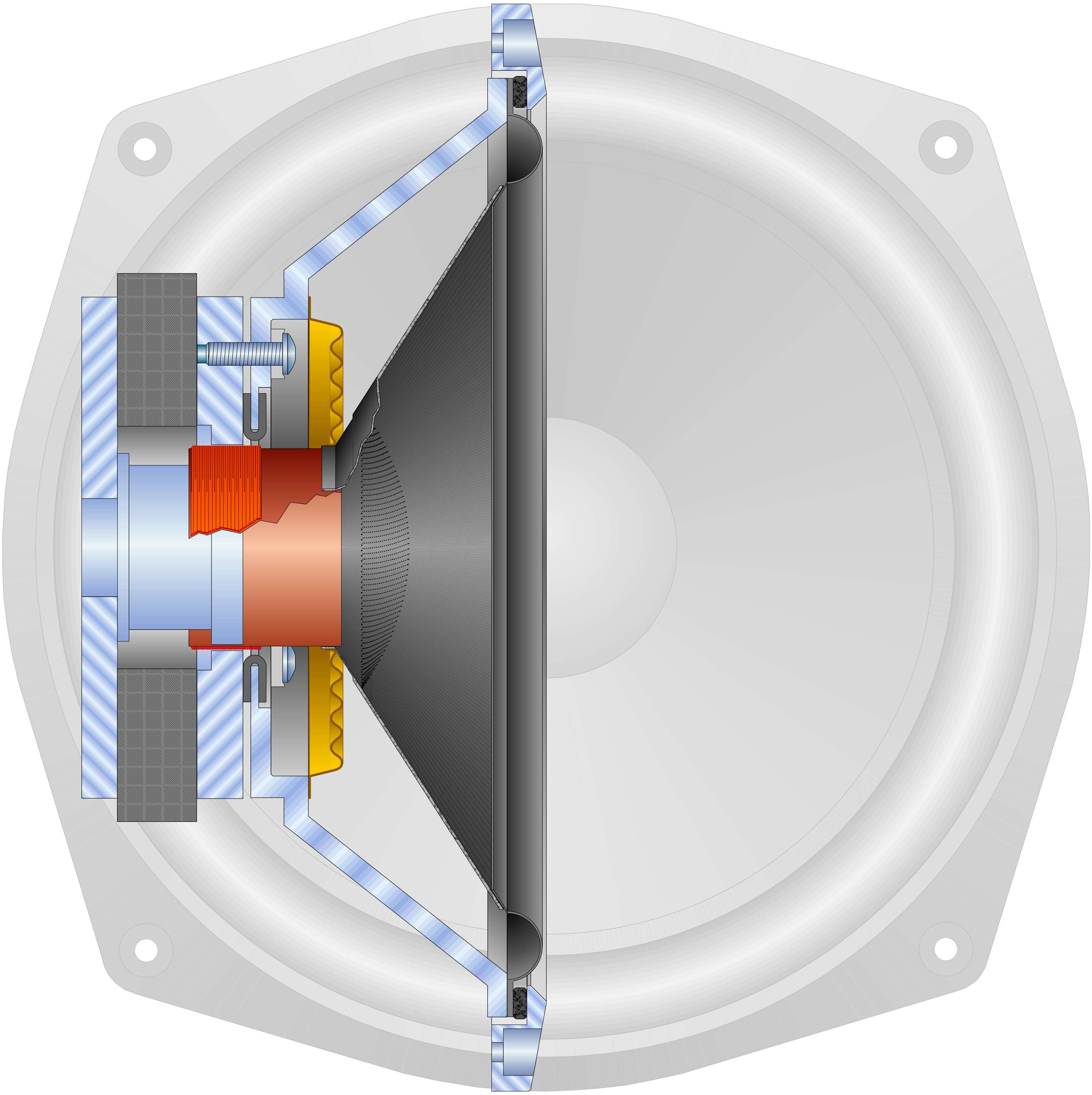
Loudspeakers and acoustic enclosures

Short illustrated description of speaker technologies and the challenges placed on them

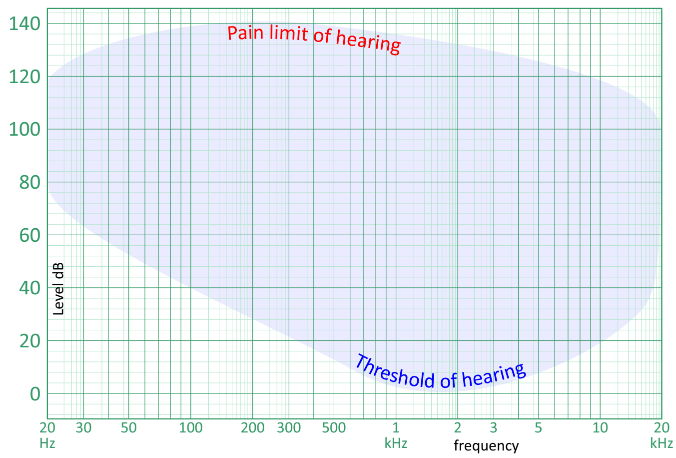


April 2025

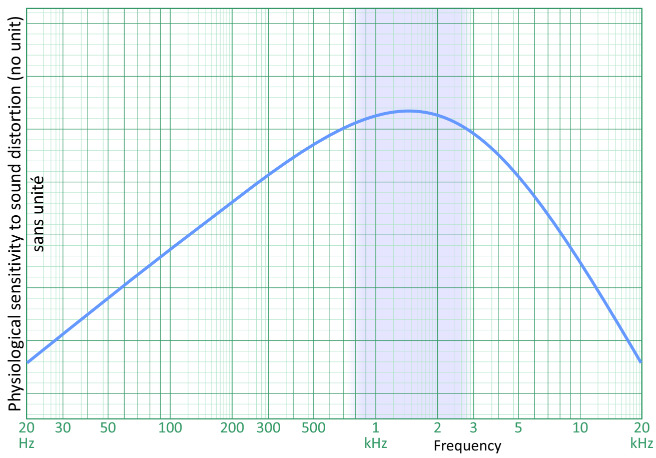
Human physiological sensitivity

Human physiological sensitivity is the basis of all constraints on the reproduction of high-fidelity music. This sensitivity is not linear; neither as far as frequency nor as far as amplitude is concerned.

Between 78 and 102 decibel (dB), a 30 year old person can still hear sound signals with frequencies between 20 and 20,000 Hz. When getting older, people's ears lose some sensitivity at higher frequencies; at over 60 years of age, frequencies above 12 or 14 kHz are attenuated, but not completely lost. During a concert, a music lover of this age hears the music with the ears of his or her age. However, the analysis system of his or her brain has better abilities than the majority of 20-year-old persons. This cultural ability allows him or her to know how to listen better, even if he or she actually hears less well. We could say that we hear with our ears but listen with our brain. Or we could say - forgive this comparison - that our ears are comparable to hardware (microphones) which ages over the years, and that our brain works thanks to continually updated software that has been culturally acquired.



Hearing range of human physiology

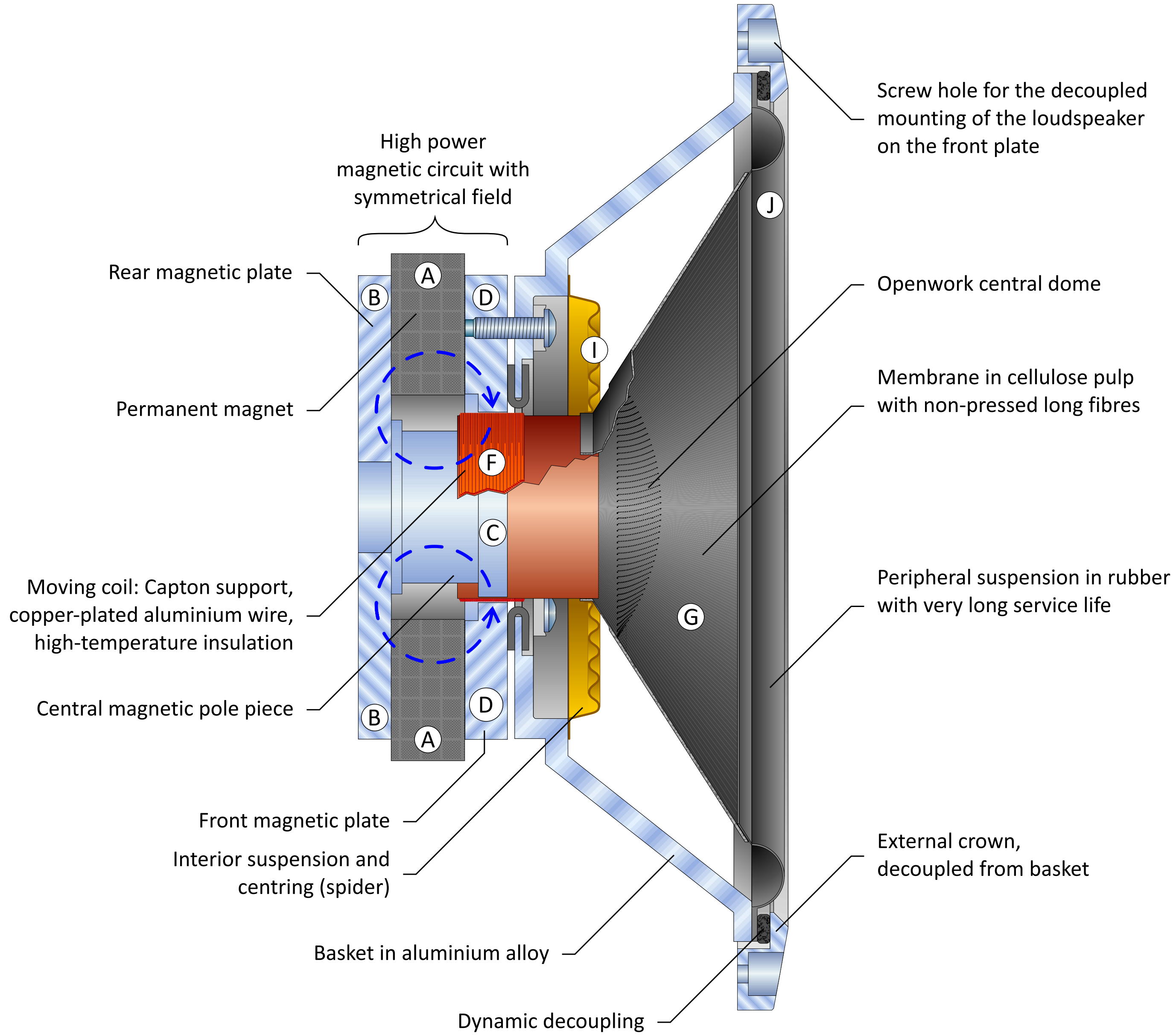


Hypersensitivity zone of human physiology centred at 1,400 Hz

Our physiological sensitivity is thus not linear as far as frequency is concerned, far from it. There is a zone of hypersensitivity located around 1,400 Hz. The frequency range of 2 octaves around 1,400 Hz, i.e. 700 to 2,800 Hz, is a "sacred area", where even the most draconian measures must be taken to reduce any form of distortion in the reproduction of the sound (distortions, decay-time broadening and directional error).

