Passive Crossover Networks

One of the most common errors in choosing a crossover for a speaker system is to think of it in its basic outward appearance: nothing more than a heap of components, often in a plastic case, to put between the amplifier and the speakers and that's all.

Actually a crossover is not an independent device but rather a circuit of which the speaker constitutes a pure component. To arbitrarily replace the speaker with a different one expecting that this doesn't alter the functioning of the crossover/speaker circuit, would be like randomly replacing a component inside a radio and hoping that it still works. If this by chance happens it shouldn't be attributed to your technical cleverness but rather to a piece of good luck.

For this and other reasons, before we examine crossover networks it is important to have a close look at the speaker

THE SPEAKER

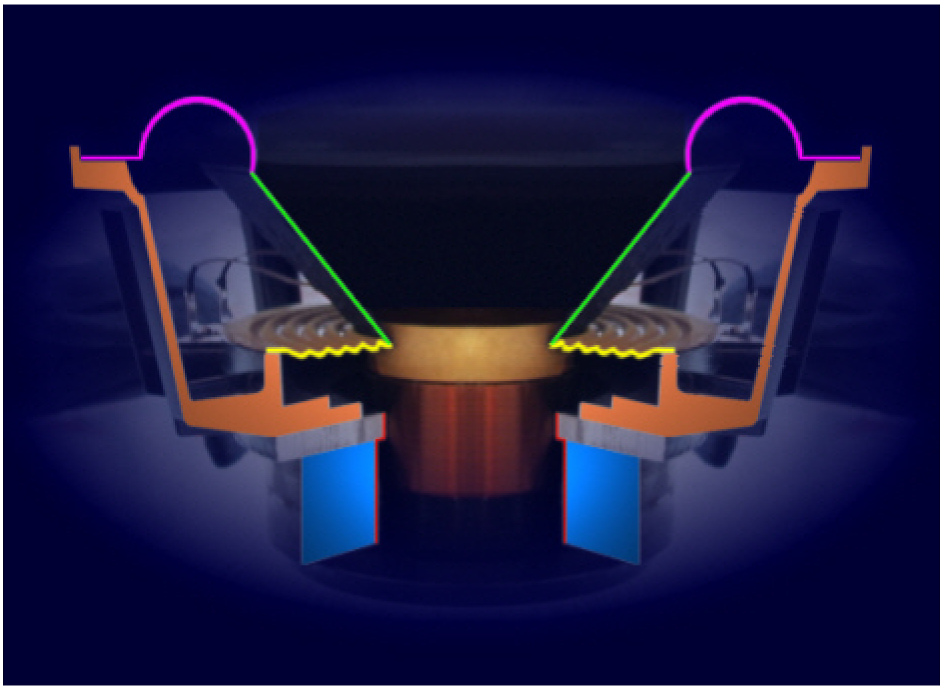
The speaker is an electromechanical device that has the task to convert the electric signal coming from the amplifier in pressure waves that our auditory system perceives as sounds. For this function it is also called a **transducer**.

The most common speakers are classified as **orthodynamic** and **isodynamic,** according to the principle of operation. In orthodinamic speakers the force that sets the radiant diaphragm in motion is applied in one point only — perpendicular to the plane of the diaphragm itself. Only an accurate geometry allows every part of the diaphragm to move in phase. In isodynamic systems the force is equally distributed on the whole radiant surface. A few car-audio manufacturers only have produced isodynamic speakers, among them Bohlender-Graebener, Eminent Technology, ESS, Fostex, Infinity and Sony.

The orthodinamic speakers are evidently the most diffused and to them we'll refer from now on. They are substantially divided in two families, **cone** and **dome,** according to the shape of the radiant diaphragm. Generally speaking, it can be said that cone membranes are more suited for low and mid-low frequency reproduction (woofers and low-mids). Dome profiles are unbeatable in reproducing mid-high and high frequencies (mids and tweeters). There are obviously some variations on the theme, represented by cone-dome profiles of a lot of tweeters and some midranges (Morel) and by inverted-dome profiles of some tweeters of the recent past (Epicure).

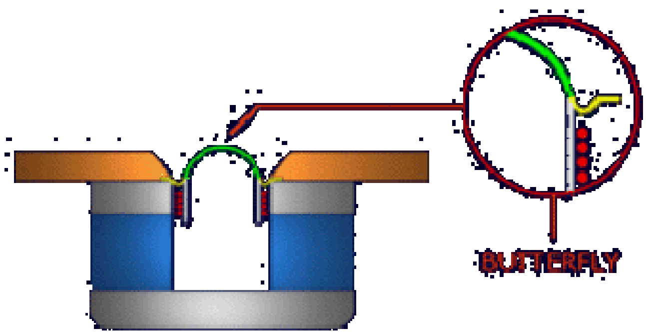
In reference to the materials used in making diaphragms, they range from cellulose and polypropylene to carbon and kevlar for the cones. The domes are usually silk and polyamide along with aluminum and titanium and many other materials, often exotic.

Below you can observe the structure of a generic woofer: the frustum-of-cone diaphragm (green) results suspended inside the basket (orange) by two suspensions, one external called **surround** (pink) — generally foam, rubber or corrugated canvas — and one internal called **spider** (yellow) — almost always corrugated canvas. The only movement allowed by this configuration is that axial. Rigidly connected to the vertex of the cone there is the **voice-coil,** immersed in the middle of the pole piece of the magnet (light blue), in a narrow crack called **air-gap** (red).



As soon as current from the amplifier flows in the voice-coil it produces a magnetic field that interacts with that of the magnet. This forces the voice-coil to move, together with the cone with which it is integral. The shift and direction of this moving element are determined by the intensity and direction of the flow in the coil.

Conceptually the structure of a tweeter differs from that of a woofer in suspension, which is unique instead of dual. It is at the base of the dome, which is also the junction point with the voice-coil, and often — but not always — it is made of the same material of the dome to prevent glueing problems among different materials.



The whole dome/suspension/coil of a tweeter is called **butterfly** and this should tell you how much it is light, really more than even the smallest cone. The extremely close project and workmanship tolerances in tweeter manufacturing justify the fact that only a few builders in the world have developed the technological know-how to manufacture them.

So why are speakers morphologically so different according to the frequency they are asked to reproduce? The answer to this question brings us to analyze better multi-way systems

MULTI-WAY SYSTEMS

An ideal hi-fi speaker should have the following requirements:

- its frequency response should show a horizontal trend over the whole spectrum of the audible range, with a tolerance of ±1dB;
- its dispersion should be constant with the frequency;
- the subjective level given back without distortions should be similar to that of the original event to play.

A single transducer that contemporarily meets all these requirements has not been invented yet. A speaker able to correctly reproduce 20Hz cannot correctly reproduce 20Khz but this is true even if we tighten the range from 100Hz to 10Khz, at least until we are speaking of hi-fi. In fact to produce high frequencies it is necessary to move a small amount of air but with high linear acceleration values, and this requires a small and extremely light transducer. On the contrary, for low frequencies it is necessary to move greater and greater volumes of air the lower the frequency. This involves the use of bigger membranes — therefore heavier — and capable of high excursions. It is plain to see that we are speaking of antithetical demands.

