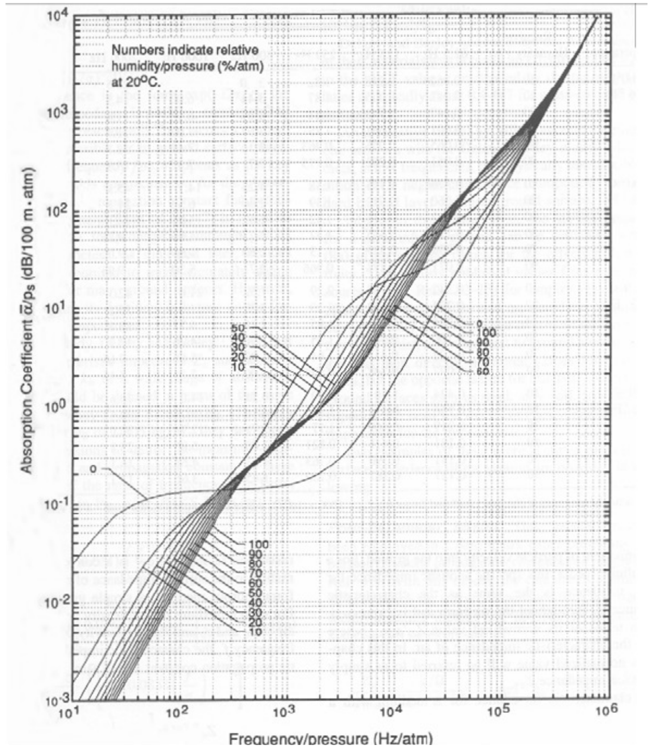
$$Att_{air} = a \cdot r \quad [dB]$$

$$I(r) = I_o e^{-mr}$$

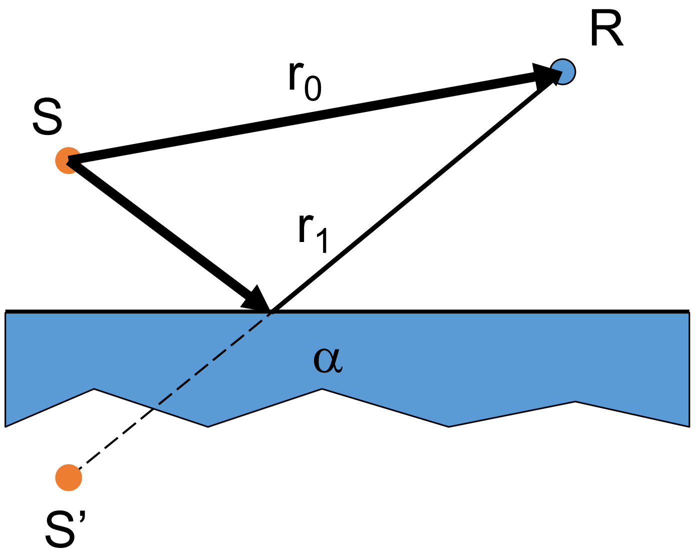
$$Att_{air} = -10\log_{10}\frac{I(r)}{I_0} = -10\log_{10} e^{-mr}$$

$$\log_{10} x = \ln x \,/\, \ln 10$$

$$Att_{air} = -10(\ln e^{-mr} / \ln 10) = 4.34m \cdot r$$



Sound propagation



$$I_0 = \frac{W}{4\pi {r_0}^2} \quad \left(\frac{W}{m^2}\right)$$

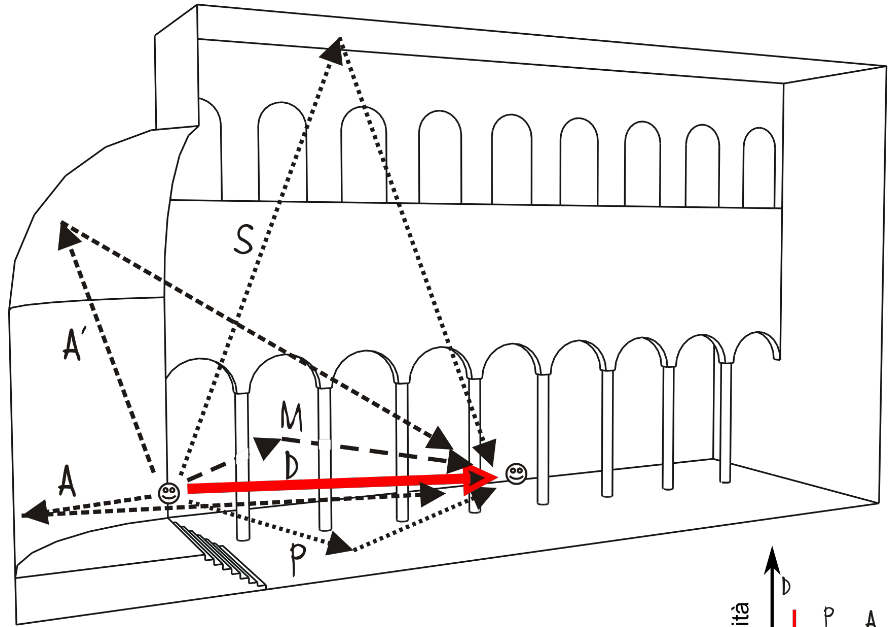
$$I_1 = \frac{W(1-\alpha)}{4\pi {r_1}^2} \quad \left(\frac{W}{m^2}\right)$$