This frequency range therefore requires, as far as possible, the design of an acoustic enclosure starting from and around its midrange loudspeaker. We require extremely precise work from this loudspeaker, whilst taking into account that the field of activity is reduced to just these 2 octaves. In order to complete the spectral reproduction as a whole, the acoustic system must be complemented with a bass-frequency speaker (woofer) and a high-frequency speaker (tweeter). Consequently, a high-performance acoustic enclosure must be fitted with no more (or less) than three speakers to be a real sound reproduction system.

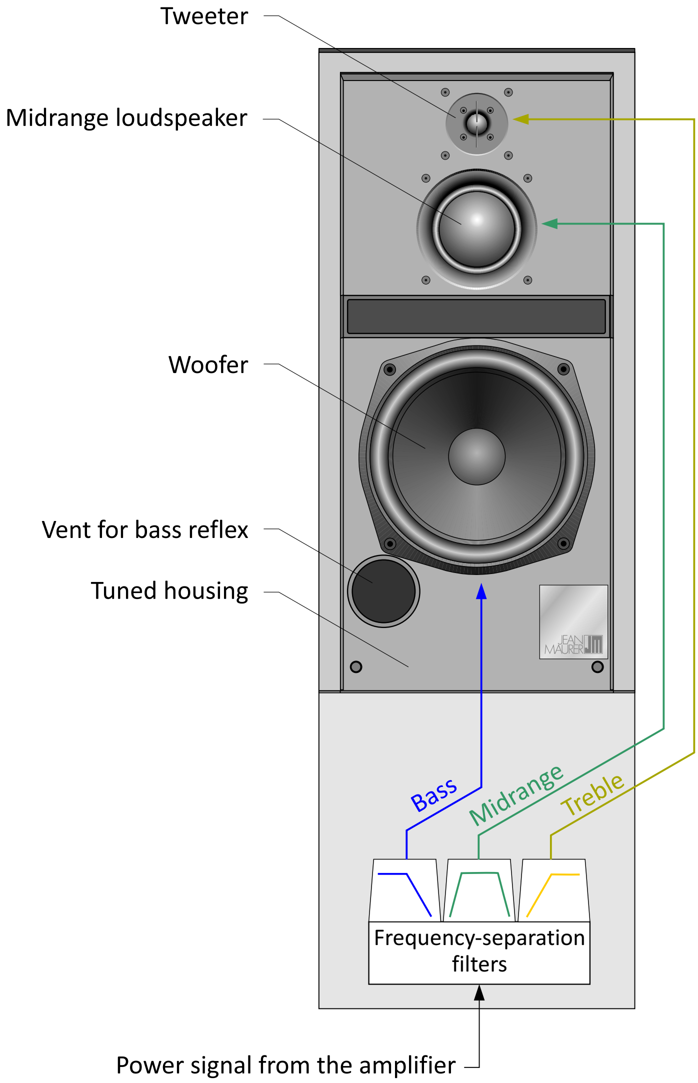
Principle of operation of a loudspeaker



B245-5 woofer

The magnetic circuit, consisting of the circular ferrite magnet A, the rear magnetic plate B, the central core C and the front magnetic plate D, creates a very strong magnetic field in the narrow circular slot located between D and C, called the air gap. The moving coil F and the conical membrane G are suspended by the suspensions I and J, so that the coil is centred in the air gap without touching the sides. The electrical current coming from the amplifier passes through the moving coil and, under the influence of the intense magnetic field in the air gap, pushes the membrane forwards or pulls it back, depending on the direction of the current. By moving forwards and backwards in the rhythm of the electrical signal, (which is the electrical image of the sound captured by the microphone during recording), the music is recreated with more or less true fidelity.

The construction of an acoustic enclosure and the distribution of power load



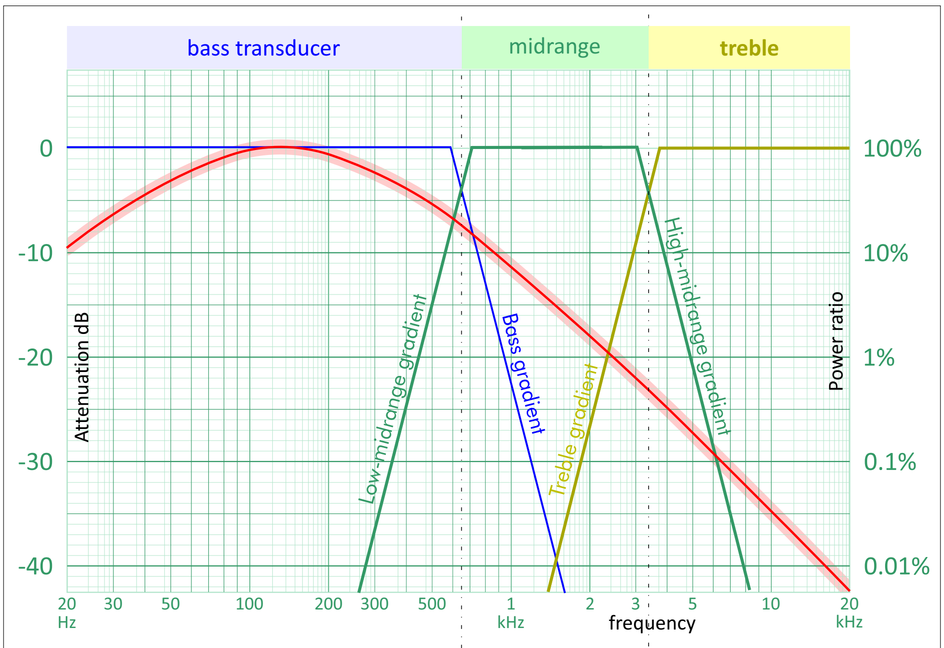
An acoustic enclosure comprises three groups of elements that are of equal importance for overall high-fidelity performance:

- *The loudspeakers, or transducers, which transform the electrical signal from the amplifier into mechanical movement and thus sound.*

- *The filters which direct the electrical signal to each of the speakers, depending on the signal's frequency content*

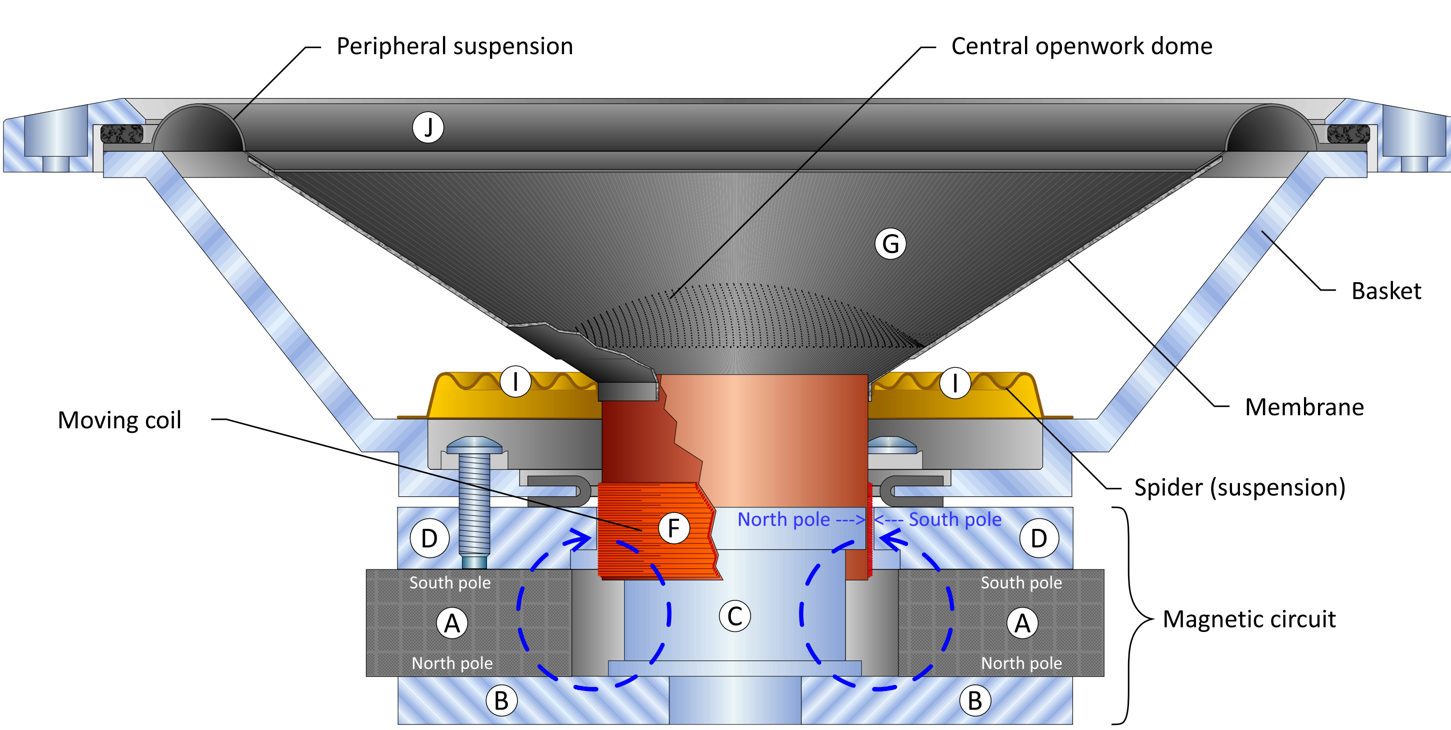
- *The housing or, rather, the housings of the loudspeaker system, which must treat the acoustic signals behind each speaker (sound waves inside the enclosure created by the membranes). This is in order to ensure that the back-wave does not disturb the direct audible signal.*