The only way to have your cake and eat it is to split the audio spectrum in at least two **frequency bands,** one including the lowest frequencies only, the other the highest ones. This way the low frequencies can be played by a speaker designed for this purpose and the high frequencies by another specific transducer. These frequency bands are called **ways** and such a system is called two-ways.

If the spectrum is splitted in three bands then we'll get a three-ways and we'll need three dedicated speakers: a woofer for the lows, a tweeter for the highs and a midrange for the mids, which can be dome or cone according to the project choices. And so on.

The number of speakers always equals the number of ways. However, this is not inversely true. In fact it is possible that one or more ways end in two or more speakers, each reproducing the same frequency interval.

To operate this delicate splitting in frequency bands just described are called really them, the filters

THE FILTERS

Filters are devices that accepting in entry a whole signal return in exit a part of it rejecting the other. This definition won't result so tricky if you think that any filter accomplish this task, even that of your everyday coffeepot.

They can be **active** or **passive.** In active ones the signal splitting occurs before it is amplified. This way we'll have a number of dedicated amplifiers equal to the number of ways, each connected to its own speaker. An active filter is so-called because it requires electric power for its operation and it can provide, besides attenuation of the single ways, to raise its gain if necessary. Contrarily, a passive filter is placed before the speaker becoming an integral part of it and drawing with it energy from the amplifier for its operation. It cannot raise the gain on the single ways rather introducing attenuations.

There exists a third type of filter, the **line-level** passive filter. Like the active one it performs the splitting before the amplifiers, but it is much more economical. The (high) price to be paid is represented by the inevitable attenuation introduced on the signal. This is the consequence of the use of passive components. In car audio those kind of devices are — or were — produced, among others, by Audio Connection, Cerwin-Vega and Harrison Lab.

From now on we'll refer to passive filters only, even though some principles are common to both types. These circuits use as components resistors, inductors and capacitors.

RESISTORS, INDUCTORS AND CAPACITORS

Resistors are components that offering a certain resistance to the current flow attenuate the signal that passes through them but don't modify its response. For this reason, as we'll see later, they are used for aligning transducers with different values of **sensitivity** — the parameter indicating how much louder a speaker plays than another under the same applied electrical tension. Resistors values are expressed in **Ohms** (Ω), the electric resistance unit. In an electric scheme they are identified by the following symbol and letter:

Inductors look like spools of enamelled copper wire — from here the common name of **coils** — that can be wound on plate or on ferrite core. Often they may not contain any core at all, in such case it is said that they are wound in air. Unlike resistors, they are reactive components that oppose a very high impedance against high frequency signals, assimilating to a short-circuit while going down toward the lowest frequencies. The transition point is determined by their value, that is expressed in **Henrys** (H), the inductance unit. For the same inductance, core-inductors require less coils than those wound in air and they are therefore smaller. However they tend to saturate in presence of strong signals, thereby introducing distortions. In an electric scheme inductors are identified by the following symbol and letter:

Reactive, but inverse in comparison to inductors, is also the behavior of capacitors. For this reason both of them are defined as **dual elements.** Capacitors are measured in **Farads** (F), the capacitance unit. In an electric scheme they are identified by the following symbol and letter:

It would be wasteful to fully analyze the operation principle of capacitors: it is enough to say here that they are devices able to store electrostatic charges. They are constructed with two electrodes connected to plain conductive plates separate between them by an insulator named d**ielectric**. They are grouped in families according to the type of dielectric used: so we'll have teflon capacitors, polypropylene, polystyrene, polycarbonate, polyester, mica or ceramic. All these materials are characterized by a permeability more or less hard depending on their relative **dielectric constant,** a number that practically quantifies their insulation ratio. To increase capacitance it will be indispensable to reduce the thickness of the dielectric that separates the plates, but only some materials allow that without risks of perforation, due to their dielectric constant, that's it.

A separate family is represented by electrolytic capacitors, whose dielectric is constituted by an electrolytic gelatinoid solution that, if subjected to polarization and as long as this polarization is maintained, produces a layer of insulating oxide so thin that it allows very elevated capacity values. Electrolytic capacitors are therefore polarized components — that' to say with a positive pole and a negative one — and they can be used only in circuits where the direct component is greater than that alternate since it is just the voltage that allows the formation of the dielectric layer. They are practically irreplaceable in power supply filtering circuits, but they shouldn't be used in crossovers where direct components are pratically absent.

There are for sale however — or they easily can be made — the so-called **bipolar** capacitors, or **non-polar,** consisting of two standard electrolytic capacitors with two homonym electrodes connected together: starting from two capacitors with equal capacitance will result a single capacitor of halved capacitance and doubled size, but non polarized and therefore suitable for crossovers. The point is that they are not the ideal items for this type of use and those in polypropylene and polyester must be preferred.

Pondering on the behavior of inductors and capacitors and remembering the definition we've given about crossover filters in the previous lesson, you can see that they are elementary filters by themselves, although from their mutual combination more complexes filters are gotten, as we'll see later on. But first it's important to know those damn filters by name. In fact, according to their function they are distinguished in lowpass, highpass and bandpass

LOWPASS, HIGHPASS AND BANDPASS

A filter is called **lowpass** when it lets pass unchanged the inferior portion of a signal, rejecting the superior one — the terms inferior and superior are intended in the domain of frequency. Imagine appropriately connecting the inductor of the previous lesson to a speaker. Referring to its behavior it's easy to understand how offering a high impedance to the highest frequencies, it actually obstructs them the way; instead offering no impedance to the lowest frequencies, it allows their transit toward the speaker. This way the speaker will utter only grave sounds and you won't have done anything else than make a simple lowpass filtration.

A filter is called **highpass** when it lets pass unchanged the superior portion of a signal, rejecting the inferior one. Replace the inductor with the capacitor — always that of the previous lesson — and as if by magic the speaker will stop speaking threateningly and will turn into a lambkin. What has happened? Simple, the capacitor has opened the door to the highest frequencies only, obstructing the path to the lower ones owing to the increase of its inner impedance. There is made a simple highpass filter.

It's worth making a brief aside here, in order to clarify the difference between **resistance** and **impedance.** Generalizing a great deal — a very great deal — we can say that both are measured in ohms and they express a similar concept, but we'll refer to resistance when talking about direct current while we'll speak of impedance in presence of alternate currents, such as musical signals are.

Forget now about connecting inductor and capacitor in turn on the same speaker. Instead imagine connecting them simultaneously on two different speakers: we would get then both highs and lows — that's the whole signal — but played by two different transducers. Moreover if we imagine these two transducers each one dedicated in reproduction of the frequency band delivered to it, then we'll have made a simple two-way system, composed — as the whole two-ways — by a lowpass and a highpass.

A filter is called **bandpass** when it lets pass unchanged the middle portion of a signal, rejecting the contiguous ones. It consists of the opportune combination of a highpass filter with a lowpass filter. Preceding this filter with an additional lowpass and following it with an additional highpass, we'll get nothing but a three-way crossover, composed — as the whole three-ways — by a lowpass, a bandpass and a highpass. Putting in the middle another bandpass we'll get a four-way crossover and so on.