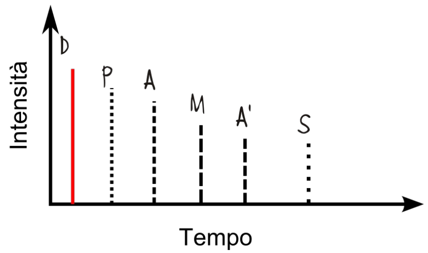
$$I_0 = \frac{W}{4\pi {r_0}^2}\exp(-mr_0) \quad \left(\frac{W}{m^2}\right)$$

$$I_1 = \frac{W(1-\alpha)}{4\pi {r_1}^2}\exp(-mr_1) \quad \left(\frac{W}{m^2}\right)$$

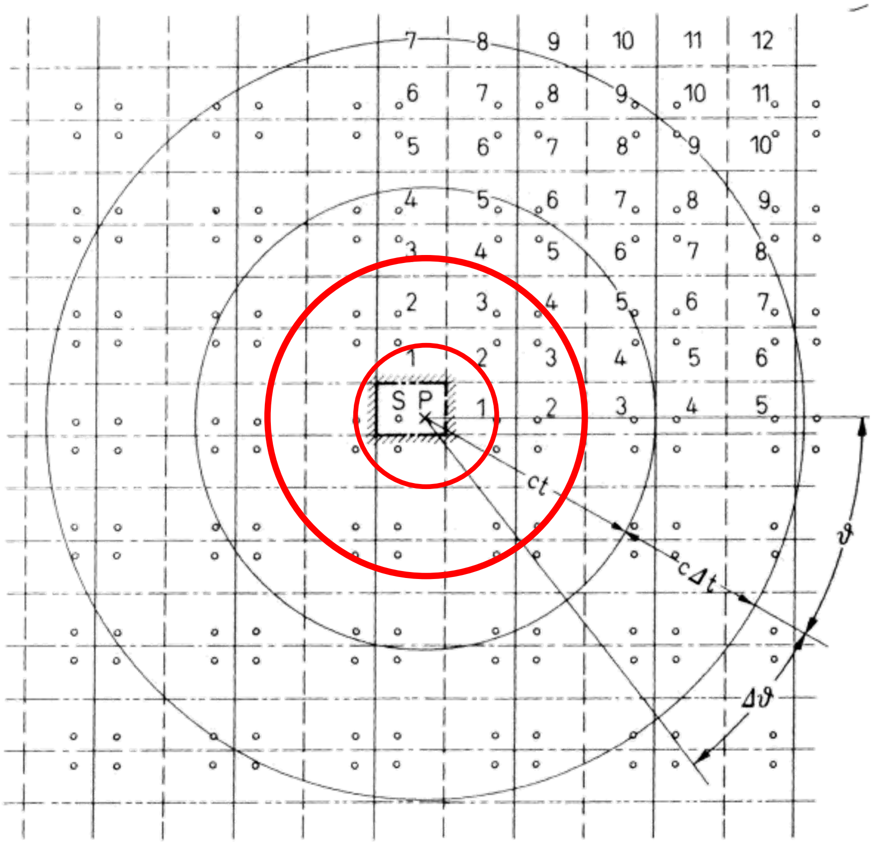
N.B.: Large surface compared to wavelength!