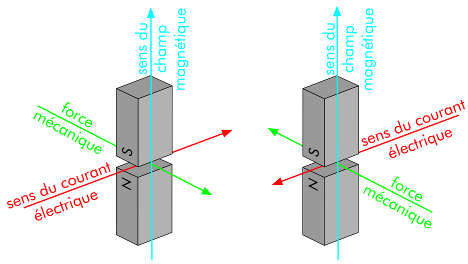
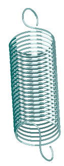
In the case of a symphonic orchestra, the largest portion of the musical energy will be reproduced by the bass loudspeaker (around 80%), which is responsible for the depth of the musical image and its instrumental body. The highest fidelity of reproduction is required from the midrange transducer, which is responsible for the intelligibility of the musical rendering. The sonic space is provided by the treble loudspeaker from which the highest reaction speed is required in order to reproduce the signals well beyond 20 kHz.



In red, the distribution of orchestral power as a function of frequency against energy transmitted by the filters to the three loudspeakers.

The four main loudspeaker parameters



Force-factor: B·ℓ	**Mobile mass: m**	**Compliance: C**	**Surface: S**
This corresponds to the magnetic field in the air gap (circular slot located between C and D) multiplied by the length of the winding of the moving coil immersed in this magnetic field	(commonly designated as Mms) corresponds to the sum of the masses of the moving coil (F) + membrane (G) + part of suspensions (I+J) i.e. those that are in motion	(commonly designated as Cms) corresponds to the sum of the elasticity of peripheral suspensions (J) + internal centring (I)	corresponds to the total emissive surface, i.e. the surface of the membrane (G) + part of the peripheral suspension (J)
			
Tesla·metre (T·m)	**gram** **(g)**	**millimetre/Newton** **(mm/N)**	**metre²** **(m²)**

and **Redc** which is the DC resistance of the moving coil (F)

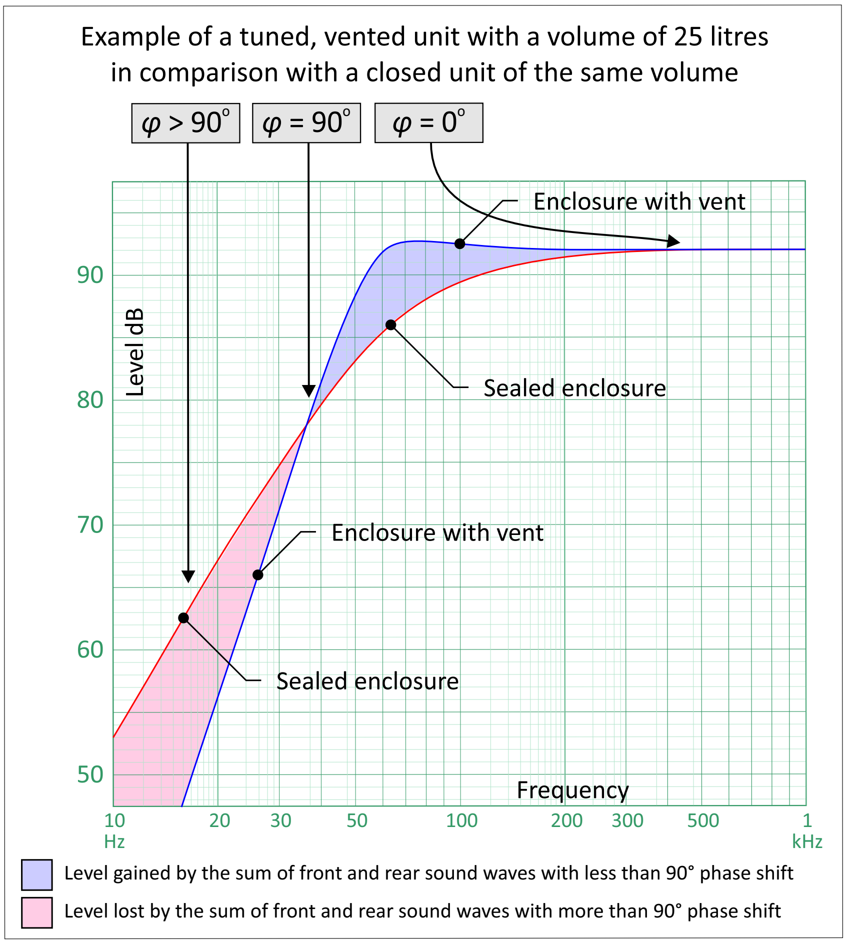
The performance of the loudspeaker is proportional to:

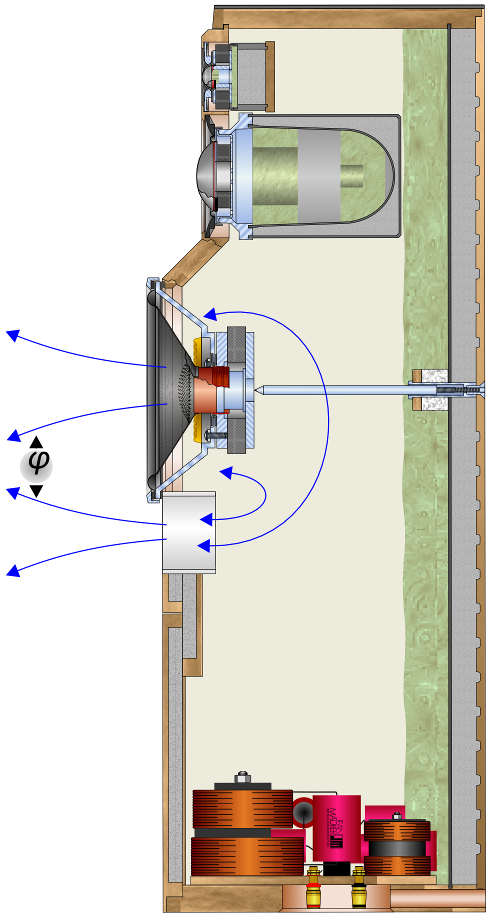
$$\eta \approx \frac{S \cdot (B \cdot \ell)^2}{m^2 \cdot R_{edc}}$$

In this equation, the compliance **Cms is not involved.**

From this formula, however, we can note that the moving mass has an influence proportional to the square of its value: For example, in the case of the B245-5 speaker, with a moving mass of 33 grams, the performance of the loudspeaker will be reduced by 6.15%, i.e. by 0.52 dB, if 1 gram too much of glue is used during the mounting of the moving parts.

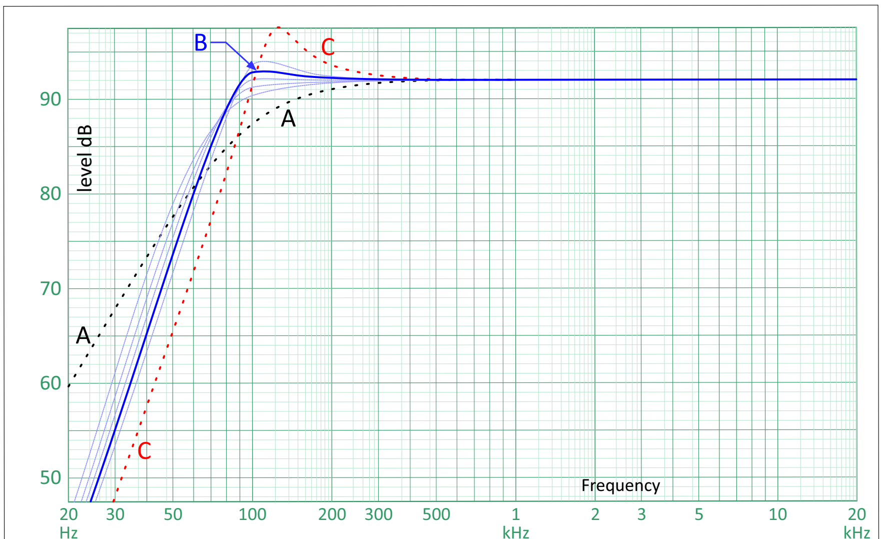
Tuning the housing by the use of vents





For extremely low frequencies, the vent transmits a lot of sound energy from inside the housing to its exterior. This energy, which decreases with increasing frequency, becomes virtually zero above 250 Hz.

In comparison to a closed unit, a well calculated vented bass-reflex unit therefore allows a certain quantity of energy to be recovered from the rear wave of the woofer in order to retrieve it to the listener's room (with a favourable phase relation) and to attenuate the extremely low frequencies that are barely audible and harmful to the loudspeaker (resulting from an unfavourable phase relationship). This system therefore allows the linearization of the response at low frequencies.