[Despite the common term crossover being used as a synonym of filter, we prefer to observe a certain strictness and will refer to it as a coupling of filters — complementary among them, as in the cases just described — that produces intersections among adjacent ways. The various types of filter constituting a crossover will then be named **rows** and a crossover will be formed by at least a lowpass row and a highpass row, to include one or more bandpass rows in multiway configurations.]

A fourth kind of filter also exists. This is the **notch-filter,** with inverse function in comparison to bandpass. It can be tuned on a particular frequency and reject it completely. It's useless for our purposes and we won't dwell over. However it's nice to know it exists.

The story's not ended here. A filter has many more distinctive parameters. It's marked by an order — to which is associated an attenuation slope — a cut frequency and a merit factor. Nevertheless it's better not to have too many irons in the fire, so we'll proceed by degrees with the cut frequency

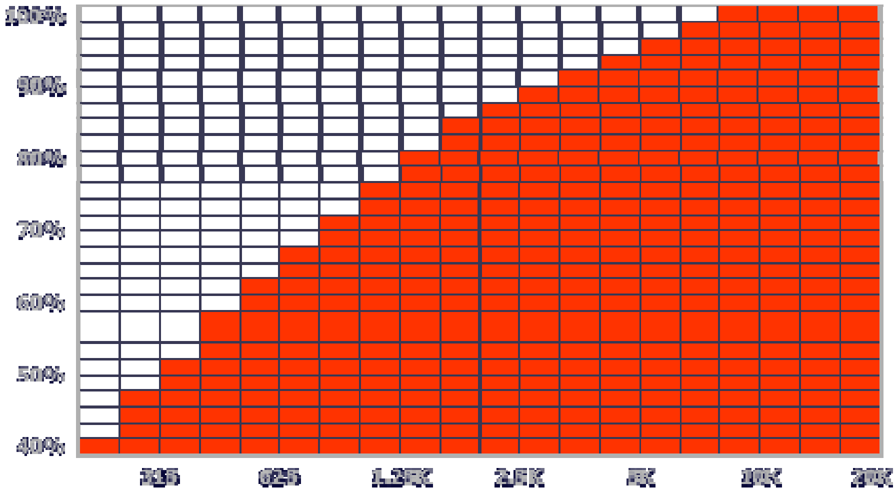
THE CUT FREQUENCY

The **cut frequency** is so-called because it is the frequency from which a filter starts to operate the rejection of the unwanted frequency bands. Although the term cut gives the idea of something sharp, be aware that this barring isn't sudden at all, but more or less gradual depending on the filter type.

Imagine being at the cinema and seeing the sheriff get off his horse while a lap-dissolve — it's called this way — opens up on the coach running in the desert. That's exactly the way the transition happens from a band to the adjacent one by means of the crossover. The lowpass gradually gets rid of the sheriff, about whom it doesn't care anything about, while gradually the highpass brings us on the coach, which all were asking where the hell it was! Of course this rule works for other types too, as thriller and spy-story. Even in porn this technique is used — although you'll agree with us it's better to continue using the example of the sheriff. Well, why did the director choose to fade while Wyatt Earp gets off horse, and not in any other scene? (Read: how to choose the cut frequency in a speaker system?)

As a rule of thumb it can be affirmed that this has to be central compared with the frequency range that both involved speakers can play without a filter. This implicitly presupposes that the response of both speakers overlaps for a good part. Having a midbass able to rise up to 3Khz and a tweeter that goes down to 1Khz, the cut frequency could be easily located in 2Khz, a central frequency in comparison to the range that both speakers are able to play. On the contrary, a woofer that reaches 2Khz cannot be coupled with a tweeter that starts from 3Khz, under pain of strong alterations in the general response of the system.

Another aspect to be held in serious consideration is the ability of the speaker to accept power. Every speaker, of any kind, brand and quality, the more it moves the more it distorts. For the same emitted acoustic level, any diaphragm suffers an **excursion** as much strong as lower the frequency, distorting accordingly. To control this distortion, therefore, its crossover frequency will have to be shifted upward according to the applied power. But the energetic content of a musical signal is not constant with the frequency, so the evaluation of the power applied to a single component at a certain crossover frequency is not so immediate. For this reason we have provided you with this chart that shows the energetic distribution of the musical signal depending on the frequency:



How to use it: locate a certain cut frequency on the abscissa and find its correspondent percentage value. That will be referred to lowpass. Subtract it from 100 to get the value referred to highpass. Very easy, isn't it?

But 'cause all seems not so easy, the cut frequency — or crossover frequency, as we have indifferently named it up to now — is also called **-3dB frequency,** and so you'll have to call it to cut a fine figure in circles that count. This eccentric name is due to the fact that it is identified as cut frequency that given back with an attenuation by 3dB compared to the average level of the **wave band.** The concept of wave band will be clearer after we'll have spoken of order and slope of filters

ORDER AND SLOPE OF FILTERS

Filters are grouped by **orders** according to the number of reactive elements that make it up. It is named first-order filter if it is made by a reactive element only: an inductor, as we have already seen, is a first-order filter. We'll have a second-order filter combining together an inductor and a capacitor, a third-order filter inserting a second inductor, and so on. Then we can attribute to a filter its order by counting the reactive elements that compose it. But beware, often instead of an only element it is preferred for various reasons to use a combination of two or more elements, connected in series or in parallel, according to rules that we'll best illustrate later on. In this case we'll refer to the combination as only a single component.

Every order is characterized by its own specific attenuation slope or **rolloff.** This represents in decibel per octave (dB/oct) the rate with which the filter operates the rejecting of the unwanted frequencies. Try to remember that the **decibel** is the unit of acoustic intensity, and that **octave** is generically called the space that runs between a certain frequency and its double or its half.

To better understand, try imagining any filter. As we have already seen, it will allow the transit of a certain portion of frequencies only. This portion takes the name of **wave band** and to it is assigned the conventional value of 0dB. At the cut frequency point the signal will be subjected to an attenuation of -3dB while beyond this frequency it will be attenuated gradually as much as the filter order is higher.

A first-order filter produces a rolloff of 6dB/oct. beyond the cut point. For example, a first-order lowpass filter with cut frequency tuned to 500hz, will let pass intact the lower frequencies, will return the 500s attenuated by 3dB, after which it will attenuate by 6dB the first octave (1Khz), by 12dB the second (2Khz), by 18dB the third (4Khz), and so on, with a constant — you'll say **asymptotic** to make a good impression at a cocktail-party — slope of 6dB for every next octave.

A second-order filter produces a rolloff of 12dB/oct beyond the cut point. Referring to the previous example, we'll always have the 500s attenuated by 3dB, but already the 1000s will be 12dB down while the 4000s even 36dB below. A third-order filter produces a rolloff of 18dB/oct beyond the cut point. A fourth-order filter produces a rolloff of 24dB/oct and so on, with an increase ratio in attenuation slope equal to 6dB/oct for every next order.

