ACOUSTICS
part - 2
Sound Engineering Course

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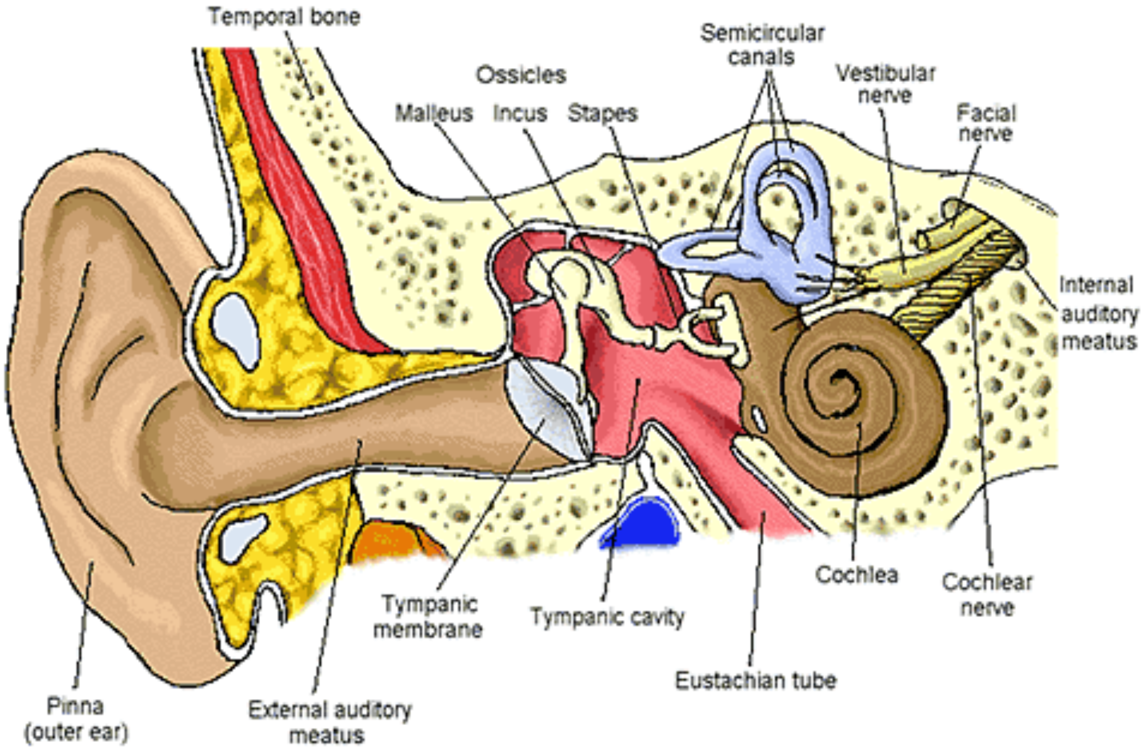
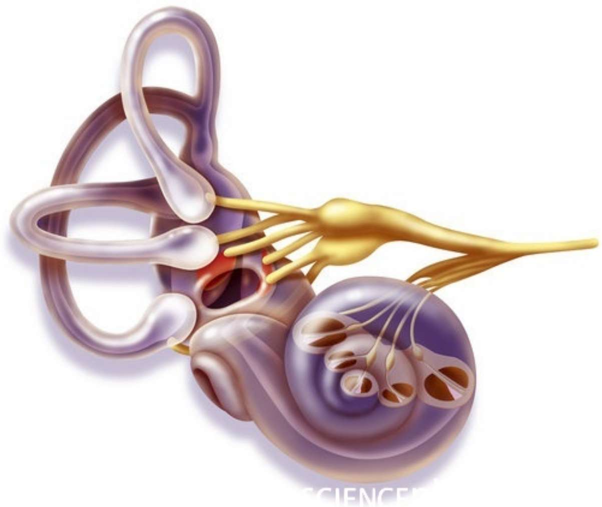
The human auditory system

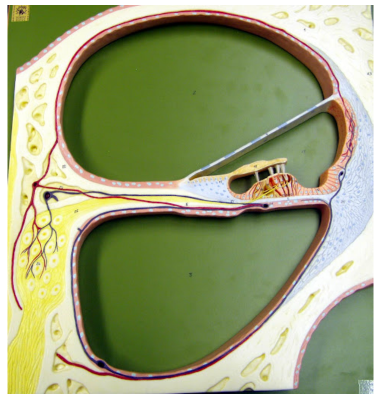
The human ear



Structure of human ear, divided in external ear, medium ear and internal ear



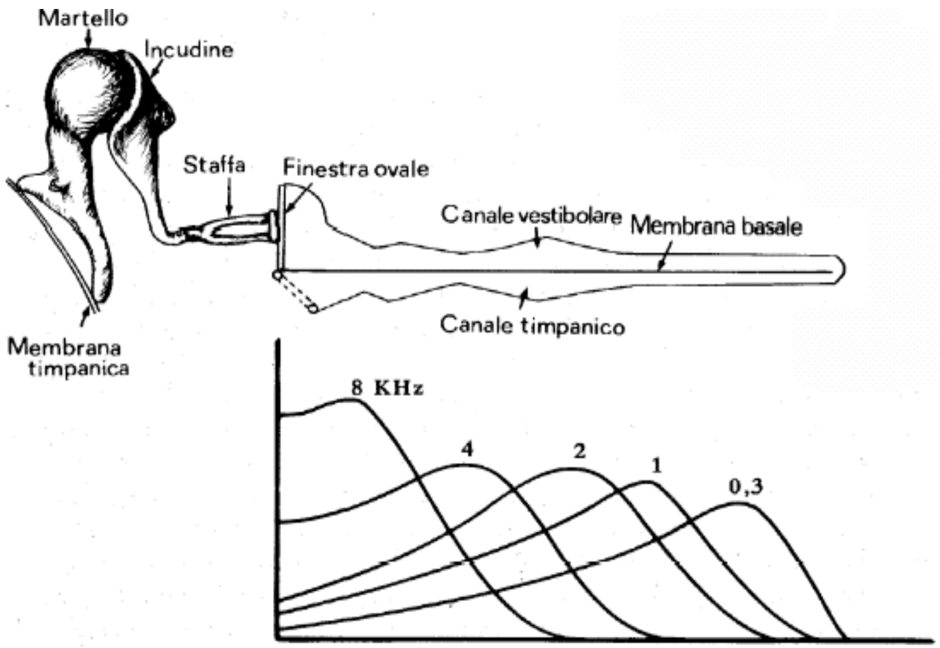
Internal ear

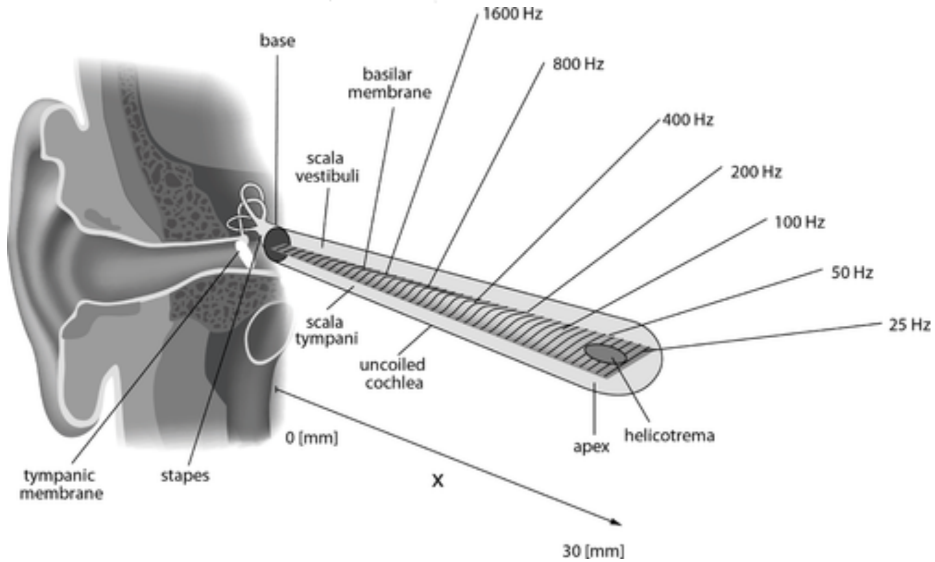


Cochlea

Frequency selectivity of Cochlea

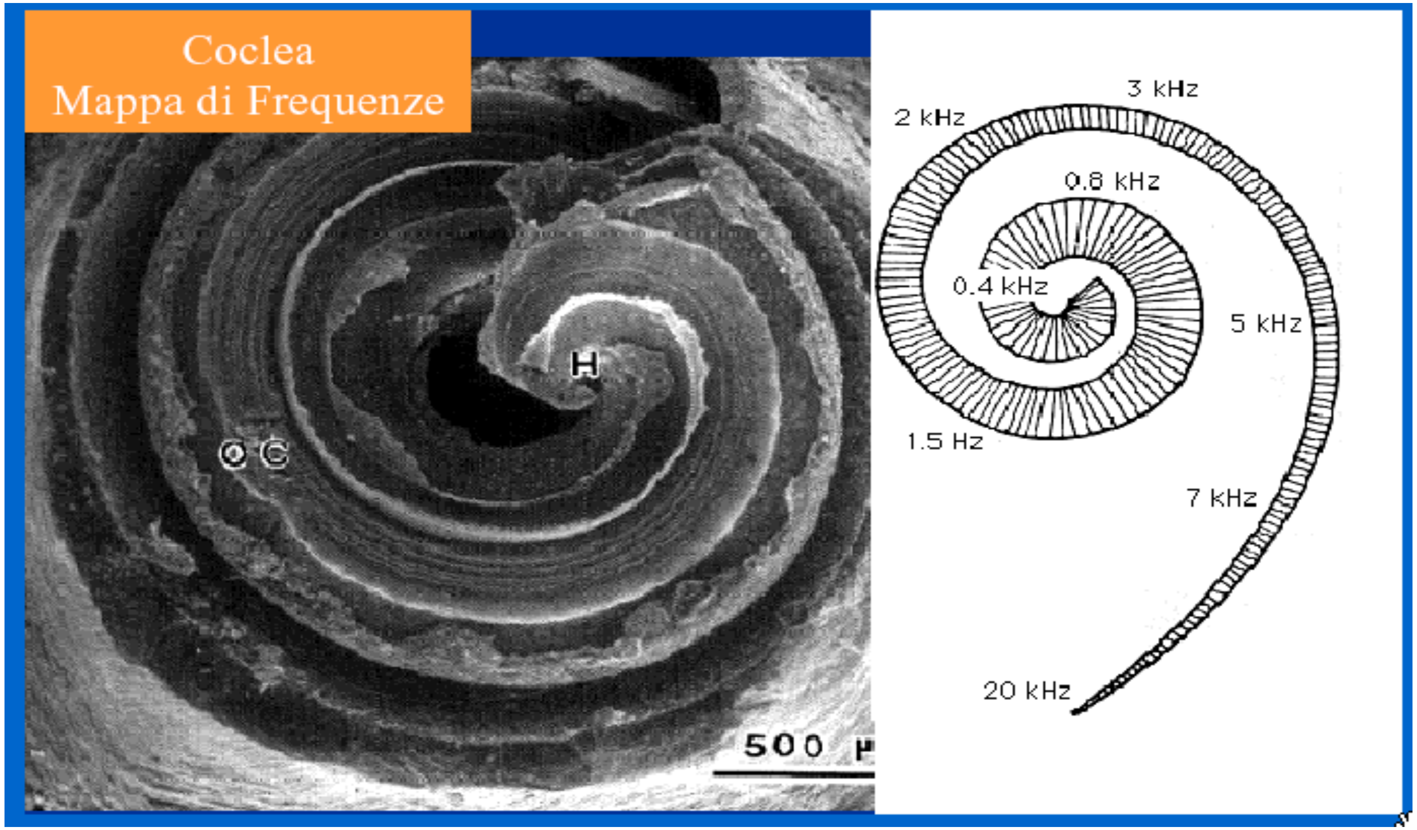
- A cross-section of the cochlea shows a double membrane dividing it in two ducts
- the membrane has the capability of resonating at different frequencies, high at the begininning, and progressively lower towards the end of the ducts.
- However, a low frequency sound also stimulates the initial part of the cochlea, which si sensible to high frequency. Also the opposite occurs, but at much lesser extent. This is the **frequency masking** effect.