Family of curves, for various vent variants in a 25-litre unit:
A : closed unit (without vent)
B : ideally tuned vent
C : vent characteristic tuned to too high frequency = flattering, but very annoying in the long term

The basic criteria of the reproduction of bass frequencies

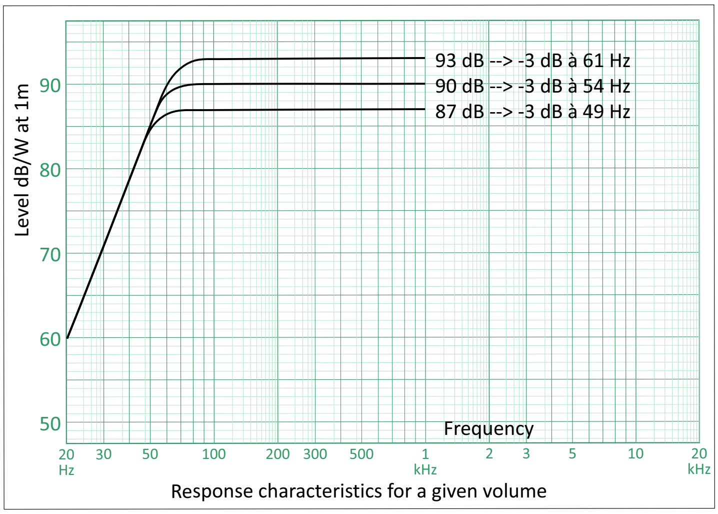
1 - **Efficiency** (dB/W at 1m)

2 - **Volume** (litres) and the type of load (closed volume, vented system, or passive load)

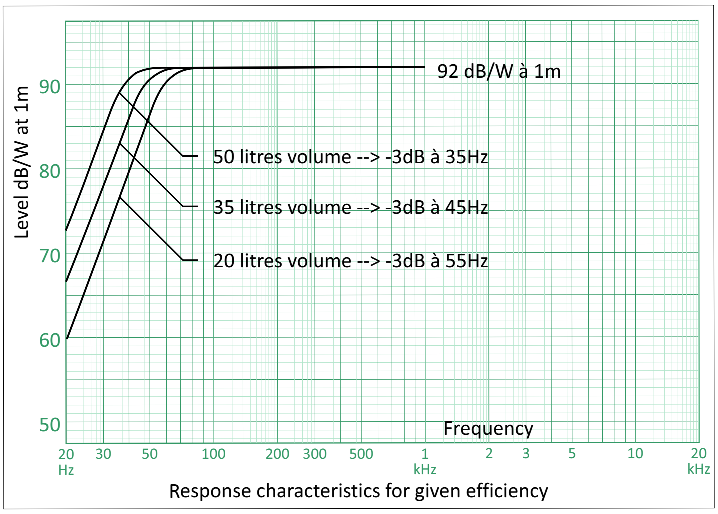
3 - **Bandwidth** for extremely low frequencies

These three criteria for bass frequencies may not be allowed to contradict the fundamental rules of physics, i.e.:

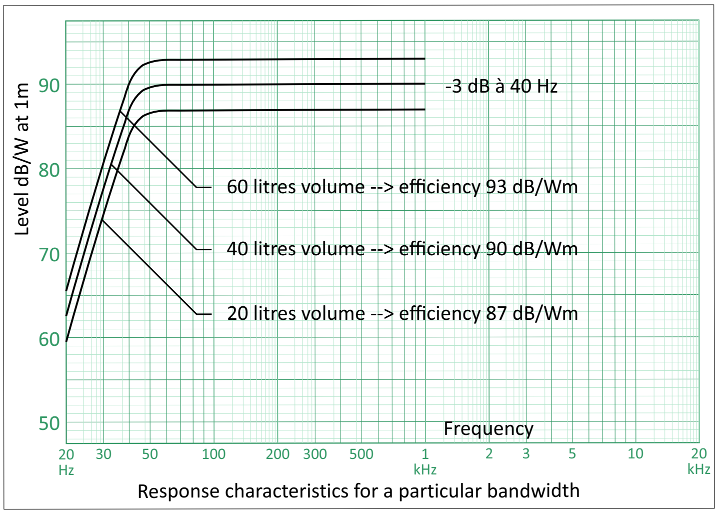
For a particular volume:
The efficiency and bandwidth are defined



For a particular efficiency:
The volume and bandwidth are defined



For a particular bandwidth:
The efficiency and volume are defined



4 - **The maximum allowable level** in decibels (dB)

5 - **Other musical performance factors** (distortions, decay-time broadening, etc.)

6 - **The reliability** (in load limiting and in aging)

The origin of a loudspeaker's directivity

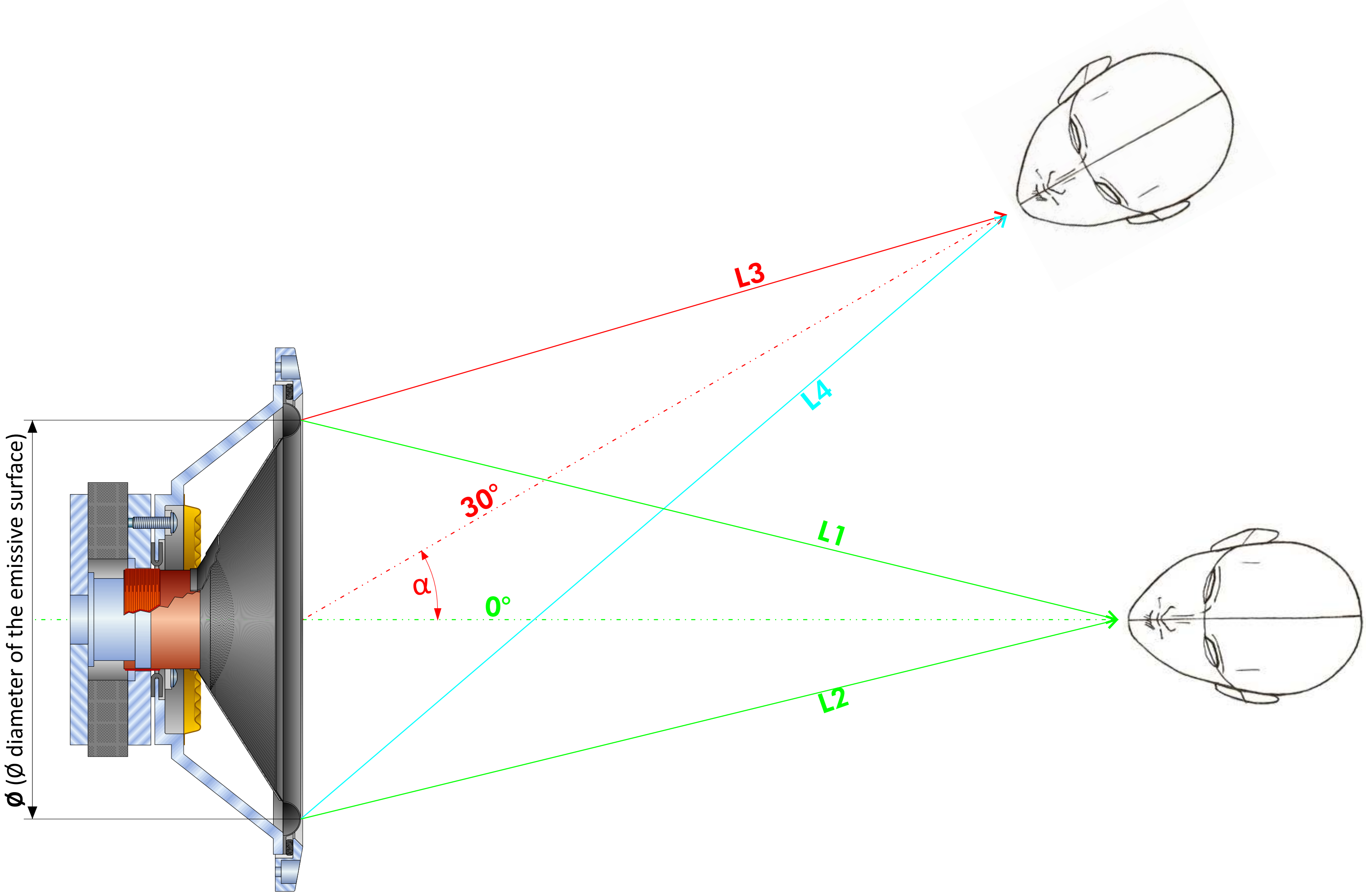
The directivity of a loudspeaker is linked to 3 parameters:

1- the size of its membrane (more precisely its emissive part), i.e. its diameter Ø (in m)

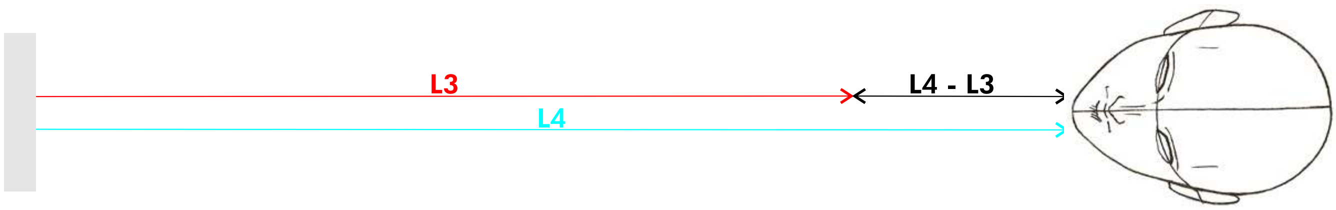
2- the frequency F (in Hz)

3- the position angle of the listener in relation to the axis of the loudspeaker: α (in degrees)

The propagation speed of sound in air (Vair=344m/s) must be taken into account in this phenomenon.



Directivity is due to the difference in sound propagation lengths from the different emitting parts of the membrane to the listener.



Directivity in detail

In the case of a listener situated in the axis of the loudspeaker, the distances L1 and L2 are equivalent. The sounds emitted by all parts of the membrane will reach the listener at the same time.

In the case of a listener situated, as in the drawing presented here, with an angle deviation $\alpha = 30°$, the distances L3 and L4 are not identical. The difference in length (L4 - L3) results in a delay in the sound signal that has travelled the longest distance (L4) compared to the one coming from L3.