How do you decide a preference to any one order over another? It can be many reasons, from building easiness and musicality of lower orders, to the ability of the most daring ones to guarantee protection to overcharged transducers. But choice criteria, won't get tired to repeat it, never have to disregard the analysis of the characteristics of the speaker on which the filter must be closed. Among these one deserves to be analyzed in detail, the speaker dispersion

THE SPEAKER DISPERSION

Dispersion is the off-axis emission of a speaker. Speakers play because of the to-and-fro oscillation of their radiant diaphragm, which impresses the air with which it makes contact in sudden variations of pressure that are propagated at speed of 344 mt/sec. This propagation can be omnidirectional or prevailingly frontal, according to the relationship between the speaker diameter and the frequency it plays, or in short, its **wavelength.** Wavelength (λ) is the distance a sound covers in the air in a time equal to its cycle. By formula, the relationship between the speed of sound (c) and the cycles per second, that's the frequency (f):

$$\lambda = c/f = 344/f_{(hz)}$$

For any circular speaker, cone or dome it doesn't matter, the transition point from an omnidirectional emission to a directional one is represented by that frequency to which half the wavelength equals the speaker diameter. This is calculable with the following formula, where speaker diameter (D) is expressed in millimeters:

$$f_{(hz)} = (344/2/D)*10^3 = 172{,}000/D_{(mm)}$$

For instance, taking a 165mm speaker (6.5") we'll locate its transition point at around 1Khz. This means that up to this frequency its emission will be as far as possible omnidirectional, that is it will be characterized by a good dispersion. From this frequency on, instead, its off-axis emission will become more and more damaged. For a 50mm (2") dome midrange the transition point will be locatable in around 3.5Khz and for a 25mm tweeter (1") in around 7Khz. For all, beyond these frequencies the dispersion will start to lessen, at least in theory. In reality everything would exactly work in these terms only if the diaphragm was a rigid and crushproof piston, but it's not this way.

Increasing the frequency, also increases the motion speed and with it the acceleration imparted by the coil to diaphragm. What happens, especially with cone diaphragms, is that the more external sections don't succeed in following the sudden variations of movement imposed by the coil to vertex, so the actual surface involved in emission has the tendency to decrease. With it decreases the diameter realizing therefore that, yes, the dispersion of a cone speaker decreases as the frequency increases, but less than if it were a rigid piston.

Designing a cone with proper material and shape it's possible to get a progressive uncoupling with the peripheral sections at the frequency increase, to maintain dispersion the more constant possible. What's more, since the response of a cone would have the tendency to drop at highest frequencies because of its weight, the virtual reduction of emission diameter, involving a reduction of the moving mass, also produces beneficial effects on the in-axis response of the component. There is nevertheless a collateral effect to be checked, but it concerns the designer of the speaker, rather than you who only have to choose it: if the percentage reduction of the moving mass was superior to that of the emitting surface, the emission level would unpleasantly tend to increase, introducing more serious problems of response equalization.

But how can the dispersion of a speaker condition the results of a filter network? To explain it, suppose a typical two-way system consisting of our 6.5" woofer and our 1" tweeter, with crossover frequency tuned to 2.5Khz and all the other compatibility conditions complied. If you were exactly in front of the speakers the overall response of the system would presumably be satisfactory, but trying to listen in an off-axis position — such as when driving a car — the woofer dispersion above 1Khz will be more and more lacking, to almost zero near the crossover frequency. Result: an awful hole so big in the frequency response, visible to spectrum analysis and what's worse audible to ear. It's not a crossover nor spectrum analyzer fault. It's your empty pate 'cause you hadn't considered the speaker dispersion.

Apart from this, another aspect not to be disregarded is that inherent the load reactivity

THE LOAD REACTIVITY

Now we should speak to you of reactance, phase angles, argument and compound numbers. But if you already throw up in front of real numbers, what a frozen heart we could have to speak of imaginary ones? We shall seek to explain it with an oversimplified example hoping that such a simplification won't cause Albert — may he rest in peace — to turn in his grave.

Imagine pushing a wheelbarrow: it represents your load. While on the plain there will be full correspondence between the force applied to it and the achieved speed, going uphill you'll see the wheelbarrow inexorably slow down. On the contrary, going downhill the speed will progressively increase and you'll have to do your best not to strike your snout to the ground. Almost likewise, in order that whole power is dissipated by the load this has to be purely **resistive** — the barrow on the plain. If the load presents also **reactive components** — uphills and downhills in the example, inductive and capacitive components in real life — then only a part of the power will be transferred to the load, the other one will be stored by these components. Where it will end up, you do not need to care a damn! What you do need to know is that the presence of these reactive components brings a shift between the tension applied to load and the current flowing in it. This **phase-shift** — so it is called in full, take a note in case they'll invite you again to that cocktail-party, thing that we doubt — in many cases can cause serious perturbations to listening.

If the speaker alone is reactive, think what it has to be with tons of capacitors and inductors to its connecting terminals! For this reason first-order filters are also called **minimal-phase** filters. They are preferred by many designers to cross multiway systems in which the overlap margin of unfiltered responses is adequately wide.

Filters of even order generally output a signal shifted by 180° — btw, phase shift is measured in sexagesimal degrees, from -180° to +180°, depending if the reactive component is capacitive or inductive respectively. To restore the correct phase it's advisable to alternatively reverse filter/speaker's connection polarities between contiguous ways.

Filters above second-order are definitely more difficult and they require skill and experience. What's more, their use isn't so frequent after all.

Now a brief return to attenuation slopes. Is it possible that in some cases this attenuation, that we have described as soft or steep depending on the filter order but constant with the frequency, does assume characteristics varying with the frequency? The answer is yes, and it calls into question a particular configuration, the cascading filters

THE CASCADING FILTERS

This is called **cascading** the arrangement of two or more circuits one next to another on the signal path, to multiply the effect that would be achieved with a single one. We deliberately leave out of consideration the effects that such a configuration would produce on the phase relations between the input signal and the output signal, dealing with the question theoretically.

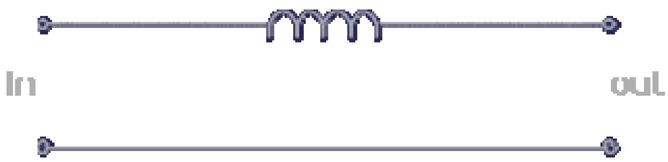
Anticipating a matter that will be developed later, we'll say that a fourth-order filter is formed by the cascading disposition of two second-order filters. This produces, as already seen, a rolloff doubled if compared to that allowed by a single filter of half the order. But this event can be verified only if the two cascading filters are tuned on the same cut frequency. Suppose instead the case of two filters which cut frequencies are close but not coincident. For example, a second-order lowpass tuned to 150Hz next to a similar one tuned to 80Hz. The output signal will be characterized by a constant rolloff of 12dB/oct. immediately above the cut frequency of the first filter, becoming even steeper — exactly doubled — gone beyond 150Hz.

The cascading configuration , although of little utility in practical cases, helps to better understand the working principles of filters, tracing complex structures to the combination of elementary parts that have produced them. It's really with this attitude that, without getting too frightened, we can get to the heart of the tutorial examining in each type of filter the circuitry patterns

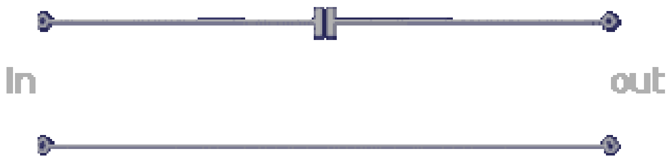
THE CIRCUITRY PATTERNS

We will examine the circuit diagrams of filters up to fourth-order, the most used in crossover circuits. Higher orders don't bring substantial benefits to listening, rather they introduce more than a few complications.