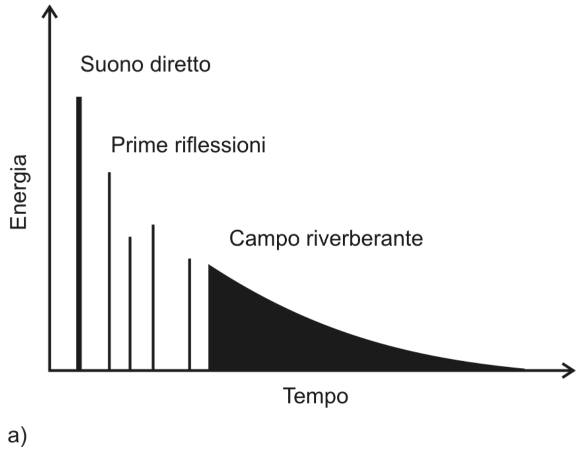
Sound propagation



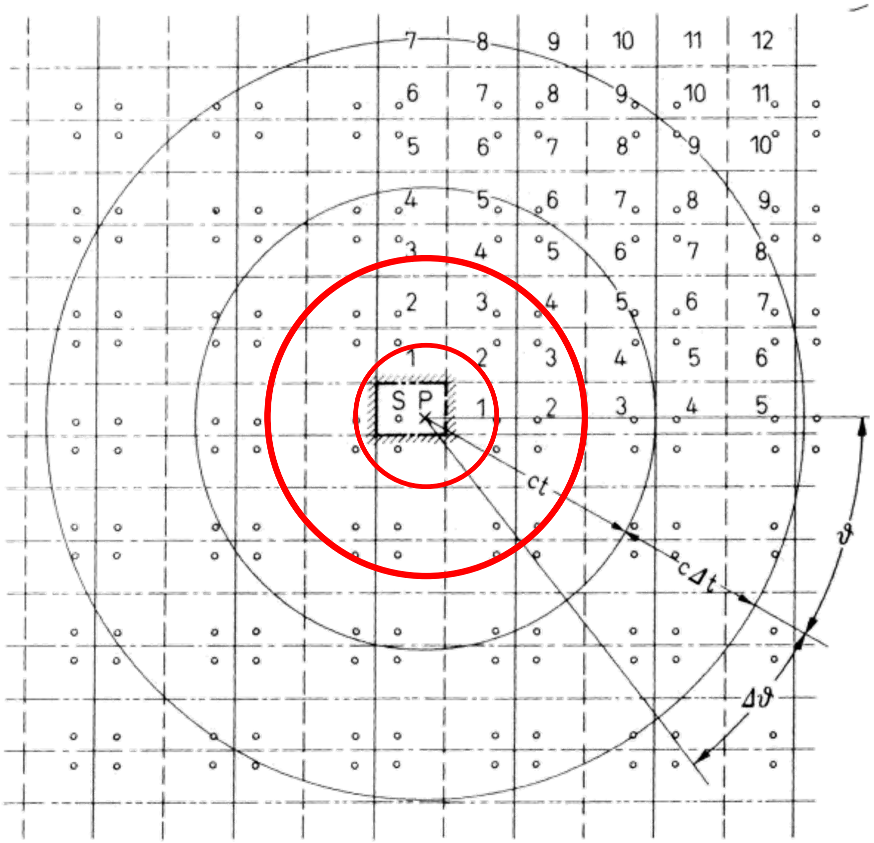


Sound propagation





Sound propagation



$$I(t) = \frac{W(1-\alpha)^{nt}}{4\pi (ct)^2} \exp(-mct) dN$$

$$dN = \frac{4\pi (ct)^2 \cdot cdt}{V}$$

$$I(t) = \frac{W(1-\alpha)^{nt}}{4\pi (ct)^2} \exp(-mct) \times \frac{4\pi (ct)^2 cdt}{V}$$

$$I(t) = \frac{Wcdt}{V} (1-\alpha)^{nt} \exp(-mct)$$

$$I(t) = I_0 (1-\alpha)^{nt} \exp(-mct)$$

Sound propagation

Assumptions:

- The **Mean Free Path,** can be derived from kinetic theory of gases

$$l = \frac{4V}{S} \quad [\mathrm{m}]$$

Mean time interval between two susequent reflections

$$t = \frac{l}{c} = \frac{4V}{cS}$$

Mean number of surfaces hit per second

$$n = \frac{1}{t} = \frac{cS}{4V}$$

Sound propagation

$$n = \frac{1}{t} = \frac{cS}{4V}$$

$$I(t) = I_0 (1-\alpha)^{nt} \exp(-mct)$$

$$(1-\alpha)^{nt} = \exp[\ln(1-\alpha)^{nt}] = \exp[nt \cdot \ln(1-\alpha)]$$

$$I(t) = I_0 \exp[-mc + n\ln(1-\alpha)]t$$

$$I(t) = I_0 \exp\left[-ct\,\frac{4mV - S\ln(1-\alpha)}{4V}\right]$$

Sound propagation

$$L_I(t) = L_0 - 4.34\left[ct\frac{4mV - S\ln(1-\alpha)}{4V}\right]$$

$$T_{60} = \frac{60}{4.34c}\frac{4V}{4mV - S\ln(1-\alpha)} = 0.161\frac{V}{4mV - S\ln(1-\alpha)}$$ Eyring's formula

$$\ln(1+x) = x - \frac{1}{2}x^2 + \frac{1}{3}x^3 - \ldots \Rightarrow \ln(1-\alpha) \approx -\alpha$$

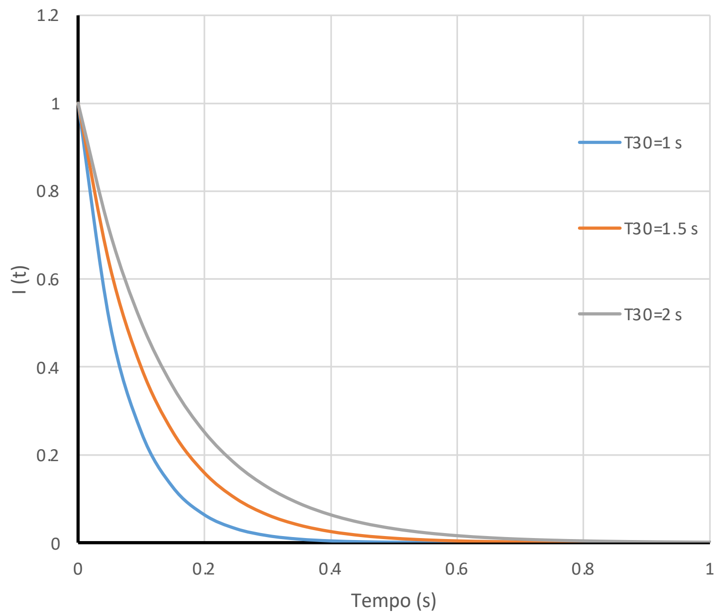
$$T_{60} = 0.161\frac{V}{4mV + S\alpha} = 0.161\frac{V}{A}$$ Sabine's formula

Sound propagation

$$T_{60} = 0.161 \frac{V}{4mV - S\ln(1-\alpha)}$$

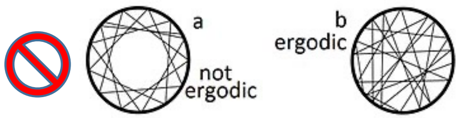
$$I(t) = I_0 \exp\left[-ct\frac{4mV - S\ln(1-\alpha)}{4V}\right]$$

$$I(t) = I_0 \exp\left[-ct\frac{0.161}{4T_{60}}\right] = I_0 \exp[-13.8t/T_{60}]$$