Here, for a diaphragm of ø 20cm and an angle of 30°:

$$L4 - L3 = \frac{\varnothing \cdot \cos 30°}{2} + \frac{\varnothing \cdot \sin 60°}{2} = 0{,}1 \text{ m}$$

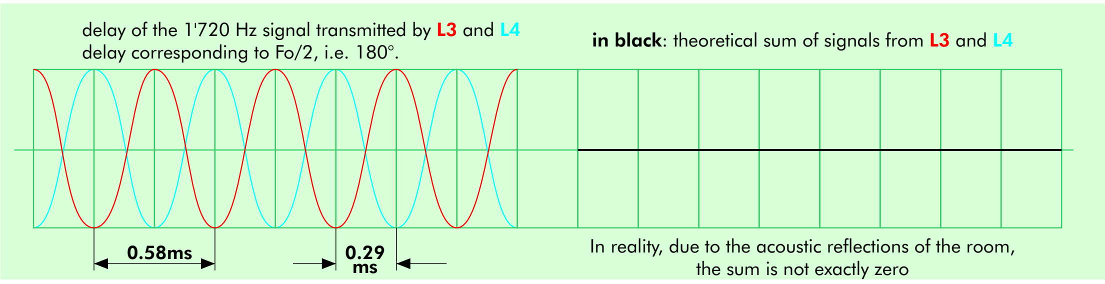
The delay T resulting from a difference in length of 0.1m is equal to :

$$T_{(s)} = \frac{L4 - L3_{(m)}}{Vair_{(m/s)}} \quad \text{soit :} \quad T = \frac{0{,}1}{344} = 0{,}29 \text{ ms}$$

The frequency corresponding to this wavelength of 0.1m is equal to :

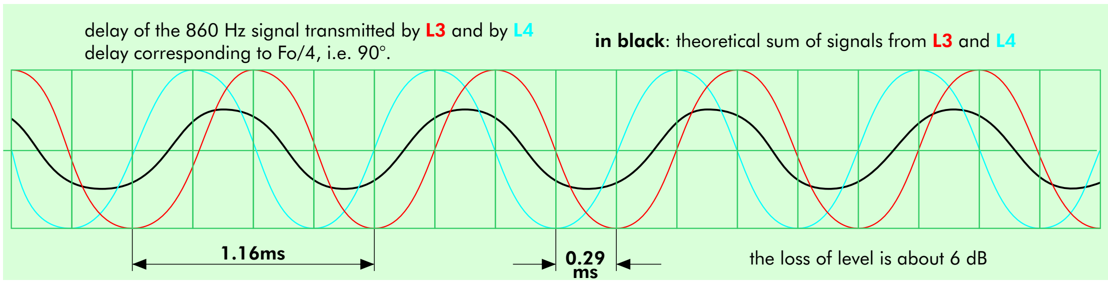
$$Fo_{(Hz)} = \frac{Vair_{(m/s)}}{L4 - L3_{(m)}} \quad \text{soit :} \quad Fo = \frac{344}{0{,}1} = 3'440 \text{ Hz}$$

To reproduce a frequency of 1'720 Hz (i.e. Fo/2), the signals from L3 will be cancelled by those from L4 by 180° phase opposition.



à 1'720 Hz

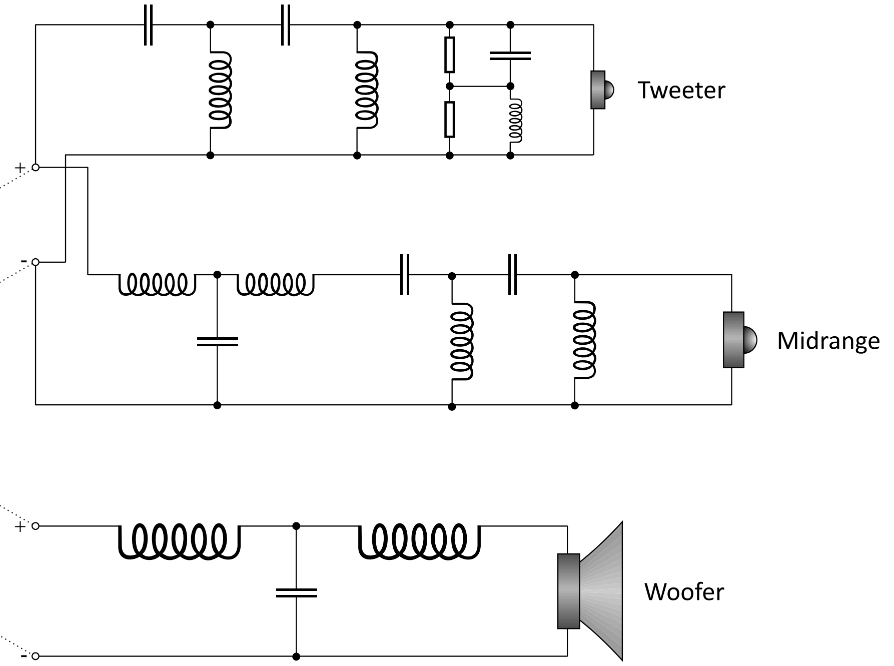
In reality, at a frequency corresponding to Fo/4 (i.e. 860Hz in our example) attenuation is significant. In our case, with a 20cm diaphragm and a 30° angle, from 860Hz a directivity is already not negligible.



à 860 Hz

The directivity diagram of a loudspeaker is not related to its musical qualities, but to the size of its membrane: the larger it is, the more directional it is in the frequency range concerned.

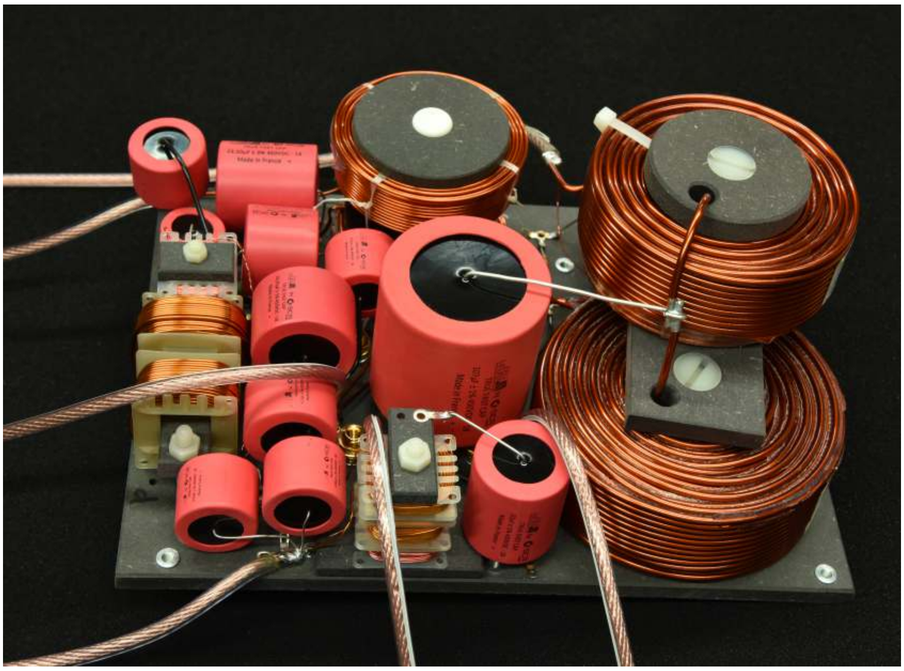
Conventional LC distribution filters



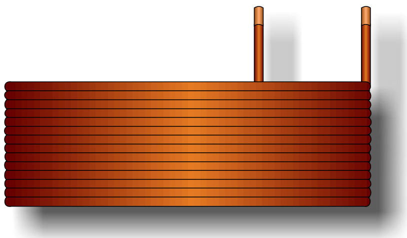
The bandpass filters transmit the signal coming from the amplifier to each of the speakers, depending on its frequency content.

The high-performance passive filters with their steep attenuation gradients offer decisive advantages compared to any other system. Consisting of self-induction coils with a large copper cross-section and along with no magnetic core, as well as of polypropylene capacitors, they are non-saturable and exhibit high linearity as well as an exceptionally long service life.

Components of passive filters and their construction



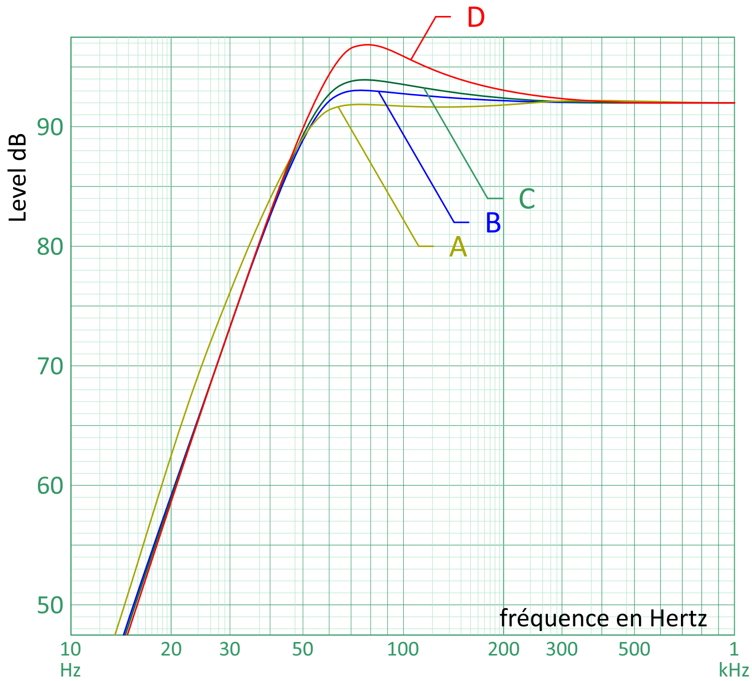
Bandpass filters with steep attenuation gradients and low serial loss, without chemical capacitors and without magnetic cores in the centre of the windings: 6.5 kg of components



Self-induction winding with no core and very large cross-section: 1.68 mH - 0.11 Ω - 3 kg

