11. Loudspeakers

If you wanna play music, you gotta move some air. For the operation of the acoustic guitar, it is predominantly the vibrating body that generates this air-movement (commonly called sound wave), while in the framework of the electric guitar, that job is done by the loudspeaker. That's the dynamic loudspeaker, specifically, because other transducer types [3] are not called into action as guitar loudspeakers. The diameters of these speakers are specified in inches (1" = 2.54 cm). Most guitar loudspeakers sport 10" or 12", and occasionally also 15"; in small practice amplifiers, 8"-speakers are also common. The guitar loudspeaker is part of the overall instrument – it is supposed to contribute to forming the sound. To put it another way: the guitar speaker should have an atrocious frequency response, and it should distort dreadfully. Okay, maybe not dreadfully – but at least it should distort "adequately". Playing an electric guitar using a HiFi-system will result in a very special sound that is not entirely unusable but not at all reminiscent of Hendrix, Clapton, Beck and Page, either. In the typical sound of an electric guitar that we are accustomed to, not only the guitar player takes part (indeed, that role should never be underestimated), and not only guitar and amplifier contribute – but the loudspeaker, as well. While this book has concentrated so far on guitar and amp, some room shall now be also given to the loudspeaker and its cabinet.

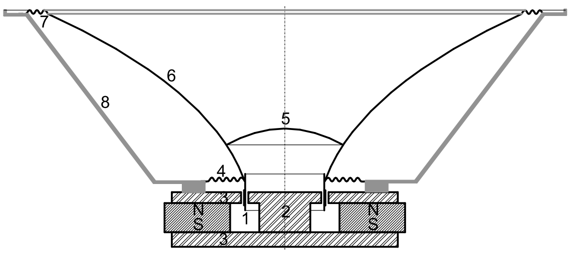
11.1 Build and function

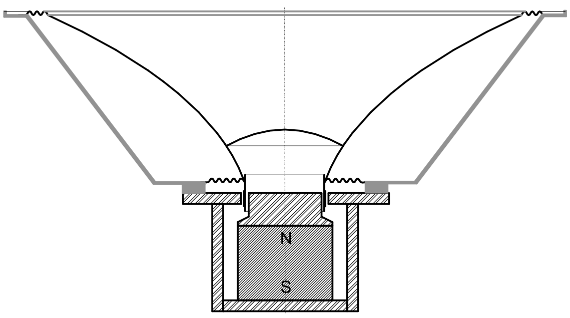
The principle of the dynamic transducer finds its scientific essentials in two simple linear mappings: 1) In a magnetic field, the force acting on a wire conducting a current is $F = B \cdot l \cdot I$, with B = magnetic flux density (induction), I = strength of the current, and l = length of the wire. 2) Moving this wire (in the magnetic field) generates an electric voltage across it: $U = B \cdot l \cdot v$, with v = speed of the movement. The force is termed **Lorentz-force** after the Dutch physicist HENDRIK ANTOON LORENTZ (1853 – 1928), the **induction voltage** usually is linked to the British scientist MICHAEL FARADAY (1791 – 1867). However, not forgotten should be the American physicist JOSEPH HENRY (1797 – 1878) who – independently of Faraday – described the mechanisms of induction, too.

The above-mentioned mapping between electrical quantities (U, I) and mechanical quantities (v, F) is a linear mapping – at least as long as the system parameters B and l remain signal-independent. The latter will of course not be the case anymore for large drive levels. Still, a linear and time-invariant model proves a useful entry point into the description of the transmission behavior of dynamic loudspeakers. That especially for the guitar loudspeaker non-linearity will be essential, that the transmission not only needs to reach a single point in space but an infinite number of these, that in the end time-invariance will not hold – all this foreshadows how complex a model for a speaker can become if we seek to describe "all" characteristics. So let's not go there – the extent of a profound literature search alone would go beyond the scope intended here. The theory presented in the following therefore is limited to the basics, and the examples and measurement protocols given are judiciously selected but not statistically conclusive.

 Translated by Tilmann Zwicker

Fig. 11.1 represents a cross-section through a membrane-loudspeaker. The build variant shown on the right is deployed for Alnico magnets (very high flux density) while the one on the left is conducive when ceramic magnets are used – they require a larger cross-sectional area of flux due to their not-quite-so-high flux density.





1 = air gap w/voice coil and its carrier, 2 = pole core,
3 = pole plate, 4 = centering (spider), 5 = dust cap,
6 = membrane, 7 = suspension, 8 = basket. N/S = magnet.

Fig. 11.1: Cross-section through a membrane-loudspeaker. Left: ceramic magnet; right: Alnico magnet. The shape is largely rotationally symmetric, the ceramic magnet is disc-shaped; the Alnico magnet is cylinder-shaped.

The permanent magnet generates a radial magnetic field in the air gap, and the ring-shaped current-flow in the voice coil has the effect of an axial drive-force on the membrane. The flux density achievable in the air gap is rather high: typically 1 – 1.6 Tesla and occasionally just above that. Both the law of induction and the Lorentz-force require, as a system parameter, the product of flux density *B* and wire length *l*; this is the **transducer coefficient** *Bl*. For an 8-Ω-loudspeaker, *Bl* often has a value between 10 and 20 N/A indicating that a direct current of *I* = 3 A is transformed into a force of *F* = 30 – 60 N. 60 N will hold up a weight corresponding to 6 kg – quite surprising given the fragility of the materials used: the membrane is made of paper, the voice coil of thin copper wire. The geometric data of this **voice coil** are: its diameter *D"* (usually given in inches), its axial length *H*, its turns number *N*, its wire length *l*, and its wire diameter *d* (often termed conductor diameter). If the insulation is included in the consideration, *d* increases by about 10%. The electrical coil parameter is the resistance *R*, at least as long as only low frequencies are discussed. **Fig. 11.2** shows, for a two-layer winding, the dependency of wire diameter *d*, wire length *l*, and turns number *N* on the voice coil diameter *D"* and the voice coil length *H* – given that the copper resistance remains always at *R* = 6 Ω. For a 1.5"-coil of 10 mm length, 11 m of wire (∅ = 0.22 mm) are required; with *B* = 1.5 T, this yields a transducer coefficient of *Bl* = 16 N/A.

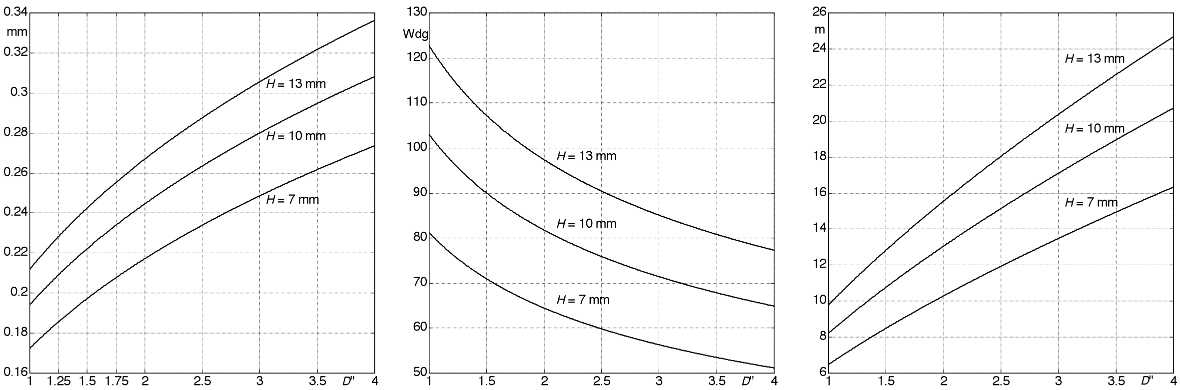


Fig. 11.2: Wire diameter *d* (left), turns number *N* (middle), and wire length *l* (right) depending on voice coil diameter D". DC resistance *R* = 6 Ω. Parameter in the family of curves: *H* = axial voice coil length.

The wire-length may easily be calculated from the winding diameter and the number of turns; however, it is the magnetically **effective wire-length** that is of significance to the *Bl*-product, and not the geometric length. **Fig. 11.3** depicts three different cases: coil-length = air-gap-length, as well as a relatively longer and a relatively shorter variant. The magnetic field is focused in the air gap and grows weaker towards the outside. A coil of a length equal to the air gap (formed by the upper pole-plate) will start to leave the (reasonably) homogenous range of the field as soon as the flowing current deflects the coil. This could formally be considered by defining either the flux density or the coil-length as dependent on the displacement. In the second case, the coil is longer than the air gap – here, the length of the air gap would approximately have to serve as the magnetic coil-length. In the third example, the geometric and the magnetic coil-length correspond. For linear operation, the cases b) and c) would have to be chosen because they feature a coil-penetrating flux that remains approximately constant when displacement occurs. With regard to the efficiency, a disadvantage makes itself felt in case b) in that a part of the coil mass needs to be moved that can contribute only little force because it is located in the weak fringe-field. For c), the whole coil is always positioned within the strong field, but additional magnetic energy is required to generate the – little used – fringe-field. Case a) appears to be the efficiency-optimal, as long the non-linear distortion is not under scrutiny. Since minimizing this distortion does not get top billing for guitar loudspeakers, the latter often feature coil-lengths that approximately correspond to the air-gap-length. Conversely, case b) is commonly found in HiFi-speakers.

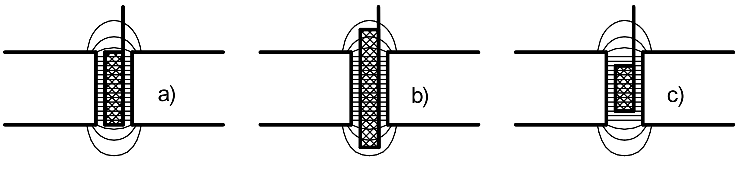


Fig. 11.3: Different voice-coils in the air gap.

In order to obtain a large transducer coefficient *Bl*, flux density and wire-length need to be large. However, because the flux-guiding pole pieces will saturate, it is not possible to indefinitely increase the flux density. A simple solution appears to present itself for the wire-length: large diameter of the voice-coil and/or large (effective) voice-coil-length seems attractive. However, both these approaches cause an increase in the vibrating mass, and thus a decrease in efficiency. On the other hand, a large transducer coefficient will increase the motive force and therefore also the efficiency. The latter is important, but not the one single criterion: power capacity and high-frequency behavior need to be up the desired overall performance. The manufacturers have found their own ways to develop marketable speakers. There is the British philosophy that guitar loudspeakers should have a membrane diameter of 12" and maximum voice-coil diameter of 2". And then there is the approach found on the other side of the Atlantic that demands (among other things) that nobody – and especially not the Brits – will tell an American how to do things. And so – with a sneer of superiority – 12"-speakers with 4"-voice-coils are produced. Nowadays, there is some restraint to dump Brit-ware into the Boston harbor, but the stuff still somehow feels trashy. Or so advertising tells us. Still, despite the 600-W-behemoths with the loud-n-proud 4"-voice-coil fabricated (or at least designed) under the Stars & Stripes, Yanks (and Rebels – and those from the West-Coast, as well) – as far as they play guitar – scour the Internet for that legendary blue British Celestion that will take no more than a measly 15 W. Well, eight of those standing united in a Marshall stack will easily deal with 120 W, after all. Also, if the real original blue ones are not available anymore: allegedly, Celestion has unearthed the olde machinery and produces *original-replicas* on it. In A.D. 2000, those replicants were offered at the steal of 584 Euro. Per unit, that is. 8 x 584 = 4672 Euro … you should be able to beat that down to 4500. Then, only be careful that your roadie – after a particularly smoky night – does not solder a mains-cable to the newly-acquired treasure …

12"-loudspeakers are manufactured with very different voice coils: customary are diameters between 1” and 4”, with a resulting moving mass of 25 to 75 g. Indeed, a larger voice coils is naturally heavier – but it allows for a larger transducer coefficient, as well, and it can dissipate more heat. In the low-frequency domain, these are already the essential parameters, while in the higher frequency range, the voice coil will influence the partial oscillations of the membrane (Chladni♣).

The involved quantities will be exemplified in the following: a 12”-speaker is operated at 200 Hz – this is above the resonance frequency and therefore we have mass-control, and it is below the cutoff-frequency of the radiation – thus there is mass-loading [3]. Simplifying the loudspeaker impedance to 8 Ω, a current of 0.35 A is required for an operation at 1 W. With a transducer coefficient of Bl = 14 N/A we get a motive force of 5 N. This force generates, in conjunction with the moving mass (e.g. 28 g), a membrane acceleration of a = 177 m/s^2 – mind you, that’s no less than the 18-fold gravitational pull of the earth! In fact, this is not unusual for a loudspeaker; at full power, these values will be much higher. From the acceleration we calculate (via integration) the membrane velocity (0.14 m/s), and another integration yields the displacement: 0.11 mm. Since we have been using RMS-values so far, the displacement needs to be multiplied by 1.4 to obtain the maximum displacement of 0.16 mm. Increasing the current 10-fold (to 3.5 A), the power rises from 1 W to 100 W, and the displacement grows to 1.6 mm (given linearity). Now, before we classify the displacement as an unproblematic quantity, let’s quickly recall that the displacement has a low-pass characteristic (with the speaker driven from a stiff current source): reducing the frequency will increase the displacement. With the power of two, that is! At 100 Hz we already have 6.3 mm, and at 20 Hz that would make … 16 cm. No, not really, because here the resonance enters the game: if the loudspeaker would have its main resonance at 100 Hz, it would operate stiffness-controlled below that frequency, with proportionality between force and spring stiffness. But back to 200 Hz: with the membrane velocity as calculated above, we can call in the effective membrane area (530 cm^2) and the real part of the radiation impedance, and compute the effective power radiated onto a half-space: P_{ak} = 48 mW. Distributing this acoustic power over a hemisphere of a radius of 1 m, a sound intensity of 7.8 mW/m^2 results, which yields a **sound pressure level** of L = 99 dB. This value applies to a non-beaming radiation into a half-space.

Fig. 11.2 has already shown that, for a given DC-resistance (e.g. 6 Ω), the wire-length, the wire-diameter and the turns-number may not be chosen independently from each other. One of the parameters is the length of the voice coil, another is the number of layers. Fig. 11.2 was calculated for a **two-layer winding**, but a four-layer winding would be possible, as well, resulting in an increase of the wire-length and –diameter. The transducer coefficient, and correspondingly the efficiency, would profit from the greater length. At the same time, however, the mass that needs to be moved would increase, and a wider air-gap would be required to contain the double-thickness winding. Increasing the width of the air-gab reduces the magnetic flux density i.e. the transducer coefficient. To compensate for the B-decrease, the magnet – the most expensive component of the loudspeaker – would have to be made larger. For the power capacity, the relations are not entirely trivial, either. The power fed to the voice coil needs to be dissipated for the most part via convection (= heat transfer) through the coil surface. However, a four-layer winding has almost the same surface as a same-length two-layer winding –the corresponding gain would be insubstantial. Every manufacturer needs to find their own strategy of optimization; there are two- and four-layer coils on the market, and even coils with rectangular wire, all in order to push for that last further bit of efficiency.

♣ Ernst Chladni (1756 – 1827), pioneer in experimental acoustics.

In fact, it is quite astonishing that a wire area of 25 cm^2 can withstand 200 W, and that 5 A can flow through a thin enameled copper wire without melting it. The current capacity of corresponding wires in a transformer amounts to 3 - 5 A/mm^2 – in a loudspeaker, this value is easily exceeded by a factor of ten. It is the **current density** that usually is seen as the load-limit: current per cross-sectional surface – apparently, there is a line that should not be crossed. If too many Amperes flow through one square-centimeter, the wire goes kaput? No, that's not the case. Across the wire-resistance, the current causes a voltage drop that, when multiplied by the current, represents the absorbed power. 2.83 V · 0.35 A = 1 W, for example (without any phase shift between *U* und *I*). Instead of the unit Watt, we may also use the unit kilo-calory as customary in thermodynamics: 1 W = 0.86 kcal/h. If an electrical resistor is fed with 1 W for an hour, this corresponds to an energy supply of 0,86 kcal. This energy cannot disappear; part of it is transferred to other objects, and part of it leads to a temperature-increase in the resistor. To enable the resistor to dissipate any caloric energy, its temperature *needs* to be increased. From the temperature difference relative to the surrounding air, the caloric energy dissipated via **convection** is calculated, and from the temperature difference relative to surrounding objects the energy transferred via **radiation** can be determined. The former is more important than the latter. A resistor (or in the present case: an enameled copper wire) that cannot dissipate heat well enough will heat up strongly, and it is here where the danger lies: if it gets too hot, it will go kaput, after all. First, the insulating lacquer and the glue will burn, and at too high a temperature the copper will even melt (melting point is 1083 °C). Therefore, it is not the cross-sectional area of the wire that is of importance but rather the surface of the heated object (together with further parameters). The value of the current density thus is not an adequate parameter to estimate the power capacity. Copper traces in printed circuit boards bear testimony to this, too: here, 200 A/mm^2 are not a rarity.

The voice coil needs to pass the energy fed to it predominantly as heat; indeed the share converted into oscillation energy (and sound) may almost be disregarded in comparison. The flowing current heats up the voice coil which heats up the surrounding air; the latter in turn needs to pass its caloric energy as well as at all possible to the field-focusing pole-plates. For that reason, too (i.e. not only in order to achieve a high flux density), a narrow air gap is advantageous. If the voice coil is longer than the air gap, the protruding part is in particular danger to overheat, because the distance to the cooling-providing pole plates is larger. An added extension (necessarily made of non-magnetic material, e.g. aluminum) serves well in this case (**Fig. 11.4**). This extender has no bearing on the static magnetic field but it does on the heat transfer. The dynamic magnetic field will be affected – however, this may indeed be desirable: the eddy current induced in the extender pushes the AC-field out of the magnetic circuit (low-pass), and decreases the non-linearity caused by the field's modulation. Whether a **pole piece vent** is helpful can only be determined in the individual case: given an airtight dust-cap (calotte), a pump results that pumps cooling air into the air gap. However, the effect of a non-linear spring is created also. The vent decreases the non-linearity, and the cooling effect, as well [Klippel W., JAES Vol 52, 2004].

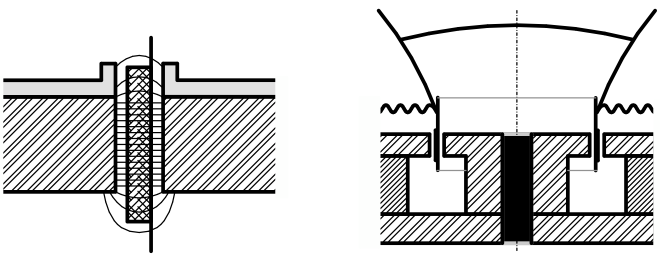


Fig. 11.4: Pole plate with non-magnetic cooling extension (left), pole-core with vent for ventilation (right).

In operation, the voice coil gets very hot, but its material (usually copper, sometimes aluminum) can deal with this issue quite well. Not so insulating material, glue and bobbin. Early on in the era of loudspeakers, the voice-coil carrier was made of paper: thin and lightweight – but not very temperature-resistant, with about 100 – 120°C being the limit for continuous operation. Accordingly, the first 12”-speakers were specified at a power capacity of merely 15W. As new plastics were developed, materials with higher resilience appeared, for example **Nomex** (meta-aramid) consisting of polyamide fibers and enduring up to 220°C. **Kapton** can withstand even higher temperatures: the manufacturer (DuPont) specifies 230°C, but loudspeaker manufacturers readily rely on the short-term specification of up to 400°C. If that is still not good enough: bobbins made from aluminum would take even higher temperature loads. They did not catch on for guitar loudspeakers, however.

Kapton has proven itself as standard material in more recent loudspeakers, but Nomex and even paper are still deployed, as well. The main reason is the sound. Manufacturers such as Eminence attest the paper-bobbin a slightly warmer sound while Kapton allegedly produces a somewhat more brilliant sound. Nomex supposedly gives an intermediate result. In any case, these would not be big differences – shape and build of the membrane have a much more considerable effect here. Eminence offers a 12”-speaker (L-122) optionally with paper- or Kapton-bobbin, with – of course – different power capacity: 20 W and 35 W, respectively, which is a common value for 1”-voice-coils. At the same time, Eminence also offers five further 12”- guitar speakers, among them a 100-W-speaker with a 2”-voice-coil on a Kapton bobbin. Options include “British” membranes, on paper- or Kapton-bobbins.

Temperature-resilience and efficiency are without doubt important features of a loudspeaker, but the main criterion is the sound. Even if the voice-coil may have a small share in this, the membrane (also termed diaphragm) is what takes care of the sound radiation, and it is the component most crucial to the sound. Following simple piston-membrane theory, we have frequency-independent power radiation between the resonance- and the cutoff-frequencies (e.g. between 90 and 600 Hz). Above this, the radiated power drops off with $1/f^2$. At low frequencies, the speaker radiates the sound power into a half-room; from about 600 Hz, beaming sets in, and the power decreasing with $1/f^2$ is increasingly focused onto a smaller section of the room. This piston-membrane theory holds, however, only for a rigidly oscillating membrane not changing its shape at all. At middle and high frequencies, the real membrane vibrates not rigidly but it “breaks up”, i.e. it vibrates in **eigenmodes** (standing waves, partial oscillations). This “life of its own” of the membrane (not initially covered by the simple theory) is undesirable for HiFi-speakers but positively welcome in guitar loudspeakers: it does enrich the guitar sound with invigorating high-frequency interferences. As already noted: color-free, neutral reproduction is not the objective in a guitar loudspeaker. And so the loudspeaker designer batters up the membrane with many a corrugation – such that it may generate as many partial oscillations as possible up to about 5 kHz. In **Fig. 11.5**, one of these circumferential corrugations is shown. Loudspeakers made by Celestion (a brand often used in guitar amplifiers) in most cases include 8 corrugations; in speakers by Jensen (another highly popular brand) we find up to 12 corrugations. More details regarding membrane oscillations are to follow in Chapter 11.3.