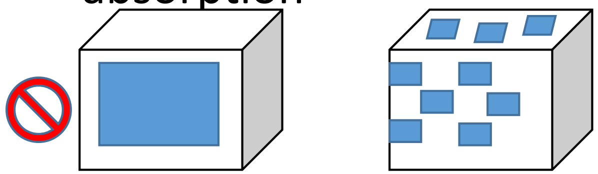


Sound propagation

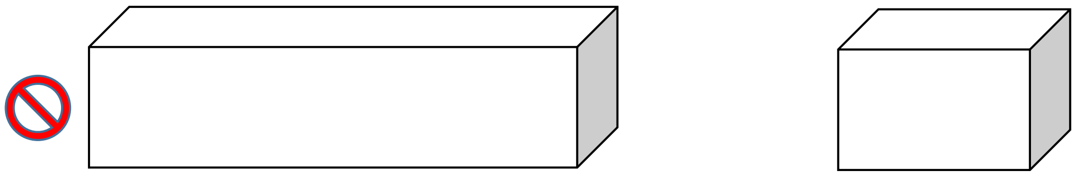
Ergodicity



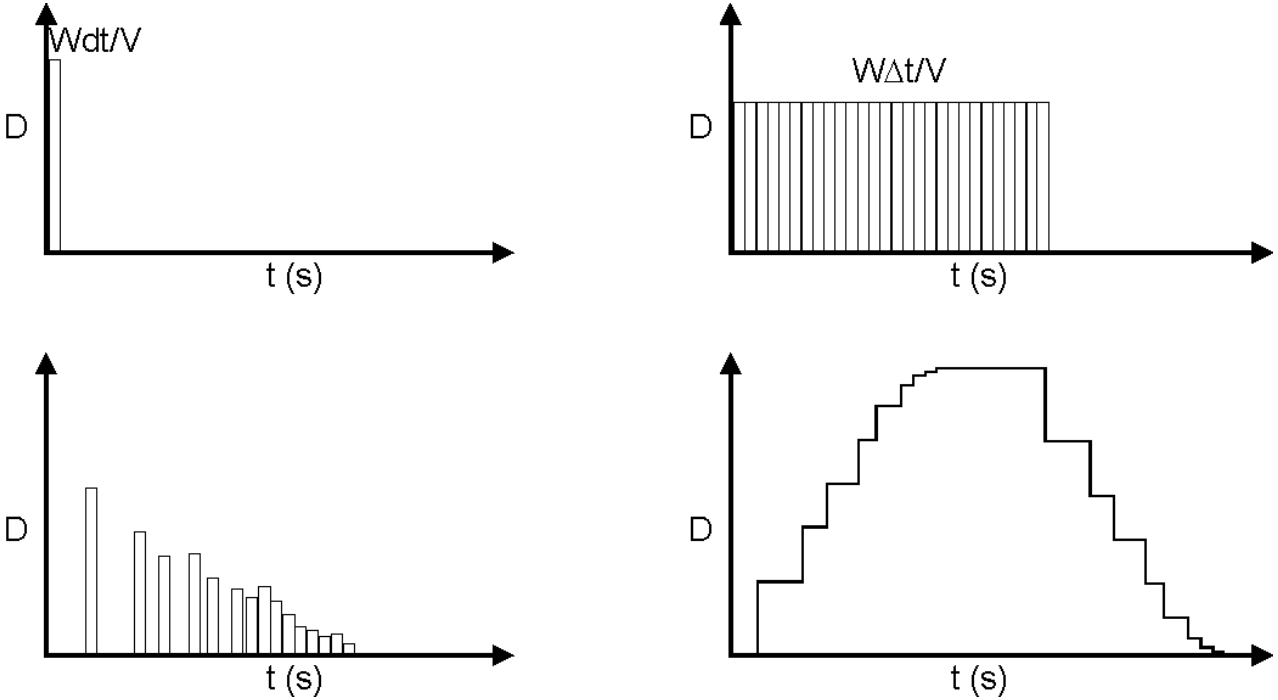
Mixing and distributed absorption



Proportionate



Sound propagation



Sound propagation

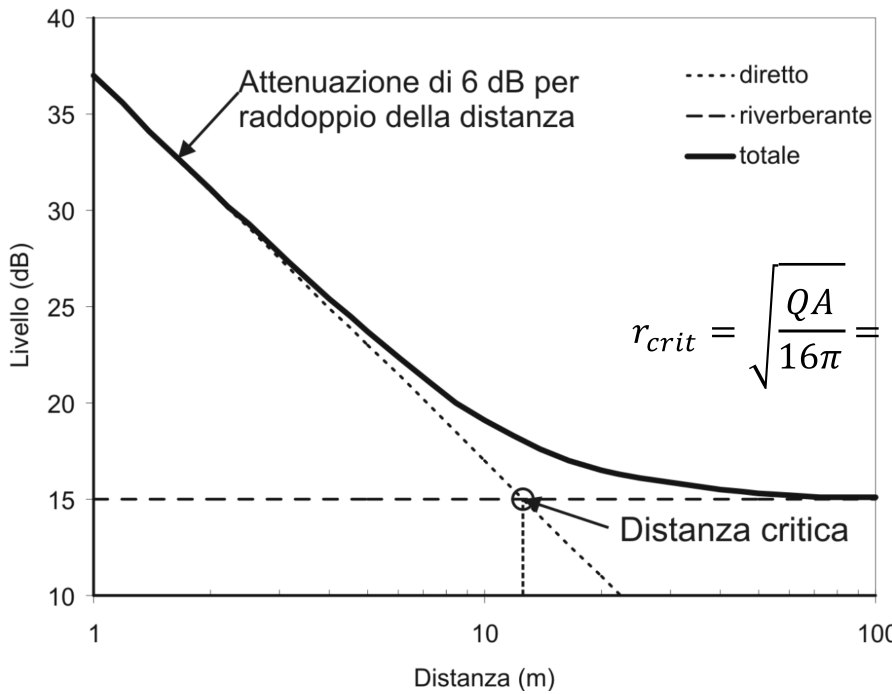
$$w_D = \frac{W}{4\pi r^2 c} \qquad w_R = \frac{4W}{cR} \qquad R = \frac{\bar{\alpha} S}{1-\bar{\alpha}} \approx A$$