Highly stable polypropylene capacitor with very fast impulse response : 12.7µF - 3% - tgδ<0,06%

The resistive losses of the bandpass filters and their consequences



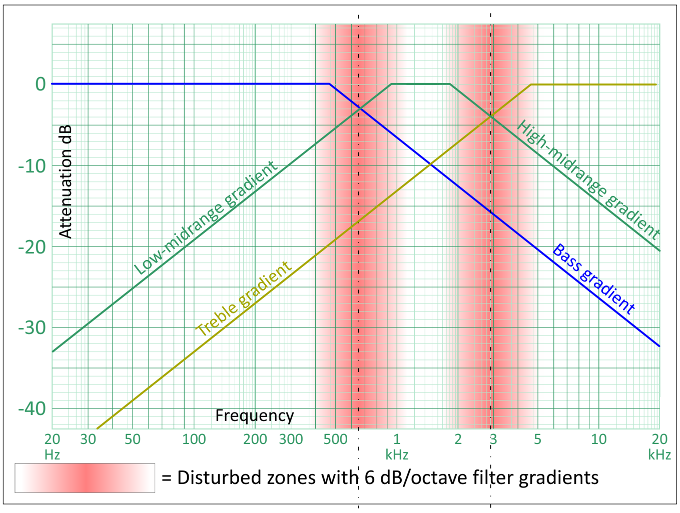
In continuous, non-transitory operation mode, the series resistances between the amplifier and loudspeakers result in significant changes in bass response:

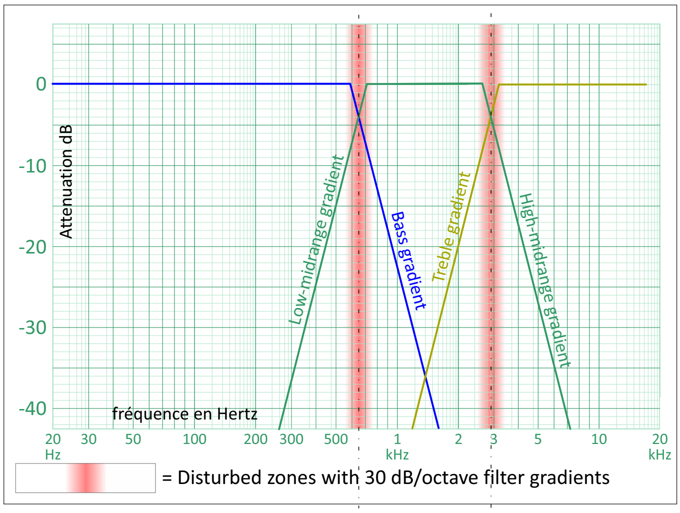
A: 0 Ω (theoretical case)
B: 0.5 Ω (ideal case)
C: 1 Ω (correct)
D: 3 Ω (bad, but unfortunately too often observed)

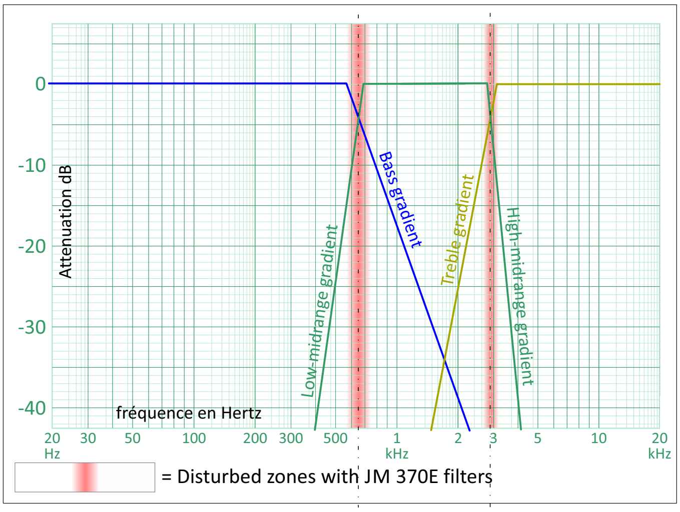
In a transitional regime (music), these series losses will generate a decay-time broadening / leading edge broadening between 50 and 300 Hz, which degrades (even more) the clarity of the musical message.

Influence of resistive losses between amplifier and woofer on bass clarity

The attenuation gradients of filters







The gradients of the filters determine the common operating zones of the loudspeakers. These zones are obviously disturbed by <u>the trampling of one transducer on the other</u>, transducers that are geometrically offset and physically dissimilar. The faster the crossover slopes, the smaller these overlap zones. The reproduction ranges resulting from a single membrane will obviously be cleaner.

The impedance of the loudspeaker and acoustic enclosure: a very complex load and a big challenge for the amplifier

Like a heating element, a resistance is a simple load. The current flowing through it depends on the voltage that is applied to it; this current will be completely consumed and transformed into heat. This is valid independent of the form of voltage applied.

An electric motor for alternating current, or a loudspeaker, transforms electrical energy into motion with a certain efficiency, dependent on the losses of the system. That part of the energy lost will be transformed into heat. The part of the energy converted into movement presents a non-resistive but reactive load to the voltage source. This implies, for reasons quite difficult to be simply explained, that this motor, or loudspeaker, is going to request more current than the amount it will actually consume, and will return the excess energy to the voltage source. This returned energy is called the electromotive counterforce (emcf).

Depending on the loudspeaker, current demand can exceed up to 7 times the energy required to move its membrane. The excess energy has to travel back through the crossover filter and the loudspeaker's cables to the amplifier. This proportion of excess current returned depends on the content of electrical musical signal (variations in frequency and amplitude at each given moment). We can speak of a "dynamic cosine phi" (cosφ).

When dimensioning the amplifier, the loudspeaker load presents two challenges:

- To provide sufficient current provision capacity: In ampere (A) and not in watt (W). This ability will allow the amplifier to avoid clipping the dynamic peaks of every large current demand coming from the loudspeaker enclosure. Clipping is not only very unpleasant for the listener but dangerous for the health of the speakers themselves.

- To have a good absorption capacity for the energy returning from the speakers (emcf). If this is not the case, a rebound of current in the direction of the speakers will disrupt the spatial acoustics by having a very detrimental echo effect. These dynamic "dips" will be drowned by this return of energy and musical detail will be filled in with very unpleasant resonances, and even be tiring. The dynamics provided by the amplifier/speaker system will become poor.

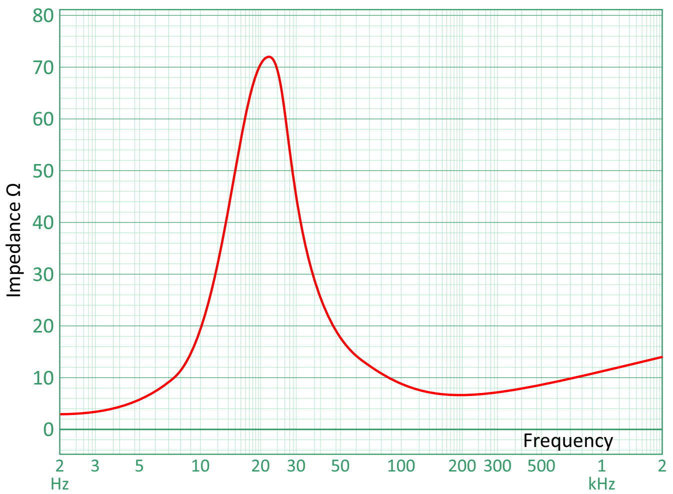
Certain transistor amplifiers are able to provide a lot of current. Their ability, for example, to accept a resistive load of 2 ohms or even less is a pleasing feature in this regard. Those transistor amplifiers which cannot operate with loads below 4 ohms should be eliminated as their designers have not understood what a loudspeaker load actually represents.

By contrast, taking up the electromotive counterforce (emcf) causes many problems for the outputs of direct transistor amplifiers, i.e. those without output transformers. The re-absorption of energy is poorly controlled, with all the consequences already mentioned above.

The technology of tube-driven amplifiers requires an output transformer to adapt the working impedance of the tubes to that of the speakers: The transformer is a component which is large, heavy, expensive and as difficult to develop as to manufacture if one wants to have excellent performance. However, if correctly dimensioned, it is a real gift for the loudspeaker, which will be able to receive not only all the current needed but also return − with a very good efficiency − this emcf, which will be short-circuited by the secondary winding of the transformer. At the same time, the low series resistance of the loudspeaker's filters and cables facilitates the return of energy to the amplifier.

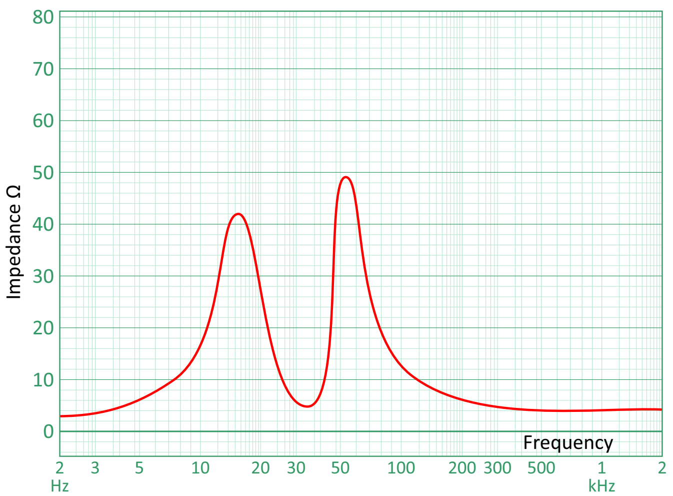
A few transistor amplifiers are fitted with output transformers and their quality can approach that of good tube amplifiers.