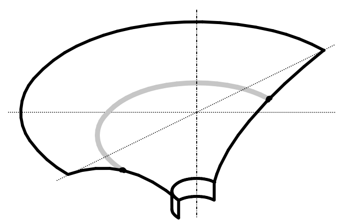


Fig. 11.5: Membrane with corrugation

11.2 Electrical two-pole characteristic

By definition, every loudspeaker is an electro-acoustic transducer i.e. a two-port device with an electrical and an acoustical port [3]. The electrical port (the two connectors) represents a relatively complicated electrical resistor that may be described by its **impedance** $\underline{Z}$. As a rough approximation, the complex impedance $\underline{Z}$ consists of a series connection of a resistor R (real part) and a coil-impedance pL (imaginary-part), with the inductance L and the complex frequency $p = j\omega$. Both components result from the **voice coil**, a cylindrically wound copper- or aluminum-wire positioned in the air-gap of a strong magnet and taking care of the motive force acting on the membrane. The movement of the membrane has the effect that an (additional) voltage is induced into the voice coil, and for this reason it is necessary to consider, within the framework of a more precise model, the mechanical elements transformed onto the electrical side as well. In fact, membrane-movement and –displacement are factors of mechanical energy that cannot appear out of nowhere but have to have their source on the electrical side of the transducer – which is why these quantities need to factor in the electrical impedance [3].

On the mechanical side, the simplest equivalent circuit diagram (ECD) of the transducer considers a mass (membrane incl. suspension and voice coil), a spring (membrane-suspension), and also a friction resistance modeling the energy losses due to deformation of membrane and suspension. The loading by the radiation impedance may be neglected in the simple model. In **Fig. 11.6**, the frequency responses of the impedance of two typical 12”-speakers (not mounted in any cabinet) are shown.

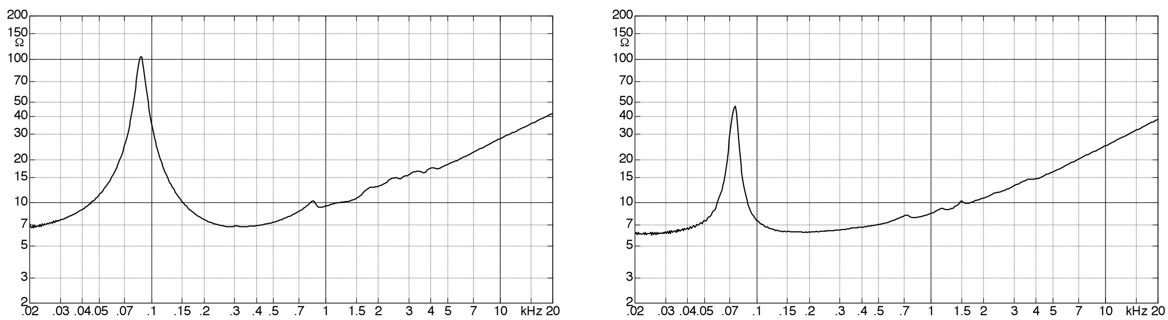


Fig. 11.6: Frequency response (magnitude) of the electr. impedance; left: Celestion Blue, right Eminence L122.

Both frequency responses include a characteristic maximum at low frequencies: together with the spring stiffness s, the mass m forms a velocity-resonance that generates a large counter-active voltage via the transducer-coupling ($U = \alpha \cdot v$, [3]): the current decreases, the loudspeaker is of high impedance at this frequency. For most guitar loudspeakers, this resonance is in the range of about 70 - 100 Hz; for bass speakers it will be somewhat lower. In the impedance-increase at high frequencies, we can recognize the inductive component of the voice coil; however, it is not a simple, frequency-proportional increase but a flatter one. This is due to the fact that it is the magnetic circuit that causes a considerable share of the voice-coil inductance, and in this circuit we find induced eddy-currents that cause a $\sqrt{f}$ -characteristic. For this reason it is not possible to model (in a more exact approach) the inductive increase with a single inductance; rather, we require an RL-network. Given less requirements, a single inductance will suffice; this is often set to 1 mH. The small impedance fluctuations around 1 kHz result from partial oscillations of the membrane, i.e. standing waves that preclude the membrane from maintaining its shape. In HiFi-speakers, designers seek to suppress this kind of behavior – conversely, it is not undesired in guitar loudspeakers.

Fig. 11.7 depicts an equivalent circuit for a loudspeaker-impedance. The resistor designated with R_{Cu} represents the ohmic voice-coil resistance while the LR-array generates the high-frequency increase of the impedance. The parallel-circuit models the three mechanical elements of the membrane. If needed, this circuit may be extended or modified without great effort. At resonance, the impedance of the mechanical membrane-resonator is purely ohmic (W), and it is mapped with $(Bl)^2$ onto the corresponding (ohmic) resistor of the parallel circuit: $R_W = (Bl)^2 / W$. Herein, Bl is the transducer coefficient based on the magnetic flow density B and the length of the voice-coil wire l. Therefore, the resonance-maximum of the loudspeaker impedance is determined mainly be two parameters: the membrane dampening and the transducer coefficient. For this reason, high-value resistances at resonance are often found in speakers with strong magnets.

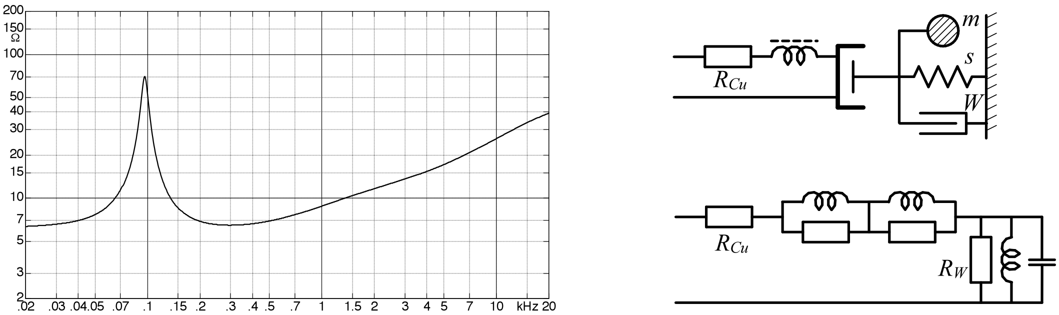


Fig. 11.7: Frequency response of the impedance, and schematic of an equivalent circuit for a loudspeaker [3].

As already mentioned, the membrane movement induces a counteracting voltage, and therefore in a more exact model, special attention needs to be paid to the radiation impedance. At low frequencies, the membrane is predominantly loaded by the **co-vibrating mass of the air** – this will amount to about 7 g for a 12”-speaker (operated without baffle). In absolute terms, that is not much, but it is of considerable magnitude relative to the membrane mass (20 – 50 g). Changing the mounting conditions (baffle, enclosure), this air mass will also vary and detune the resonance (**Fig. 11.8**) Merely adding a baffle will have not much of an effect (the air-mass approx. doubles), but mounting the speaker in an **enclosure** considerably modifies the impedance. Of course, not only the impedance changes – the behavior of the radiation will vary drastically, too. In principle, every change in the electro-acoustical efficiency needs to find its match in the frequency response of the electrical impedance. However, in practice this will, especially in the high-frequency range, not be noticeable because the corresponding changes in the radiation impedance become small compared the mass of the membrane. Moreover, the ohmic resistance of the voice coil will see to it that these small load-variations are practically invisible in the frequency response of the impedance.

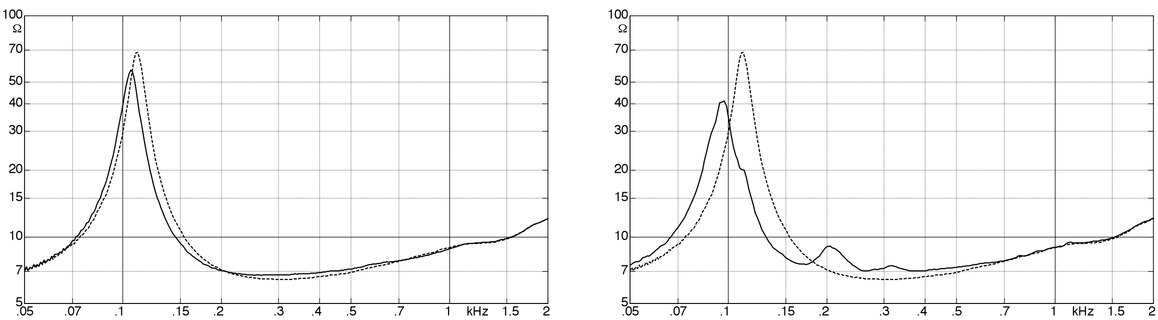


Fig. 11.8: Impedance: loudspeaker without (----) and with baffle (——); right: with and without open housing.

We can easily derive the most important membrane parameters from the *electrical* frequency response of the impedance – this is without any mechanical measurements. With s and m, the resonator has two degrees of freedom but only a single known quantity: f_{Res}. However, detuning of the resonance by applying a small additional mass to the membrane yields two further known quantities and only one additional unknown variable. The system therefore has a solution [3]. In practice, difficulties may be encountered, though: for example if – due to a large dust-cap – a relatively big mass-ring needs to be laid onto the membrane. In this case, it may be that the membrane-stiffness between voice coil and additional mass already has disturbing effect such that the frequency response of the impedance does not merely show a detuned maximum but two maxima. This scenario requires an extension of the equivalent circuit diagram. It may also help to work with two additional masses. The typical **membrane mass** of a 12"-speaker will be in the order of 20 – 50 g, typical **stiffness** will be about 5 - 10 kN/m (without the stiffness of the air inside an enclosure) – in singular cases a bit more.

To determine the **transducer coefficient** (*Bl*), measuring a transmission-quantity is necessary. The membrane-acceleration $\ddot{x} = g$ can be ascertained relatively easily: if $\ddot{x}$ is only even slightly above the earth's gravitational pull, small particles (e.g. sand) set on top of the membrane will start to dance. Typical transducer coefficients are found to be in the range of $Bl = 10 – 20$ N/A.

As the figures presented so far show, the DC-resistance of an 8-Ω.speaker is not actually 8 Ω but less: about 6 – 7 Ω may be seen as customary. This is at room temperature! In operation, **the voice coil heats up** to above 200°C under certain conditions, and the resistance rises correspondingly by up to 80% (for example from 6.5 Ω to 12 Ω). If the speaker is operated from a stiff voltage source, the power taken in by the loudspeaker decreases be a third, as does the radiated sound♣! Likewise, with a tube amplifier having no negative feedback (that in principle is similar to a current source) the received power will drop, as well, if the amplifier is pushed to the drive limit. This volume-drop caused by the heating-up of high-power loudspeakers is system-immanent – undesirable but unavoidable. For ceramic magnets, a further effect may manifest itself: their flux density may noticeable drop off with rising temperature. Alnico magnets show this behavior only at temperatures that considerably higher than the operating range of guitar loudspeakers; the flux density of these magnets is practically independent of temperature.

It is understood that an amplifier needs to feature stable operation (i.e. no RF-oscillations) not just with an ohmic nominal resistance but with a complex speaker load, as well. Therefore, measurements with a real loudspeaker loading need to be taken in fact not just because otherwise any instability would not be noticed, but because only that way the typical output signals occur. Irrespective of whether we have operation with a stiff voltage source or a stiff current source, the electrical impedance of a loudspeaker is crucial for its transmission behavior. The power fed from an amplifier is dependent on the actual loudspeaker impedance, and the nominal value (e.g. 8 Ω) only offers an orientation value. Combined with tube amplifiers with their transformer coupling at the output, we get a particularly complicated system with non-linear source- and load-impedances. Swapping the loudspeaker may cause considerable changes in the transmission behavior especially around 100 Hz – these changes are caused already at the interface output-transformer/loudspeaker. Further contributions are made by the radiation characteristics of the individual loudspeaker.

♣ The exact value will depend on the internal impedance of the power supply.

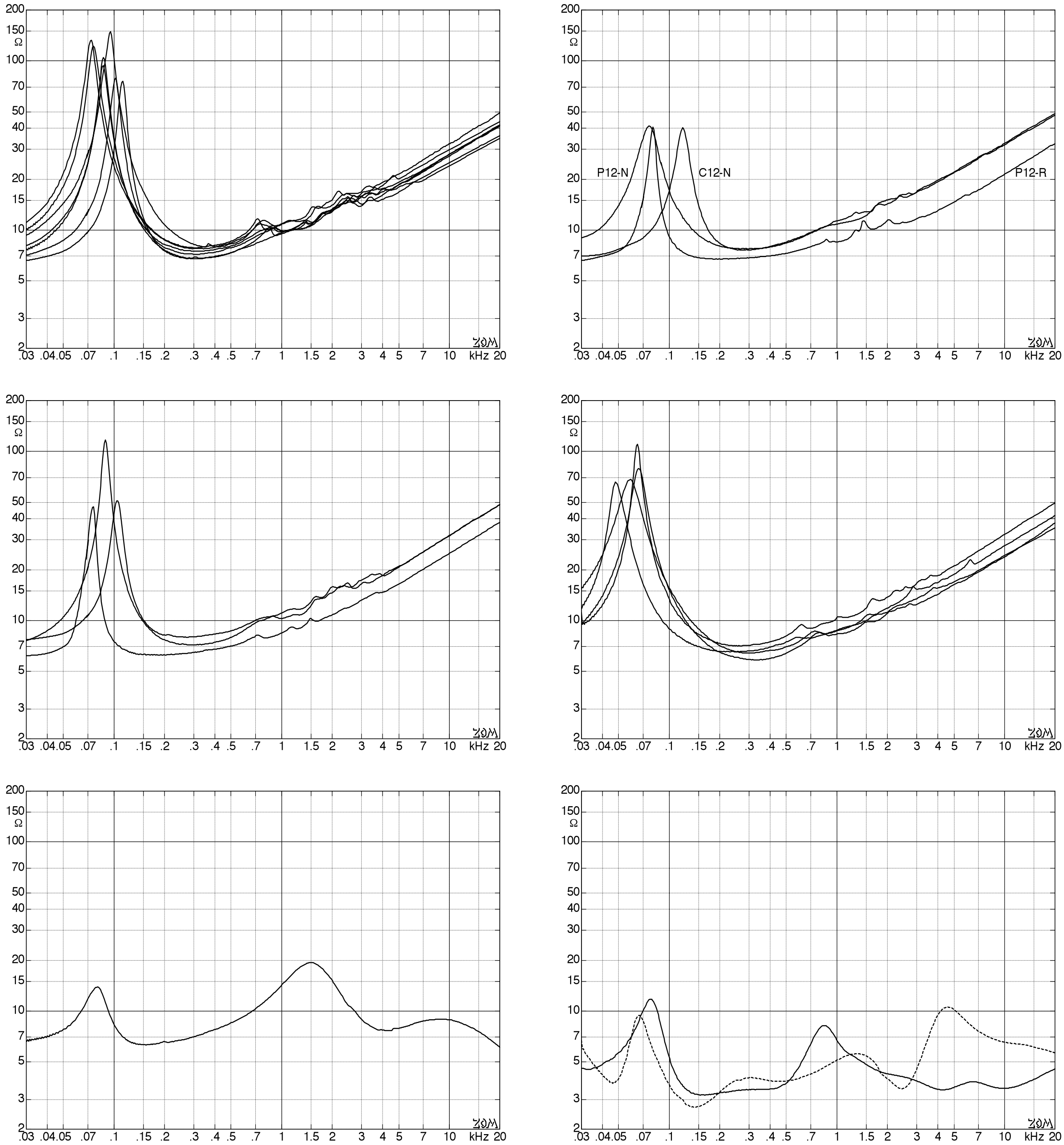


Fig. 11.9: Frequency response of the impedance of 8-Ω-speakers. Upper left: Celestion, upper right: Jensen. Center left: Eminence, center right: 12"-loudspeakers with resonance frequencies below 70 Hz. Lower left: 2-way-speaker (Canton, 8 Ω), lower right: 3-way speakers (Canton, 4Ω).

In **Fig. 11.9** the frequency responses of the impedances of a number of 12"-speakers are shown. All measurements were taken in the anechoic chamber and with un-mounted speakers (i.e. without enclosure). The curves are in principle similar but differences show up in the details. The lower two diagrams show a comparison to HiFi-speakers. All impedance curves were taken with low voltage i.e. in the linear range. Chapter 11.6 will discuss that the voltage/current correspondence may be non-linear, as well.

11.3 Frequency response of sound pressure level

A linear and time-invariant system can unambiguously be described by its magnitude- and phase-characteristics. However, if we take the transmission behavior of a loudspeaker to be approximately linear and time-invariant (for reasonable drive levels this assumption is certainly allowable), *one single* magnitude/phase characteristic is completely inadequate. This is because the loudspeaker is not an *electrical* two-port! While it does include an electrical input-port (the connectors), at its output it radiates a special field formed by sound pressure (a scalar) and particle-velocity (a vector). Both these quantities are location-dependent within the three-dimensional space, and thus an indefinite number of transmission functions exists.

In order to still handle this issue in a reasonably manageable way, we limit transmission behavior to special cases (subsets): the analysis of the frequency response in a single direction, and/or analysis of directionality at a single frequency. In particular, measurements of the frequency response "on axis" (i.e. with the microphone positioned centrally ahead of the speaker) belong to the former group; the latter group includes directional (polar) diagrams.

Trying to appreciate all details will render the frequency response of a loudspeaker infinitely complicated; therefore a rigorous simplification is called for. Starting point for many observations is a loudspeaker mounted in a very large baffle, and with a membrane that is simplified to a flush plate (a so-called piston-diaphragm) [3] to begin with. Assuming linear behavior, the current is proportionally mapped into a force acting onto the membrane and moving it. The spring-like membrane-suspension and the mass of the membrane in conjunction form a resonator with a pole frequency at 70 – 110 Hz. Below this pole- (or resonance-) frequency, the membrane acts approximately like a spring, and above it acts like a mass. Alternatively, we may say that below resonance, the membrane is spring-controlled, and above resonance, it is mass-controlled. Given a sinusoidal current, the three movement-quantities displacement, velocity and acceleration are generated; they can be converted into each other via differentiation or integration. Since the membrane is mass-controlled above the resonance frequency, a stiff current source will imprint the acceleration in the corresponding frequency range (Newton: $F = m \cdot a$). With the linear model, it is no problem that loudspeakers are not always driven from a stiff-current source: the electrical impedance links voltage and current.

Integration of the membrane acceleration yields the membrane velocity from which – using the real part of the radiation impedance – the radiated **effective sound power** may be calculated [3]. In the simple model, this effective sound power is frequency-independent between the resonance frequency and the upper **cutoff frequency**. The latter is at about 600 Hz for a 12"-speaker; above that, the radiated power decreases with $1/f^2$. Or so the simple theory says. The frequency responses measured on-axis do show that your typical guitar speaker will radiate frequencies up to 5 kHz with a rather decent level – only above this limit, the frequency response drops off quite abruptly. This is, however, no contradiction to the theory, because sound-level and sound-power are not equivalent: upwards of 600 Hz, the radiated power decreases, but beaming-effects focus it increasingly to the area in front of the membrane. In fact, power-decrease and beaming compensate each other in the simple model such that on-axis there is no high-frequency drop-off at all. Still, this is where grave differences between theory and practice become visible: the real membrane deviates particularly in the high-frequency range from the idealizing theory. While the theory of the axially oscillating piston-diaphragm requires a rigid-shape membrane, the real membrane shows partial oscillations changing the shape: it "breaks up" and forms nodal lines with partial areas radiating in opposite phase.

Fig. 11.10 shows measurements taken with a loudspeaker installed in a **baffle**. This was not an infinite baffle as required by the theory of piston diaphragms, but a square baffle of 3m by 3m, or a circular baffle of 1 m diameter. Its finite size has the effect of a **diffraction wave** generated at the rim that reaches the microphone and superimposes itself with the direct sound wave radiated by the loudspeaker. As a result, interferences appear i.e. frequency-dependent amplifications (same-phase superposition) and cancellations (opposite-phase superposition) in the sound pressure. For the circular baffle the distance of all points on the rim to the membrane center is equal – a pronounced *comb filtering* occurs. For the square baffle, the path-lengths of the sound wave (around the baffle) are dependent on the direction, and also the wave diffracted around the baffle needs to pass a longer distance: its amplitude therefore is much smaller than that of the direct sound wave, and the interferences are much less distinct.

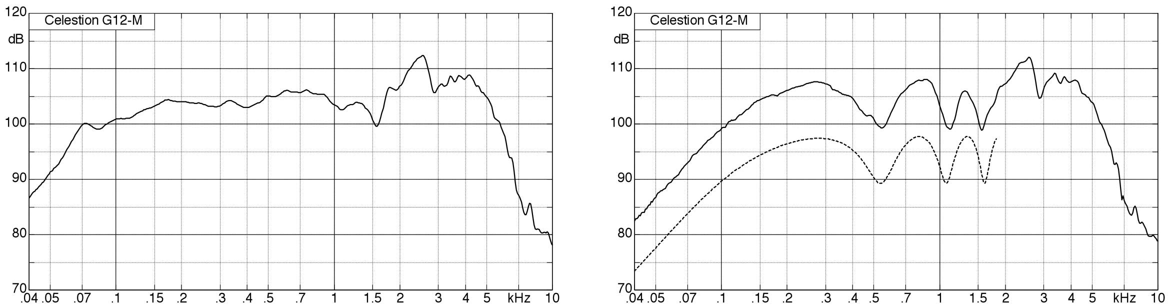


Fig. 11.10: Frequency response of a 12”-loudspeaker installed in a baffle. Microphone position: 0.5 m from the speaker (axially). Left: baffle of 3m x 3m. Right: circular baffle ∅ = 1m. Theoretical interference (----).

In the simple model, two opposite-phase half-spherical waves are radiated on the two sides of the baffle (**Fig. 11.11**). As the wave front reaches the rim of the baffle, its shape changes because now a diffraction wave enters the space behind the baffle. This diffraction has the character of a low-pass: low-frequency sound runs around the baffle without significant attenuation but with increasing frequency the amplitude of the diffraction wave diminishes such that in the high-frequency range only the primary sound dominates – no interference effect remains.