The Cochlea

- Each point of the cochlea reacts maximally to one given frequency, as shown here for the human cochlea:



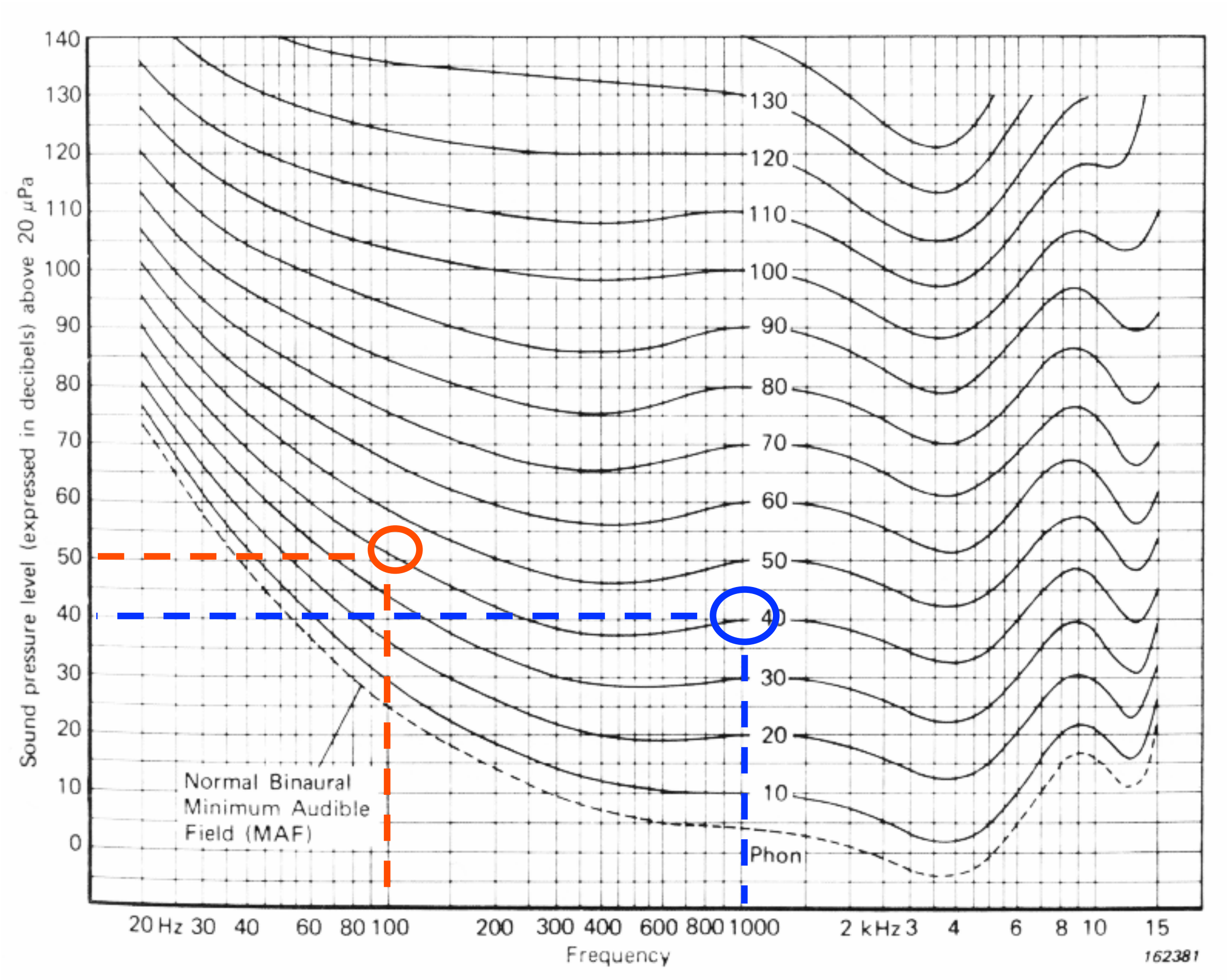
Frequency-dependent sensitivity of human ear:

The sensitivity of the human hearing system is lower at medium-low frequencies and at very high frequencies.

The diagram shows which SPL is required for creating the same loudness perception, in phon, at different frequencies

⇓

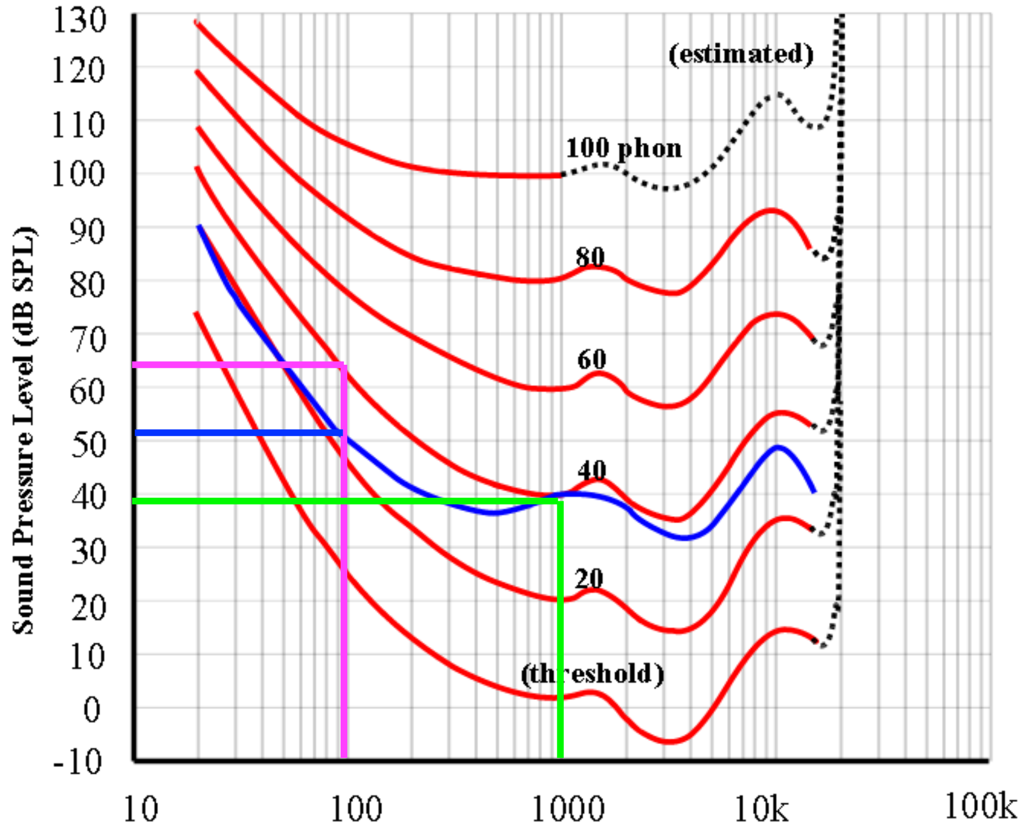
The human ear perceives with diffrent loudness sounds of same SPL at different frequencies.



The new “equal Loudness” ISO curves:

In 2003 the ISO 226 standard was revised. In the new standard, the iso-phon curves are significantly more curved:

With these new curves, a sound of 40 dB at 1000 Hz corrisponds to a sound of 64 dB at 100 Hz (it was just 51 dB before).

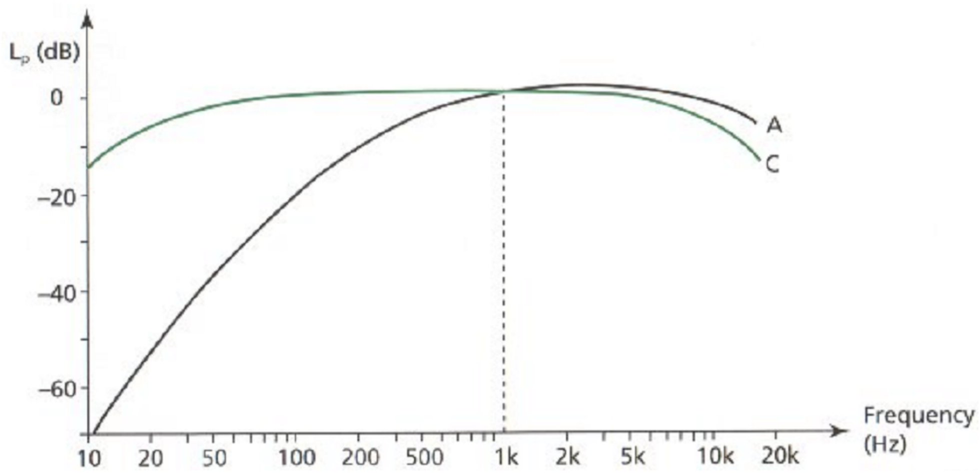


Equal-loudness contours (red) (from ISO 226:2003 revision)
Original ISO standard shown (blue) for 40-phons

Weighting filters:

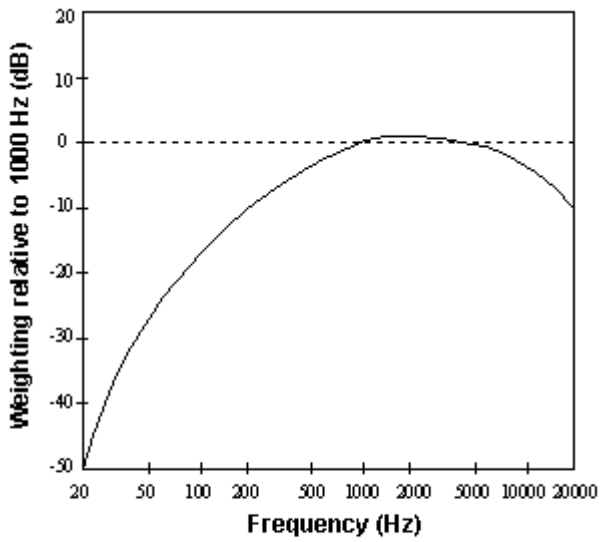
For making a rough approximation of human variable sensitivity with frequency, a number of simple filtering passive networks were defined, named with letters A through E, initially intended to be used for increasing SPL values. Of them, just two are still in use nowadays:

- **"A" weighting curve,** employed for low and medium SPL values (up to 90 dB RMS) [dB(A)].
- **"C" weighting curve,** employed for large amplitude pulsive sound peaks (more than 100 dB peak) [dB(C)].