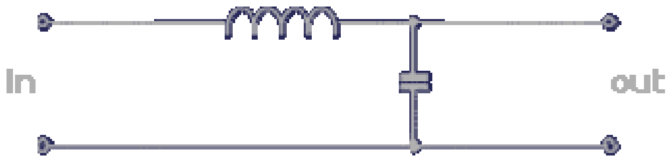
A first-order lowpass filter is simply formed by an inductor in series with the speaker. It produces a rolloff of 6dB/oct.:



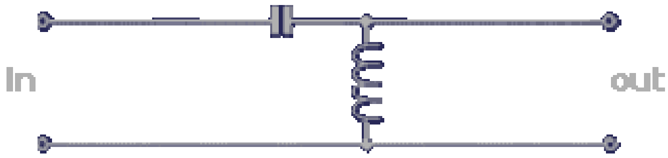
Do you remember what we said about the behavior of capacitors in the fourth lesson? Well, replace the inductor with a capacitor and you'll get a first-order highpass filter with the same attenuation slope:



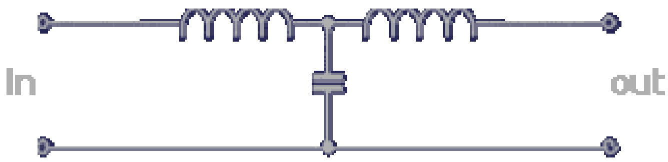
Now try to add a capacitor to a first-order lowpass, but in parallel with the speaker this time. The capacitor — which is totally transparent to the highs — actually diverts them toward ground, helping the inductor to reject unwanted frequencies with better effectiveness. This way you'll have gotten a second-order lowpass filter, with a characteristic rolloff of 12dB/oct.:



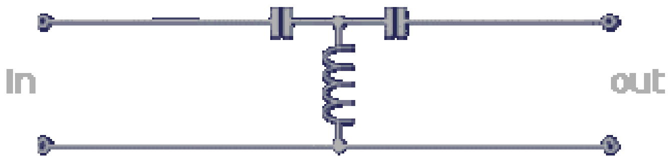
Now start from a first-order highpass filter and reverse the above statements. There, the second-order highpass you asked for:



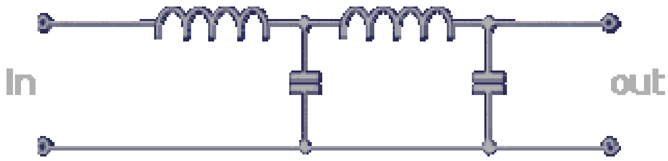
Schematically you can think of a third-order lowpass filter as a second-order one with the addition of a second inductor in series. Its characteristic rolloff is equal to 18dB/oct.:



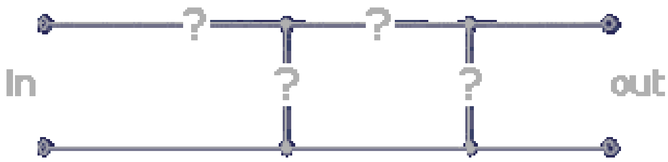
Inversely, the third-order highpass will be formed by a capacitor in series, an inductor in parallel, plus a second capacitor in series:



The rolloff of a fourth-order lowpass is equal to 24dB/oct. It is formed by an inductor in series, a capacitor in parallel, plus a second inductor in series and a second capacitor in parallel. Surely the most attentive of you will realize it is similar to two second-order lowpass cascading filters:



The list is one short without the fourth-order highpass, but at this point of the story you should be able to guess what it is by yourself:



Who's not clever enough to solve the problem is strongly suggested to take a deep breath and start all over again. In the meantime we'll be drawing the numbers

DRAWING THE NUMBERS

Now if you want to design a filter it won't be enough to know which components to use, you also need to know their values, won't you? Well, with the following formulas you'll be able to calculate them according to the desired cut frequency. But keep in mind that results are true only if the filter is closed on a resistive load, that's to say a resistor. A speaker, we'll have the opportunity to better explain it if you'll survive this lesson, is everything but resistive.

In the following formulas, and in those that will come further on, we'll use some symbols:

C	capacity of capacitor, in microfarads (µF)
L	inductance of inductor, millihenrys (mH)
f_c	cut frequency, in hertz (Hz)
Z	speaker impedance by the cut frequency, in ohms (Ω)
π	pi = 3.141592654...
√2	root of 2 = 1.41421356237...
a	√(4+2√2) = 2.61312592975...
b	2+√2 = 3.41421356237...
d	b-1 = 2.41421356237...
e	a*(1-1/d) = 1.53073372946...

And now, without delaying over, out with the abacus (yeah, a pocket calculator is worth the same!):

I order LP	L =	$(1/2\pi f_c)*Z*10^3$	[mH]
I order HP	C =	$(1/2\pi f_c)*(1/Z)*10^6$	[µF]
II order LP	L =	$(1/2\pi f_c)*(\sqrt{2}Z)*10^3$	[mH]
	C =	$(1/2\pi f_c)*(1/\sqrt{2}Z)*10^6$	[µF]
II order HP	C =	$(1/2\pi f_c)*(1/\sqrt{2}Z)*10^6$	[µF]
	L =	$(1/2\pi f_c)*(\sqrt{2}Z)*10^3$	[mH]
III order LP	L =	$(1/2\pi f_c)*(3Z/2)*10^3$	[mH]
	C =	$(1/2\pi f_c)*(4/3Z)*10^6$	[µF]
	L_2 =	$(1/2\pi f_c)*(Z/2)*10^3$	[mH]
III order HP	C =	$(1/2\pi f_c)*(2/3Z)*10^6$	[µF]
	L =	$(1/2\pi f_c)*(3Z/4)*10^3$	[mH]
	C_2 =	$(1/2\pi f_c)*(2/Z)*10^6$	[µF]
IV order LP	L =	$(1/2\pi f_c)*(eZ)*10^3$	[mH]
	C =	$(1/2\pi f_c)*(d/eZ)*10^6$	[µF]
	L_2 =	$(1/2\pi f_c)*(aZ/d)*10^3$	[mH]
	C_2 =	$(1/2\pi f_c)*(1/aZ)*10^6$	[µF]
IV order HP	C =	$(1/2\pi f_c)*(1/eZ)*10^6$	[µF]
	L =	$(1/2\pi f_c)*(eZ/d)*10^3$	[mH]
	C_2 =	$(1/2\pi f_c)*(d/aZ)*10^6$	[µF]
	L_2 =	$(1/2\pi f_c)*(aZ)*10^3$	[mH]

You will notice as in second-order filters the values of C and L are the same for both rows.