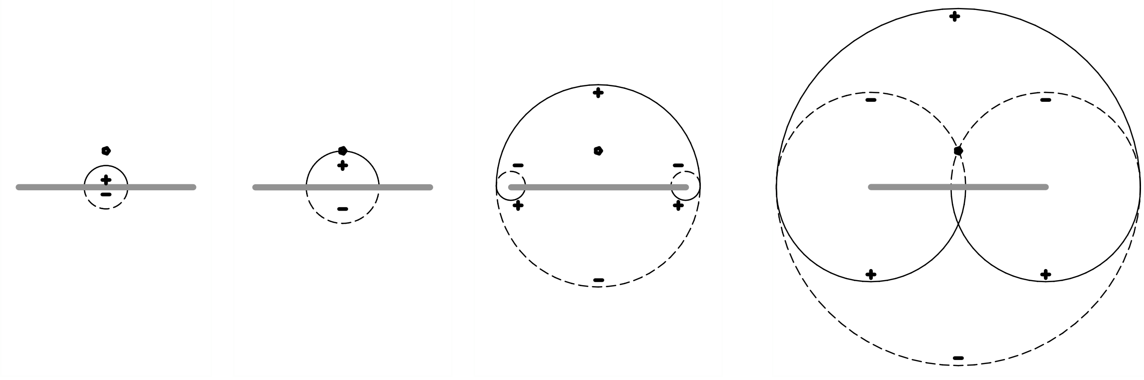


Fig. 11.11: Generation of an opposite-phase diffraction wave at the rim of the baffle. The dot above the baffle designates the position of the microphone; the two opposite-phase diffraction waves follow the primary wave.

In Fig. 11.11 we see the wave at four subsequent points in time. In the second picture, the primary wave just reaches the microphone. In the third picture the wave has reached a bit beyond the baffle, and in the fourth picture the opposite-phase diffraction wave reaches the microphone.

As Fig. 11.10 has shown, the large baffle prevents an acoustic short between the opposite-phase sound waves radiated by the front and the rear of the membrane – however, this approach is not actually stage-worthy. Alternatively, the propagation of the wave radiated from the rear may also be stopped via mounting the loudspeaker in an airtight **enclosure**. That will have three main effects: 1) The radiation of the wave from the rear is stopped, 2) in the enclosure, resonances occur that influence the membrane oscillation and thus the sound radiated from the front, 3) the stiffness of the air encased in the enclosure increases the frequency of the main-resonance. Before we look into the specifics of enclosures, we first still need to investigate the frequency response measure with baffle-mounting in more detail.

According to the theory of piston-diaphragms, the SPL measured on axis rises with a slope of 40 dB/decade up to the resonance (e.g. 100 Hz), and remains at a constant level above the resonance frequency. We have already seen from Fig. 11.10 that reality does not reflect this: from 1.5 kHz, ripples cannot be overlooked anymore, and from 5 kHz, the curve takes a nosedive. The reason for these deviations from the idealizing theory are **partial oscillations** of the membrane; the latter indeed does not manage to rigidly keep its shape but develops a position-dependent pattern of oscillation. **Fig. 11.12** depicts a cutaway view of a typical loudspeaker membrane. From the cylindrical voice-coil bobbin (in the picture at the bottom), the slightly curved membrane extends, with the dust-cap glued to it a few millimeters out. The upper half of the membrane includes circumferential corrugations representing a mechanical filter designed to decouple the peripheral parts of the membrane at high frequencies. At the positions indicated by numbers, the axial membrane velocity was measured dependent on the frequency using a **laser-vibrometer** – see **Fig. 11.13**.

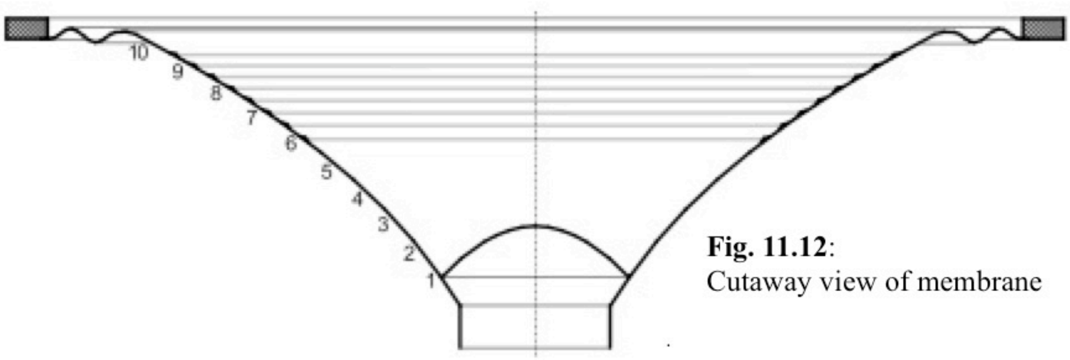


Fig. 11.12:
Cutaway view of membrane

The analysis of the velocity shows that only in the frequency range up to about 300 Hz, the membrane manages to maintain its shape rigidly. In this frequency range, the frequency response of the velocity follows the theoretical band-pass curve. At higher frequencies, a vast variety of eigen-oscillations of the membrane show up.

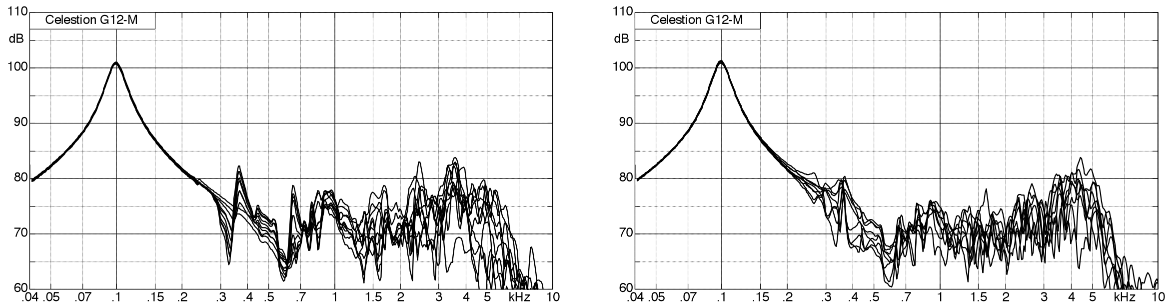


Fig. 11.13: Frequency response of the membrane-velocity at various locations; radial (left), circular (right).

Particularly striking, however, is the fact that the membrane corrugations actually do not form a band-pass, after all! In the fringe areas, the membrane does not at all vibrate less compared to close to the centre – rather contrary is the case: the rim vibrates more strongly. The **low-pass theory** is quite old and stems from a time when it was not possibly to do an on-the-fly quickie-scanning of the membrane with a laser vibrometer. It is easily imaginable that the loudspeakers investigated back in the day with simple methods had such efficient corrugations that the effective membrane-diameter indeed became smaller with rising frequency – as it was desirable in order to optimize beaming and efficiency of the speaker. For the loudspeaker investigated here, however, a multitude of relatively weakly dampened eigen-oscillations are created, the amplitude of which is larger than that of the actuation. Two each of the frequency responses of the velocity from Fig. 11.13 are shown in **Fig. 11.14**: one for a measuring point at the glue-seam of the dust cap (----), and another one for a measuring point close to the rim. Neither for the Celestion speaker (with 8 corrugations) nor for the Fane speaker (smooth membrane), a low-pass filtering is evident.

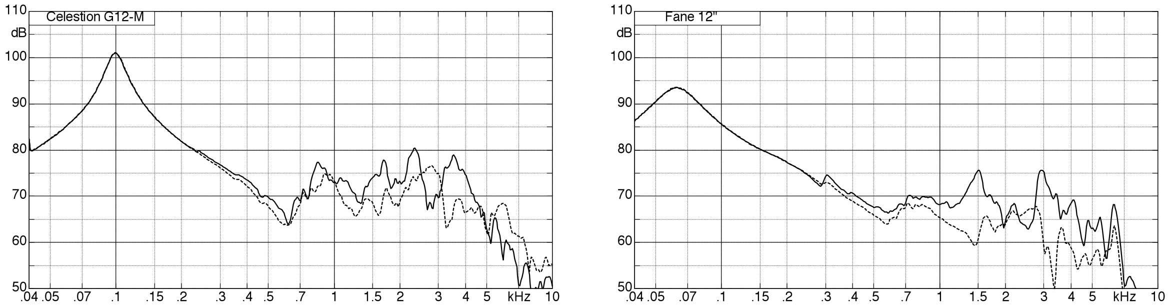


Fig. 11.14: Comparison of the membrane velocities: close to the center (----), close to the rim (——).

Comparing the two loudspeakers, it seems not far-fetched to assume that in fact the corrugated membrane may even be resonance-happier than the even one. That would not actually be a big surprise: every movement actuated by the voice coil (or the magnetic force) starts at the inner rim of the membrane and propagates across the latter as a bending wave. Any change in the wave-impedance – as it is introduced by the corrugations or at the rim – creates reflections. In the end, a multitude of primary and reflected waves run across the membrane. In specific membrane-areas, many waves superimpose with the same phase leading to particularly strong oscillations (anti-node), while in other areas the waves cancel each other out to a large degree, resulting in nodes in the vibration (**nodal lines**). These nodal lines may have the form of concentric circles – as we measure along a radial line, this would be captured as a minimum (Fig. 11.13, left section). However, the nodal lines may also run on a radial course, which would require a circular measuring path (Fig. 11.13, left section).

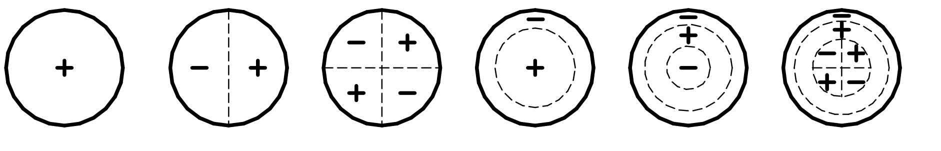


Fig. 11.15: Vibration nodes of the membrane

In **Fig. 11.15** we see a few typical patters of the membrane vibration. The left-hand picture stands for a membrane rigidly maintaining its shape: all points move in the same direction. In the second picture, a nodal line separates the right and left halves: while the point on one half vibrate in one directions, the points on the other half move in the opposite direction. This standing wave does not need to be fully distinct – an additional traveling wave may well be superimposed. The other pictures show vibration nodes of increasing complexity as it may well occur already at frequencies as low as 1 kHz.

The multitude of maxima and minima shown in Fig. 11.13, and also their extreme dependency on the location, proves that in the middle- and high-frequency ranges many different modes come into existence the exact calculation and verification of which was not intended as the subject of the present investigations. A more precise analysis was done only for the G12-M – in this speaker, a location dependent membrane movement occurs already at 300 – 400 Hz. The reasons are probably two 21-modes. **Fig. 11.16** illustrates the vibrations that occur at two relatively close frequencies. As outlined clearly by Fleischer♣ in 1994, this behavior is often found in approximately rotation-symmetrical structures. The eigen-values of anti-symmetric vibration always occur in pairs for the *ideally* rotation-symmetric shape (e.g. 21-mode). For approximate rotation-symmetry they break down into two different values with two corresponding, slightly different eigen-frequencies. The corresponding eigen-shapes are of equal type but differ in the angular position of their node-diameter, as shown by Fig. 11.16: the eigen-shapes occurring at 350 Hz and 374 Hz are shifted relative to each other by 45°.

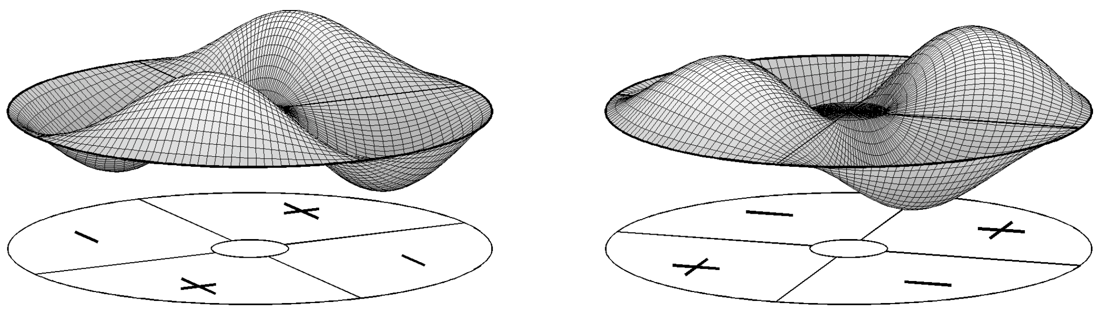


Fig. 11.16: Vibration modes of the Celestion-membrane. The mode shown on the left occurs at 350 Hz, the one shown on the right at 374 Hz. These two modes are the ones of lowest frequency for this 12”-membrane.

Theory and practice concur in that the membrane vibrates – however, it vibrates in such diverse fashions that refining the theoretical models could not be a subject of the presently planned investigations. Therefore, practical measurements were conducted in the anechoic chamber (AEC), generally at a distance of 3 m, with 2.83 V (for an 8-Ω-loudspeaker) fed from a stiff voltage source, or in the reverberation chamber (RC), also using a stiff voltage source (pink noise, 2.83 V per third-octave for an 8-Ω-speaker). For the first measurements, a 12”-speaker was mounted in a small wooden enclosure (39x39x25 cm^3) and a somewhat larger wooden enclosure (39x75x25 cm^3). **Fig. 11.17** shows the corresponding frequency responses of the impedance: as expected, the additional stiffness of the enclosed air increases the frequency of the main resonance. The corresponding effect is relatively strong for the small enclosure and less pronounced fort the larger enclosure.

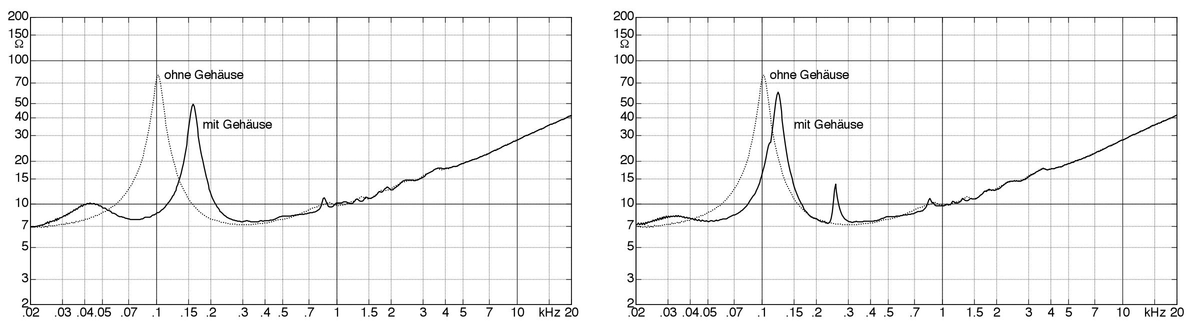


Fig. 11.17: G12-M, impedance-frequency-response, sealed enclosure (“mit Gehäuse”) 39x39x25 cm^3 (left), 39x75x25 cm^3 (right). “Ohne Gehäuse”: without enclosure.

♣ H. Fleischer: Spinning Modes. Research report UniBW Munich, ISSN 0944-6001.

The increase in the resonance frequency amounts to slightly more than 41% for the smaller enclosure and somewhat less in the larger one. Consequently, the **stiffness of the air** is a little larger than the membrane-stiffness♣ for the former case and a little less in the latter. For adiabatic change, the stiffness of the air is $s_L = 1.4 \cdot 10^5$ Pa $\cdot\, S^2 / V$. In this formula, S is the effective membrane area, and V stands for the net-volume of the enclosure. From the effective membrane mass m and the overall stiffness $s' = s_L + s_M$, the **resonance frequency** is calculated: $f_{Res} = \sqrt{s/m}/2\pi$. Mounting the speaker in a sealed enclosure will, however, not only shift the resonance frequency towards higher values but also generate a secondary maximum at about 45 Hz that can be traced to leaks. This "**leakage-resonance**" (as the secondary maximum is often called) stems from the mass of the air moving in the fissures, and the air-stiffness s. And of course, the co-vibrating membrane will – strictly speaking – also contribute. In a completely airtight enclosure, the leakage resonance should disappear. Should it, really? Not necessarily, because that would take an airtight loudspeaker, as well. Any ventilation hole will change the leakage-resonance, too.

Further maxima in the impedance curve are visible above the main resonance, for example at 250 Hz for the larger enclosure. They may be attributed to cavity resonances appearing due to reflections occurring within the enclosure (standing waves, Chapter 11.8). In the higher frequency range (above about 1 kHz), the enclosure loses any influence on the electrical impedance; the rising value of the latter has its main source in the voice-coil inductance.

Fig. 11.7 clearly showed an impact of the enclosure (a sealed box) on the frequency response of the impedance; however, essential for the sound is the frequency response of the sound pressure level (SPL). In this context, **Fig. 11.8** shows the differences between mounting the speaker to a baffle, and mounting it in an enclosure. Two characteristics stand out: the enclosure is unable to easily radiate sound in the bass range, and it generates a series of resonance-peaks in the range of 200 – 2000 Hz that can be traced to **standing waves**. This is particularly evident in the right-hand picture: 240 Hz matches the wavelength of 1.43 m, thus half the wavelength fits exactly into the enclosure (internal length is 72 cm). The peaks at 800 Hz found for both enclosures fits the depth of the enclosure (internal spacing is 21 cm).

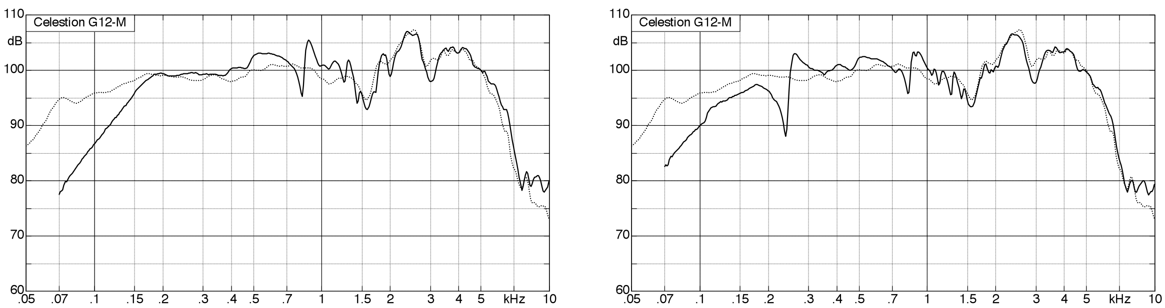


Fig. 11.18: 12"-speaker. Left: baffle vs. 39x39x25-box; right: baffle vs. 39x75x25-box.

Cavity resonances can be fought with a tried and tested remedy that no HiFi-box can do without: dampening material, e.g. quilting cotton, or glass wool, or mineral wool. Standing waves are effectively dampened by loosely filling the enclosure with it, and the frequency response becomes more even. There is, however, a loss of efficiency that is undesirable for guitar-speakers, and any padding is usually dispensed with here. In contrast to its acoustic cousin, the electric guitar has no adequate body that would take care of introducing cavity resonances, and therefore loudspeaker resonances are indeed rather welcome.

♣ This term always actually refers stiffness of the membrane suspension.

Fig. 11.19 shows frequency responses of enclosures with dampening fitted in the form of porous absorbers. The latter represent a real load-impedance to the membrane and transform effective power into heat. This is not desired, since corresponding dampening of the membrane from the rear reduces its movement and thus also the radiation of sound.

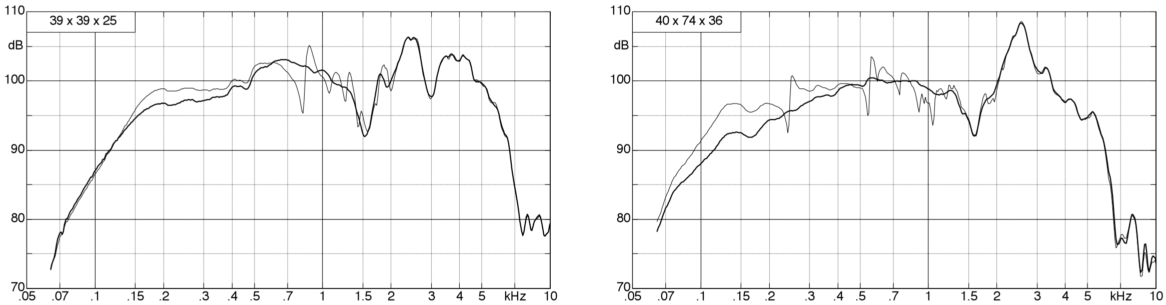


Fig. 11.19: Two different loudspeakers mounted in enclosures with (——) and without (——) absorber.

Besides absorption, the **opening up the box** is another possibility to reduce resonance-effects. In the un-dampened enclosure (closed box), the sound waves generated by the rear of the membrane are efficiently reflected back and forth; standing waves of high Q-factor can manifest themselves. In a box open towards the rear a large part of the sound energy generated by the rear of the membrane leaves the box after only a few reflections. Desirable side effect: both sides of the membrane contribute to the sound arriving at the listener's location. Undesirable side effect: ditto. That is because of course the two involved sound waves will not generally superimpose on each other with the same phase, and destructive interference (cancellation) is bound to occur, as well. The membrane acts as a **dipole**: as one side generates a positive pressure, the other side will generate a negative pressure. Still, the same happens for the vented enclosure (bass-reflex box), and that does work quite well. The reason is that phase shifts [e.g. 3] are introduced via acoustical filters and different-length travel paths of the sound wave. For sound reproduction of very low frequencies, the open box certainly is sub-optimal – in this frequency range, the sound waves generated by the font and the rear of the membrane, respectively, will cancel each other out to a large degree. In the closed box, this cancellation is prevented – but problems in the very low frequency range still appear due to the high air-stiffness increasing the resonance frequency. Luckily, very low frequencies are not that important for the guitar and often even unwelcome. The first guitar amplifiers thus were of the open-box "combo" design – to this day a tried and trusted variant.

Fig. 11.20 shows frequency responses with an open box. The way the sound is guided increases the rear mass loading of the membrane and the resonance frequency decreases slightly. Cavity resonances are still present but more strongly attenuated than in Fig. 11.19.