$$w = w_D + w_R = \frac{W}{c}\left(\frac{1}{4\pi r^2} + \frac{4}{R}\right)$$

$$L_p = L_w + 10\log\left(\frac{1}{4\pi r^2} + \frac{4}{A}\right)$$

$$L_p = L_w + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4}{A}\right)$$

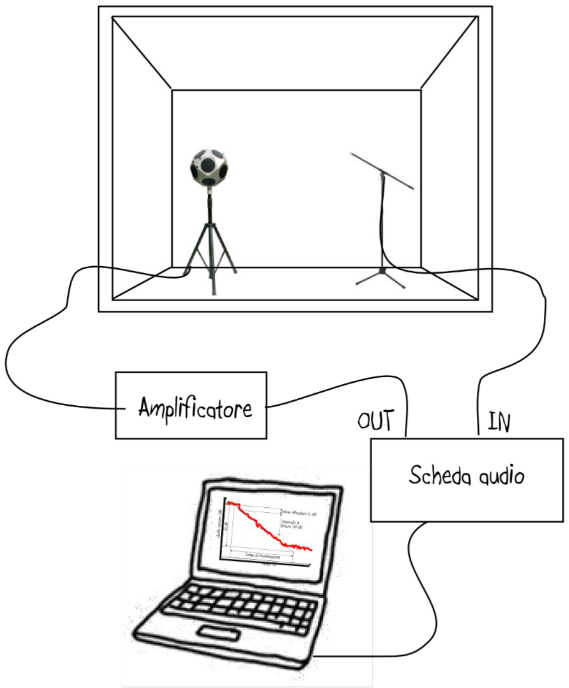
Sound propagation

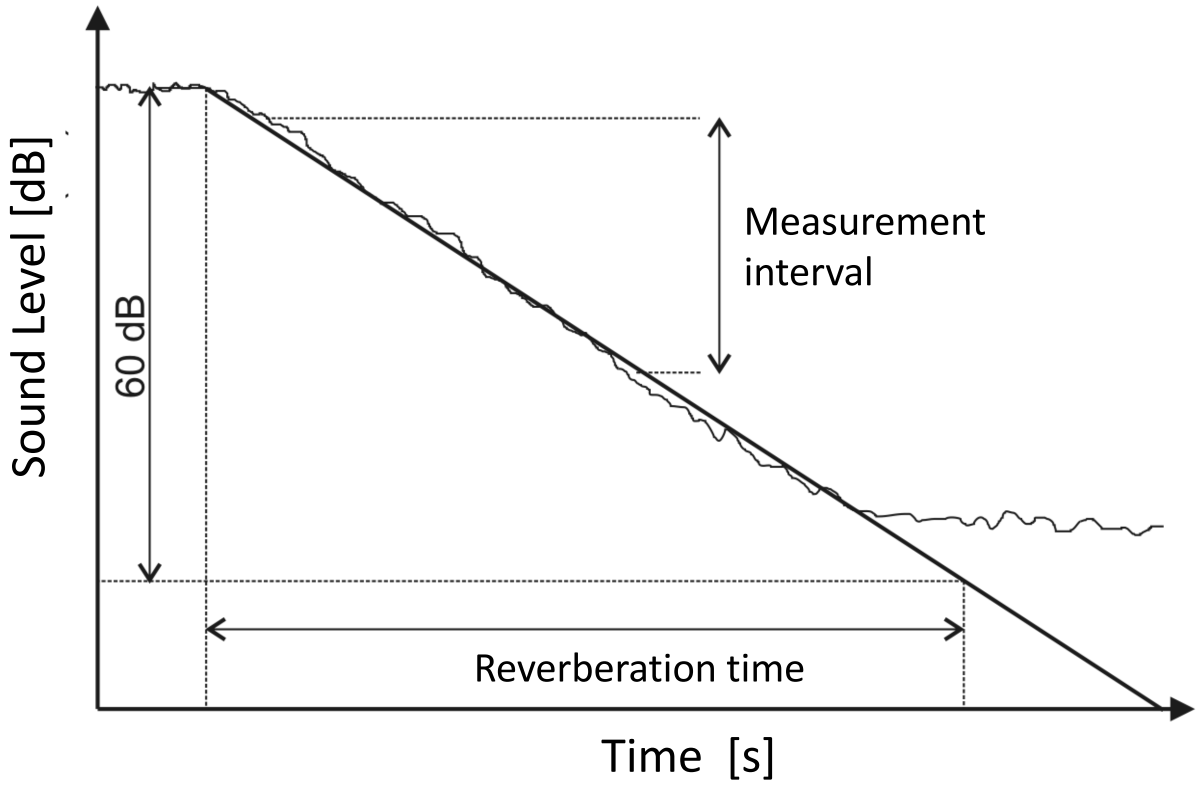


$$r_{crit} = \sqrt{\frac{QA}{16\pi}} = 0.057\sqrt{\frac{QV}{T}}$$

How to measure reverberation time (and impulse responses)

- Measurment techniques
  - Noise emission (standing)
  - Impulsive source
  - Electro acoustic excitation (deconvolution)
- ISO 3382-1:2009