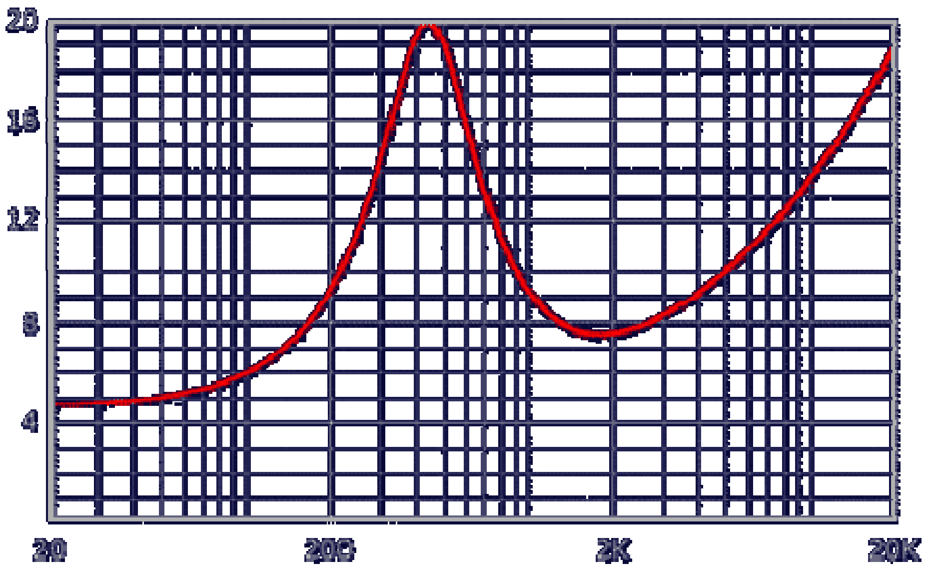
As we've already said above, these formulas guarantee an absolute correspondence between the simulated model and the real model on condition that the filter is closed on a resistor. A speaker is really assimilable to anything but a resistor, unless you accept a considerable degrade in filter performances. You'll understand therefore the importance in giving a closer look to the impedance of speakers

THE IMPEDANCE OF SPEAKERS

In this lesson we'll try to destroy the myth which the impedance of a speaker would correspond to as stated by the builder. In our opinion, to believe in rubbish as this is the same as believing that children are carried by the stork.

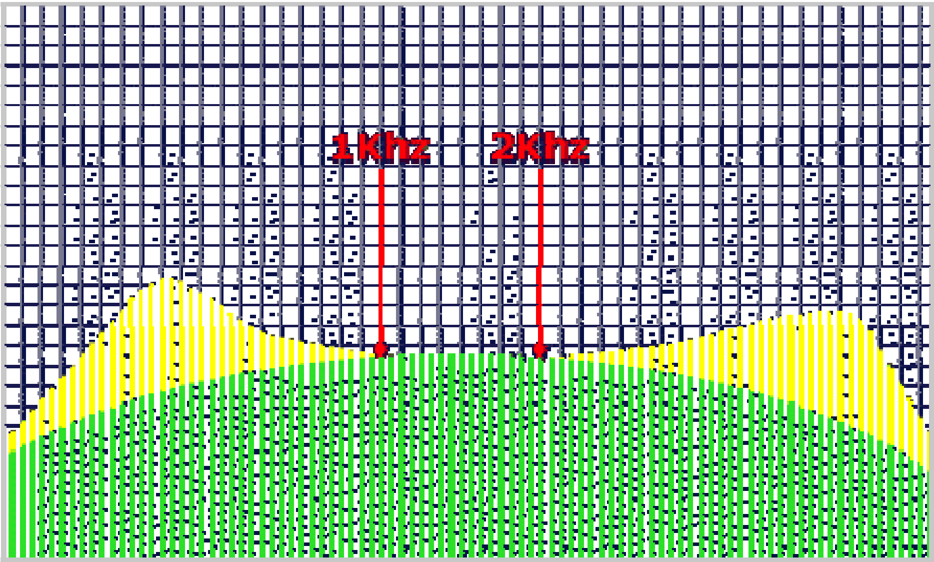
Actually the **nominal impedance,** that specified on catalogs, has the only purpose to divide speakers in families, depending on the type of use which they are intended for. So we'll have 4Ω speakers for car-audio, 8Ω speakers for home hi-fi and 16Ω or more for professional applications. The nominal datum would approximate to the closest of these values the real impedance, measured on the moving coil at certain frequencies — typically 100hz, 400hz, 1Khz, but not only, according to the builder and the type of speaker. Being the nominal datum referred to an only frequency, what's more in approximate way, and being instead the impedance variable with the frequency in real life, it stands to reason that you can't refer to this value for the calculation of the filter. If we then add it's not rare to come up against builders that pass off for 4Ω components speakers of even double impedance, the picture is complete.

It is possible to describe the impedance assumed by the moving coil at different frequencies in graphical form on a cartesian plan, with the scale of frequencies in abscissa (Hz) and the impedance values in ordinate (Ω). The drawing that results is called an **impedance curve** and visually represents the modification of the ohmic value assumed by impedance at different frequencies. See for example that of a cone midrange rated for 6Ω nominal:



The first thing you notice is a peak of impedance with an ascending front and a descending front specular between them (well, this specularity is verifiable only if the frequency scale in abscissa is logarithmic). To it corresponds the **resonance frequency** of the component, recognizable exactly because at the resonance all transducers show a sudden increase of impedance. Moving toward the right, that's going up with the frequency, we meet a saddle where it should be possible to read a value near to that declared as nominal — see what has already been said about some builders — after which the effect of the moving coil inductance begins to make itself felt, in the form of a gradual increase of measurable impedance.

Through the graph below you can realize as a generic second-order bandpass calculated with the provided formulas produces definitely different responses if closed on a resistor (green) or on a real midrange (yellow):

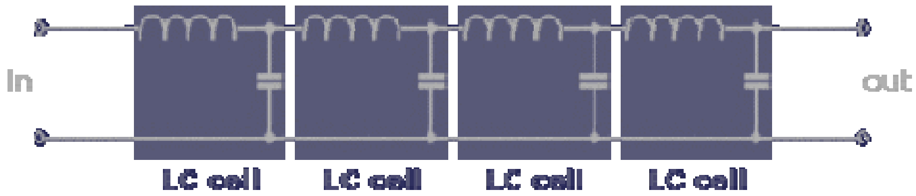


Both to resonance of the midrange and to the highest frequencies, the filter sees a greater load impedance and filters less. Result: the curves are equal only in the octave 1Khz–2Khz while the portions of frequency that the filter had to reject are instead in overbearing evidence. Really awful!