The impedance of speakers and acoustic enclosures

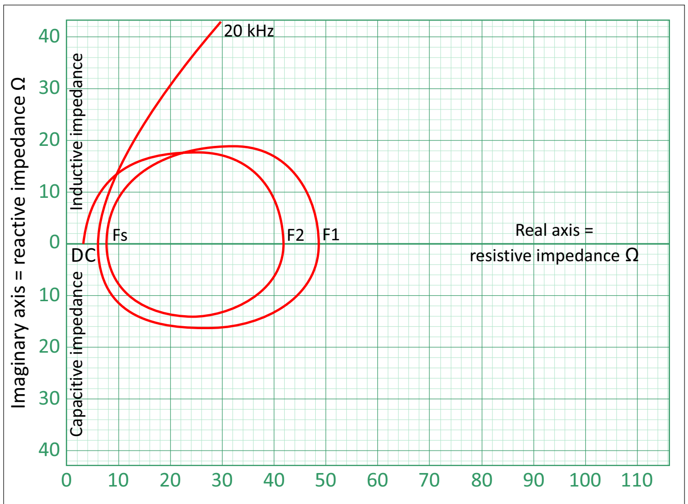


Apparent impedance of a woofer in free air

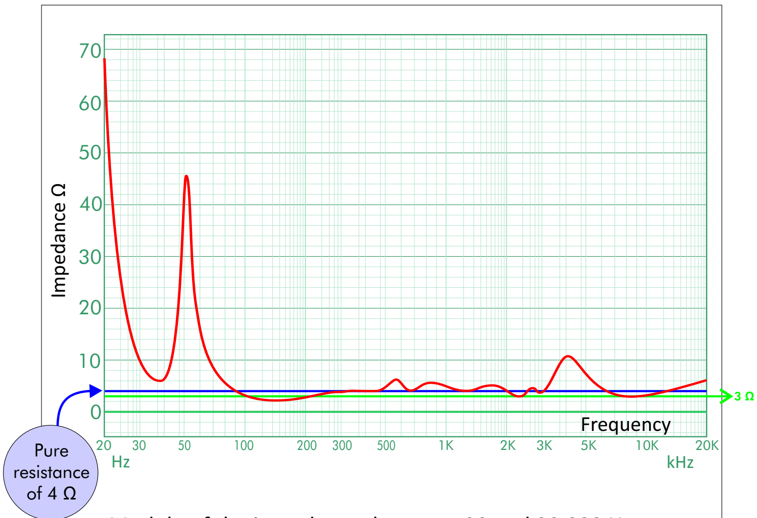
Complex impedance of a woofer in free air



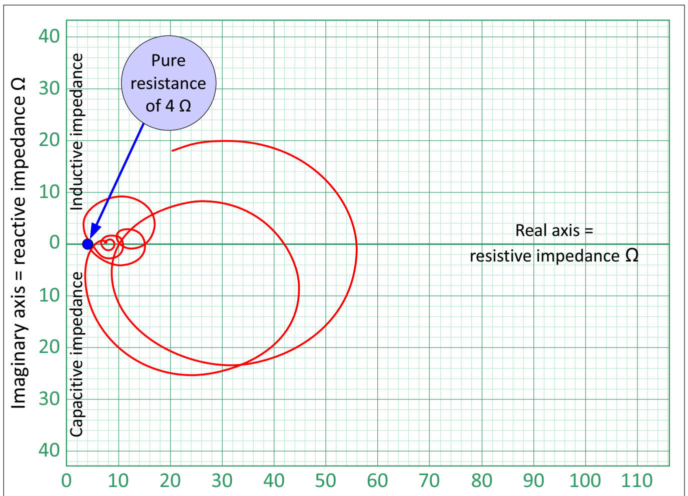
Apparent impedance of a woofer in a bass-reflex enclosure



Complex impedance of a woofer in a bass-reflex enclosure



Module of the impedance, between 20 and 20,000 Hz, of a speaker system of advanced technology: **JM 370F**
A theoretical pure resistance of 4 ohms
is represented by the blue horizontal line.



"Nyquist" diagram of the JM 370E enclosure in the complex plane. The theoretical pure resistance of 4 ohms is represented by the blue spot at the centre of the diagram: The circles that originate from the blue spot highlight the complexity of the actual load of an acoustic enclosure!

The reactive energies of the loudspeaker, the killers of dynamics

The musical dynamic is defined by the amplitude difference between a peak and a dip of a modulation curve. To be exact, this represents the dynamics, expressed in decibel (dB).

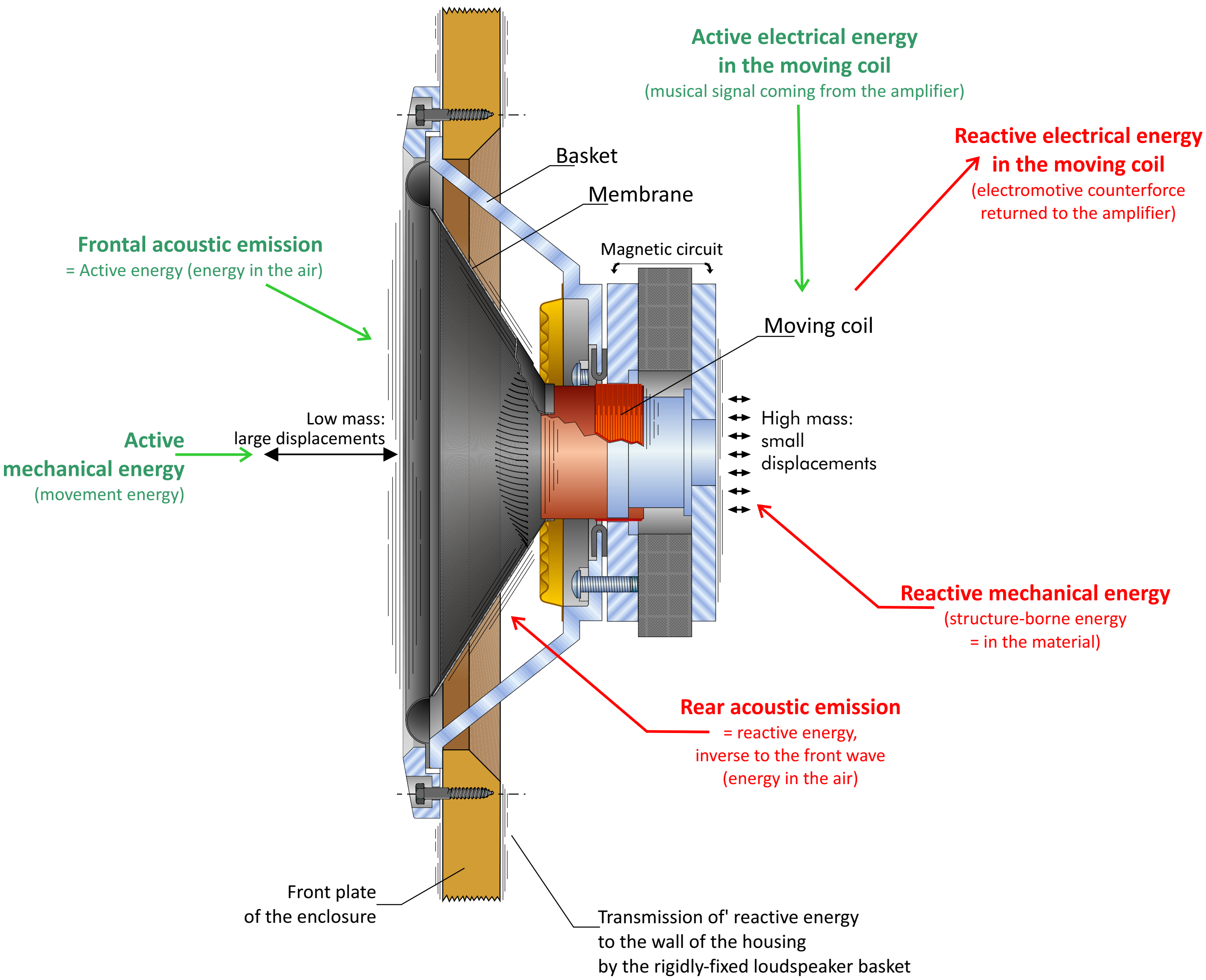
It is fairly easy to reproduce a dynamic peak; it is far more difficult, however, to reproduce the dynamic drop that follows, because a quantity of vibratory residues (of mechanical and electrical nature) will tend to drown a musical signal of low amplitude. For example, a recording of a symphonic orchestra easily has 60 dB of dynamics, in other words a power ratio of 1 to 1,000,000. This difference in power has to be reproduced in a fraction of a second.

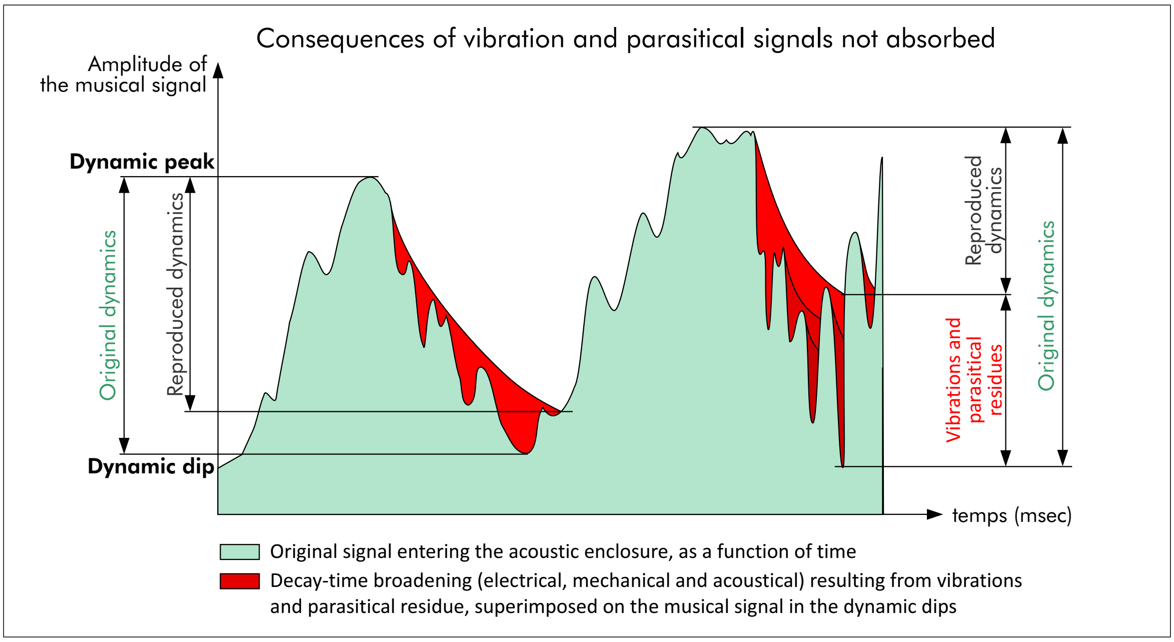
These parasitical residues (vibrations in the speaker box, electrical echoes, badly damped membranes, etc.) obviously result when a fortissimo is followed by a moment of very low intensity, sometimes even close to silence (or, simply, a dynamic drop) and this drop should actually be reproduced at this precise instant. One designates these signals which are indelicately prolonged as "the effects of decay-time broadening". The crux of the problem lies in what is called "the respect of silence".