“A weighting” filter:

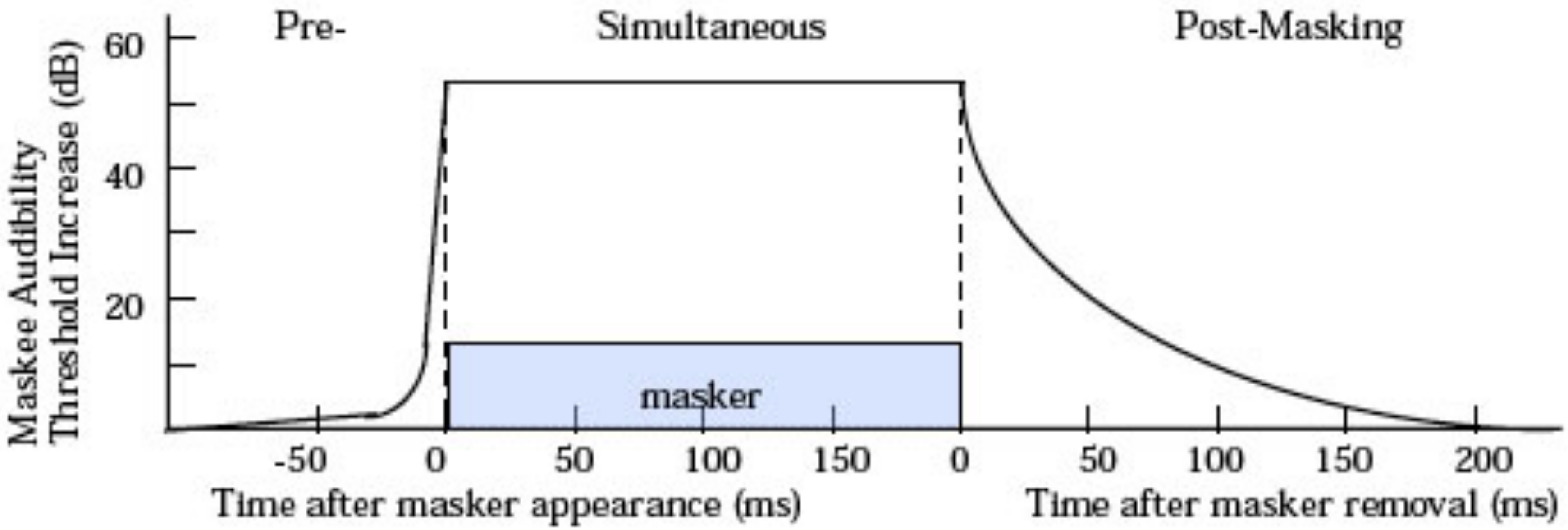
Table of A-weighting factors to be used in calculations



$$A = 10 * Log\left(\frac{3.5041384 * 10^{16} * f^8}{(20.598997^2 + f^2)^2 * (107.65265^2 + f^2) * (737.86223^2 + f^2) * (12194.217^2 + f^2)^2}\right) \quad [dB]$$

f (Hz)	A (dB)	f (Hz)	A (dB)
12.5	-63.4		
16	-56.7	16	-56.7
20	-50.5		
25	-44.7		
31.5	-39.4	31.5	-39.4
40	-34.6		
50	-30.2		
63	-26.2	63	-26.2
80	-22.5		
100	-19.1		
125	-16.1	125	-16.1
160	-13.4		
200	-10.9		
250	-8.6	250	-8.6
315	-6.6		
400	-4.8		
500	-3.2	500	-3.2
630	-1.9		
800	-0.8		
1000	0.0	1000	0.0
1250	0.6		
1600	1.0		
2000	1.2	2000	1.2
2500	1.3		
3150	1.2		
4000	1.0	4000	1.0
5000	0.5		
6300	-0.1		
8000	-1.1	8000	-1.1
10000	-2.5		
12500	-4.3		
16000	-6.6	16000	-6.6
20000	-9.3		

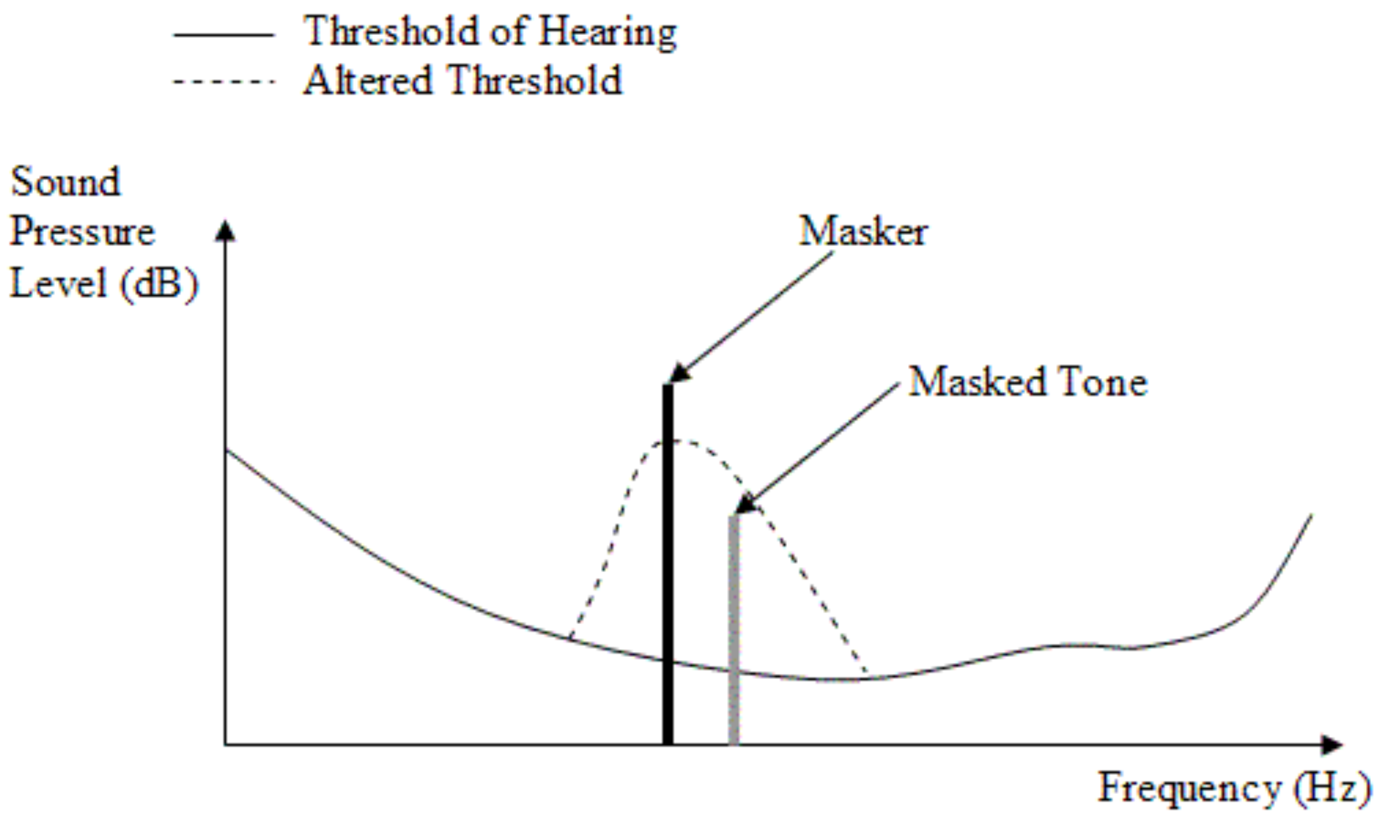
Time masking



After a loud sound, for a while, the hearing system remains “deaf “to weaker sounds, as shown by the Zwicker masking curve above.

The duration of masking depends on the duration of the masker, its amplitude and its frequency.

Frequency masking



A loud pure tone create a “masking spectrum”. Other tones which fall below the masking curve are unadible. The masking curve is asymmetric (a tone more easily masks higher frequencies)

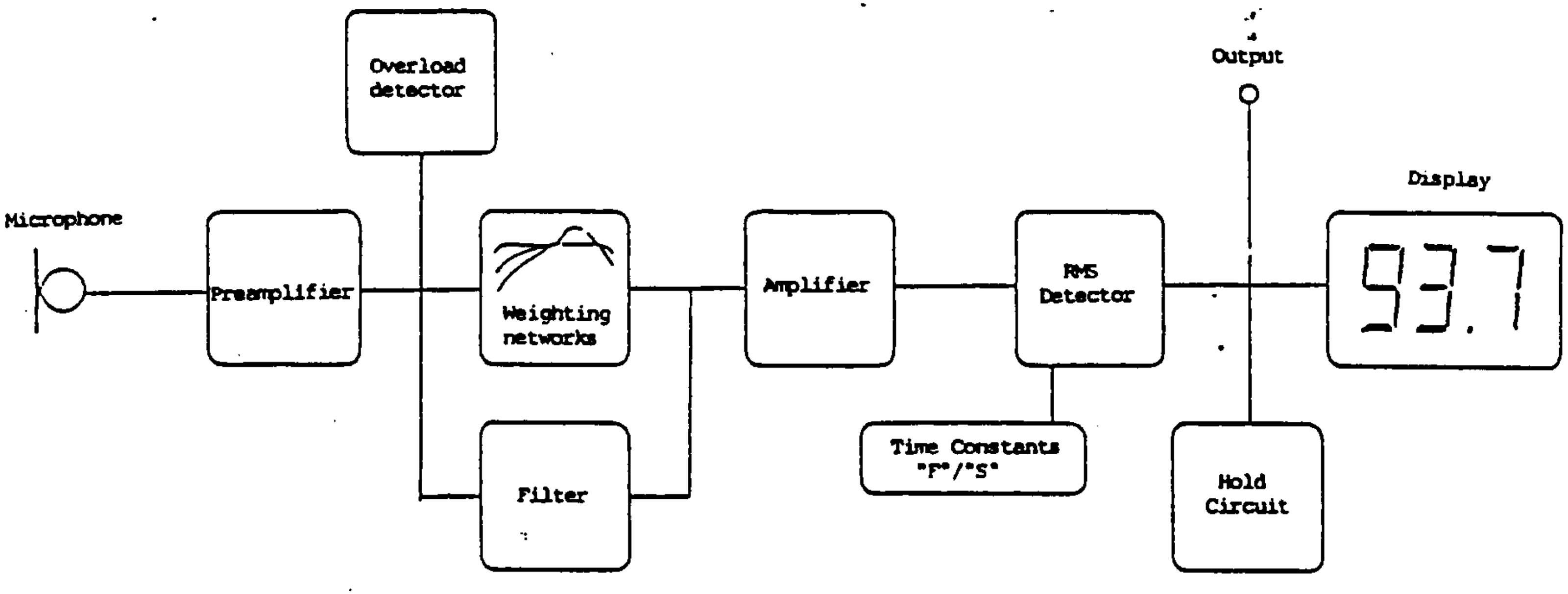
Sound pressure measurement:
sound level meters

The sound level meter

A SLM measures a value in dB, which is the sound pressure level evaluated by the RMS value of the sound pressure, p_{rms} averaged over the measurement time T:

$$Lp = 10\log\left(\frac{p_{rms}}{p_0}\right)^2 \qquad \text{with} \qquad p_{rms} = \sqrt{\frac{1}{T}\int_0^T p^2(t)\,dt}$$

Structure of a sound level meter:



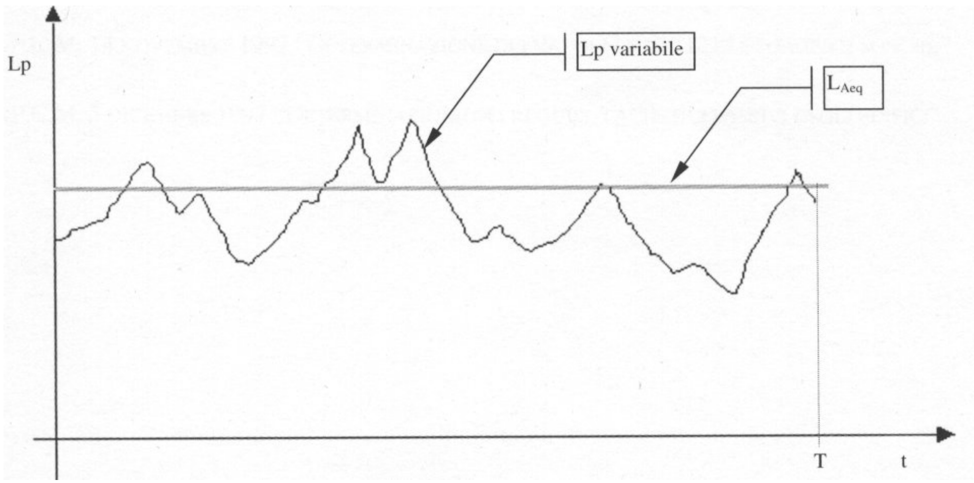
The SLM contains a preamplifier for adjusting the full scale value, a weighting network or a bank of pass-band filters, a "true RMS" detector which can operate either with linear averaging over a fixed measurement time, or a "running exponential averaging" with three possible "time constants", and a display for showing the results.