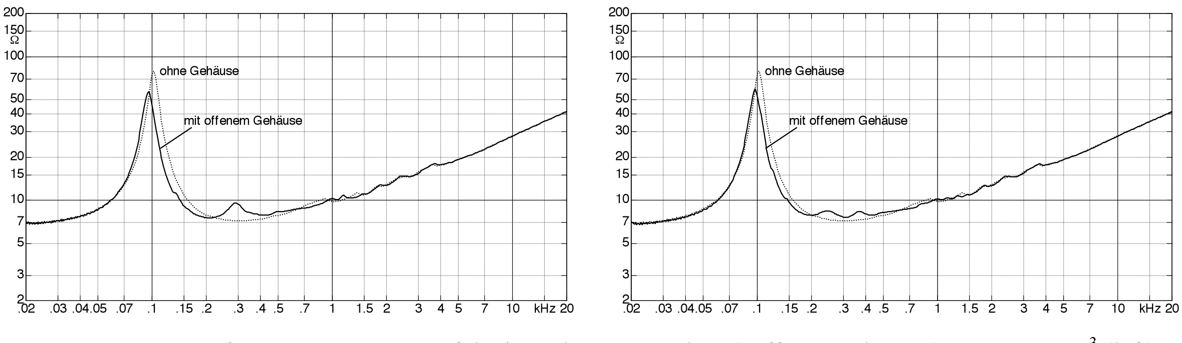


Fig. 11.20: G12-M, frequency responses of the impedance; open box ("offenes Gehäuse") 39x39x25 cm^3 (left), 39x75x25 cm^3 (right). "Ohne Gehäuse" means "without enclosure".

Fig. 11.21 depicts the SPL frequency response relating to Fig. 11.20. Compared to the reproduction using a baffle, the ripples clearly increase but with a different characteristic compared to a sealed enclosure: they are less narrow-band but more global and come in broader arches. The figures in the second row hold information on the sound power radiated in the diffuse sound field: from 200 Hz and 160 Hz, respectively, the open cabinets radiate more sound; only in the frequency range below, selective attenuation occurs. **Conclusion**: compared to the closed cabinet, the open-cabinet design is louder but also somewhat weaker in the bass. Again, it remains a matter of taste, which one you prefer.

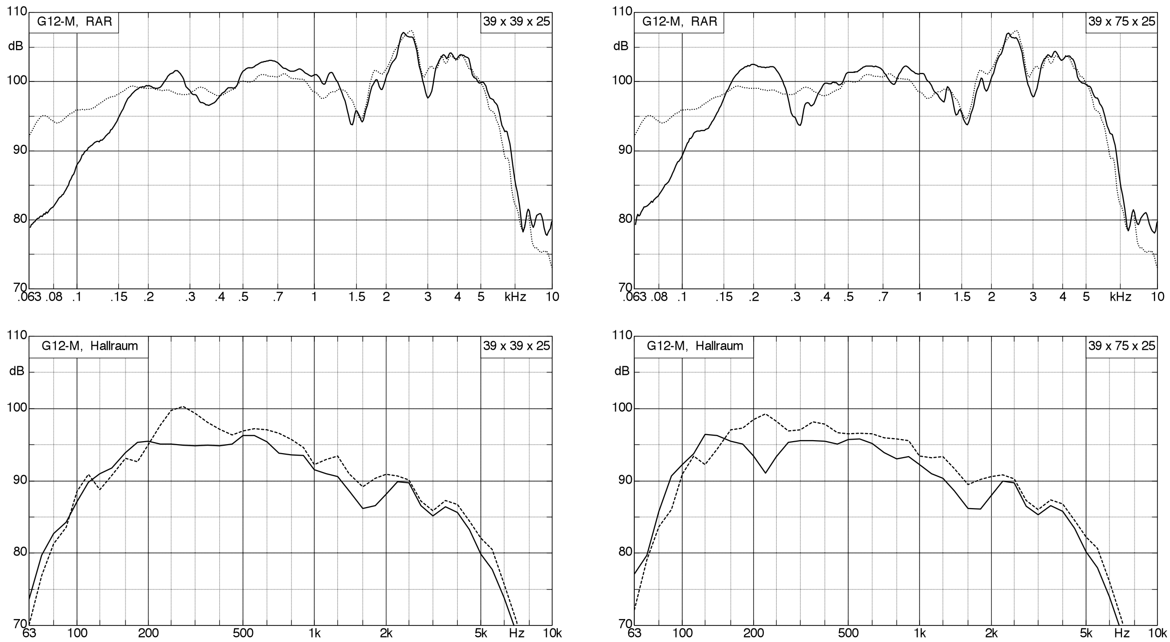


Fig. 11.21: Top: baffle vs. 39x39x25-open-cabinet (left) and 39x75x25-open-cabinet (right).
The lower row shows frequ. responses in the diffuse field, w/rear panel of the cabinet (—) and w/out (---).

One could argue that in modern times with super-powerful signal processors, the frequency response of the loudspeaker is insignificant because any desirable frequency response may be "designed" with a few rows of program code. Again, the guitar amplifier breaks rule: if power-amp distortion is favored (as it is by many guitarists), digital filtering is not possibly anymore. The loudspeaker directly follows the power amplifier, and – as irrevocably postulated by systems theory – the sequence of circuit sections may not be changed in non-linear systems. Only the loudspeaker and its cabinet can filter the signals generated by the power amp, after the speaker there is only the space … the final frontier. The speaker, or rather the membrane, filters mechanically, and the cabinet acoustically – and not insignificantly, either. In the dimensions of the loudspeaker cabinet, the designer has effective parameters at his/her disposal to kick the frequency response into shape one last time – after that the sound leaves the production area. Presumably, the size of the loudspeaker that had to be accommodated was the main criterion for the dimensions of the first guitar combos, and even for Jim Marshall's 4x12-cabinet, that was no different: the cabinet primarily served as mount and protection. Acoustic filter design came later – if at all. Maybe it was a happy chance that the dimensions of the now legendary small combos were not far from the dimension of an acoustic guitar. The shape of a cavity determines the cavity resonances, and what sounds good in a guitar may help to arrive at the right sound color in a speaker cabinet, as well.

The lowest body resonance (the so-called Helmholtz-resonance♣) of the acoustic guitar is located between the notes of F#2 and A2 i.e. at 92 – 110 Hz. Incidentally, that is exactly the range where most guitar loudspeakers have their main resonance – unless you mount them into a small, sealed enclosure. The latter may push the resonance up to 160 Hz (see Fig. 11.17) corresponding already almost to an E – not the E2 of the low E-string but the E3 one octave higher. If we now would combine such a cabinet ("tuned high") with one of the legendary amp-forefathers (e.g. a Tweed Deluxe or an AC-15), we would obtain entirely different frequency responses than those shown on Fig. 11.18. These early amps had **no negative feedback** (NFB) **in their tube power amps**, and therefore they featured a rather special internal impedance: within the small-signal range, the terms "stiff current-source" is almost appropriate, while in overdrive conditions (clipping), they form almost a stiff voltage-source. All SPL frequency responses presented so far in this chapter had been measured using a stiff voltage-source; switching to a stiff current-source (imprinted current), the frequency response of the impedance multiplies onto the transmission factor. For example: if the impedance at 160 Hz rises from 7 Ω to 50 Ω, the SPL will increase by 17 dB! Not all (tube) power amps dispense with negative feedback: in Fender amplifiers, for example, NFB becomes a standard circuit feature from the 1960's. VOX, however, does not follow that route, and to this day the AC-30 does not have NFB. Power amps without NFB feature high output impedance with a value of 200 Ω easily reached. Introducing NFB will decrease the internal impedance – but not down to zero. For one, a high NFB-factor will decrease the gain (which is a precious commodity in tube amps), and second, phase-shifts may quickly lead to instability. Therefore even a tube amp with NFB may easily have an internal impedance of 20 Ω – which would, in the case of the above example, not lead to a resonance boost of 17 dB but still to one of 9 dB.

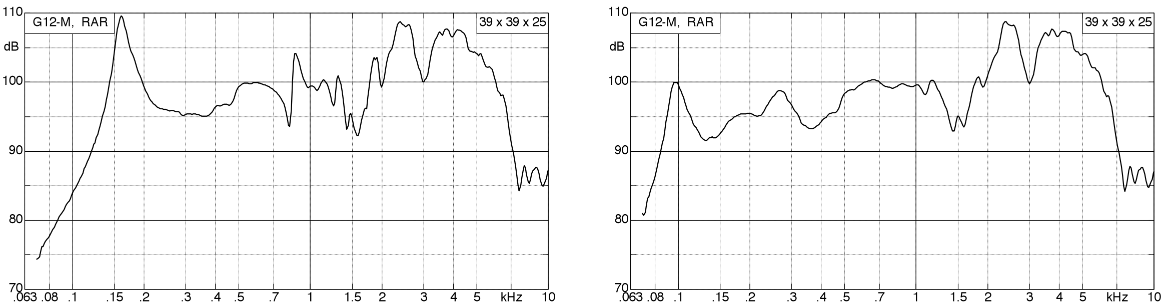


Abb. 11.22: SPL frequency response with imprinted current, 39x39x25-box; closed (left), open (right).

Fig. 11.22 shows, for the small speaker box, frequency responses resulting from driving it via a stiff current-source (**imprinted current**). In this mode of operation, any trace of a weak bass-response has disappeared in the open cabinet; the resonance frequency (lower than with the closed box) takes care of the required low-frequency-boost. In the linear range, that is, since internal impedance of the power amp becomes lower as the drive level increases. In addition, non-linearity in the output transformer makes for a rather complicated signal-shaping. Here, there is room for the developer to design – based on the combination power-amp/transformer/speaker/cabinet – a convincing product the characteristics of which surely are not describable with a few diagrams. Measuring frequency responses will help to document transmission functions – no more, no less. The final decision happens with listening/playing tests – and those are not done in the anechoic chamber. Not to forget: the eyes "listen", as well! Not unheard of is the combo that did not pass the final test in the music store because it had the "wrong" name on the front cover …

♣ This resonance is not only defined by the cavity, but also by any co-vibrating walls of the enclosure.

The loudspeaker housing alone offers many design possibilities – an impression of the diversity is found in **Fig. 11.23**. A Celestion G12-M was mounted in 5 different typical speaker boxes with the rear wall being either open, or half-open, or closed. The various peaks occurring at different frequencies (depending on the cabinet) are the result of geometry-specific cavity resonances. In comparison, the type of wood used for the cabinet does not play any role as long as the construction is not untypically fragile.

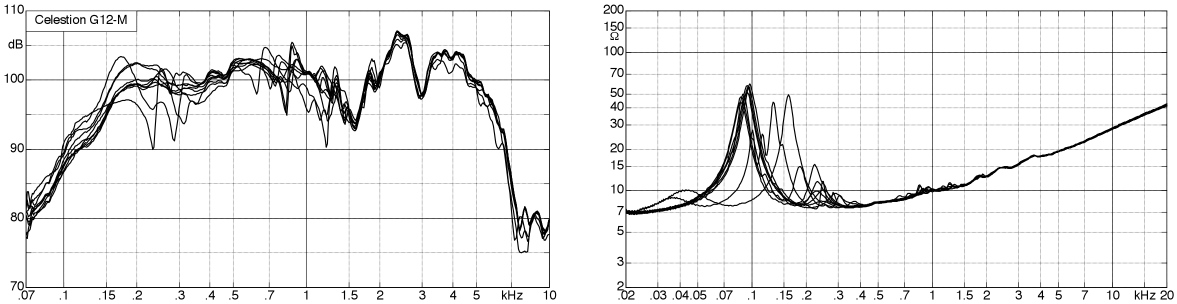


Fig. 11.23: SPL frequency response using a stiff voltage source, AEC, on axis; various boxes, 1W/1m.
On the right the corresponding frequency responses of the impedance are depicted.

To discuss **DSP-filtering** again: of course, it would be possible to approximate the shown frequency response via software. However, the amplifier/loudspeaker-interface connects two non-linear, interacting systems – a simple pole-zero design will not get you far in that context. And not to forget: the loudspeaker filters direction-dependent – something a modeling amp fitted with a DSP is not able to simulate. The filtering calculated in the DSP effects all radiation directions in the same way while every speaker cabinet will have its geometry-specific directionality (Chapter 11.4).

In order to achieve clear resonance effects, the two speaker cabinets used for Fig. 11.21 were deliberately built with special dimensions – they are, however, not entirely typical for the genre. For this reason, the following measurement results were taken with a VOX-cabinet. No the one of an AC-30 because there, two loudspeakers cause interference, but the cabinet of an AD60-VT – the modern housing for a VOX-typical 12”-Celestion. Celestion has been the purveyor to the court of VOX since the late 1950’s, despite all attempts by Goodmans and Fane. In this AD60-VT-housing, the following speakers were mounted: G12-80, G12-M, G12-H, G12-S, Vintage-30, G12-Century, Celestion Blue, and the original speaker of that amp. **Fig. 11.24** shows the measured frequency responses – again referenced to 1W/1m.

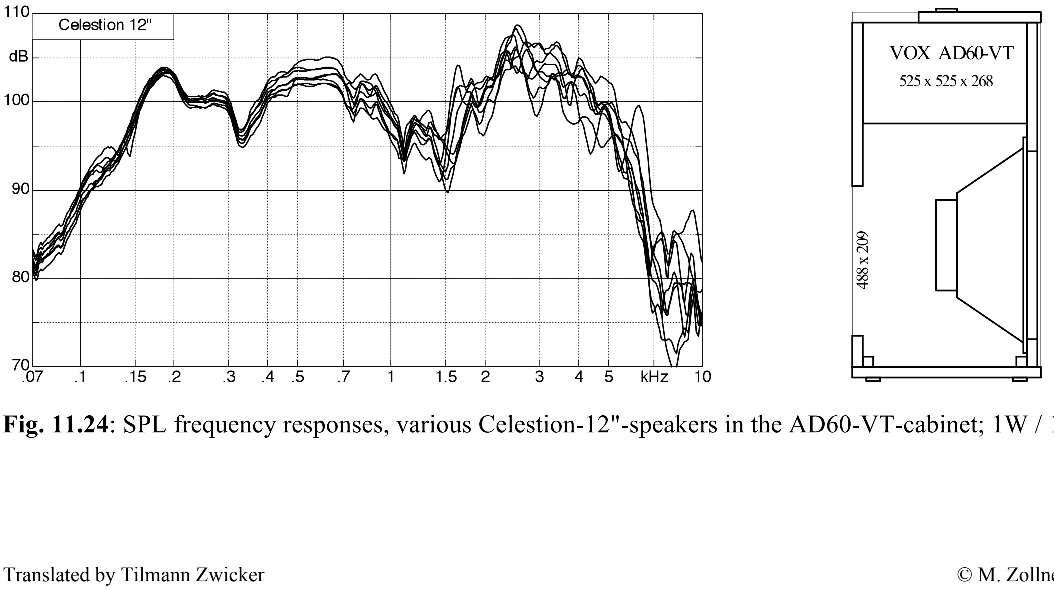


Fig. 11.24: SPL frequency responses, various Celestion-12"-speakers in the AD60-VT-cabinet; 1W / 1m.

The curves shown in Fig. 11.24 share a lot of commonalities. There are, however, also selective divergences that, in the relevant frequency range, do exceed 5 dB here and there. These are very different loudspeakers, after all, with a power capacity of between 15 W and 80 W, and a price range from 127 Euro to 584 Euro (this was in A.D. 2000, and apparently they were serious regarding the latter price). Details are shown in **Fig. 11.25** – and suddenly we are not quite sure anymore whether the same speaker is not erroneously included twice. But no, these are all different speakers, and closely inspecting the peaks reveal the deviations. The latter justify the whole effort – there must be a reason why Celestion builds so many different 12”-speakers. The term *many* might by misleading here, because this small excerpt represents merely a fraction of the allegedly much more than 100 different variants. If the Vintage-30 is not to your liking, just get yourself the Celestion Blue: the price has dropped to a yummy 349 Euro by now *Translator’s note: that’s in 2008, in 2018 it was 279 Euro street price* .

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Fig. 11.25: Comparison of different Celestion 12”-speakers. AEC, AD60-VT-cabinet, 1W/1m.

In the price list from A.D. 2000 mentioned above, the “Blue” sets you back four times the financial damage the Vintage-30 would do. That makes sense somehow, since the Vintage-30 has four times the power capacity of the “Blue”. Sales-math – it can be so simple: 155 Euro for 60 W, and 584 Euro for 15 W. That justifies a closer look at these two candidates: indeed, there are differences besides many similarities (**Fig. 11.26**) – but stop: there is (in the right-hand picture) another competitor in the race that features a similar response curve.

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Fig. 11.26: Celestion "Blue" (——), compared to Celestion Vintage-30. AEC, AD60-VT-cabinet, 1W /1m.

We can see from the lower-most curve in the left-hand picture that the magnitude differences between the “Blue” and the Vintage-30 are mostly smaller than 2 dB; larger deviations are found at only one single place. In the curves shown in the right-hand picture, the maximum differences are smaller although the average square deviation is in fact even a bit bigger than in the picture on the left. Which speaker is that? O.k. – here we go … From the point of view of the manufacturer, it may seem outrageous that, despite the deterringly high price, somebody goes out and buys no less that two specimen of that blue Celestion … and *compares them to one another*. Well, it was simply too appealing to miss. Right: 2 specimen are of course not the quantity that you would need for a reliable variance-analysis, but lets still cut to the chase (without safe statistical base): **according to the present measurements, the differences between a Celestion “Blue” and a Vintage-30 lie in the same range as the differences between two Celestion “Blue”.** The differences between the “Blue” and the Vintage-30 are just about noticeable – but the same holds for the differences between two “Blue”. If the sound pressure levels of two randomly selected Celestion “Blue” differ already by ±3 dB, it must be assumed that there will be even larger tolerances across the whole “hand-built series”. With this, the statement “the Vintage-30 sound more mid-rangy than the blue Celestion” becomes untenable. Broadening the term *intra-individual* from the individual to the same-type group (all the Blues), the rationale is: given such large intra-individual tolerances, the inter-individual tolerances are not significant; the Vintage-30 on average sounds just like the Celestion Blue does. Sure, that is speculation at this point – the sample was much too small, and it might be that one of the two acquired Blues is different from all the rest of the family. In any case: showing bottomless impudence, this author has carried out more comparative measurements with further speaker-pairs: see **Fig. 11.27**. To preempt any wrong conjecture: all speakers were bought in pairs, none was re-coned, and none had been subjected to excess power.

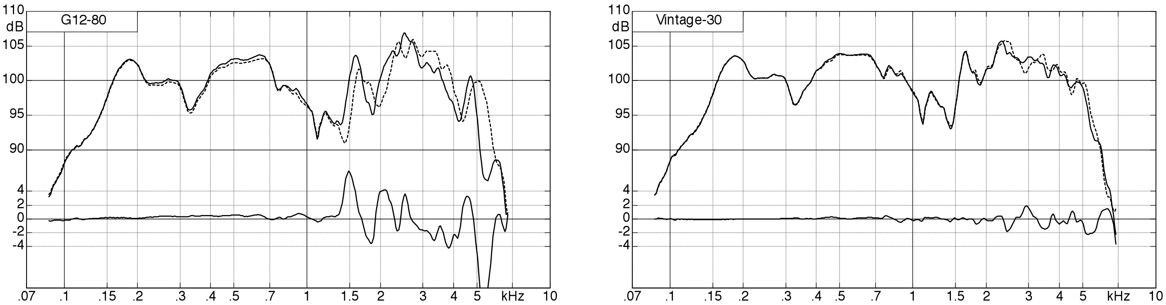

Fig. 11.27: Comparison of two same-type Celestion speakers: 2 x G12-80 (left), 2 x Vintage-30 (right).

What can happen if a loudspeaker is re-coned (i.e. if has received a replacement membrane), is shown in **Fig. 11.28**: someone has re-coned an old AC-30-speaker … with the wrong membrane, however! So much for the legendary vintage-sound …

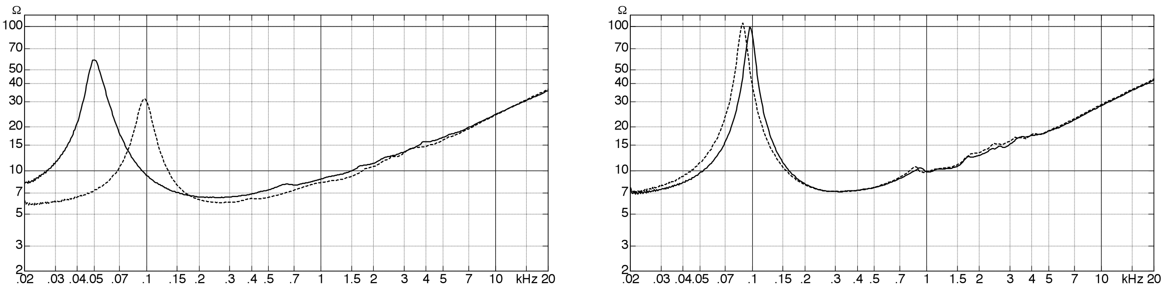

Fig. 11.28: Left: frequency responses of the two Celestion speakers of an AC-30 from the 1960’s.
Right: frequ. response of the impedance of the two Celestion Blue from Fig. 11.26 (measured w/out cabinet).

The measurement results of loudspeakers of the same type or build advise caution: even if we insinuate that speakers from modern production have negligible tolerances, it would be quite appropriate to have some doubts regarding the holy cows from the 1960's or even from back in the 1950's. That AC-30 (copper panel) offered for a whopping $ 4000 – does it sound so good because its speakers have been "played in" for so long? Or because they were re-coned at some point in time with no-name membranes … which the always-helpful Mr. Ly-Ing has discretely stamped with "T530"? Or maybe the amp features yet un-played NOS-Types♣?? Word is the latter are unearthed more and more often these days. It is also easily possible that *new replicas* are mounted: lovingly wound by British hand using old original tooling re-discovered in the back of the basement. Well, that would not be cool, though, 'cause even if they'd been "aged" by Mr. Murphy himself personally – nothing beats the real stuff. This nagging question remains: what's real, if two original G12-80 differ by ±5 dB? An answer cannot be given as long as the vita of most of the old speakers remains shrouded in the mists of time, and artificially inflated prices impede statistically relevant investigations.