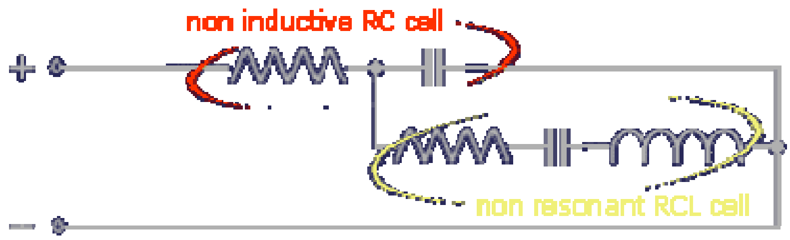
To make the response of the set filter/speaker coincide to the model theorized by formulas it must modify the speaker in order to achieve in its terminals an impedance as much resistive as possible, that's constant with the frequency. Provide to this really them, the equalization networks

THE EQUALIZATION NETWORKS

This is called **network** that part of a circuit to which is referable a specific function, with explicit allusion to the tangle represented by its electric diagram. So we'll have filtration networks — read crossovers — supply networks, conversion networks, feedback networks, etc. A network is composed in its turn of **cells,** that's groups of elementary units, the single electric components. So if for example we made reference to the diagram of an improbable eighth-order lowpass filter, we could realize as it is nothing but a combination of four LC cells, each formed by an inductor and a capacitor:



This preamble needed to say that an **impedance equalization network** is composed of a **non-resonant** cell and a **non-inductive** cell. Their name is self-explanatory to their function. The complete diagram is the following:



This correction network must be placed in parallel with the speaker, between this one and the real filter. Keep in mind also that if the filter is second-order or steeper and the crossover frequency is at least three octaves far from the resonance of the speaker, it is unnecessary to compensate the impedance peak at f_s and the RCL cell can be eliminated. The correction network will be limited therefore to the only RC cell, always in parallel with the speaker and next to the filter obviously:

But here are the respective formulas...

non inductive cell	$R = R_e$	[Ω]
	$C = (L_e/R_e^2)*10^6$	[μF]
non resonant cell	$R = R_e*(Q_{es}/Q_{ms})$	[Ω]
	$C = (1/2\pi f_s)*(1/(R_e*Q_{es}))*10^6$	[μF]
	$L = (1/2\pi f_s)*(R_e*Q_{es})*10^3$	[mH]

...with a due explanation of the symbols:

- R_e — direct current resistance of the speaker's voice-coil in ohms (Ω), also quoted as R_{dc} or R_{vc} by most american manufacturers
- L_e — voice-coil inductance in millihenrys (mH) *(warning, the figure in the formula must be inserted against conversion in henrys)*
- f_s — free-air resonance of the speaker in hertz (Hz), also quoted as f_0
- Q_{es} — electrical merit factor of the speaker at f_s
- Q_{ms} — mechanical merit factor of the speaker at f_s

Nevertheless, getting to this point is not completely a smooth ride. In fact, as everything else was not enough, ninetynine times out of a hundred you'll have a close encounter with another obstacle: some speakers will play stronger than others with which they have to cross. Typically this drawback occurs with tweeters and in general with all dome speakers. Their dedicated filtration row therefore will have to damp this exuberance through the attenuation cells

THE ATTENUATION CELLS

Among the characteristics of a speaker, is its **sensitivity** to inform us about its attitude to play more or less strong. In technical literature it is expressed by the symbol **dB_{spl}** (sound pressure level) and measure — yeah, in decibels — the level of sound pressure 1 meter away with an input of 2.83 volts. Sensitivity is tied to other fundamental parameters such as the **force factor** of the magnetic unit (Bl), the **surface of diaphragm** (S_d), the **direct current resistance** of the voice-coil (R_e) and the **moving mass** (M_{ms}) from the relationship

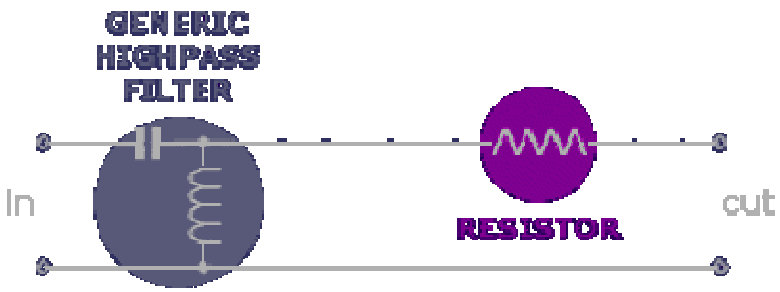
$dB_{spl} = 88.49+20*\log_{10}(Bl*S_d/(R_e*M_{ms}))$

which shows us, radiant surface being equal, that the sensitivity of a speaker can be increased by:

- strengthening its magnet,
- decreasing its moving mass,
- reducing its voice-coil electrical resistance.

This last method is definitely economical in implementation and although it dangerously forces the voice-coil toward its breaking point, some manufacturers make brazen recourse to it to sell for more what has cost less for them.

As it regards us, the true sense of the formula is that being the moving mass at denominator the more it decreases the stronger the speaker will play. The most evident result is that under the same conditions a small speaker will play stronger than a large speaker, that is speakers for the mid-highs and highs are generally the most sensitive. Don't be surprised therefore if this damn tweeter will play stronger than the woofer and don't lose hope, there's a way out. Well, there are actually two ways out. In the first case it's to interpose a resistor between filter and tweeter, in series with this one, according to the following scheme:



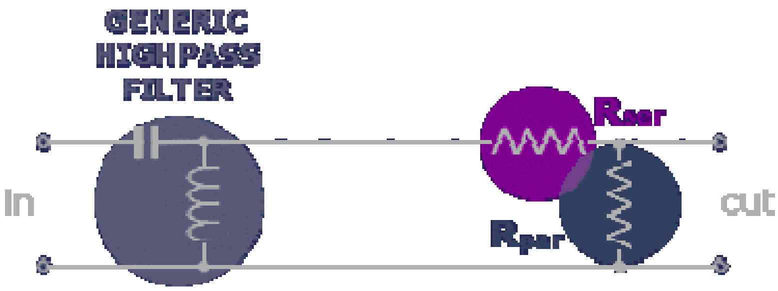
The **wattage** of the resistor will be proportional to the power it has to handle and its ohmic value is calculable with this formula:

$$R_{ser} = Z_{tw}*(10^{\alpha/20}-1) \quad [\Omega]$$

where α is the difference in sensitivity between the two speakers and Z_{tw} the impedance of the speaker to attenuate. Notice we are supposing this is a tweeter just as an example — it could be any speaker. Notice carefully that in this way the filter will not be loaded only on the tweeter but on the combination of this one plus the resistor. Therefore you'll need to re-parameterize the filtration network considering the new impedance:

$$Z_{tot} = Z_{tw}+R_{ser} \quad [\Omega]$$

A second method which has the undeniable advantage in bringing about no variations in the native filter network, is to put a second resistor next to the first one, in parallel with the speaker, according to the following scheme:



This second resistor will also be of appropriate power and its value is given by the relationship

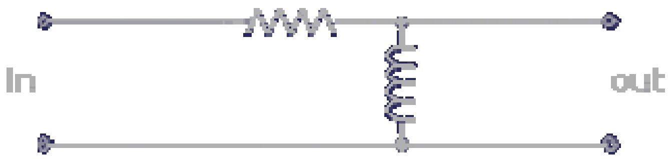
$$R_{par} = Z_{tw}*(1+Z_{tw})/R_{ser} \quad [\Omega]$$

As already said this solution allows us to keep the filter intact, as R_{par} restores its original load. For this reason it is particularly advisable when you work afterwards on already made filters.