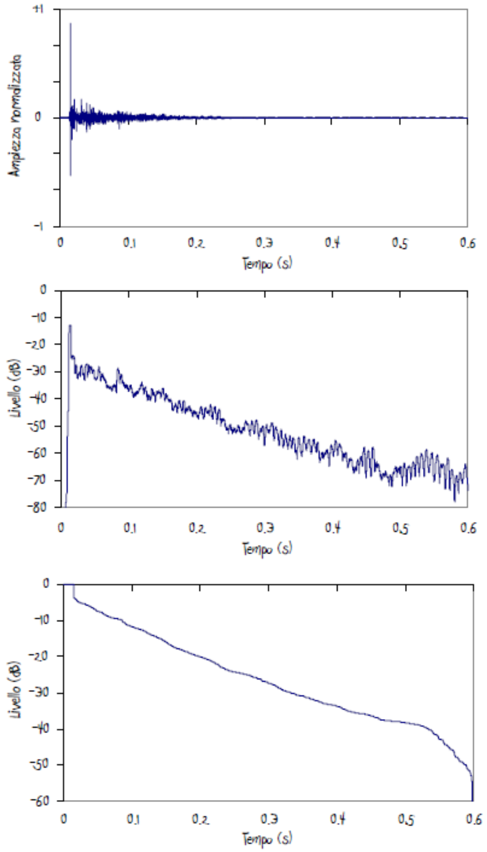



Sound Level [dB]
60 dB
Measurement
interval
Reverberation time
Time [s]

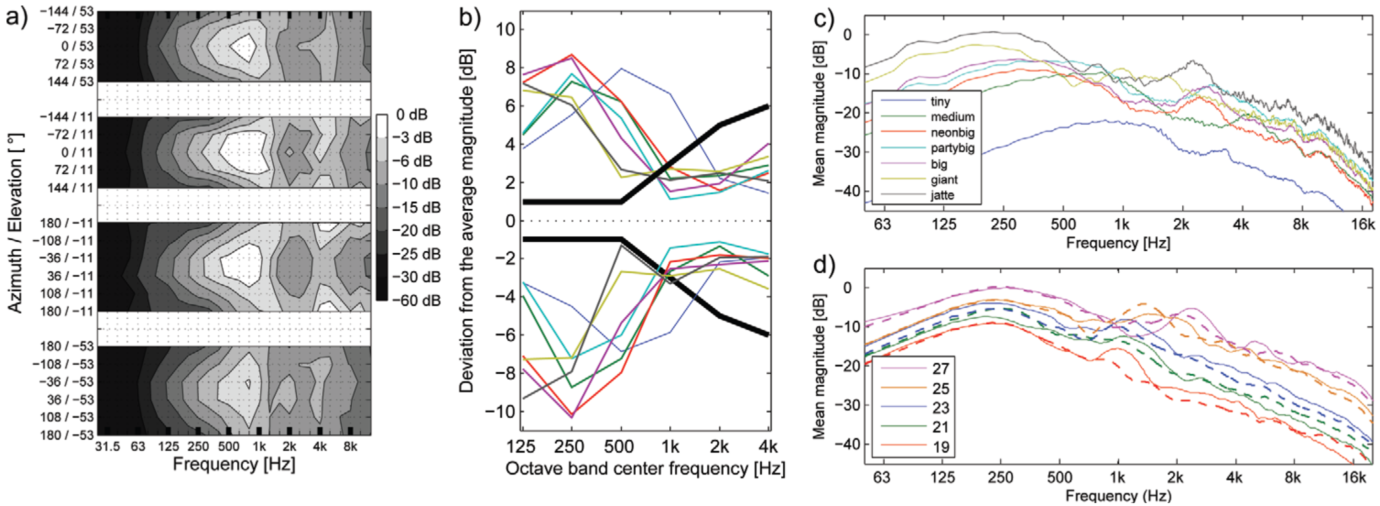
Backward integration

$$h^2(t) = Ae^{-kt} \longrightarrow \int Ae^{-kt}dt = -\frac{A}{k}e^{-kt}$$

$$\int_t^{\infty} Ae^{-kt}dt = -\frac{A}{k}e^{-k\cdot\infty} + \frac{A}{k}e^{-kt}$$



Baloons



a)
Azimuth / Elevation [°]
Frequency [Hz]
31.5
63
125
250
500
1k
2k
4k
8k
0 dB
−3 dB
−6 dB
−10 dB
−15 dB
−20 dB
−25 dB
−30 dB
−60 dB
b)
Deviation from the average magnitude [dB]
Octave band center frequency [Hz]
c)
Mean magnitude [dB]
Frequency [Hz]
tiny
medium
neonbig
partybig
big
giant
jatte
d)
Mean magnitude [dB]
Frequency (Hz)
27
25
23
21
19