So, let us dwell some more on the loudspeakers at hand, and think not just about life in general but the frequency response in particular. A measurement in the AEC is a required criterion, but not a sufficient one. Of course, beaming-effects need to be considered (we'll get to those in Chapter 11.4), and non-linearity (Chapter 11.6). In order to be able to give at least a general statement on directionality, we find measurements in the reverberation chamber in **Fig. 11.29**. No surprise there: differences of a few dB across all measured Celestion speakers, and small deviations between the Vintage-30 and the "Blue" (they do not stand out significantly beyond the – assumed – production tolerances). It certainly would be an exaggeration to attribute the same sound to all (measured) Celestions: there are differences, and they are audible. However, despite all appreciation of the odd decibel that distinguishes the frequency responses here and there, we must not overlook one fact: if you remove the combo from its stand (in Bavaria, that would be two beer-crates …) and place it directly on the floor, level-changes of the same order of magnitude will occur. And that's free of charge!

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Fig. 11.29: Measurement in the reverberation chamber: overlapping 1/3rd-octave analysis, pink noise, rotating microphone. Right: Celestion "Blue" (——) vs. Vintage-30 (----).

And with that, enough space has been dedicated to **Celestion**, manufacturer of *"the finest guitar loudspeakers that money can buy"* – there are others, after all. No, not **Goodmans,** *"the largest UK manufacturer of loudspeakers"*. And not **Fane**, *"Home of the greatest high power speakers in the world"*, either. And neither **JBL**, *"the leading loudspeaker manufacturer in the world"*. Rather: **Eminence**, *"the world's largest loudspeaker manufacturing company"*, shall be checked out, and **Jensen**, simply *"the inventor of the loudspeaker"*. What Celestion represents for VOX and Marshall, Jensen was for Fender. From the 1940's to the 1960's, Fender mounted Jensen Alnico-speakers, and until about 1967 Jensen ceramics-speakers. Optionally, the JBL D-series was available, but Jensen was the standard.

♣ NOS = New Old Stock = unused stock. Allegedly stowed away for decades.

Already at first glance, the P12-R-membrane reveals a different build, distinctly deviating from the Celestion-standard: a smaller dust-cap, and more (and differently formed) corrugations. **Fig 11.30** clarifies the differences: the Jensen is a bit less loud but puts more emphasis on the treble. The latter characteristic, at the very least, would suit the Fender community fine – “silvery treble” is expected there.

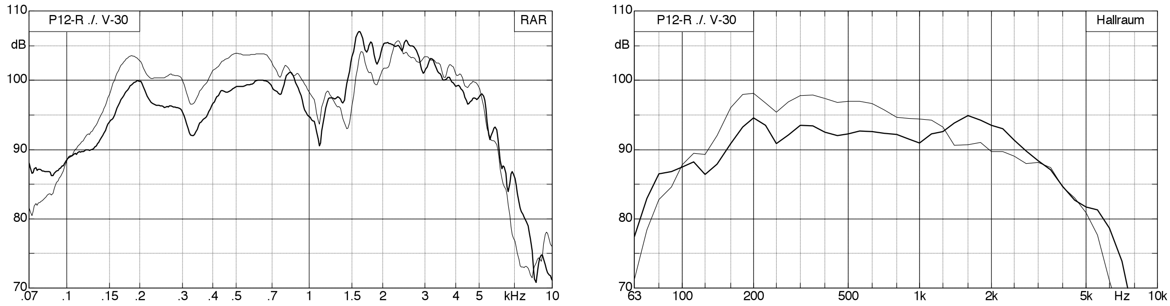


Fig. 11.30: Comparison Celestion Vintage-30 (——) vs. Jensen P12-R (——) in the AD60-VT-cabinet.

You may naturally ask right away what sense there would be in installing a typical Fender-speaker into a VOX-cabinet. Indeed … but how else would you do a comparison? Both in a Fender-cabinet? That would not work either, for the same reason. Each speaker in its own proper cabinet? In that case you would not only compare two loudspeakers but also two different enclosures. Each speaker in a baffle? That would be absolutely not stage-typical. From the almost indefinite number of possible enclosures, we very arbitrarily picked the AD60-VT – a choice had to be made, eventually. Also, in order to enable us to compare to the other measurements presented so far, all further speakers were analyzed mounted in this cabinet.

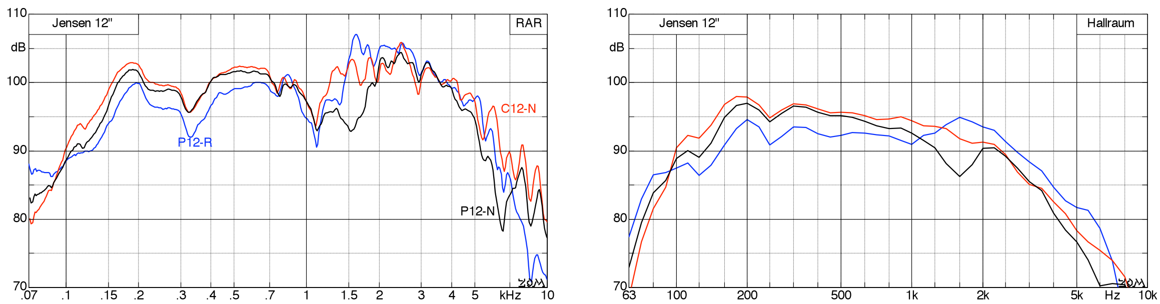


Fig. 11.31: Three Jensen-speakers in comparison: P12-R, P12-N, C12-N in the AD60-VT-cabinet.

Fig. 11.31 depicts the comparison of three Fender-typical Jensen 12”-speakers. Unlike the Celestions – which gave very similar measurement curves – the Jensens show pronounced differences. It is not only the power capacity that is distinct but in fact diverging sound-philosophies are realized: we have the treble-emphasizing P12-R, the Celestion-like P12-N with the marked 1.5-kHz-dip, and the balanced C12-N … as far as you actually want to use the word “balanced” in the face of ±6-dB-fluctuations. These peaks in the frequency response are, however, typical for the genre – none of these loudspeakers could be termed “better” or “worse”. Yes, we could wish for a little better efficiency in the P12-R, but that’s it. Everything else is a matter of taste. Guitarists that appreciate a treble-laden sound with little distortion often opt for the Jensen. Distortion-rockers tend to go for the Celestions. And then there are those players that seek a not-quite-so-trebly sound without much distortion – it takes all sorts to make a world, doesn’t it …?

And on to **Eminence**, Fender's choice in speakers after 1967. As shown by **Fig. 11.32**, these loudspeakers prove to have their own character, too – both in the direct sound (AEC) and in for the diffuse sound field (RC). Jensen and Eminence each offer about a dozen guitar-suitable speakers; only three from each manufacturer were selected and analyzed.

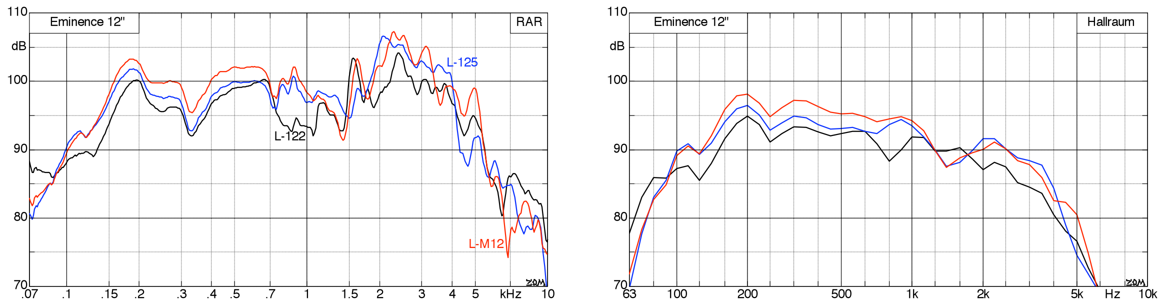


Fig. 11.32: Three Eminence-speakers in comparison: L-122, L-125, L-M12 in the AD60-VT-cabinet.

As a supplement, we will now call in the **Fender-cabinet**, after all, in order to at least once operate Jensens and Eminence-speakers on their home turf: a Tweed Deluxe (**Fig. 11.33**) shall now serve. The small 14-W-amp in the 5E3-Deluxe would not ask too much of any of these speakers; the authentic choice would be a Jensen P12-R. The measurements (as always not with the guitar amp, but with a stiff voltage source) reveal differences that occurred in a very similarly manner with the AD60-VT-cabinet, as well – no surprise there. Up to about 2 kHz we see significant, enclosure-typical divergences that – depending on your mentality and sense of mission – could be called "huge" or "marginal". Some significance should be attributed to at least the 190-Hz-peak in Fig. 11.32 that is followed by a 320-Hz-dip: that's quite typical for VOX alright.

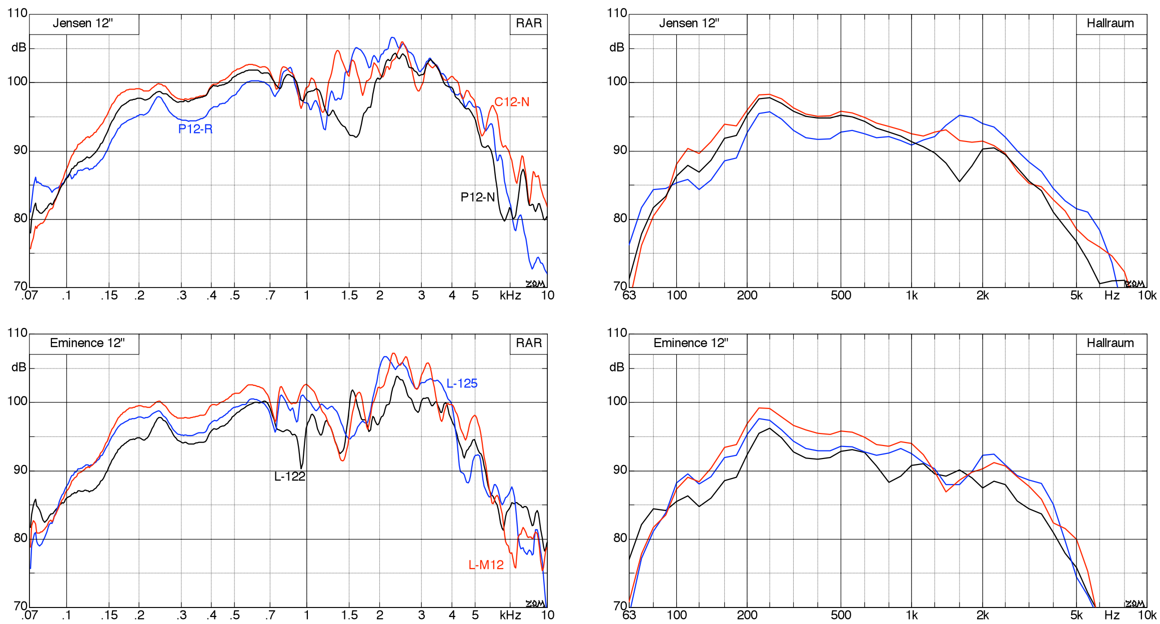


Fig. 11.33: Measurements in the Tweed-Deluxe-cabinet: Jensen and Eminence 12"-speakers.

We have given a relatively large amount of space to the 12"speakers. Before we – more concisely – get to their 10"- and 15"-colleagues, let us try to come to some kind of evaluation – a classification of significant differences (Fig. 11.34):

In **Fig. 11.34**, three different characteristics are emphasized. They were not elaborated using sophisticated factor-analysis but constructed (hopefully not all too arbitrarily) on the basis of visual criteria from the frequency responses. The first criterion is found in the efficiency achieved at 1 kHz, i.e. level (dB-value) of the frequency response. The investigated 12”-speakers exhibit differences up to 4 dB, and that is indeed noteworthy: a level difference of 4 dB corresponds to a power-increase of 150% i.e. for example form 10 W to 25 W. Relating that to **loudness**, as it is readily done from the side of psychoacoustics, is permitted but requires some special caution: the simple rule of “double loudness necessitates 10 dB level increase” is valid for (sufficiently loud) 1-kHz-tones that are not subject to masking! A guitar-tone having to assert itself against competing sounds is not of that category! (For more on this see “masked loudness” in the psychoacoustics-textbooks). 4 dB – in everyday life on stage, that is the difference between “always a bit too soft” and “that’s it!”.

As a second criterion, we picked the range of the mids that was defined for this comparison from about 600 Hz to about 4 kHz (i.e. incl. the so-called “presence”-range). Here we have speakers with and without a **middle-dip** (or “mid-scoop”): it is generally strongly pronounced in the Celestions and rather less in the Eminence L-125. As the last criterion, a distinction is made between a more even level-curve at middle and high frequencies, and a more **resonant curve**. The corresponding first group includes e.g. the G12-H or the P12-N, while the G12-80 and the original speaker of the AD60-VT are to be counted in with the second group. The main differences between all measured Celestions manifest themselves in these resonance peaks: their markedness (damping, or Q-factor) shapes the sound – but it is subject to pronounced manufacturing tolerances, as we have seen in Figs. 11.26 and 11.27.

The question “which is the best loudspeaker, then?” has to remain unanswered for two reasons: if manufacturing-induced variances of speakers of the same type are larger that type-specific differences, classifying becomes rather problematic. And then: beyond the efficiency, sound evaluations are subjective. There are more speakers between …

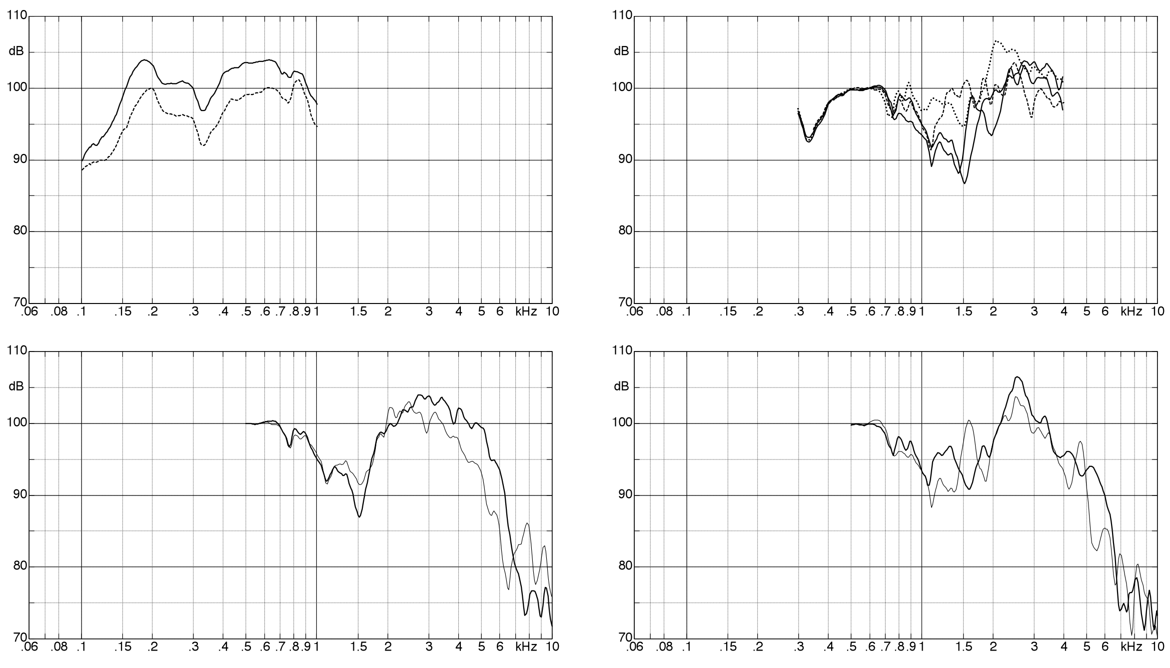
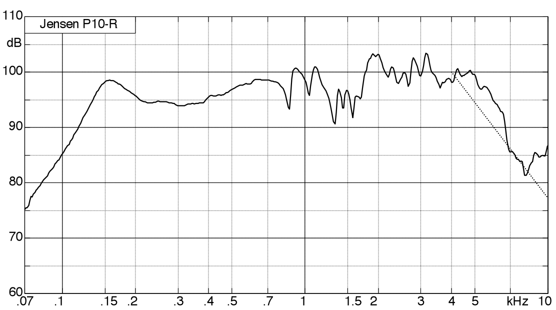


Fig. 11.34: Attributes for distinguishing loudspeakers: 0.1 – 1 kHz (upper left), 0.6 – 4 kHz (upper right), relatively even treble-range (lower left), resonant treble-range (lower right).

At last, we now turn to the **10"-loudspeakers**, as they are found mounted in single units in small combos (e.g. Princeton), but also installed as 4x10”-quartetts in rather grown-up amps (Super-Reverb, Bassman). Compared to the 12”-speaker, the membrane-surface of the smaller 10”-cousin is reduced by 30%; at the same vibration-amplitude, the membrane can thus set less air in motion. More precisely: with the same membrane-movement, the smaller membrane radiates less sound power. Given the equal input power, a 10”-speaker does not need to be less loud than a 12”-speaker but it is in many cases. The “golden rule” of: *the larger the loudspeaker, the louder it is* has a seductive rationale: as the surface of the membrane approaches zero, the efficiency also needs to go down to zero. In reality, however, the surface of a membrane is never close to zero, and the reasoning is misleading. In fact, the efficiency depends not only on the membrane surface but also on the membrane mass and the force-factor (transducer coefficient *Bl*), and these normally differ from speaker to speaker. Eminence, for example, specifies the L-B102 (10") with 101 dB (1W @ 1m) and the KAPPA-18 (18") with 97 dB (1W @ 1m). And a counter-example from the same manufacturer: the L-102 (10") has a 97-dB-spec while the L-151 (15") lists 100 dB.

There is only one safe statement for the difference between 10”- and 12”-speakers. The 10”-loudspeaker is smaller. The 10”-spealer is not generally of lower power capacity, not generally less loud, not generally lighter and not generally brighter in its sound. Regarding the power capacity: both Jensens P12-R (12") and P10-R (10") are specified at 25 W, the C10-Q (10") is listed with 35 W, the C12-R (12") with 25 W, and the NEO-10 (10") with 100 W. As to the weigh: L-122 (12") = 2.5 kg, L-B102 (10") = 5.5 kg. Things become more complicated concerning the treble response, because the upper cutoff frequency of the power-radiation indeed depends on the diameter of the membrane, and on the other hand an ideally form-rigid membrane would be the corresponding pre-requisite to make a corresponding simple statement. Completely wrong is the often-voiced justification that the larger membrane would be too heavy to vibrate at high frequencies. Very basically: as the mass of the membrane is enlarged, the efficiency drops in the whole range above the resonance frequency (e.g. 100 Hz) and not merely at high frequencies [3]. There are other reasons for the fact that in many cases the larger membrane does not sound as trebly as the smaller one: the former has more beaming at high frequencies and generates less diffuse sound in the treble range. In the end, it is always the membrane that plays the pivotal role: its shape and thickness, its corrugations, its damping and its dust-cap determine the transmission characteristics. Eminence specifies 3.5 kHz as the upper cutoff frequency of the DELTA-10, but a whopping 4.5 kHz for the larger GAMMA-15. The basis for this info is, however, an on-axis measurement – presumably the power bandwidth is larger for the DELTA-10 (the datasheet is silent about that).

In **Fig. 11.35** we see the frequency responses (measured in the AEC) of a 10"- and a 12"-speaker. Both were mounted for the measurement in a sealed 39x39x25-enclosure. The P10-R generates, on average, a smaller level but relatively more treble than the P12-R.



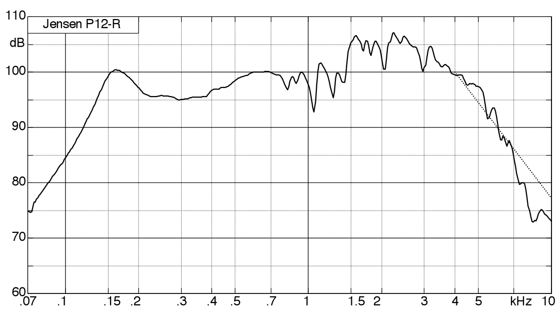


Fig. 11.35: Comparison between a 10"-loudspeaker (P10-R, left), and a 12"-loudspeaker (P12-R).

Fig. 11.36 compares **15"-loudspeakers,** all mounted in the 106-ℓ-enclosure (1 ℓ = 1 liter = 0.264 US-gallons) and measured in the AEC at 1W/1m, with no damping introduced to the enclosure so that the cavity resonances at 230 Hz and 500 Hz are clearly visible. An enclosure of the given size is relatively small for a 15"-speaker. But then almost everything is possible for guitar setups: the Vibroverb, for example, only makes a scant 88 ℓ (gross) available to its 15"-speaker in an open-back configuration, while the Showman is much more generous at 163 ℓ in a ported box. We shall not concentrate on the bass-range here, however – rather the focus shall be directed to the range upwards of about 300 Hz: the G15-100 and the Fane display an even level-response (save for the enclosure resonance) but differ by more than 6 dB. Reminder: in order to increase the level by 6 dB, the input power needs to be quadrupled. The Powercell, on the other hand, is not designed with an even frequency response in mind but shows the typical S-curve of instrument-loudspeakers. The measurements prove a 15"-speaker does not generally generate a higher SPL than a smaller loudspeaker, and document that the upper cutoff frequency (measured on axis) can readily by at 5 kHz – just like for a 12"-speaker. The differences in the beaming-behavior will be examined in Chapter 11.4.