The Equivalent Continuous Level (L_{eq}):

The **continuous equivalent level** *Leq* (dB) is defined as:

$$L_{eq,T} = 10\log\left[\frac{1}{T}\int_0^T \frac{p^2(t)}{p_{rif}^2}dt\right]$$

where T is the total measurement time, *p(t)* is the instantaneous pressure value and p_{rif} is the reference pressure



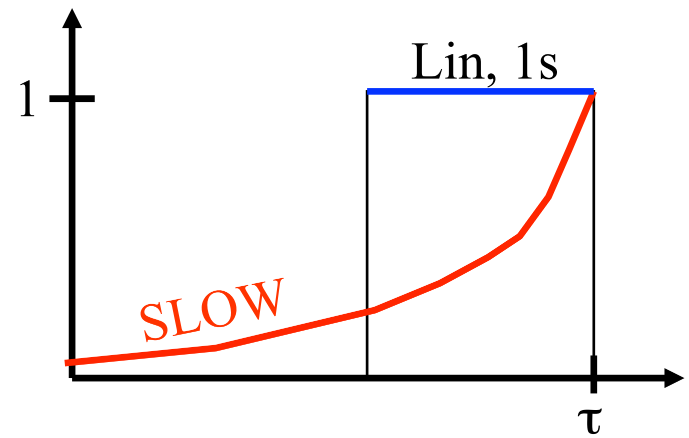
- $L_{eq,T}$ ⇒ dB (linear frequency weighting)
- $L_{Aeq,T}$ ⇒ dB(A) ("A" weighting)

- Please note: whatever the frequency weighting, an Leq is always measured with **linear time weighting** over the whole measurement time T.

"running" exponential averaging: Slow, Fast, Impulse

Instead of measuing the Equivalent Level over the whole measurement time T, the SLM can also operate an "exponential" averaging over time, which continuosly displays an updated value of SPL, averaged with exponentially-decaying weighting over time according to a time constant T_C :

$$p_{rms}(\tau) = \sqrt{\frac{1}{T_c}\int_0^\infty e^{-\frac{t}{T_c}} \cdot p^2(\tau - t)\,dt}$$



in which the time constant T_C can be:

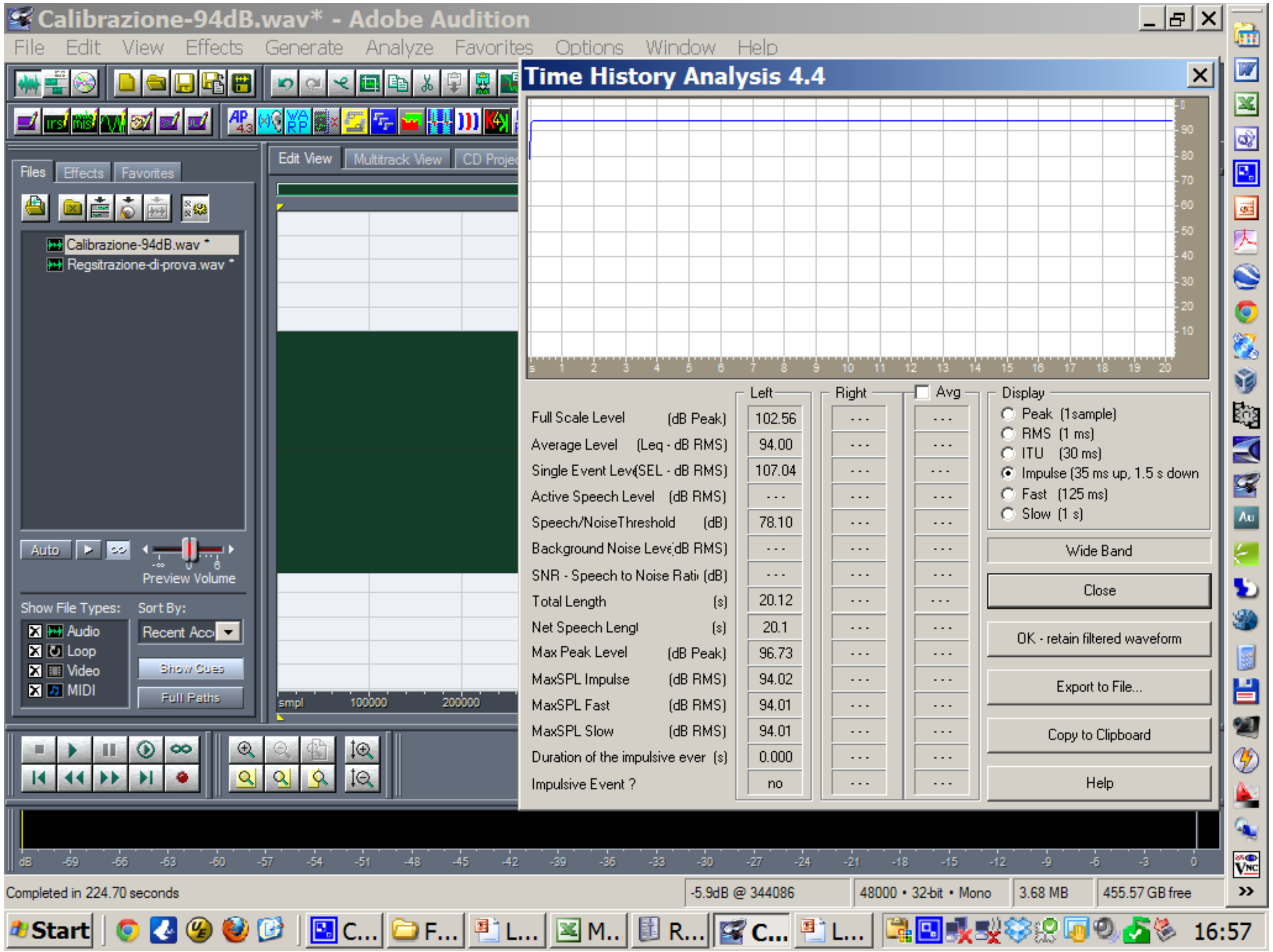
- T_C = 1 s – SLOW
- T_C = 125 ms – FAST
- T_C = 35 ms for raising level, 1.5 s for falling level – IMPULSE

In exponential mode, a SLM tends to "forget" progressively past events……

Instead, in linear mode, the result of the measurment is the same if a loud event did occur at the beginning or at the end of the measurement time

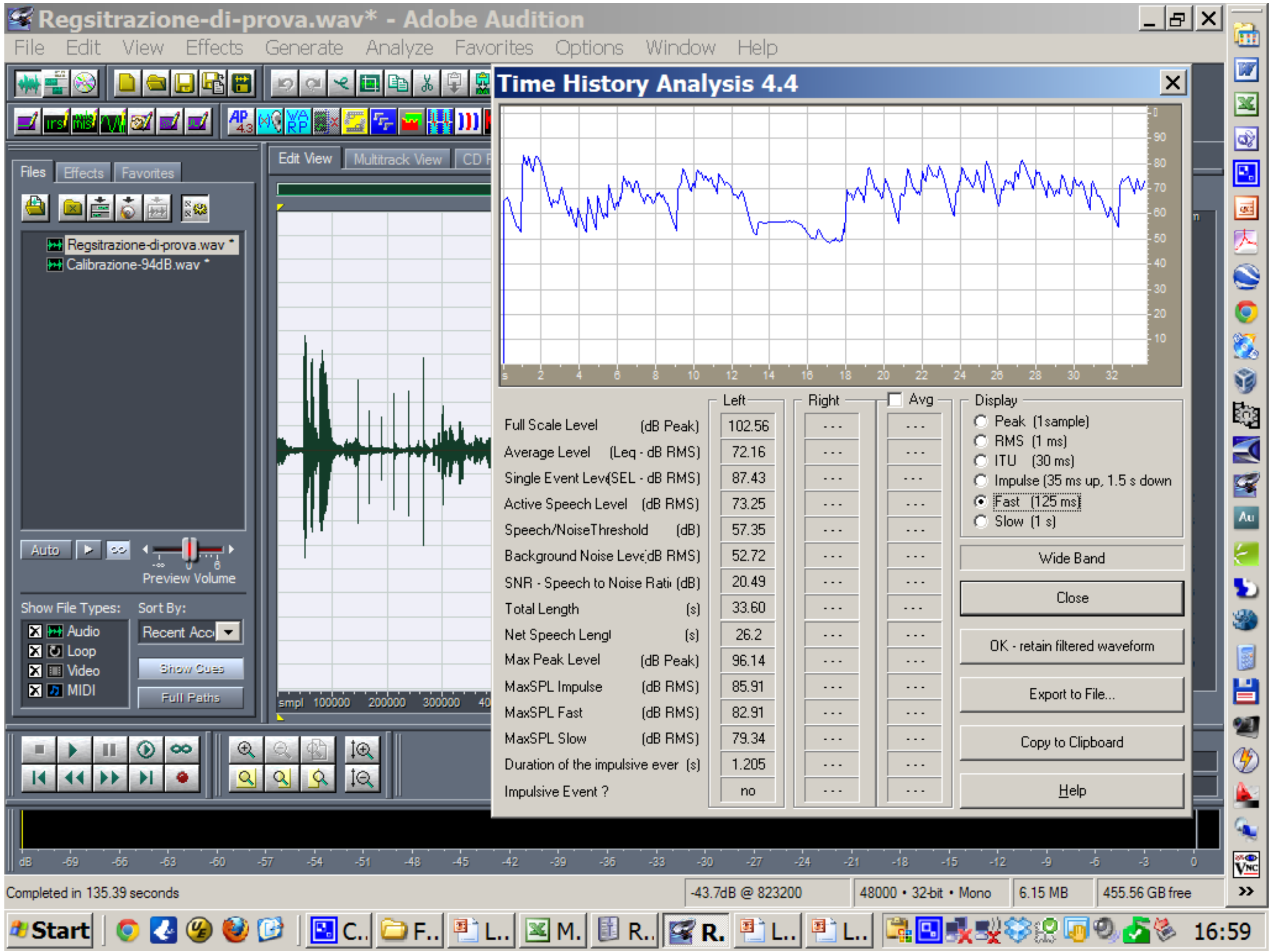
Calibration at 1 Pa RMS (94 dB)

The calibrator generates a pure tone at 1 kHz, with RMS pressure of 1 Pa:



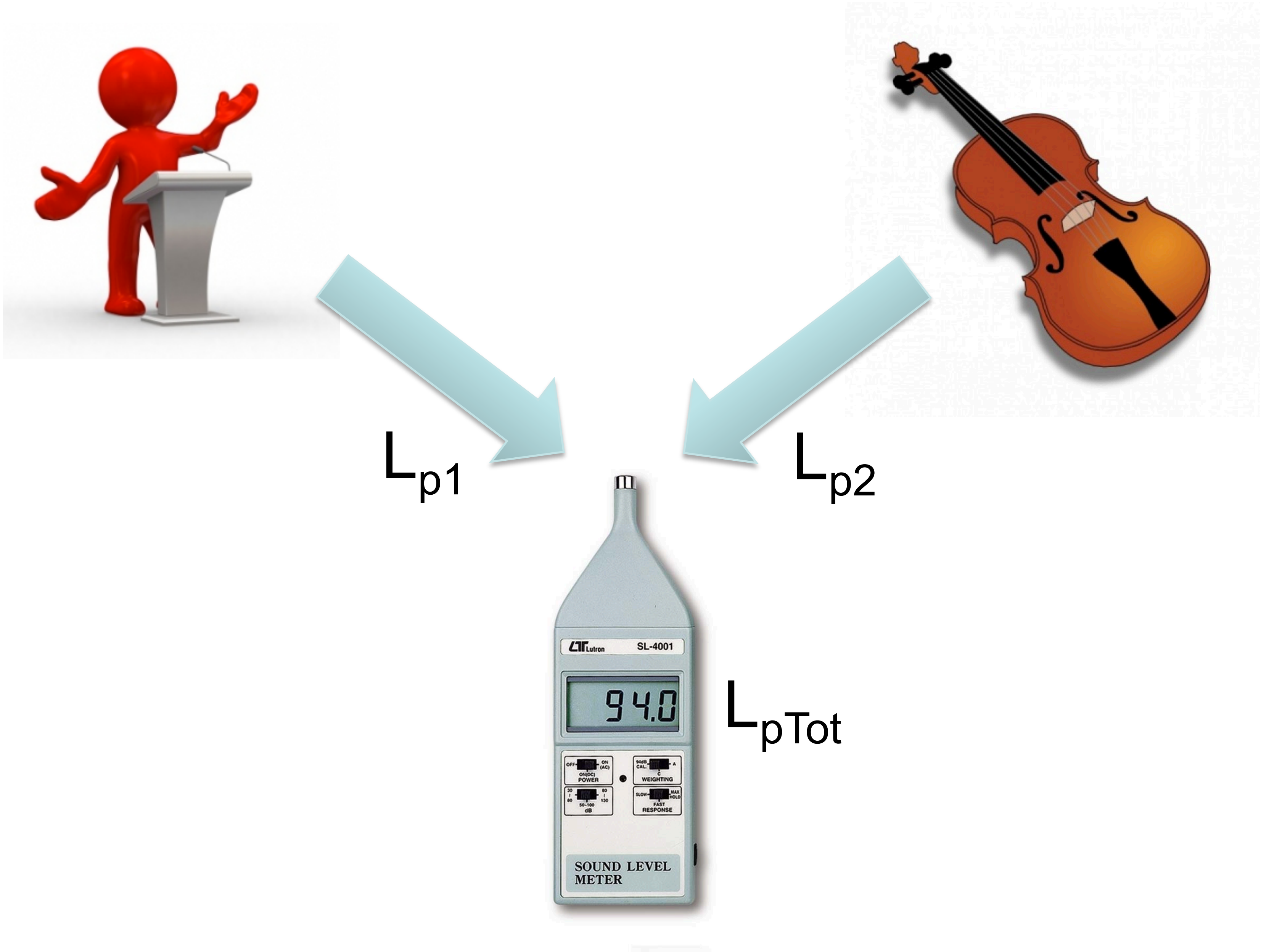
SPL analysis of a calibrated recording

The software computes a time chart of SPL with the selected time constant:



Sum and difference of levels in dB

Sum of two different sources



Sound level summation in dB (1):

<u>"incoherent" sum of two "different" sounds</u>:

$$\mathbf{Lp_1} = 10 \log (p_1/p_{rif})^2 \qquad (p_1/p_{rif})^2 = 10^{\,Lp1/10}$$

$$\mathbf{Lp_2} = 10 \log (p_2/p_{rif})^2 \qquad (p_2/p_{rif})^2 = 10^{\,Lp2/10}$$

$$(p_T/p_{rif})^2 = (p_1/p_{rif})^2 + (p_2/p_{rif})^2 = 10^{\,Lp1/10} + 10^{\,Lp2/10}$$

$$\mathbf{Lp_T} = \mathbf{Lp_1} + \mathbf{Lp_2} = 10 \log (p_T/p_{rif})^2 = 10 \log (10^{\,Lp1/10} + 10^{\,Lp2/10})$$

Sound level summation in dB (2):

<u>"incoherent" sum of two levels</u>

- <u>Example 1</u>:

$L_1 = 80$ dB $\quad$ $L_2 = 85$ dB $\quad$ $L_T = ?$

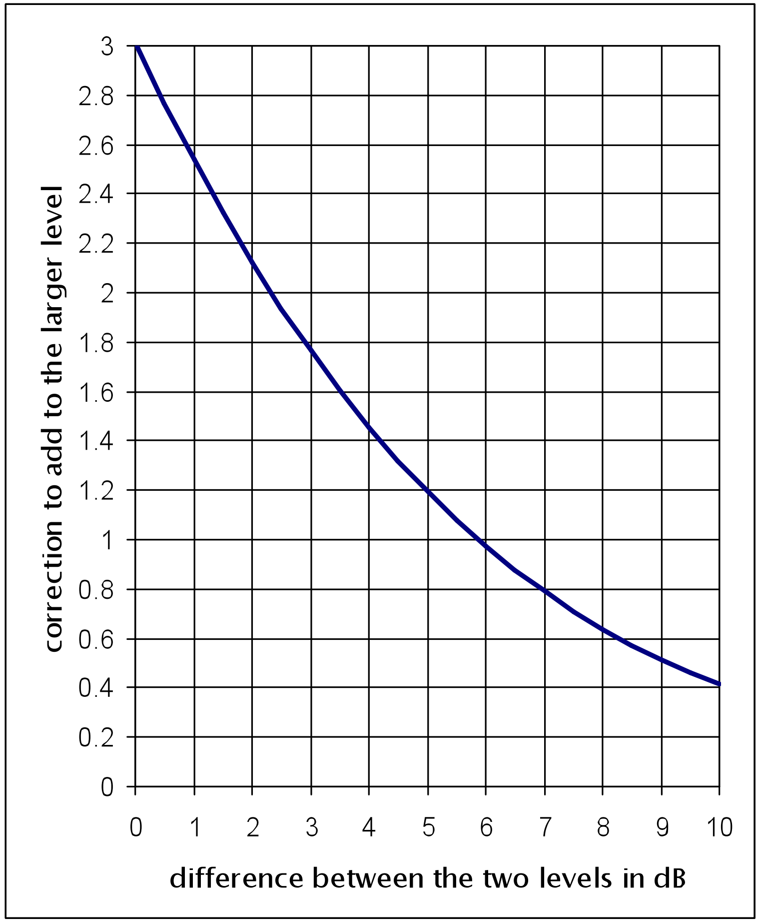
$L_T = 10 \log (10^{80/10} + 10^{85/10}) = 86.2$ dB.

- <u>Example 2</u>:

$L_1 = 80$ dB $\quad$ $L_2 = 80$ dB

$L_T = 10 \log (10^{80/10} + 10^{80/10}) =$

$L_T = 80 + 10 \log 2 = 83$ dB.



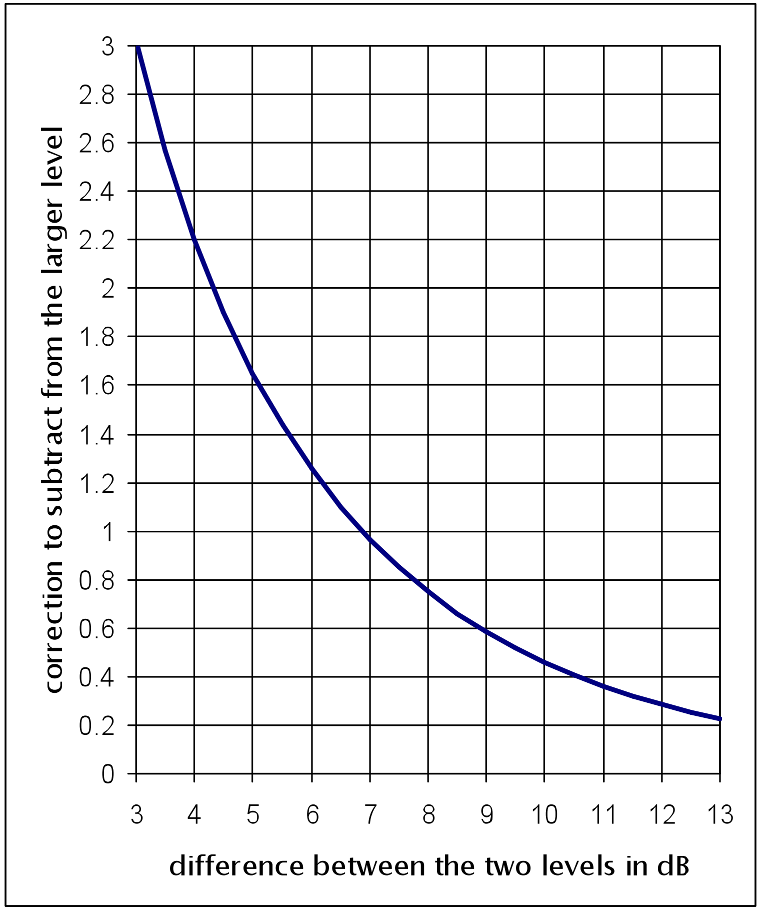
Sound level subtraction in dB (3):

<u>"incoherent" Level difference</u>

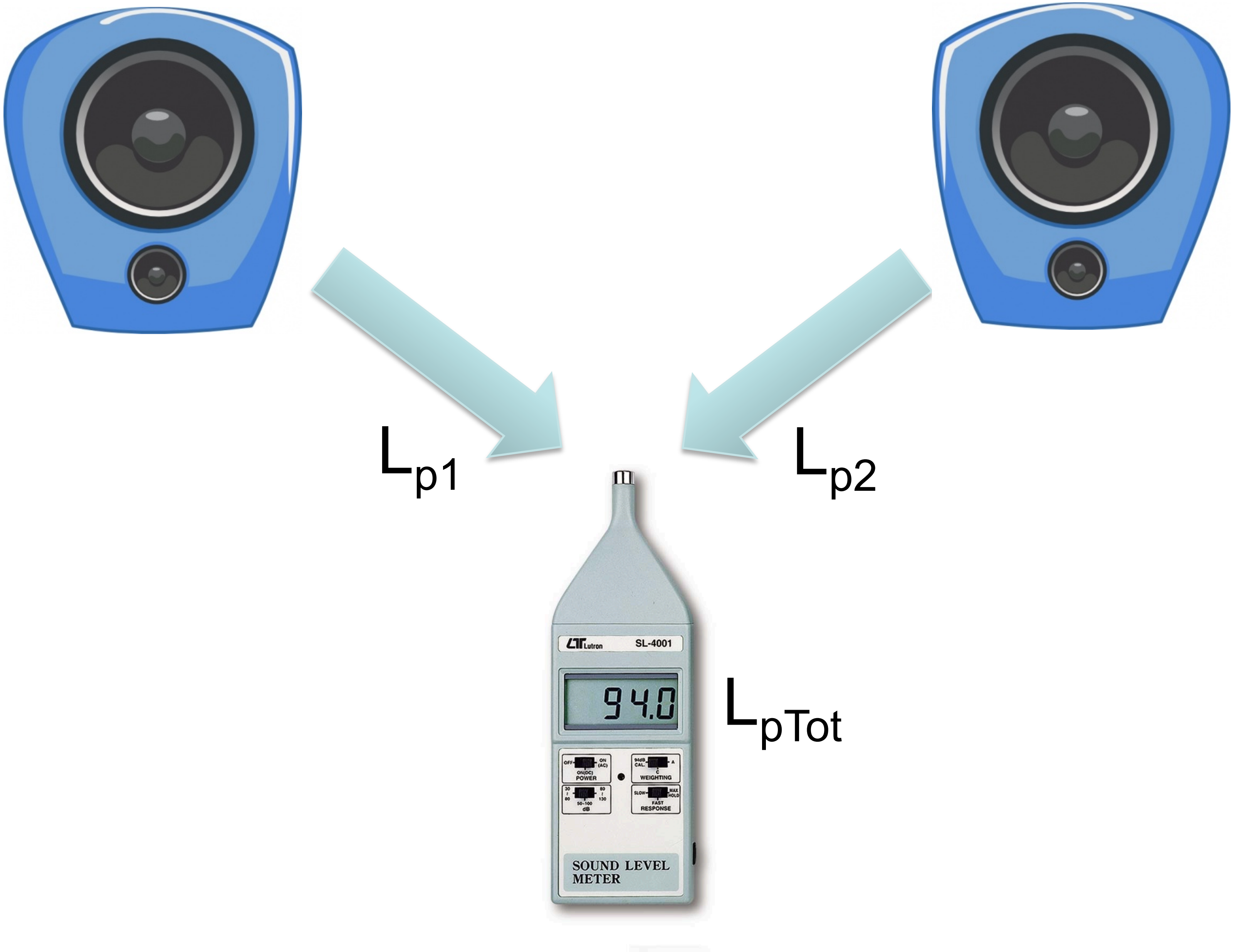
- <u>Example 3</u>:

$L_1 = 80$ dB $\quad L_T = 85$ dB $\quad L_2 = ?$

$L_2 = 10 \log (10^{85/10} - 10^{80/10}) = 83.35$ dB



Sum of two identical sources



Sound level summation in dB (4):

"coherent" sum of two (identical) sounds:

$$\mathbf{Lp_1} = 20 \log (p_1/p_{rif}) \qquad (p_1/p_{rif}) = 10^{\,Lp1/20}$$

$$\mathbf{Lp_2} = 20 \log (p_2/p_{rif}) \qquad (p_2/p_{rif}) = 10^{\,Lp2/20}$$

$$(p_T/p_{rif}) = (p_1/p_{rif}) + (p_2/p_{rif}) = 10^{\,Lp1/20} + 10^{\,Lp2/20}$$

$$\mathbf{Lp_T} = \mathbf{Lp_1} + \mathbf{Lp_2} = 10 \log (p_T/p_{rif})^2 = 20 \log (10^{\,Lp1/20} + 10^{\,Lp2/20})$$

Sound level summation in dB (5):

<u>"coherent" sum of levels</u>

- Example 4:

$L_1 = 80$ dB $\quad L_2 = 85$ dB $\quad L_T = ?$

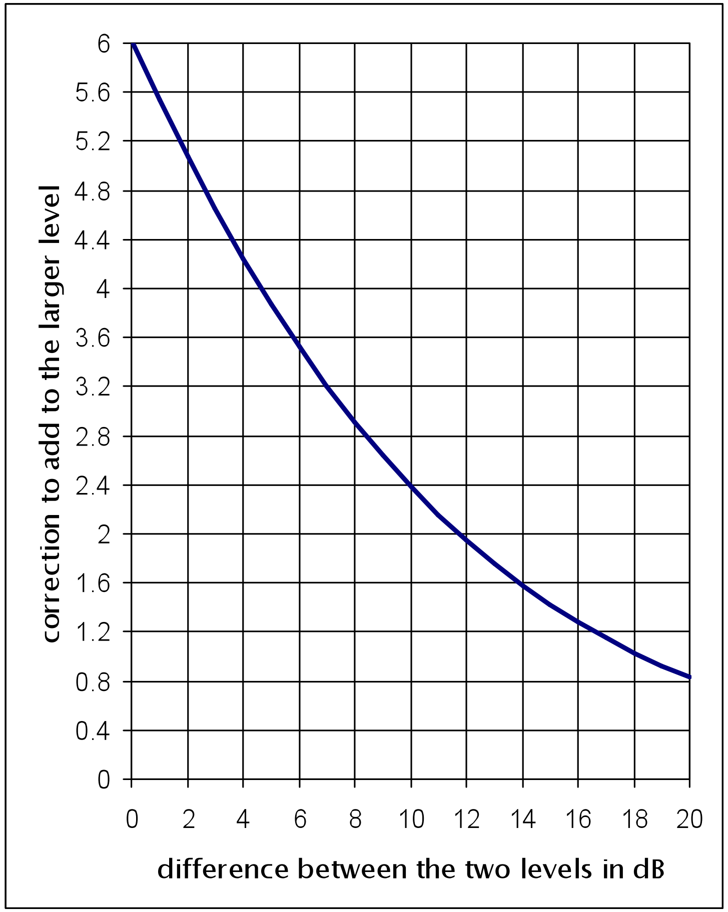
$L_T = 20 \log (10^{80/20} + 10^{85/20}) = 88.9$ dB.

- Example 5:

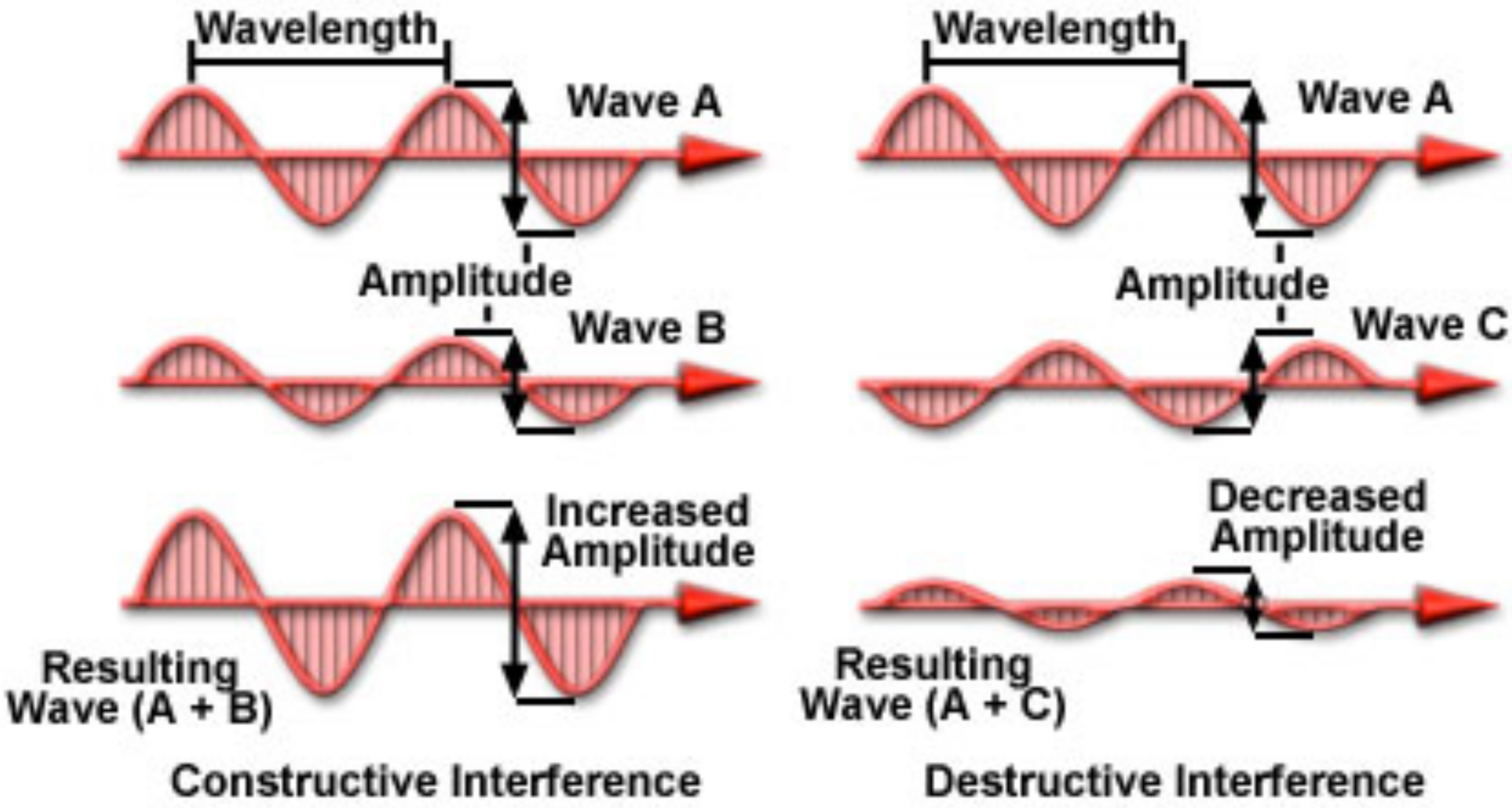
$L_1 = 80$ dB $\quad L_2 = 80$ dB

$L_T = 20 \log (10^{80/20} + 10^{80/20}) =$

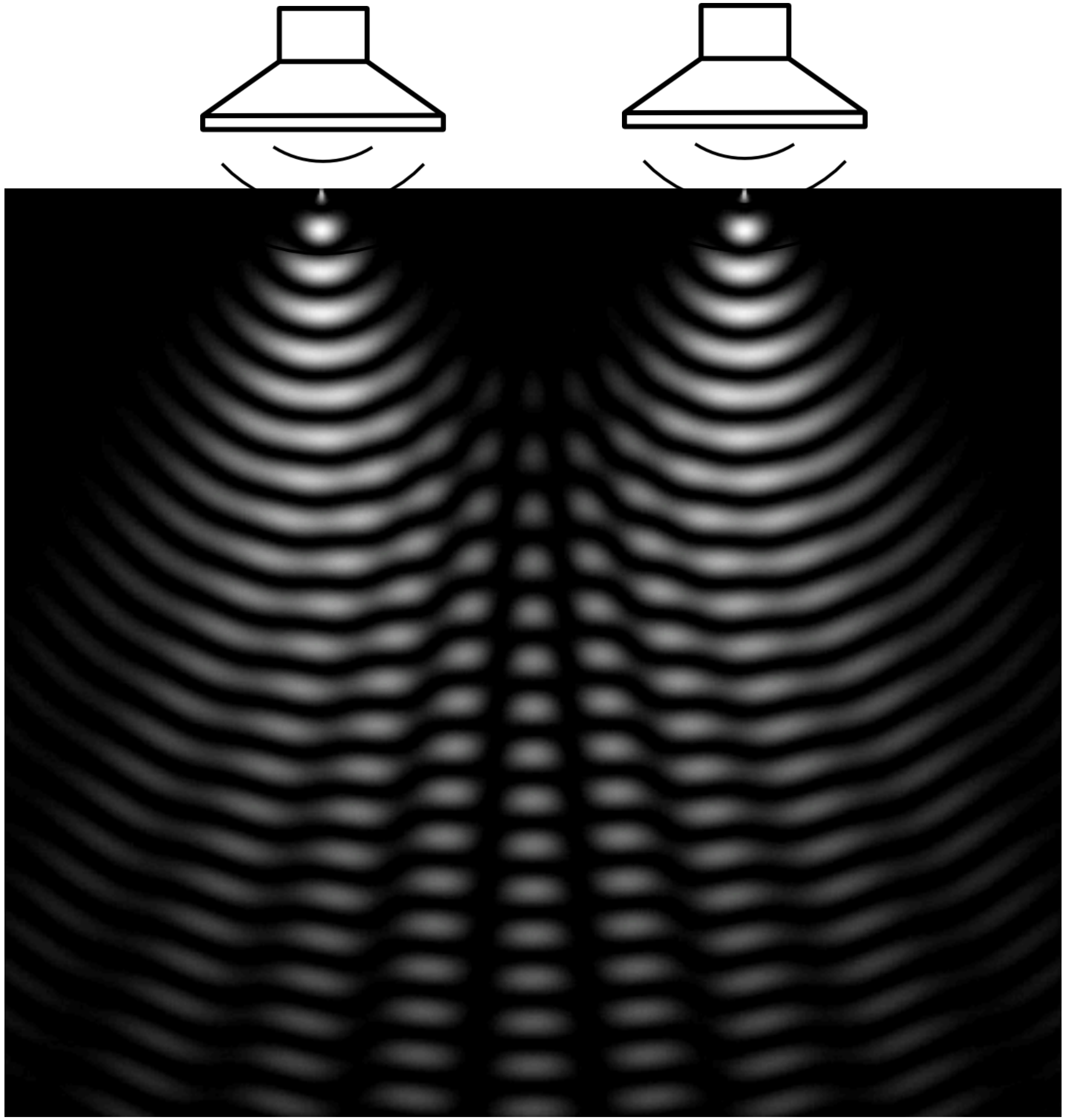
$L_T = 80 + 20 \log 2 = 86$ dB.