There is however a particular filter that all in one go operates both the cut and the attenuation, that's the RL filter

THE RL FILTER

Since in almost all cases a tweeter will require a corrective action on its emission level as well as the cut, to opt for this solution allows us to kill two birds with one stone. It is to put a resistor in series with the speaker and an inductor in parallel. Such a configuration allows the elimination of the whole highpass filter as it itself operate a 6dB/oct. highpass filtering. This is the electric representation of the RL cell:



In this case too the value of the resistor is given from

$$R = Z_{tw}*(10^{\alpha/20}-1) \quad [\Omega]$$

but pay attention to its wattage because being lacking of the filter ahead it has to handle the whole power sent by the amplifier and it is interested in a very high flow of current. It also gets very hot so take care of its positioning.

The value of inductor is calculable through

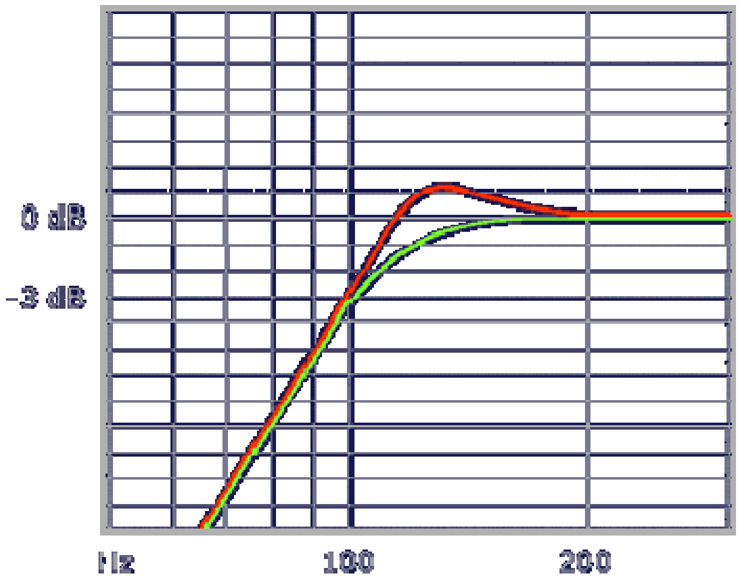
$$L = (R*Z_{tw})/(R+Z_{tw})/2\pi f_c \quad [mH]$$

A filter of this kind is not free from side effects. In combination with lowpass row it constitutes a very critical load for the amplifier, that sees an impedance formed by the parallel of the woofer with the resistor. Speaking from experience, the RL filter requires a great amount of knowledge and ability to experiment. Don't all of you begin to build them starting tomorrow unless you have some idea of what you are doing.

In these last lessons we have described a series of corrective actions to implement on the speaker so that it correctly interfaces itself with its dedicated filter. As a matter of fact, with all we've done till now, you should be able to design and build a passive crossover for any speaker system and even expect it to work. Nevertheless, before we go to the end, we would like to return on the topic filter to develop a concept that in this tutorial we have only touched on: the filter "Q"

THE FILTER Q

In electrotechnics and electroacoustics the letter Q is synonymous with **merit factor**. The word merit can induce the more witty among you to think of a mark that distinguishes a well made filter from a badly made one, but it's not this way. The merit factor is, yes, a number, but it describes the trend of response in proximity of the cut frequency. See how, conventionally reporting to second-order filters. In the illustration below you can observe the overlapped responses of two generic highpass filters, both tuned to 100hz:



You'll notice they are evidently different, although associated by the fact they exhibit an equal attenuation slope below f_{-3}

According to their profile around the cut frequency the curves take a name, which is the one of the mathematician that first got its related equations. You'll hear therefore about **Bessel** curves, **Butterworth, quasi-Butterworth, Chebyshev,** all imbeciles who had their balls shaken on filters instead to go fishing. To each curve corresponds a different merit factor. For example the second-order Butterworth is characterized by a Q of .707 and is also called **maximally flat** because it tends more than all the others to reach the ordinate 0dB without never trespassing it — as a rule, 0dB is assumed to be the mean level of the wave band. Quasi-Butterworth and Bessel curves show merit factors lower than .707 and they too never overcome the level of the wave band. On the contrary Chebyshev and others have higher merit factors and produce a **ripple** more or less accented in the crossover region — particularly, the second-order Chebyshev shows a Q equal to 1.

In second-order filters the Q value is given by the ratio between the capacitor and inductor values, according to the formula

$$Q = Z*\sqrt{(C/L*10^3)}$$

The formulas we have provided you for second-order flters are associated to Q's equal to .707 — try it, you'll see it works. It's evident however that altering C and L values also changes the filter Q. Take the second-order formulas, both lowpass and highpass, and replace the 2 under the radical sign with the integer

L =	$(1/2\pi f_c)*(2Z)*10^3$	[mH]
C =	$(1/2\pi f_c)*(1/2Z)*10^6$	[μF]

and you'll always get a second-order filter even though with different values of components, but with a Q of .5 related to a crossover point of -6dB. This is not a trick to force you to do some more calculations, oh no, it's the expression of a precise theoretical model known as **all-pass filter** — another word to write down if you wish to create a good impression with your friends.

If now you think you have finally reached the end, set your mind at rest. In solving all the formulas you would find very often that there don't exist in the marketplace components for the values you are looking for. The problem is marginal for inductors that are generally custom-made but it is decidedly felt for capacitors and in less measure for resistors. Your choices would then be two. Use a component value closest to the one resulting from calculations, but in this case it would be mandatory re-calculate the new cut frequency reversing the terms in the formulas. The other option would be to make up yourself the desired value appropriately combining two or more standard elements with series and parallel connections

SERIES AND PARALLEL CONNECTIONS

Do you remember what we have said about bipolar capacitors in their lesson? They are formed by two elements connected together and if these elements are equal in capacity it would result in a single capacitor of halved capacity — and double size. Well, it was just a preview of what now we'll go to develop by formulas:

capacitors in parallel	$C_{tot} = C_1+C_2+C_n$
capacitors in series	$C_{tot} = 1/(1/C_1+1/C_2+1/C_n)$
inductors in parallel	$L_{tot} = 1/(1/L_1+1/L_2+1/L_n)$
inductors in series	$L_{tot} = L_1+L_2+L_n$
resistors in parallel	$R_{tot} = 1/(1/R_1+1/R_2+1/R_n)$
resistors in series	$R_{tot} = R_1+R_2+R_n$
speakers in parallel	$Z_{tot} = 1/(1/Z_1+1/Z_2+1/Z_n)$
speakers in series	$Z_{tot} = Z_1+Z_2+Z_n$

Let's stop here. You'll have realized that somehow the topic crossovers opens up to indirect themes that we have deliberately avoided so we don't complicate the exposure. It will be clear therefore that even the complete acquisition of the previously enunciated concepts won't be enough to make an electroacoustic engineer out of you. The hope is instead that we have succeeded in making things just a little bit clearer, using simple language and minimal technical jargon to make the concepts more comprehensible to the majority. It's certainly possible that to some of you the topic was so unpleasant as to be irksome and bothersome, despite our best efforts. To those people we would fraternally suggest considering something other than crossover design. To all the rest, our thanks for your participation and a sincere wish for a good job.