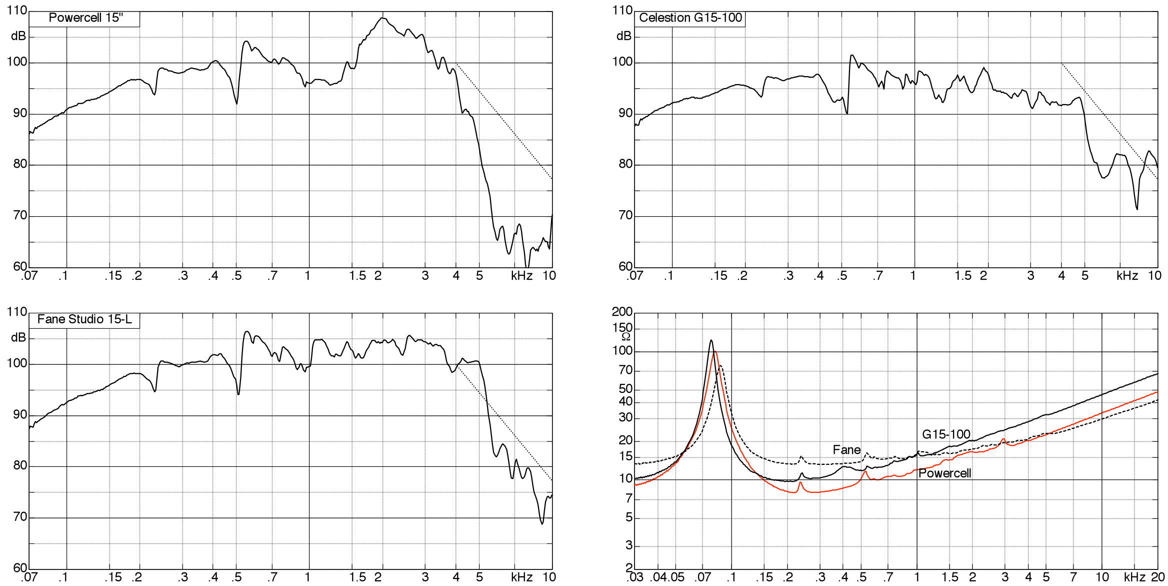


Fig. 11.36: Frequency responses of SPL and impedance, 15"-loudspeaker in a sealed 106-ℓ-enclosure, AEC, 1W @ 1m. All three speakers are manufacturer-specified at 8 Ω.
Maximum power input (manufacturer specs): Celestion Powercell = 250 W, G15-100 = 100 W, Fane = 200 W.

If **two loudspeakers** are mounted in an enclosure instead of one, the on-axis sound pressure theoretically doubles. Compared to the doubled power input, this implies a gain of 3 dB (**Fig. 11.37**). The efficiency does, however, not simply continue to rise proportionally with the number of speakers but depends on the individual geometry.

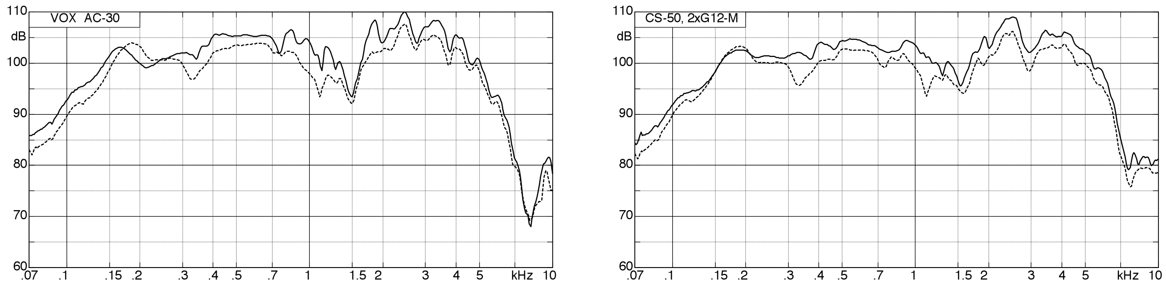


Fig. 11.37: Comparison 1x12" (---) vs. 2x12" (——); ordinate values are referenced to the same power input.

11.4 Directional characteristics

As a loudspeaker radiates sound, it gives rise to a **sound field** around it, i.e. a section in space to which physical quantities can be assigned as a function of location and time. In a sound field, these quantities are the *sound pressure* (p, a scalar), and the *(sound particle-) velocity* (v, a vector). Both quantities are not only dependent on time but also on the location. For frequency responses of loudspeakers, usually the SPL measured "on axis" is given, i.e. the SPL that occurs e.g. 1 m ahead of the membrane. Deviating from this measurement point by a specific angle (e.g. 30°) from the axis, the result is a different frequency response. The reasons for these differences in the frequency responses are differences in travel time (and corresponding interferences) between the sound waves emitted from different sections of the membrane. These are effects summarized with the term **beaming,** or **directionality**.

As a first simplification, the loudspeaker membrane is described as a circular plate (piston diaphragm) vibrating without changing its shape. To explain the directionality, Huygen's principle (well known from optics) is called into action: every differentially small part of the membrane emits a spherical wave, and all these spherical waves superimpose in the free sound field resulting in the radiated sound wave [3]. At a measurement point located axially, all sound waves will have to travel approximately the same distance, and arrive at the same time (with the same phase). However, as we move the measurement point off-axis, the sound paths will differ, and phase shifts – and thus cancellations and beaming – will occur. At low frequencies (= long wave-length), the travel path differences are relatively small and the beaming is less pronounced. However, as the wavelength becomes smaller with rising frequency ($\lambda = c / f$), already small differences in path-length (e.g. 5 cm) give rise to a noticeable phase-shift (elaborated in [3]). Consequently, the loudspeaker will radiate without beaming (spherically) in the low-frequency range, but as the frequency rises, so will the beaming effect. Usually, the frequency with a wavelength just fitting into the circumference of the loudspeaker is taken as limit from which beaming occurs. For an effective diameter of 27 cm, this results in f_g = 400 Hz. A 12"-speaker therefore features *approximately* (!) two different radiation characteristics: without beaming below 400 Hz, and above 400 Hz a frequency-proportional beaming. So much for the simple piston diaphragm theory, anyway.

Measurements with lasers (Chapter 11.3), however, show that the membrane already "breaks up" (i.e. it fails to keep its shape) upwards of 350 Hz. Therefore the piston-diaphragm theory also breaks: it breaks down, though. To formulate this more obligingly: from 350 Hz, we leave the range of validity of the piston-diaphragm theory. Now, it is simple to shoot down a theory but much harder to present a better theory instead. Of course, there are powerful formula the global significance of which can hardly be shaken, e.g. $\mathrm{rot}(\boldsymbol{v}) = 0$. Given the (location-dependent) membrane velocity, we may – now already more specifically – formulate the radiated wave as an integral that can be solved at least numerically. In approximation, that is, without saying. However, *one* differential equation won't do the job because the pattern of partial vibrations on the membrane may strongly change already with small frequency variations (e.g. +5 Hz). Also, to put together a directional diagram, the solution is required not only for *one* point in space. Because numerical algorithms for calculating the sound radiation are effortful (and require even more effort in corresponding measurements), the approach using purely metrology can still hold its own next to analytical descriptions. So let's go ahead, and let's measure frequency responses in various directions, put together polar diagrams for various frequencies, and determine frequency dependent directional indices in the AEC or the RC. The following characterizations use the piston diaphragm theory as a basis and compare its teachings with measurement results.

The directional gain Γ of the piston diaphragm is calculated from the Bessel-function J_1:

$$\Gamma(k,\Theta) = 2 \cdot J_1[ka\sin(\Theta)] \,/\, ka\sin(\Theta) \qquad \text{Directional gain [3]}$$

Γ is dependent on the wave-number $k = \omega/c$, on the effective membrane radius a, and on the angle Θ defined relative to the loudspeaker axis. The logarithm (with the base 20) of the directional gain is the **directional index** D. For low frequencies, D is approximately zero, as the frequency rises or as the angle Θ increases, D becomes negative. The left-hand section of **Fig. 11.38** shows the directional index, the right-hand section shows the directivity. Directional indices are bi-variant quantities; they depend on frequency and angle. To obtain the directivity, the envelope integral is calculated ("averaged") across all angles – only a frequency-dependency remains. Since the theory of the piston diaphragm is based on an infinite baffle, sound is only radiated into one half-space – and thus d = 3 dB at low frequencies.

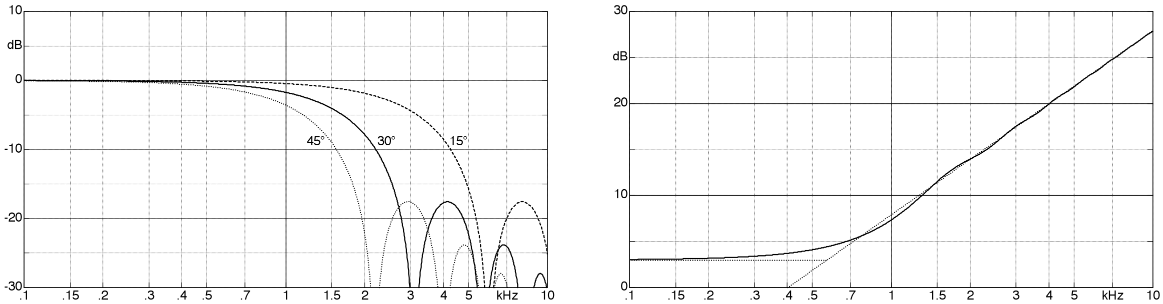


Fig.11.38: Directional index D of the piston diaphragm, a = 13.5 cm, Θ = 15°, 30°, 45°. Right: directivity d.

So much for our (simple) theory – how do measurements in the anechoic chamber compare? For that assessment, a 12"-Celestion-speaker (G12-M) was mounted in a small sealed enclosure (39x39x25 cm^3), and measurements of the SPL were taken at 0° and 35° (**Fig. 11.39**). Easily recognizable is how nicely the curves run in sync up to about 150 Hz – from then on the 35°-curve increasingly deviates from the 0°-curve. However, it is also clearly evident that this deviation corresponds only with a very coarse approximation to the piston-diaphragm theory. In the right-hand section of the figure, the calculated directivity for 35° is included (dashed line) – the curves do take a rather different course. Particularly evident: the figure holds three measurements taken with the enclosure turned by ±90° around the speaker axis (as indicated by the small sketch). We would expect rotationally symmetrical behavior from a single speaker, requiring the membrane to vibrate exclusively in rotationally symmetrical fashion. Which in fact it does – but not exclusively, as shown in Fig. 11.6. In particular in the high-frequency range, a multitude of complex modes occurs that certainly are not all rotationally symmetric. The radiation behavior is correspondingly complex.

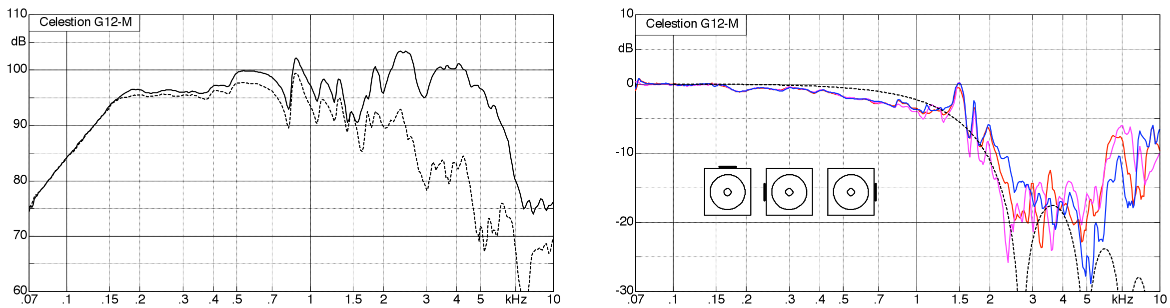


Fig. 11.39: 12"-loudspeker measured in the AEC, with 0° (——) and 35° (---). Right: directivity.

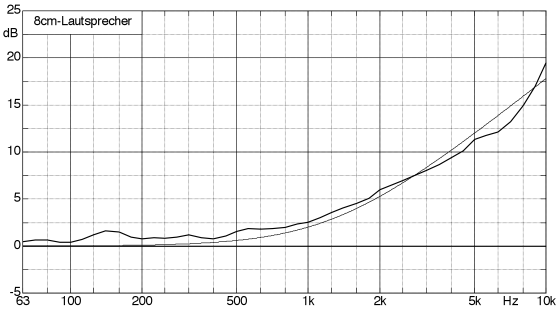
The directional characteristic found in Fig. 11.39 clearly deviates from that of an ideal piston diaphragm. Still, it should not be inferred that only the sound radiated off-axis is going to buck the theory; some effects (e.g. at 1.5 kHz, but also in the high treble range) originate in the axial SPL that for the ideal piston diaphragm should be frequency-independent. Evidently, this is not the case, and for that reason alone the directivity deviates from the nominal curve.

Why in fact is the directional characteristic of the radiation behavior that important? An often-heard comment is that most of the listeners are seated in front of the loudspeaker, and therefore the sound radiated to the side would be insignificant. Well, it is significant, because in the listening room (or hall), the sound radiated off-axis will be reflected by floor, ceiling and walls, and it will reach – as room sound – the ears of the listener with only little delay. It is impossible to exactly describe all individual reflections in a real room because already simple objects (chairs, lamps) feature a highly complex reflection behavior. That's why we make do with the directivity. It is quite useful as an approximation: a high directivity means much direct sound and little room sound. Sure: that room … its special absorber-distribution ... the position of the listener … and much more. Still, we need to simplify in order to push forward to the essentials. When operating two loudspeakers with significantly different directivity, the above statement holds as a simplification: more beaming = less room sound.

Like the directional index, the directivity d is calculated using the first-order Bessel-function (J_1). Approximately, d rises at a rate of 20dB/dec above the cutoff frequency, with the latter being defined by its wavelength $\lambda \,\hat{=}\,$ effective membrane-circumference (12" → 400Hz).

$$d = 10 \cdot \lg \frac{(ka)^2}{1 - J_1(2ka)/ka} \mathrm{dB}$$ Directivity [3]

The larger the membrane is, the lower the frequency where the beaming starts: a 15"-speaker has stronger beaming than a 10"-spekaer, but four 10"-speakers have a more pronounced beaming than a 15"-speaker because the effective membrane area of the former quartet is larger than the membrane area of the latter. **Fig. 11.40** juxtaposes theory and measurement results. As already mentioned, measuring the directivity is difficult because the "artifacts" encountered in reverberation chamber and anechoic chamber can add up. However, if we do not regard the directivity as a system-immanent quantity (which in fact it is not, anyway) but as relating to the environment, then the measurements become sufficiently reliable, and even a negative directivity appears purposeful: at low frequencies, the loudspeaker positioned in the reverberation chamber has a higher efficiency compared to the positioning in the anechoic full-space (Chapter 11.5). If we do not attribute any significance to differences as small as up to 1 dB, the basic curve can be interpreted nicely, especially when comparing several loudspeakers measured in the same room.



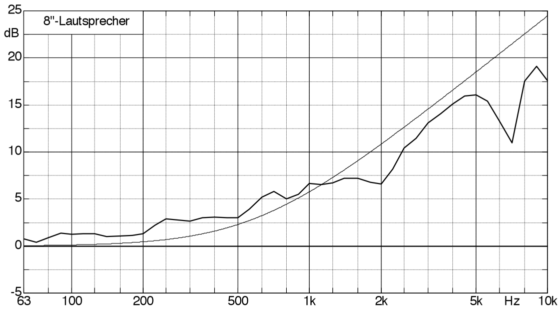


Fig. 11.40: Beaming for an 8cm- and an 8"-loudspeaker. Measurement (—), simple model-calculation (—).

The rather bad match between measurement and theory seen in Fig. 11.40 for the 8”-speaker is not likely to be a result of unsuitable instrumentation: the theory simply does not fit the speaker – in the higher frequency range, the membrane is not vibrating anymore without a change in shape. The beaming-minimum at 7 kHz has its basis in a destructive interference that leads to a minimum in the *axial* radiation. Half the wavelength amounts to a mere 2.5 cm at this frequency, and cancellations are easily conceivable. The off-axis radiation is not subject to this interference, and that leads to the effect of a minimum in the directivity. The latter does depend on two quantities: on the direct sound, and on the diffuse (room) sound. Consequently, a minimum in the directivity may be obtained via two ways: by efficient radiation of diffuse sound, or by inefficient radiation of the direct sound.

With **Fig. 11.41**, we return to the 12”-speaker that was already used for most of the previously presented measurements: the Celestion G12-M. The left-hand picture shows measurements with a small sealed enclosure. Up to 1 kHz, the beaming is somewhat stronger than calculated using the simple theory – that may be due to the enclosure: at 39cm x 39cm, the front panel is not actually infinite but already larger than the effective membrane diameter (27 cm). The curve above 1 kHz cannot be clearly attributed anymore to anomalies of *a single* sound field: both direct- and diffuse-sound deviate significantly from the simple piston diaphragm theory.

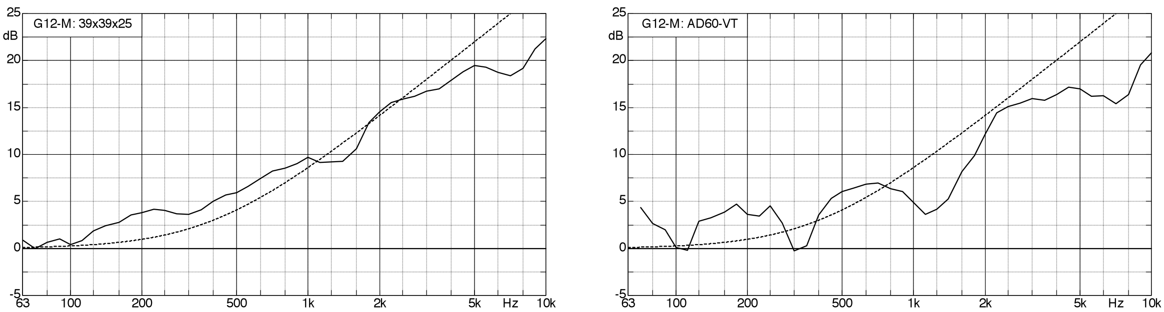


Fig. 11.41: Frequency response of the directivity: G12-M, mounted in two different enclosures.

For the right-hand graph in Fig. 11.41, the G12-M was mounted in the open-back VOX-cabinet already used in Chapter 11.3. This cabinet shows everything but a dipole-characteristic! As main effect, we recognize two beaming-minima (350 Hz, 1.2 kHz) on the one hand, and on the other hand a global widening of the treble-reproduction (reduction of beaming). Given the high-frequency beaming of the loudspeaker, it will not make a difference for the on-axis AEC-measurement whether the rear panel is open or closed. For measurements in the RC, however, a difference will show because the same amount of power is radiated from the rear of the speaker (in idealized thinking: level of diffuse sound +3dB). At low and middle frequencies the superposition of the sound waves radiated from the front and from the rear leads to comb-filter-like ripples in the directivity. Again, it is predominantly the sound radiated to the front that forces the shape of the frequency response in the beaming: the minima at 350 Hz and 1.2 kHz are found with axial AEC-measurements, as well – as e.g. Fig. 11.24 shows for all measured Celestion-speakers

Shape and type of the cabinet contribute significantly to the loudspeaker-sound. That also holds for HiFi speaker arrangements, but here the direct SPL should be as much as possible frequency-independent, and the directivity should rise evenly across the frequency such that in the end the speaker will sound good (i.e. neutrally) despite the enclosure-specifics. Conversely, for the guitar speaker the cabinet provides a distinct filter; its directionality cannot be changed electronically.

How dominant the influence of the cabinet is in comparison to variations of the loudspeaker may be seen from **Fig. 11.42** – it includes the directivities of several Celestion speakers all mounted in the VOX AD60-VT-cabinet. At first glance all the curves are of very similar shape – at second glance one speaker is strikingly different at 8 kHz: it is the Celestion "Blue". After what has been stated above regarding that speaker, at last we have an objectifiable rationale in favor of this speaker … possibly a late satisfaction for all those still paying off the debts caused by that speaker. Of course, we shall not even start questioning how significant the frequency range in question actually is (☺ Abb. 11.25).

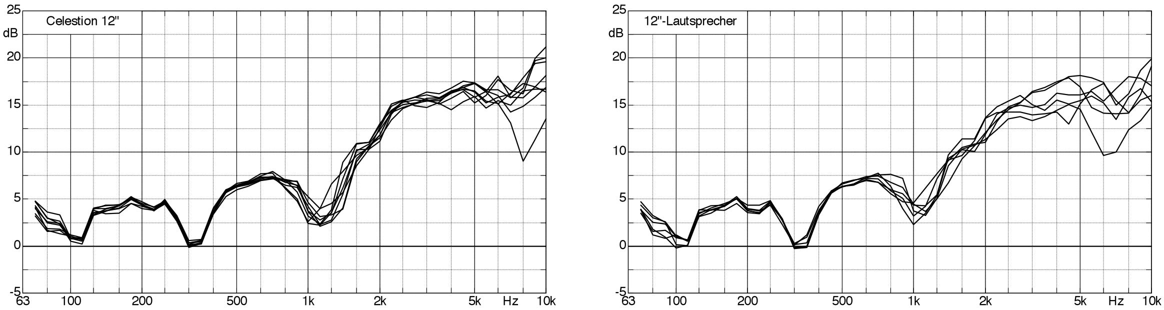


Fig. 11.42: Frequency responses of the directivity for various 12"-loudspeakers: Celestion, Jensen, Eminence.

In the right-hand section of the figure we find a few loudspeakers with a more strongly differing directivity: Jensen and Eminence. The general reason is quickly identified: their membranes show more diversity than those of the Celestion speakers: size of dust cap, corrugations, depth of membrane, diameter of the voice coil. Still, the main effect is caused by the enclosure; the opening in the rear takes care of characteristic beaming-minima. A directivity of 0 dB is often interpreted as **spherical radiation** although this is not always applicable. The degree of beaming (or beaming factor) relates the intensity radiated in the axial direction to the averaged intensity radiated in all directions [3]. If – due to an interference-cancellation (pole) – no sound is radiated axially, the beaming-factor is zero and the directivity is $-\infty$. If axially only little sound is radiated but in all other directions beaming occurs, d = 0 dB may result – despite the fact that there is no spherical characteristic.

Directional diagrams give clues regarding the direction-dependency of sound radiation. In corresponding measurement setups, the object to be measured rotates by 360° on a revolving table, and the SPL is registered dependent on the rotation angle. The resulting diagram is usually laid out using polar coordinates. **Fig.11.43** exemplifies 3 directional diagrams measured with the AD60-VT. None of the diagrams shows the dipole-typical radiation pattern – this is due to the cabinet acting as a phase-shifting filter for the wave emitted to the rear.