Interference between identical sounds



Interference between identical sounds



Coherent difference (destructive interference):

"coherent" difference of levels

- Example 6:

$L_1 = 85$ dB $\qquad L_2 = 80$ dB $\qquad L_T = ?$

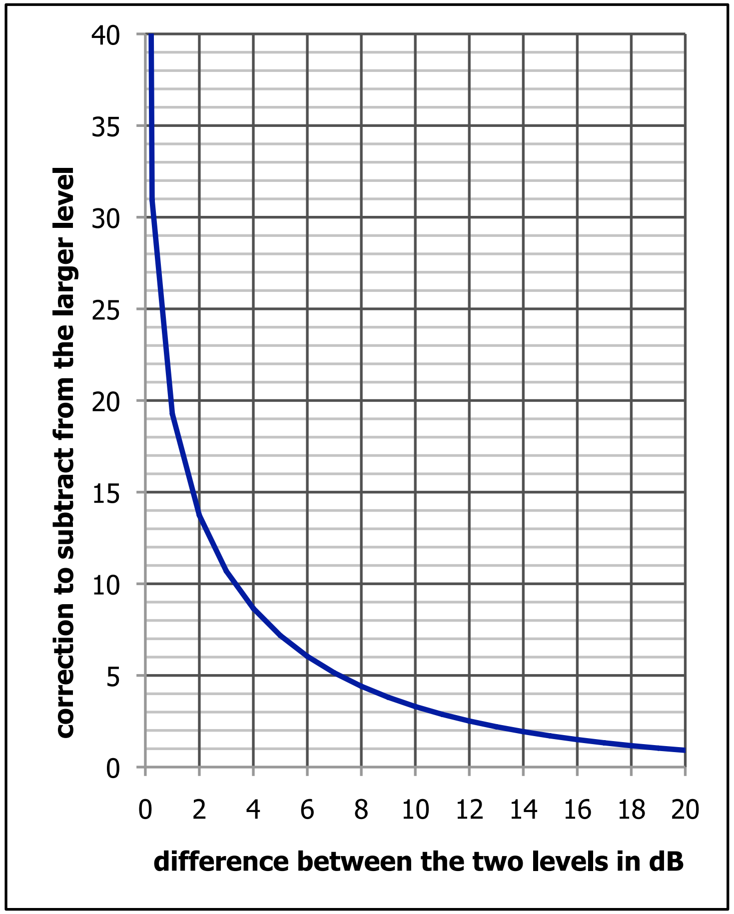
$L_T = 20 \log (10^{85/20} - 10^{80/20}) = 77.8$ dB.

- Example 7:

$L_1 = 80$ dB $\qquad L_2 = 80$ dB

$L_T = 20 \log (10^{80/20} - 10^{80/20}) =$

$L_T = -\infty$ dB.



Frequency analysis

The **sound spectrum** is a chart of SPL vs frequency.

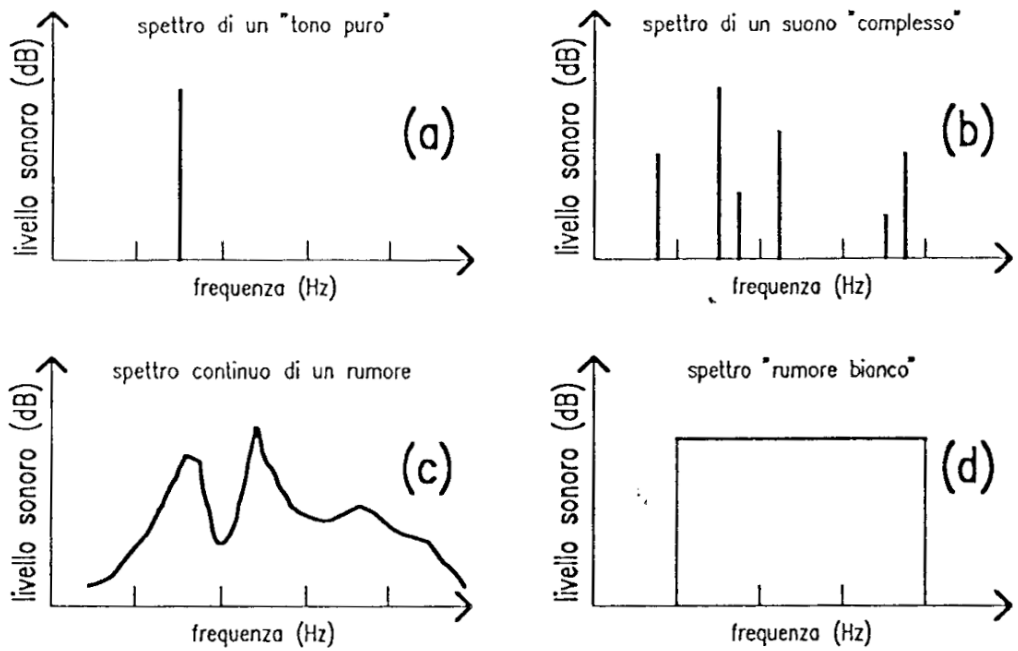
Simple tones have spectra composed by just a small number of “spectral lines”, whilst complex sounds usually have a “continuous spectrum”.

a) Pure tone

b) Musical sound

c) Wide-band noise

d) “White noise”

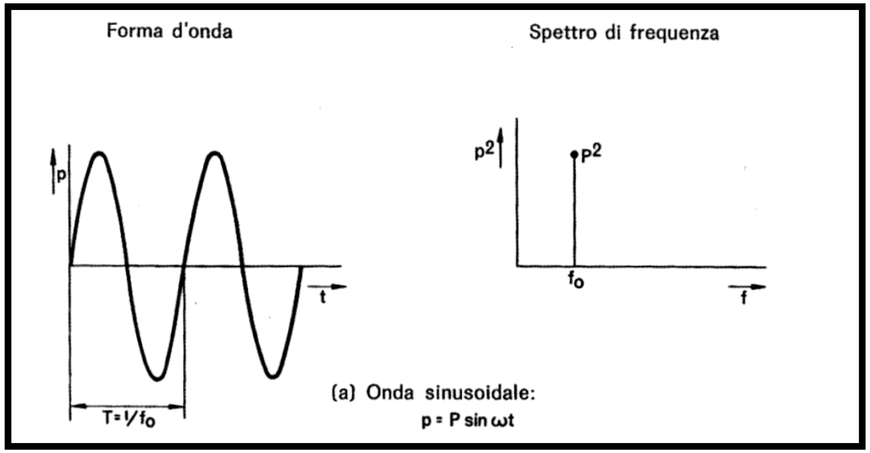


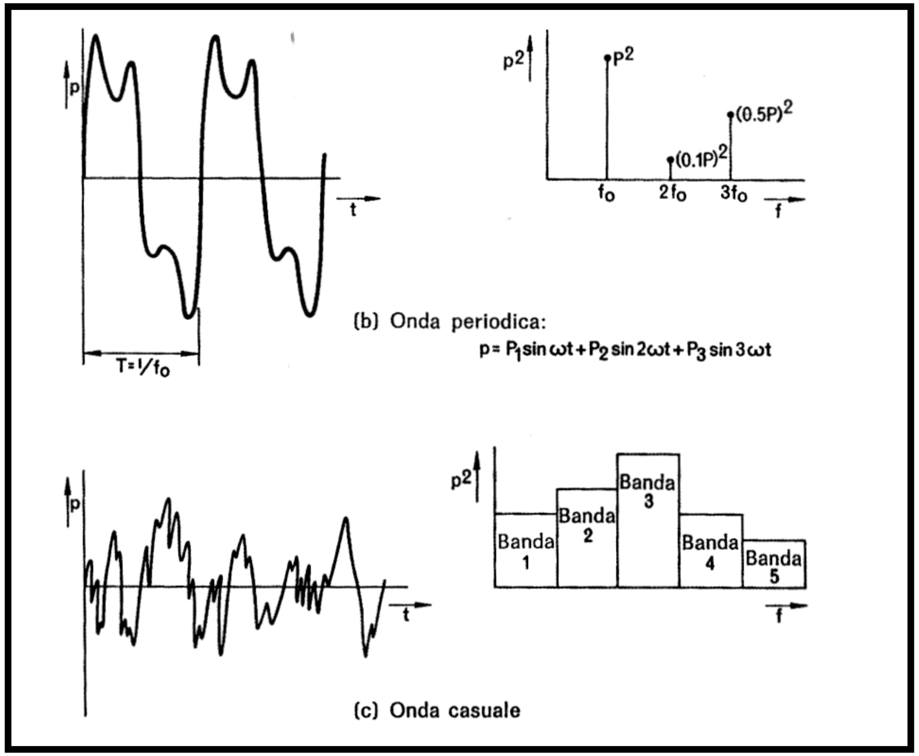
Time-domain waveform and spectrum:

a) Sinusoidal waveform

b) Periodic waveform

c) Random waveform





A practical way of measuring a sound spectrum consist in employing a filter bank, which decomposes the original signal in a number of **frequency bands**.

Each band is defined by two **corner frequencies**, named higher frequency f_{hi} and lower frequency f_{lo}. Their difference is called the **bandwidth** Δf.

Two types of filterbanks are commonly employed for frequency analysis:

- **constant bandwidth** (FFT);
- **constant percentage bandwidth** (1/1 or 1/3 of octave).

Constant bandwidth analysis:

“narrow band”, constant bandwidth filterbank:

- **$\Delta f = f_{hi} - f_{lo}$ = constant**, for example 1 Hz, 10 Hz, etc.

Provides a very sharp frequency resolution (thousands of bands), which makes it possible to detect very narrow pure tones and get their exact frequency.