The clapper

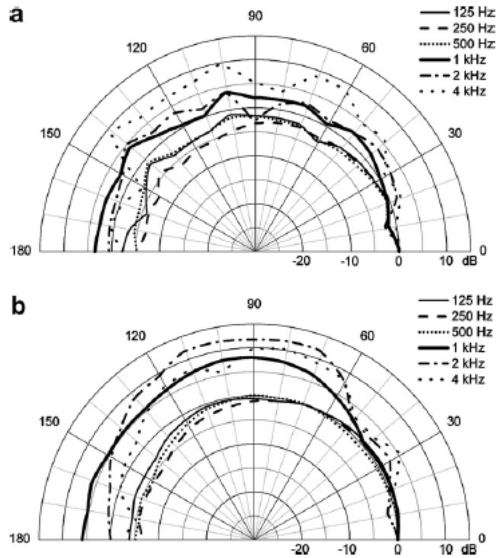
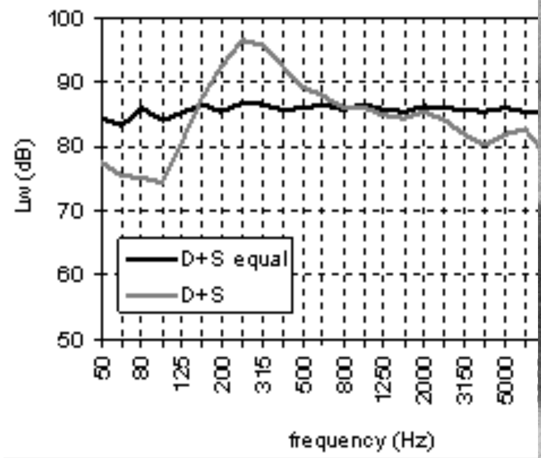


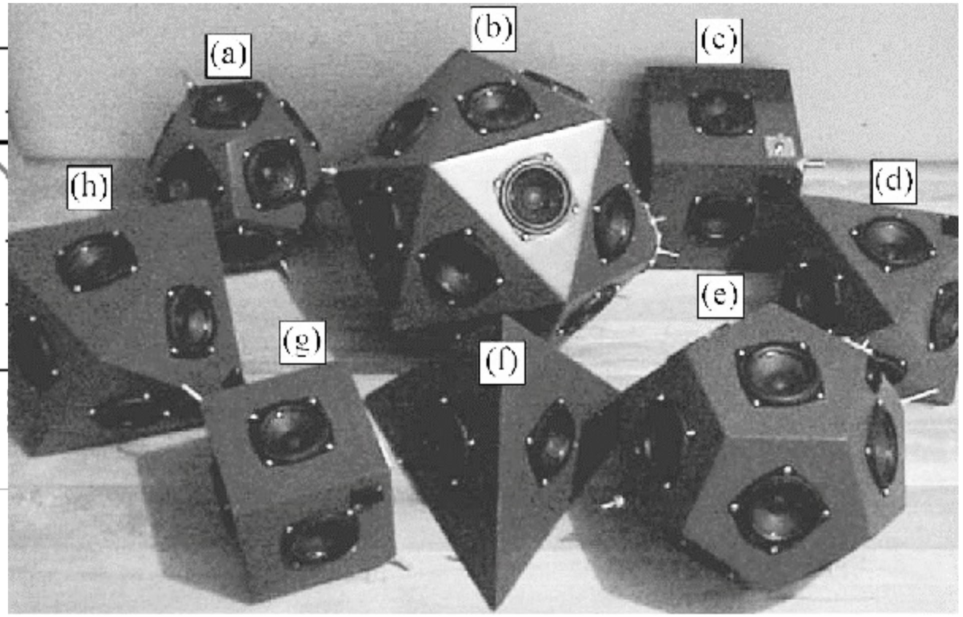
Fig. 5. Directivity of the clapper – variations in the maximum reached signal value: (a) frontal plane; and (b) medial plane.

Impulsive sound source

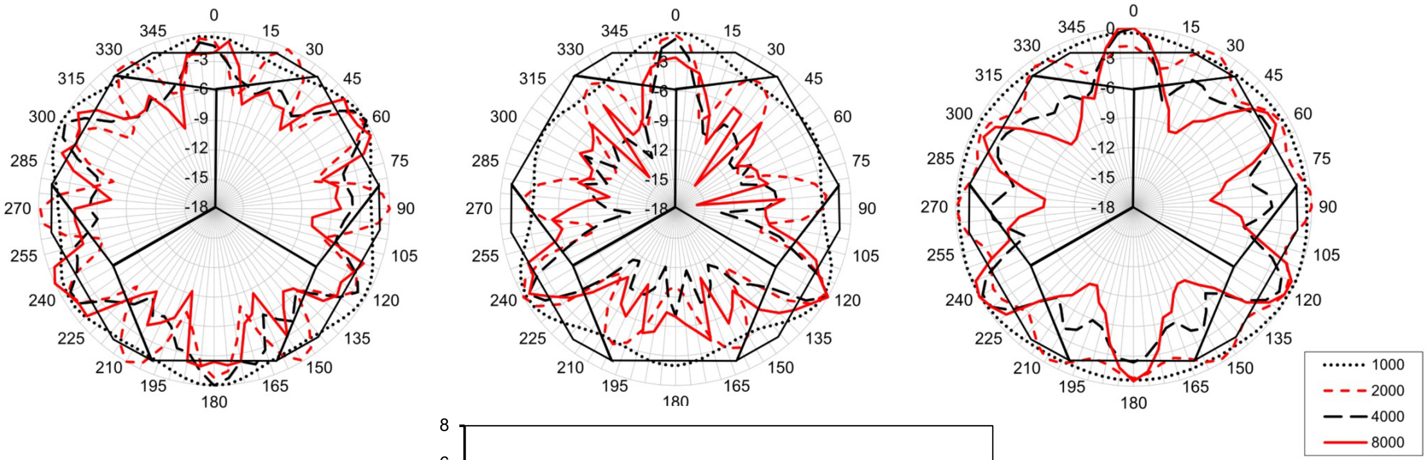
- Pros
  - Simple procedure and lightweight sources
  - Yields a usable impulse response (with limitations)
  - Smooth decay (thanks to backward integration)
- Cons
  - Poor repeatability
  - Poor control of radiated spectrum and source directivity
  - Signal-to-noise ratio depending on the source power