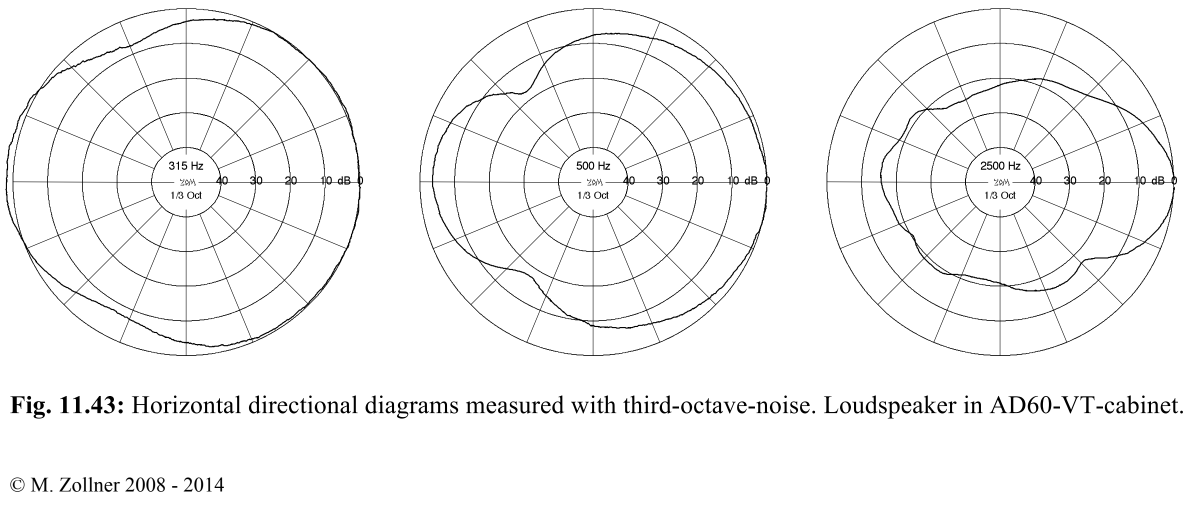


Fig. 11.43: Horizontal directional diagrams measured with third-octave-noise. Loudspeaker in AD60-VT-cabinet.

Directional diagrams have descriptive qualities but can only exemplify one single plane – as such they have limitations. For a circular membrane, often a rotation-symmetric radiation is implied, coupled to the hope that a single measurement (per frequency!) will be sufficient. Often, this is a reasonable approach, but just to be safe we should take additional measurements. **Fig. 11.44** shows horizontal directional diagrams – in contrast to Fig. 11.43, a sinusoidal test-signal was used, though. At 400 Hz a perfect symmetry exists, while at higher frequencies, any asymmetric shape may occur due to membrane resonances. Since these shapes are highly dependent on frequency, the information contained in directional diagrams needs to be drastically reduced in order to remain clear. Therefore, noise (of octave- or 1/3rd-octave bandwidth) is often employed as test signal – this has the effect of an averaging across the corresponding frequency interval. Using that approach, the small variations contained in directional diagrams are not an expression of high directional selectivity but the result of stochastic processes. Given an optimized averaging time-constant, misinterpretations are not to be expected – if necessary, fluctuations can be reduced via averaging over several turns of the rotational table.

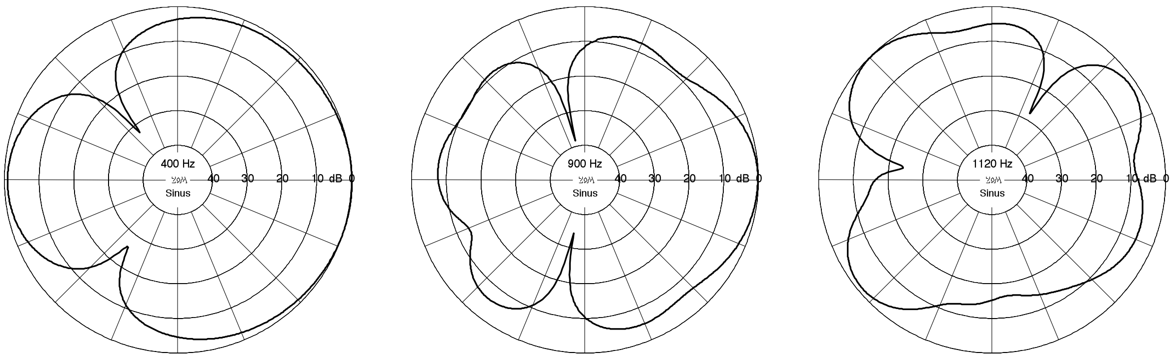


Fig. 11.44: Horizontal directional diagrams, measured with a sinusoidal signal. 12"-speaker, AD60-VT-cabinet.

Installed in a sealed cabinet, a loudspeaker will operate as a spherical source; with a rear opening in the cabinet, a dipole will result. However, the stiffness of the air contained in the cabinet forms, in conjunction with the inert (mass-dominated) radiation impedance of the opening, an acoustic filter creating phase-shifts, and therefore the directional diagrams have the shape of a (logarithmized!) eight only at very low frequencies. Already at 200 Hz, this dipole-behavior is all but gone, and the horizontal directional diagram approaches a circular shape. In **Fig. 11.45** we find a comparison between the original VOX and a variant where the rear was closed off with a board. The latter does not provide a complete seal, however: the slits foreseen to provide ventilation for the amplifier section let sound pass through. Horizontal directional diagrams for the AD60-VT cabinet with open and closed rear wall are juxtaposed in **Fig. 11.46** (measured in the AEC using 1/3rd-octave noise).

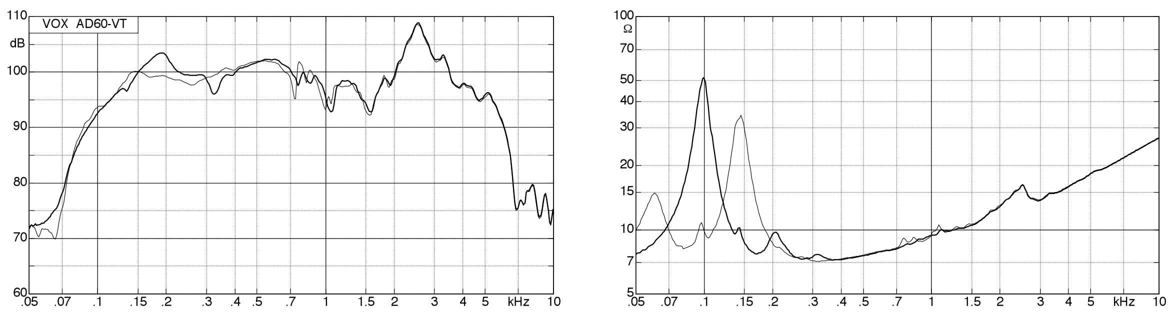


Fig. 11.45: VOX AD60-VT: rear wall closed with board (–––) vs. original condition (——).

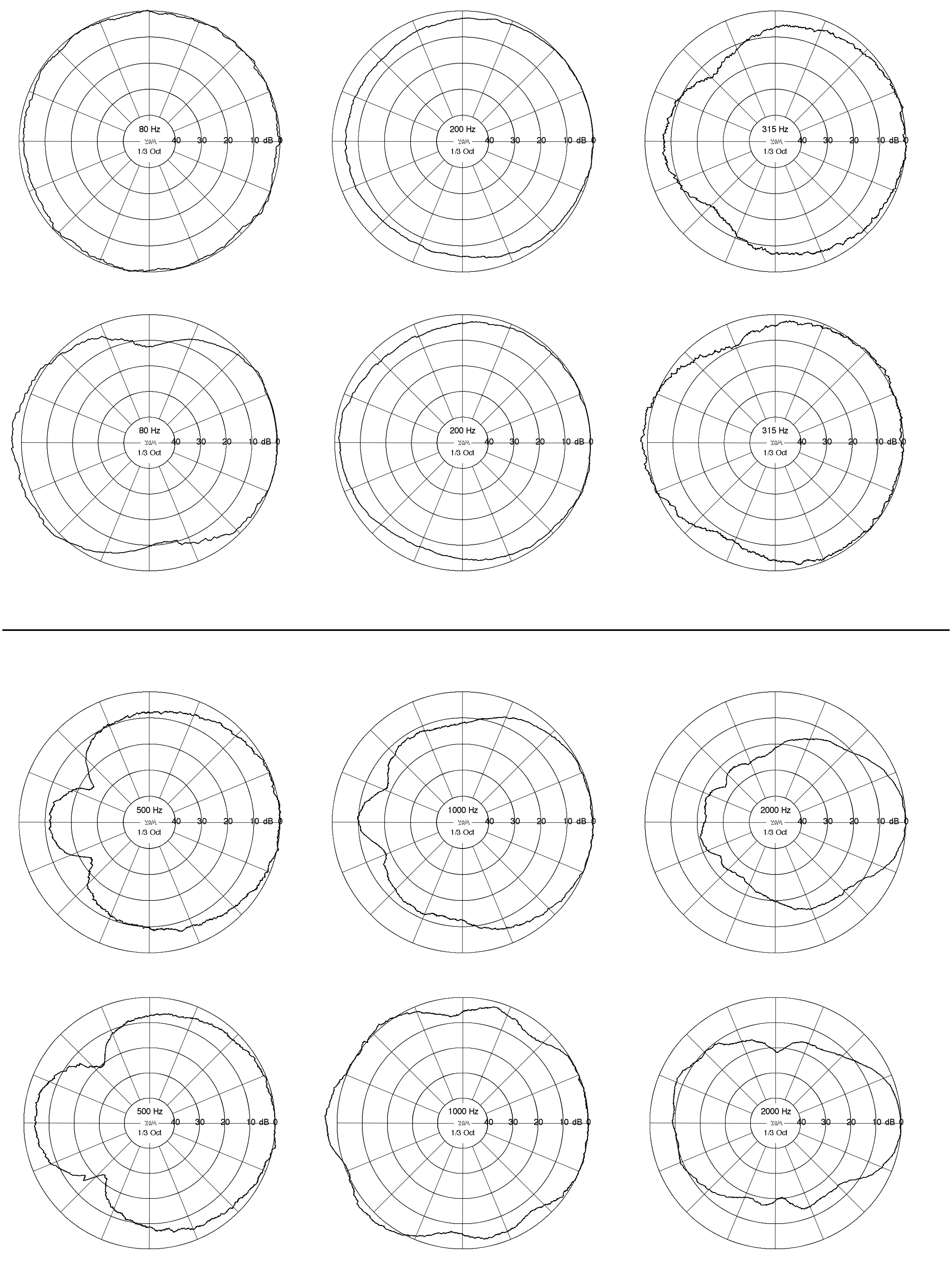


Fig. 11.46: Horizontal directional diagrams with sealed (1st and 3rd line) and open rear wall of the AF60-VT. The dipole-characteristic begins to become visible at 80 Hz, while at 200 Hz barely any differences are visible. In same ranges, more sound is radiated to the rear than to the front (e.g. at 315 Hz); this is caused by an impedance-transformation (an effect of the cabinet cavity).

Fig. 11.47 shows how strongly the cabinet influences the sound-beaming. All measurements were done using the same cabinet with or without a rear panel (i.e. closed resp. open). For both variants, the 10”-speaker features less beaming in the high-frequency range – however, the cabinet-specific differences far outweigh the loudspeaker/diameter-specific differences. The directivity is negative at 315 Hz – this again is due to the direct sound radiated to the front and showing an interference minimum (rear-ward diffraction wave) at that frequency.

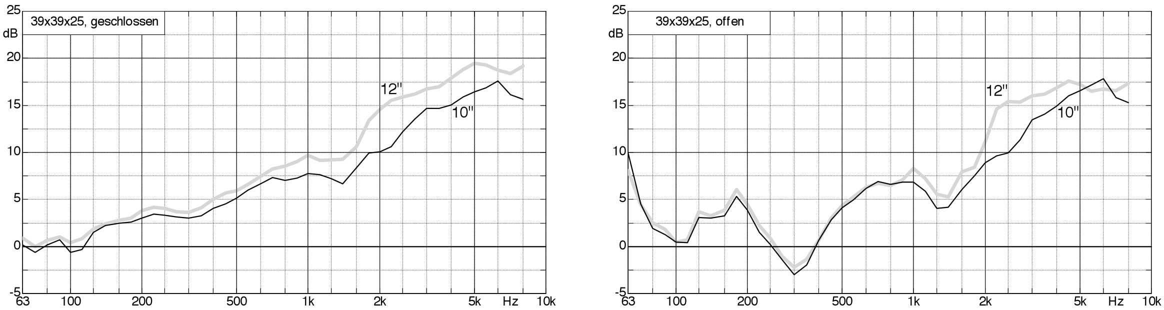


Fig. 11.47: Comparison of the directivity of the closed and the open cabinet; G12-M vs. P10-R.

If two or more loudspeakers are mounted in a cabinet, the beaming increases because the membrane area grows. Corresponding measurements that support this general statement are shown in **Fig. 11.48**. Differences are visible in the details, though: first, the enclosure shapes are different, and second, the sound power radiated to the rear is loudspeaker-specific. Finally, **Fig. 11.49** presents the directivity of loudspeaker cabinets that are designed to reproduce the whole frequency range relevant for music transmission (so-called “full-range” speakers). Their directivity should increase as evenly as possible – this is achieved quite well in the Quinto.

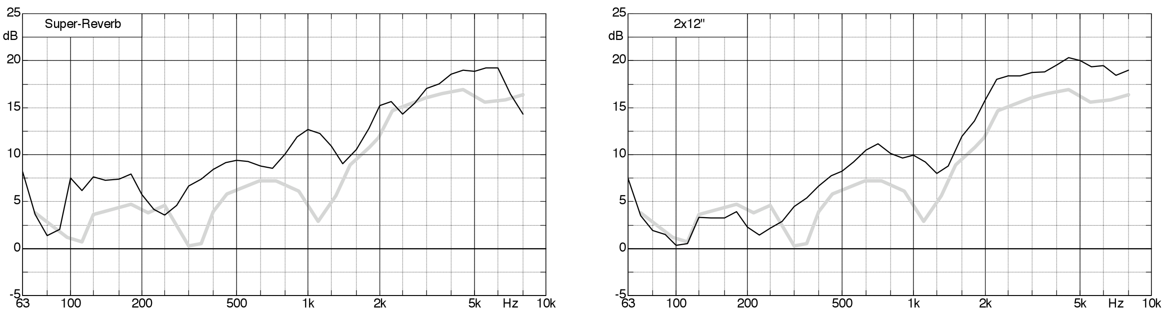


Fig. 11.48: Left: Fender Super-Reverb, 4x10", Jensen P10-R; right: typical 2x12"-Box, Celestion G12-M.
Grey curve = VOX AD60-VT (1x12") for comparison. The directivity is given as a function of frequency.

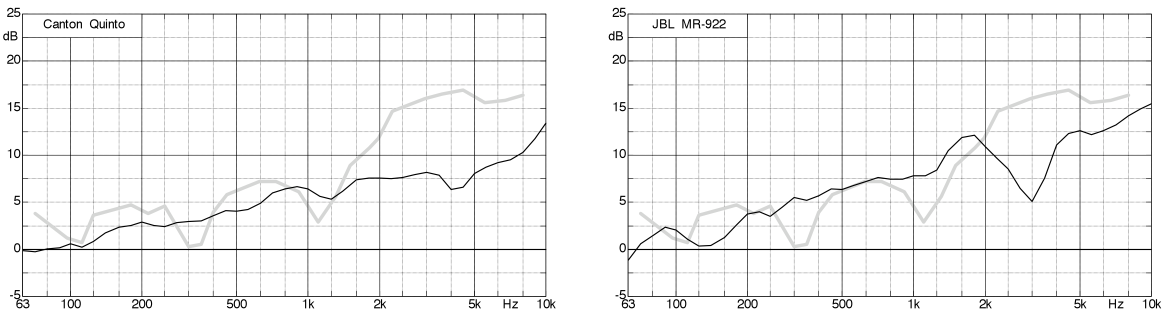


Fig. 11.49: Directivity of a HiFi-Box (left) and a small full-range-box for stage use (right).
Grey curve = VOX AD60-VT (1x12") for comparison. The directivity is given as a function of frequency.

The area-rich **15”-loudspeakers** should actually show particularly strong beaming effects – but measurements support this hypothesis only in part (**Fig. 11.50**). For both the Fane and the Powercell, the directivity decreases again in the highest frequency range. Possibly, this is connected to the large dust-caps of these speakers: both sport air-tight dust-caps acting as high-frequency emitters with a diameter of naturally not 38 cm but merely 10 cm. However, no further measurements regarding this hypothesis were conducted.

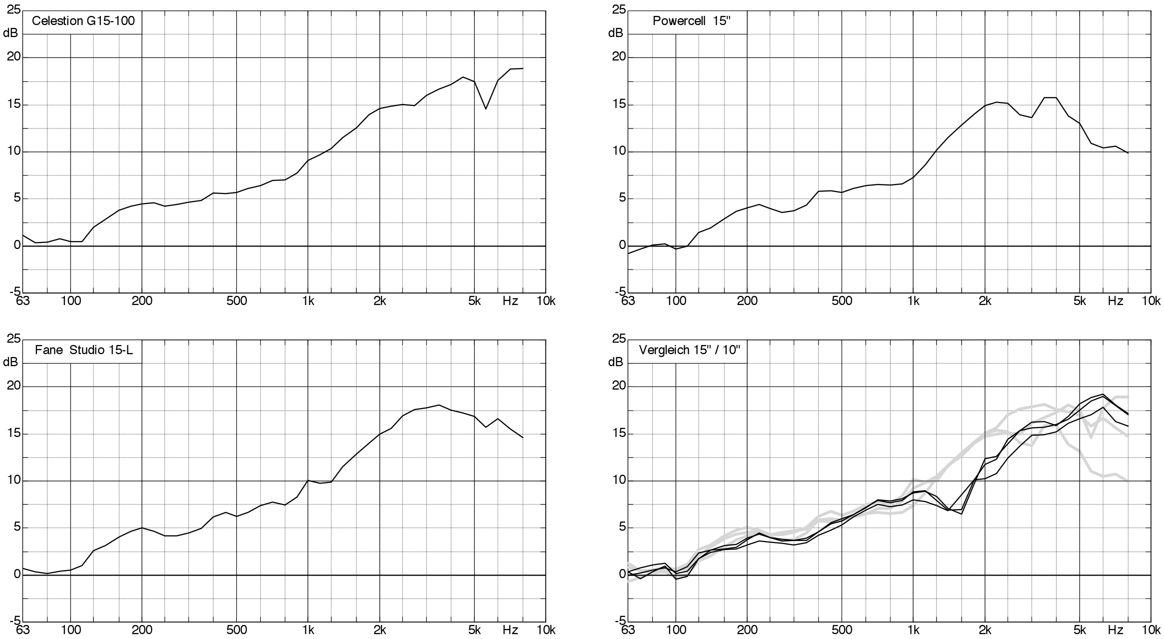


Fig. 11.50: Directivity of 15"-loudspeakers. Lower right: comparison 15” vs. 10” speaker.

The last section in Fig. 11.50 depicts a comparison between 15”- and **10”-loudspeakers**. Below 1 kHz, differences are rather limited; only above this limit, things become more specific. According to the simple piston-diaphragm theory, we should see a difference of 3.5 dB. Prerequisite would, however, be an infinite baffle – but the measured loudspeakers were installed in airtight boxes of different sizes (10" ⇒ 39x39x25 cm^3, 15" ⇒ 40x74x36 cm^3).

As a last point, a special characteristic of 2x12”-combos shall be considered: almost always, the loudspeakers in the corresponding cabinets are mounted *horizontally next to each other,* conversely to the speakers in public address systems where the speaker chassis are mounted *vertically above each other*. The vertically aligned column has the advantage that the vertical beaming is increased (less sound to floor and ceiling), while horizontally a wide-angle radiation is retained. That the 2x12”-combo is realized exactly the other way ‘round may be the result of a desired visual look and feel, but also a necessity required by the amplifier-chassis: you need quite a bit of space to line up 11 knobs (like in the Twin-Reverb). Fender’s first foray into tall cabinets and two rows of control knobs (1967, in the first Solid-State amplifier series) pretty much was a disaster (possibly not merely due to the geometric configuration…): comments included the terms “ugly refrigerators” or “TV-trays”. With regard to beaming and sound dispersion, the tall, slender design would certainly have had advantages. (Translator’s note: maybe the more vertically oriented sound dispersion is something favoured by the musician standing in front of the amp? Especially when using the “tilt-back” legs customary in these amps, the vertical spreading out of the sound makes it less crucial at what distance the guitarist stands relative to the amp, and how tall he/she is. Also, more sound is aimed at the guitarist when playing – too? – loud …)

11.5 Efficiency and maximum sound pressure level

Is a 100-W-speaker twice as loud as a 50-W-speaker? That question is asked a lot, and has at its basis a common misunderstanding. The Watt-specification of a loudspeaker only tells us about the maximum power the speaker can take, but includes no statement at all about the acoustical power-yield. Even if you just mount four 100-W-lightbulbs into an enclosure, you may label it with “400-W-Box” – but you won’t get any sound out of it.