It is performed efficiently on a digital computer by means of a well known algorithm, called FFT (Fast Fourier Transform)

Constant percentage bandwidth analysis:

Also called "octave band analysis"

- The bandwidth Δf is a constant ratio of the center frequency of each band, which is defined as: $f_c = \sqrt{f_{hi} \cdot f_{lo}}$

- $\frac{\Delta f}{f_c} = \frac{1}{\sqrt{2}} = 0.707$ $f_{hi} = 2\, f_{lo}$ 1/1 octave

- $\frac{\Delta f}{f_c} = 0.232$ $f_{hi} = 2^{1/3}\, f_{lo}$ 1/3 octave

Widely employed for noise measurments. Typical filterbanks comprise 10 filters (octaves) or 30 filters (third-octaves), implemented with analog circuits or, nowadays, with IIR filters

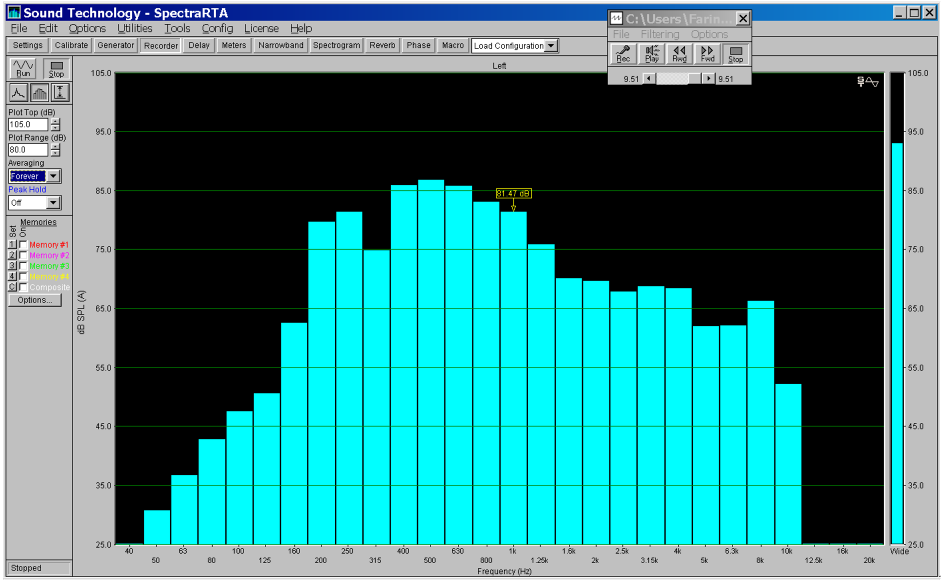
Nominal frequencies for octave and 1/3 octave bands:

- 1/1 octave bands
- 1/3 octave bands

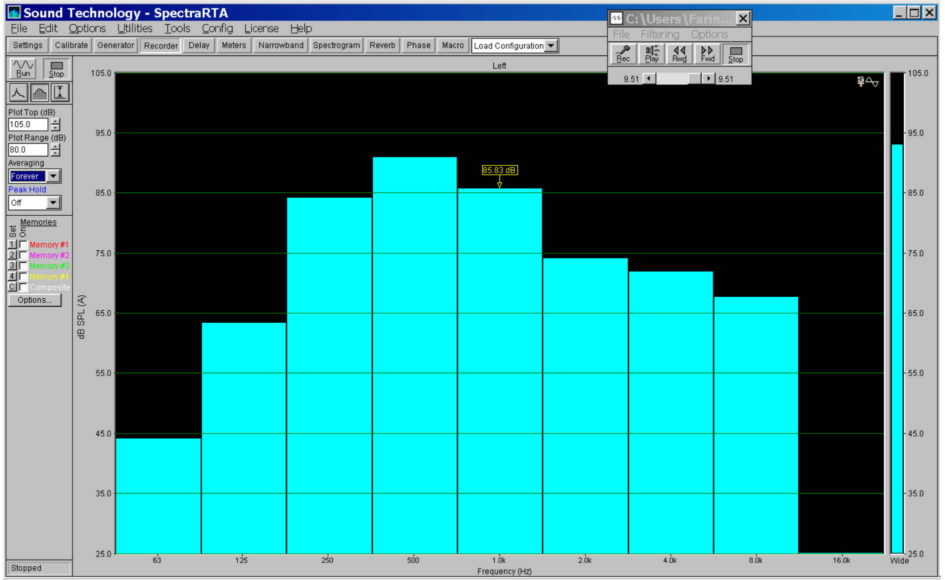
Bande di 1/1 ottava			Bande di 1/3 ottava		
Frequenza limite inferiore	Frequenza di centro banda	Frequenza limite superiore	Frequenza limite inferiore	Frequenza di centro banda	Frequenza limite superiore
11	16	22	14,1	16	17,8
			17,8	20	22,4
			22,4	25	28,2
22	31,5	44	28,2	31,5	35,5
			35,5	40	44,7
			44,7	50	56,2
44	63	88	56,2	63	70,8
			70,8	80	89,1
			89,1	100	112
88	125	177	112	125	141
			141	160	178
			178	200	224
177	250	355	224	250	282
			282	315	355
			355	400	447
355	500	710	447	500	562
			562	630	708
			708	800	891
710	1000	1420	891	1000	1122
			1122	1250	1413
			1413	1600	1778
1420	2000	2840	1778	2000	2239
			2239	2500	2818
			2818	3150	3548
2840	4000	5680	3548	4000	4467
			4467	5000	5623
			5623	6300	7079
5680	8000	11360	7079	8000	8913
			8913	10000	11220
			11220	12500	14130
11360	16000	22720	14130	16000	17780
			17780	20000	22390

Octave and 1/3 octave spectra:

•1/3 octave bands



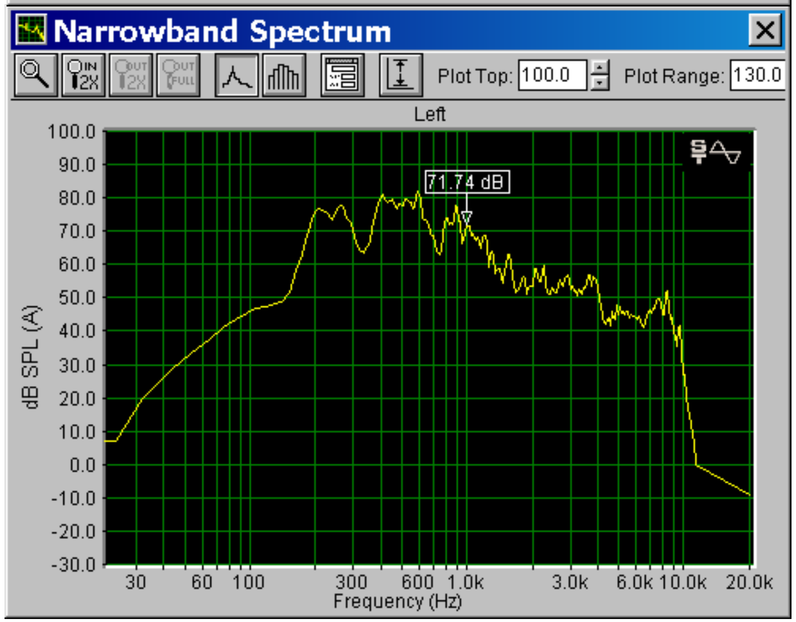
•1/1 octave bands



Narrowband spectra:

- Linear frequency axis

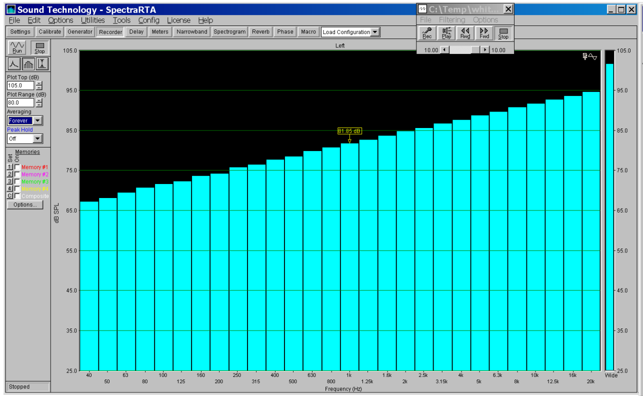
- Logaritmic frequency axis

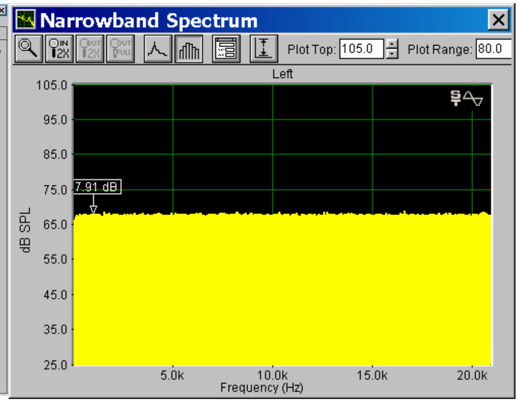


White noise and pink noise

- White Noise:

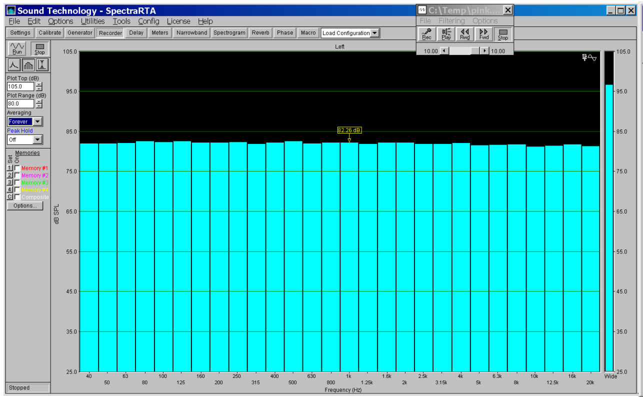
Flat in a narrowband analysis

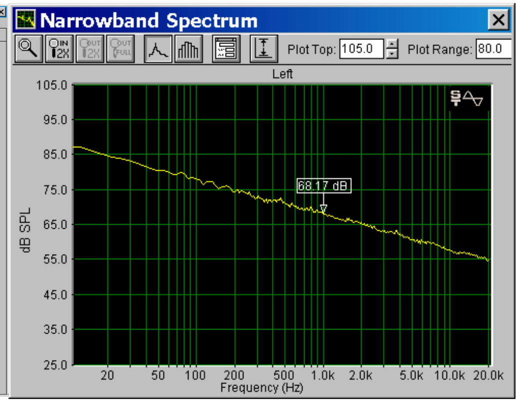




- Pink Noise:

flat in octave or 1/3 octave analysis





Critical Bands (BARK):

The **Bark scale** is a psychoacoustical scale proposed by Eberhard Zwicker in 1961. It is named after Heinrich Barkhausen who proposed the first subjective measurements of loudness

Bark

N.	Center freq.	LoFreq	HiFreq	Bandwidth
1	50	0	100	100
2	150	100	200	100
3	250	200	300	100
4	350	300	400	100
5	450	400	510	110
6	570	510	630	120
7	700	630	770	140
8	840	770	920	150
9	1000	920	1080	160
10	1170	1080	1270	190
11	1370	1270	1480	210
12	1600	1480	1720	240
13	1850	1720	2000	280
14	2150	2000	2320	320
15	2500	2320	2700	380
16	2900	2700	3150	450
17	3400	3150	3700	550
18	4000	3700	4400	700
19	4800	4400	5300	900
20	5800	5300	6400	1100
21	7000	6400	7700	1300
22	8500	7700	9500	1800
23	10500	9500	12000	2500
24	13500	12000	15500	3500

Terzi d'ottava

N.	Center freq.	LoFreq	HiFreq	Bandwidth
1	25	22	28	6
2	31.5	28	35	7
3	40	35	45	9
4	50	45	56	11
5	63	56	71	15
6	80	71	89	18
7	100	89	112	22
8	125	112	141	30
9	160	141	179	37
10	200	179	224	45
11	250	224	281	57
12	315	281	355	74
13	400	355	447	92
14	500	447	561	114
15	630	561	710	149
16	800	710	894	184
17	1000	894	1118	224
18	1250	1118	1414	296
19	1600	1414	1789	375
20	2000	1789	2236	447
21	2500	2236	2806	570
22	3150	2806	3550	743
23	4000	3550	4472	922
24	5000	4472	5612	1140
25	6300	5612	7099	1487
26	8000	7099	8944	1845
27	10000	8944	11180	2236
28	12500	11180	14142	2962
29	16000	14142	17889	3746
30	20000	17889	22361	4472

Critical Bands (BARK):