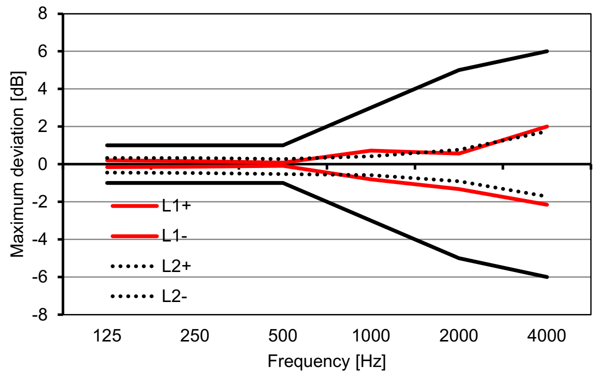
Electro acoustic source



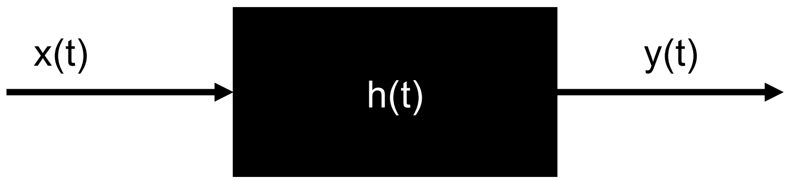


Electro acoustic source





Convolution/Deconvolution

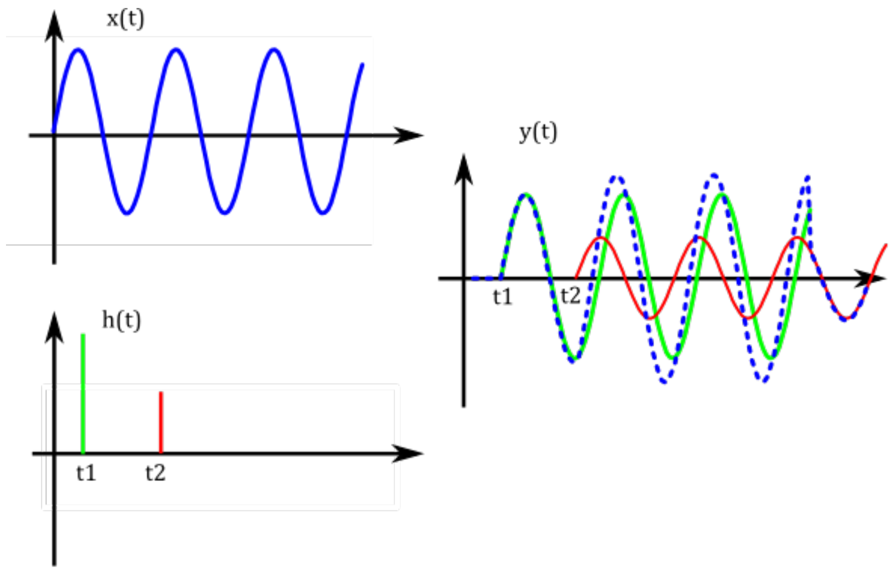


$$y(t) = x(t) \otimes h(t) = \int_0^{\infty} h(\tau) \cdot x(t-\tau)\, d\tau$$

FFT

$$Y(f) = X(f) \cdot H(f)$$

Convolution



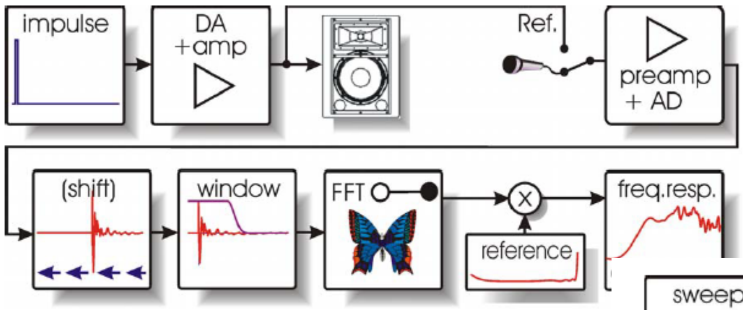


Fig. 3. Signal-processing for transfer-function measurement with i

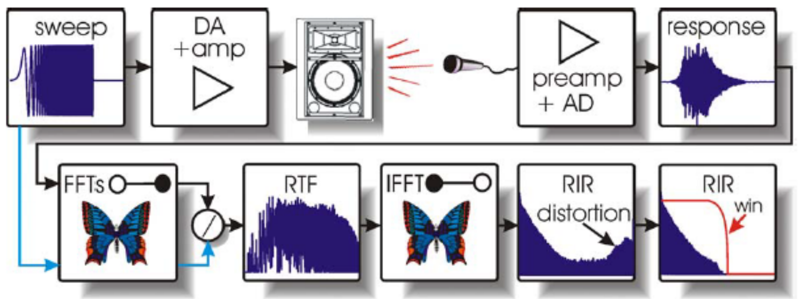


Fig. 6e. Alternatively to the linear deconvolution, a circular deconvolution using an FFT size equal to the acquisition time may be employed. In this case, however, the distortion products could smear into the decay of the IR. This means that the length of the excitation signal ha has be chosen sufficiently longer than the decay time. The distortion products will then appear in the noise floor where they can be safely discarded by windowing without affecting the reverberant tail.

Sine sweeps (or chirps)

- Pros
  - Reject distortion;
  - High signal-to-noise ratio;
  - Immune to noise (more or less);
  - Possibility to equalize the source;

- Cons
  - Needs a LTI (linear-time-invariant) system