In a loudspeaker, there are 3 major causes of decay-time broadening:

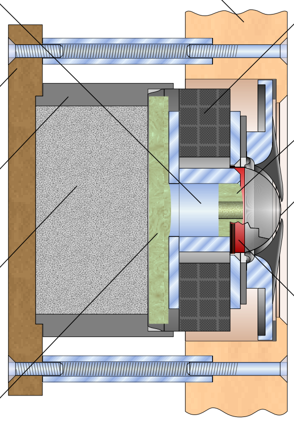
1. The electromotive counterforce (emcf) of each of the loudspeakers is a parasite current inverse to the electrical music signal from the amplifier. It has to be reabsorbed by the amplifier in the best possible way. If this absorption goes wrong, an echo effect will result such as can be obtained in the mountains when facing a rock wall: The original sound signal is rendered unintelligible by this reverberation. Now, the amplifier's transistors can poorly absorb this return energy, which then is reflected back towards the speakers and spoils the dynamics. If a tube amplifier is equipped with very good output transformers, the secondary windings of the transformers short circuit these parasitical currents, and do this highly efficiently.
2. The reactive mechanical energy caused by the movement of the loudspeaker components (winding and membrane) and by the mass of the air around them gives rise to a form of parasitic residue. These are mechanical vibrations generated by the loudspeaker itself. In order to get around this, these vibrations are transferred to the loudspeaker's enclosure by the frame of the loudspeaker's basket and excite the various surfaces both with a delay and with considerable deformation. In Jean Maurer's loudspeaker enclosures, these various residues are absorbed by the quartz sand, thanks in particular to the pre-stressed threaded pin which dynamically connects the drive of the woofer to the sand-filled double back of the enclosure, which has a very low resonance frequency. In addition, the loudspeaker's basket is dynamically decoupled from the front of the loudspeaker's housing. For the midrange and treble loudspeakers, the sand inserted in their double enclosures ensures this effect.
3. The movements of the loudspeaker's membrane produce a forward emission of sound and an inverse emission of sound at its rear. This acoustic back-wave, the only form of unwanted signal which everyone is talking about, is absorbed by the mineral wool. Part of this signal is taken up by the bass-reflex vent in order to linearize the response at low frequencies. With respect to the previous two forms of reactive energy, this is the one feature that presents the least amount of control problems as far as silence is concerned. If the speaker's enclosure is very inert, i.e. has little resonance, the spatial acoustic returned will be good. The quartz sand trapped in the double back of the case and, in the case of the JM 370F model in the sides and 2 front sections, contributes greatly to this result.

Active and reactive energies of the loudspeaker





The midrange and treble transducers: two enclosures built into the woofer enclosure

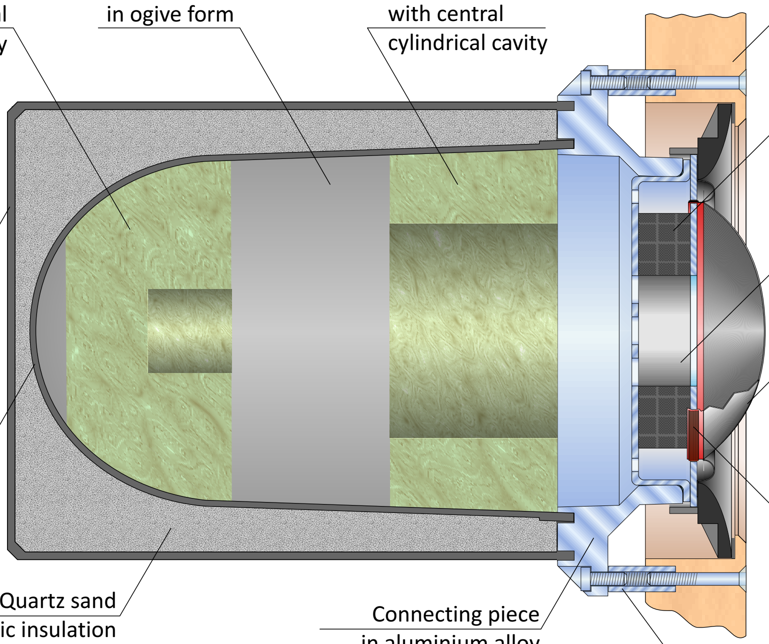


Treble enclosure, with TD 25-4 transducer

Each loudspeaker, or transducer, produces an acoustic front wave and a rear wave.

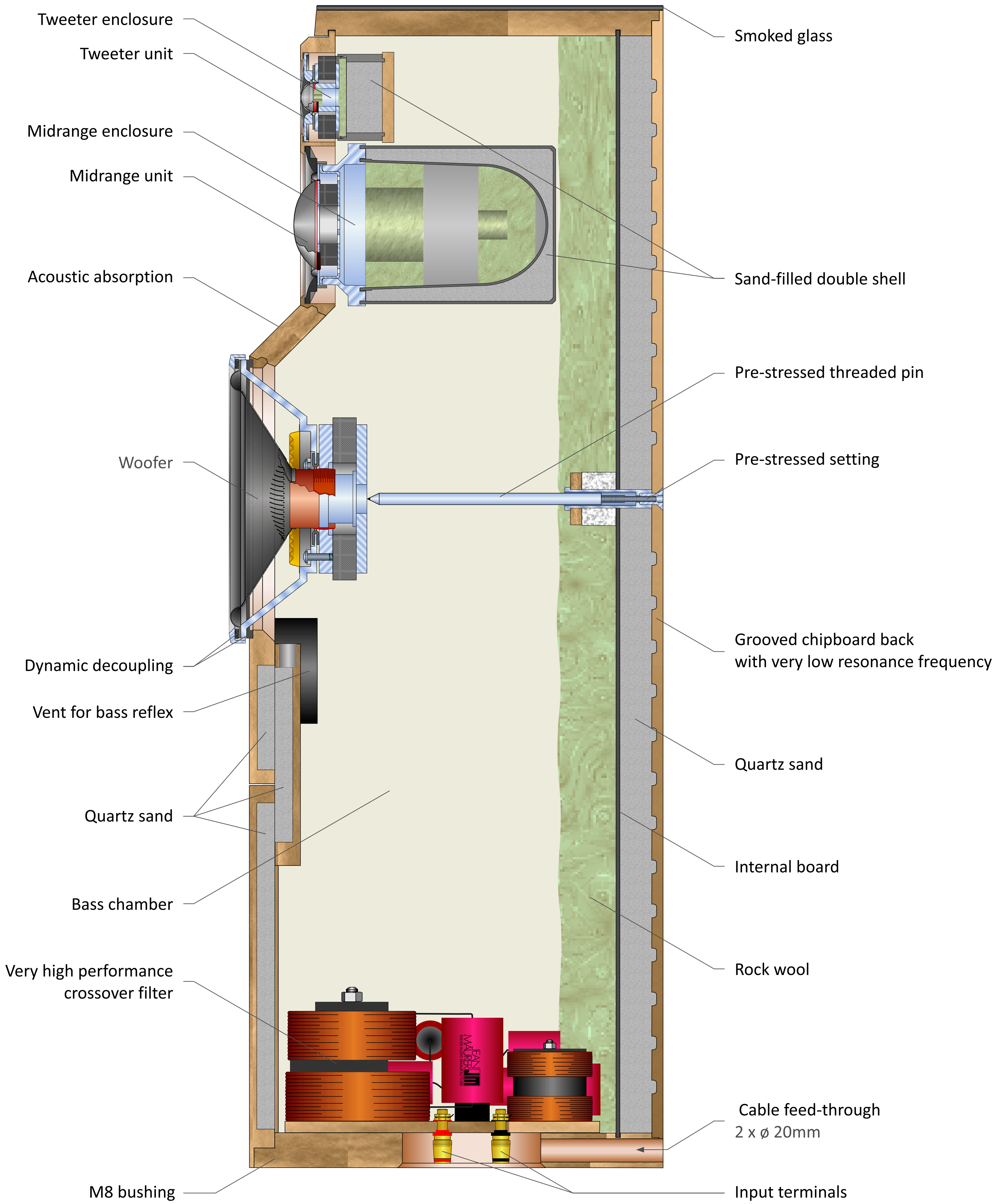
The rear energy of a loudspeaker should not under any circumstances be mixed with that of another one, as the risk of seriously disrupting the behaviour of the neighbour's membrane and thus causing sound pollution is unacceptable. The risk of mechanical destruction of the moving parts of the midrange or treble transducers by bass-energy is also considerable.

The reactive energies of these two units (midrange and treble) must also be treated seriously. In these two cases, the mass of the quartz sand will stabilize them, as does the sand at the back of the speaker for the bass.



Midrange enclosure, with MD75-3 transducer

Construction of the acoustic enclosure with energy absorption



Cutaway view of the JM 370F acoustic enclosure

Swiss manufacturer
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