Strictly speaking, we would have to distinguish the power fed to a speaker into effective power, reactive power, and apparent power. However, in practice that is simplified: every speaker is to be assigned, by the manufacturer, a **nominal impedance** R (e.g. 16 Ω), and, together with the maximum power P, the **maximum voltage** is derived: $U = \sqrt{P \cdot R}$. A 16-Ω-speaker with a maximum power rating of 100 W may be driven by an RMS-voltage of 40 V. Some limitations need to be observed here: a DC-voltage of 40 V may not be connected to the speaker, although again 100 W would be the result – however the speaker would be destroyed by this “drive signal” (the manufacturers do not specify any maximum DC-voltage at all). A typical source material would be guitar-tones, but this signal definition is too general. As a compromise, specially filtered noise-signals are often chosen, e.g. the **EIA-noise** (RS-426-A, RS-426-B), or the **IEC**-268-1-noise, or the **AES**-2-1984-noise, or the **DIN**-45573-noise, or other specifically defined signals. These are then (depending on the specification) to impact for 8 or 100 or 300 hours on the speaker without destroying it. If the loudspeaker can take e.g. 100 W according to such a standard, the sales department labels it with “100 W”. Or with “200 W”, because there may be further considerations: since, allegedly, the power load is much smaller in practice, a "CONTINUOUS PROGRAM POWER" was defined. This is a power specification 100% above the limit-power data determined with the noise. We can see: power-data are manufacturer-specific; they may not simply be grasped via U=RI und P=UI. That’s similar to the area of power amps: at the Frankfurt music fair, a French manufacturer answered – slightly irritated – to the question why his 90-W-specified amplifier would deliver no more than 55 W: “that’s French Watts”. Ah oui, monsieur, bien sur.

The **nominal impedance** is not something the knowledge-seeking person will readily understand at first glance, either. Is it the DC resistance, or the minimum- or the maximum-impedance? It’s none of these three. The impedance $Z(f)$, i.e. the magnitude of the complex resistance, for a loudspeaker depends strongly on the frequency: at 0 Hz it may e.g. amount to 6.5 Ω, at resonance (at 110 Hz) it may rise to e.g. 75 Ω, at 300 Hz, it may almost be back to 6.5 Ω again, and it will rise continuously towards higher frequencies♣ (**Fig. 11.51**). This curve cannot be specified via a single value, and so the manufacturers choose a (another?) method to arrive at *one* value. For example, the value of the impedance at 1 kHz is measured. Why is that 1 kHz? Because that’s an often-used standard-frequency. Or 800 Hz may be employed … because the recommended crossover frequency is here. Or 400 Hz: you may want to set yourself apart from the competition that way. Or the speaker is labeled right away with “Impedance: 4 - 8 Ω”. No, that doesn’t mean that the speaker features an impedance of between 4 and 8 Ω. Rather, the speaker is recommended for amplifiers the manufacturers of which on their part recommend using speakers of 4 or 8 impedance. Well then. Given all this, it comes as no surprise that the guys at Just Barely Loud frankly admit: "The JBL 2215B Professional Series Loudspeaker is rated at 16 Ω, while the LE15A Home Loudspeaker, *which is the same unit*, carries an 8-Ω-rating". Thanks a lot, then: both allowable maximum power and impedance are now precisely defined, and everybody can calculate from these values the allowable maximum voltage. In case the speaker starts to communicate via smoke signals, JBL recommends: Turn it down!

♣ For enclosure- and membrane-resonances, see Chapters 11.3 and 11.8.

When does a loudspeaker actually cross the River Styx? The most frequent reasons for malfunction are too high a voice-coil temperature (excessive effective power), or too wide a membrane displacement. Both these effects may influence each other: a strong membrane displacement may increase the cooling of the voice coil and push the power limit a bit further. Since, for a drive signal from a high impedance source (stiff current source), the excursion drops off with $1/f^2$ above the resonance frequency, large displacements are only found at low frequencies – that is one reason why the resonance of guitar speaker is not located at 20 Hz but at 80 – 110 Hz. Another reason is the fact that as guitar player, you do not want to get in the way of your bass player – that's the guy who owns the low end (not necessarily implying that guitar players are generally to be seen as belonging to the High-End-range).

For the musician, it will generally not make any difference why exactly the speaker died after the volume was cranked up from "5" to "10". *Had* to be cranked because otherwise the guitar would have been drowned out (by the keys that just went to "10", as well). Now the speaker is kaput – overloaded, as the roadie knowingly attests. That happens if the amplifier delivers more power than the speaker can take. So how much power can the speaker take? We've been there – see above. Other question: how much power does the amp in fact deliver? We should be able to at least measure that value with adequate accuracy, shouldn't we? In principle: yes … but: guitar amplifiers often dispense with (strong) negative feedback, and a power specification at e.g. 1%-THD does not make much sense. Rather, the amplification is turned up until visible clipping sets in, and from this a power-value is calculated. Maybe happily using 1 kHz, and gladly at the nominal impedance. The power that the amp can feed to a real loudspeaker, and in particular what it can generate under overdrive conditions – that remains unknown. And so we read statements from the service technician testifying that he never saw a Marshall 1959 that had a mere 100 W: it always was 140 W, or even 160 W. On the other hand, the question does pop up how an AC-30 with its continuously-under-overload power amplifier can generate 30 W if a quartet of EL-84's is specified at no more than 24 W. Let's jot this down: both the generated amplifier power, and the power capacity of a loudspeaker could be measured with good accuracy – but the market has found its own standards that "not always" coincide with the norms in metrology.

Ah - the market: that is the key to understanding. Fender's Pro-Reverb sported 40 W, so that's 5 W more than the Vibrolux. At the end of the 1960's, Celestion's G-12-speaker received the urgently expected power-upgrade from 25 to 30 W. Grown up, at last! You will recognize similarities to the car-market: isn't the 220 something entirely different compared to the 219?! On the one hand, there are classifying power-ratings that portray a 10%-difference as relevant – but on the other hand differences of 50% or more seem to be subject to pure arbitrariness. It is difficult to avoid the impression that the head of sales – just before the big music fair – quickly checks repair-statistics, and if the 12-50 has next to no failures, that speaker receives a red cover and mutates into the 12-65-S. To cite Cicero: O tempora, o mores (liberally translated: where there's a market, there's a way). No, this is not meant to say that power-upgrades happen only in the sales brochures: from the 12 W of the first 1,25"-voice-coil-carrier made of paper to the 200-W-3"-polyimid-carrier, there has been indeed a mighty development. Individual cases need to be scrutinized, however: the Vintage-30 (12", 60 W) is specified at 100 dB "average sensitivity", the Powercell 12-150 (12", 150 W) at 94 dB. Attention: "6 dB less" means that at the same power input, only ¼ of the sound power is generated. For the same sound power, the Powercell would require an input of 240 W. That is beyond its power limit – so better buy two of the guys. Powercell? Rather, it's Powersell!

Let us remain for a moment with the term "**average sensitivity**". There is – and that is not the norm for the business – consensus that this specifies the SPL that can be generated at a distance of 1 m with 1 W electrical power. However: this one Watt is not actually generated, rather a voltage is applied to the loudspeaker that would create 1 W at the real nominal impedance (for 16 Ω that would be 4 V_{eff}). If the speaker actually has 12 Ω rather than 16 Ω, that alone will result in the gift of another 1.25 dB for the specification listing – in the brochure, a measly 99 dB is turned into some stately 100 dB that way. Also, across which frequency range the averaging happens has, in case of doubt, a company-specific definition.

Let's let a manufacturer have a say: *The Sensitivity represents one of the most useful specifications published for any transducer. It is a representation of the efficiency and volume you can expect from a device relative to the input power*. Well said – that had to be defined for once. However, the text continues with: *Loudspeaker manufacturers follow different rules when obtaining this information – there is not an exact standard accepted by the industry*. Okay then… We can leave the world of datasheets for a bit and look into what theoretical **electro-acoustics** have to offer. A spherical source generating a sound pressure of 100 dB at a distance of 1m produces a sound power of about 126 mW [3]. Guitar loudspeakers reach these 100 dB @ 1m already with about 1 W power input; the efficiency therefore would be 12.6% – if indeed the radiation were spherical. In the relevant frequency range, however, on the one hand the beaming effects need to be counted in, but on the other hand many loudspeakers exceed 100 dB @ 1m, so that overall we find efficiencies of about 10% to be the approximate limit for the single membrane-loudspeaker. HiFi-speakers often reach only 0.1% whereas horn-speakers can achieve more than 25% efficiency[♣]. Thus only the smaller part of the input power is converted into sound, the larger part ends up as **heat**. No wonder that voice coils can be destroyed if from the 100 W input power, more than 90 W dedicated themselves to heat up the thin wire. As is generally known, a soldering iron of a mere 30 W generates a lot of heat; the voice coil therefore needs to be able to bear substantial strain. At full power, 200°C or more will occur; only special materials can withstand that. To decrease the temperature, there are only two possibilities: turn it down, or increase the heat-dissipation. The former approach would be up to the musician, the latter is the manufacturer's area (constructional build of the pole-pieces carrying the magnetic field, broadening of the pole-plate, pole-piece vents, etc.).

We carried out **measurements** with a number of guitar loudspeakers to obtain more precise date regarding efficiency. The instrumentation used was of high precision while the measuring rooms were somewhat more limited in that respect. The fiberglass wedges of 80 cm length in the available anechoic chamber (**AEC**) will absorb 100 Hz to a sufficient degree; disturbing room resonances will occur below this limit. With 220 m^3, the reverberation chamber (**RC**) is large enough but still sub-optimal (due to a lack of diffusers and because of unsuitable installations). The results presented in the following therefore may not generally claim an accuracy of ± 1dB, but they are still well usable to arrive at statements for orientation. Measurements in the AEC (B&K 4190) were done at a distance of 3 m to the baffle but were re-calculated for 1 m distance to make them better comparable: $L_{1m} = L_{3m} + 9.5$ dB. For sweep-measurements, the input voltage was 2.83 V_{eff} from a stiff voltage source, for 1/3rd-octave measurements, the voltage per 1/3rd-octave was kept constant (pink noise + 1/3rd-octave-filtering). Polar diagrams were taken in the AEC with 1/3rd-octave noise, revolving table B&K 3922, d = 3 m. In the RC (B&K 4135), measurements were carried out following a skewed circular path (∅ = 3 m), along which energy-averaging was performed. Most RC-measurements were done with 50%-overlapping 1/3rd -octave pink noise (IEC 1260 class 0); $U_{1/3rd\text{-}octave} = 0.5\ V_{eff}$. Employed as analysis-software: CORTEX-Viper and Matlab.

[♣] H. Fleischer: Hörner endlicher Länge (horns of finite length), research report from the Institute for Mechanics, HSBw, 1994.

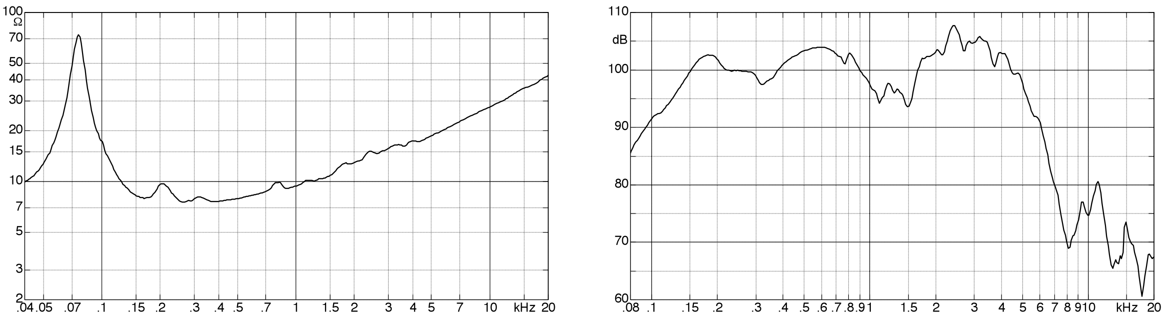


Fig. 11.51: Magnitude of impedance (left), SPL in the AEC (speaker mounted in VOX AD60-VT enclosure).

The loudspeaker analyzed in **Fig. 11.51** is a **Celestion Blue**, commonly termed "the legend" because it served in the famous early VOX-amplifiers, and the just as famous early Marshall cabinets. This speaker is said to have a fabulous efficiency that is – if you believe in statements on the Internet – due to the magnetic material (Alnico) deployed back in the day. And indeed: with 1 W as input, and at a distance of 1 m, this speaker generates up to 108 dB! Given far-field conditions, this results in an intensity of 63 mW/m^2, giving (with a sphere surface 12.6 m^2) 0.79 W sound-power and 79% efficiency. Indeed? Can that be?

Without question this is a fine loudspeaker, and it does have a high efficiency, but never 79%. At 2.5 kHz, we must not assume spherical radiation any more, so that the "efficiency" mentioned above needs to be multiplied by the beaming factor [3]. And while we are doing corrections: the real input power is not $P = U^2/R_{\text{nominal}}$, but results from the actual real part of the electrical impedance.

Let us first look at the **directional characteristic** (directional index, [3]): loudspeaker manufacturers publish (if they publish anything at all) the transmission frequency response "on axis". However, the loudspeaker radiates sound not only to the front but in all directions. This behavior is captured either via direction-dependent transmission factor, or via frequency-dependent directivities. That means: level plotted over frequency for various directions, or level plotted over direction for various frequencies (Chapter 11.4). If we insinuate rotation-symmetric sound-radiation, beaming measurements in *one* plane will suffice. In **Fig. 11.52** we see two directional diagrams from measurements of a combo cabinet, the rear wall of which has an opening of 49 cm x 21 cm. Against all expectations, an almost circular radiation pattern shows, and not the "eight" of a dipole (for details see below). At 2.5 kHz, however, we find typical high-frequency beaming – despite the opening on the rear.

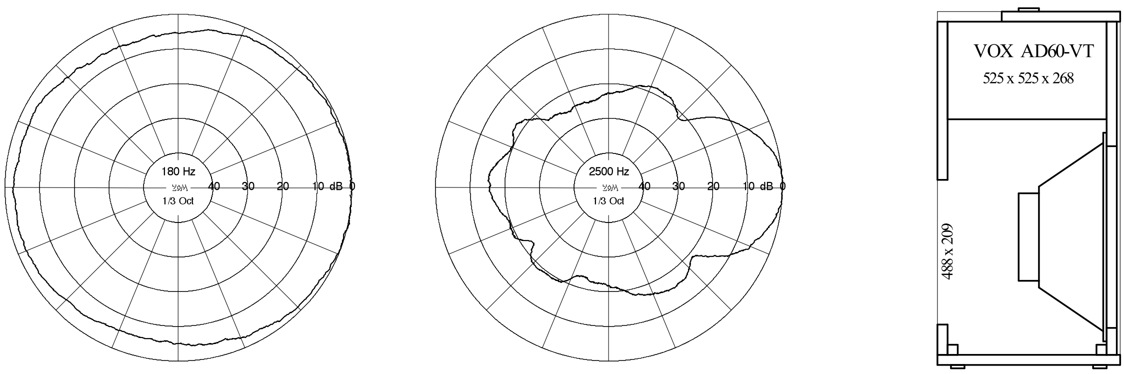


Fig. 11.52: Directional diagram in the horizontal at two different frequencies; VOX AD-60VT-cabinet.

The **efficiency** η is a quantity relative to the power: sound power / electrical power, or – more precisely – effective acoustical power P_{ak} / effective electrical power P_{el}. Operating the loudspeaker from a stiff voltage source, P_{el} is obtained via $P_{el} = U^2 / \mathrm{Re}(\underline{Z})$. It is not very difficult to determine the real part of the electrical impedance, but establishing the effective power P_{ak} emitted by the speaker does become complicated – and doubly so! The metrological investigation requires a substantial effort to begin with, and in addition this power P_{ak} is dependent on the environment of the loudspeaker, i.e. is not a constant. It's a bit like with a car: the engine may have a power output of 400 hp … but not on an icy road. The acoustic source-impedance of the membrane (defined as the quotient of sound pressure over particle velocity) is relatively high: the membrane could generate a high pressure, but only at a relatively small membrane velocity. The real part of the radiation impedance is, however, more on the low side: even for relatively high membrane velocity the forces transmitted to the air remain relatively small, and a considerable mismatch at the membrane results. *High/large* and *low/small* need to be seen task-specific; literature [e.g. 3] delivers supplementary data. The loudspeaker membrane is highly unchallenged in the typical mode of operation – just like a pitcher throwing a very small ball: whether that ball weighs 10 or 20 grams is immaterial, with the speed being approximately the same for both cases. The energy of the heavier ball will be twice that of the smaller one, the efficiency will be load-dependent. Applied to the loudspeaker: could we increase the load-impedance, the efficiency would increase, as well. The load-impedance can actually be increased by positioning the speaker enclosure directly on the floor, or even right away into a corner of the room – that increases the efficiency. Not without limit, of course, the velocity will drop with too high a load. Again there are parallels to the pitcher: a 5-kg-ball will not be able to have a higher speed than the 20-gram-ball.

Apparently it is not easy to determine the loudspeaker-efficiency – that may be the reason why the industry rarely publishes corresponding data. According to established theory, η may change by a factor of 8 (!) if the loudspeaker is taken out of the AEC and placed into a corner of a reflective room. Even if in practice the limits of the corresponding range are not reached – already a factor of 2 would represent considerable uncertainty. A way out of this dilemma is linked to comparative measurements in a special room: for example, 2 loudspeakers are measured in the AEC – however, the desired results are not so much their absolute efficiencies but the relation between the two. If we find, for example, a relationship of 5% to 3% in the AEC, a similar difference can be expected to occur in the real room. Measurements in the AEC deliver pretty accurate results but require considerable effort because of the non-spherical sound radiation that necessitates a high number of measuring points (or measuring paths). Moreover, imperfect absorption of the **absorber-wedges** in an AEC even at frequencies above 100 Hz needs to be considered. Therefore, there is still no perfect free-space field if we limit the measuring range to $f > 100$ Hz. In the available AEC, we measured level differences of ±1 dB up to 300 Hz as the positions of loudspeaker and measurement microphone were changed (axial measurement at $d = 3$ m). For the efficiency, a difference of only 2 dB represents a relative deviation of 58%, i.e. e.g. 8% instead of 5%. In addition, the instrumentation devices will have some tolerances; they may be still connected in spirit to Messrs. Brüel and Kjaer, and be of exemplary precision – but they will deviate a bit from the reference value, anyway. This author does have a bit of a queasy feeling when, after just mildly ridiculing the 35/40-W-differences in Fender amps, suddenly a measuring uncertainty of an ample 58% pops up. What the heck … other measuring rooms are not available, and things become even more inaccurate in the reverberation chamber. Seriously: of all the examined AEC-positions, the best possible was retained for all further measurements. Comparative statements can quite well made based on this situation, and above 300 Hz, the deviations already remain below ±0.5 dB. Also, this holds in general: any more precise measurement result is most welcome.

For **measurements in the AEC**, we assume that the sound wave emitted by the loudspeaker is not (or almost not) reflected anywhere; as it hits the glass-fiber wedges that constitute the borders of the room, the energy of the wave is (almost) completely transformed into heat. In this mode of operation, the **radiation impedance** (= the impedance loading the loudspeaker) may be calculated for a few idealizing cases [Beranek, Olsen, Zollner/Zwicker]. However, the loudspeaker is rarely used in such an environment – there are not that many occasions when the guitarist plays in an anechoic chamber. That does not mean that measurements in the AEC are without purpose; it's just that supplementary measurements in other rooms and, of course, listening experiments are desirable. In contrast to the walls of the AEC, regular walls do reflect the sound to a considerable extent. Sound waves (in fact an infinite number of them) return to the loudspeaker, and the membrane does not radiate anymore into a free sound field but has to work against the sound pressure of the reflections. Still, due to the fact that the membrane is not challenged anyway (see above), its movement is not weakened much by the returning sound but – if the involved phase shifts are advantageous – **the efficiency is increased**. In a real room, the loudspeaker can thus generate more sound power than in the AEC – but it may also be less depending on the circumstances, for example if the speaker is position at a pressure node.

At this point it is recommended to also take a look at the **electrical impedance**. The loudspeaker is a passive two-port, and changes in the load impedance should also change the input impedance. **Fig. 11.53** confirms that this indeed is the case – but only to a rather small degree♣. The straightforward reason: the efficiency of course influences the impedance transformation, as well. Or, more elementary: relative to the ohmic voice-coil impedance, the load impedance plays only a minor role. The relationship between magnitude and real part of the electrical impedance is depicted in the right-hand picture. The two curves more or less correspond at the extremes (the impedance is approximately real here), in between the real part is smaller than the magnitude – just as it need be with impedance functions.

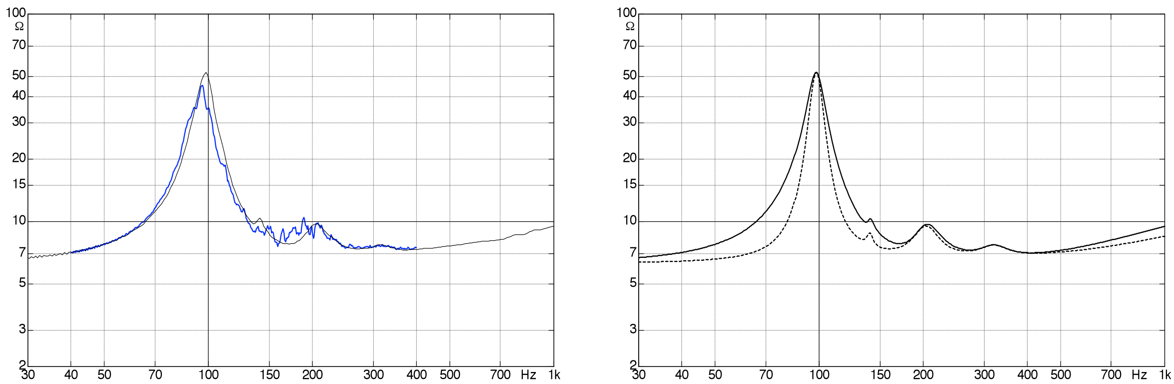


Fig. 11.53: Left: magnitude of the electrical loudspeaker impedance (AEC ——, RC ——). On the right, the comparison between magnitude (——) and real part (-----) of the electrical impedance is shown (AEC).

From these results, we may take the following **approximation**: as the acoustical environment of a loudspeaker changes, its input power remains almost unchanged; its power emission may, however, drastically change (this needs to be looked at some more).

♣ Again, a difference of 10% can easily occur here, but the focus shall remain with the main effects. Moreover, the differences are limited to the range below 200 Hz; above this limit, both curves coincide.

Now, on to the **reverberation chamber** (RC). In the ideal case, this is a room with strongly reflecting walls that lead to a **diffuse sound field** in the room (except for the space in close proximity of the sound source). This is a sound field in which the sound arrives at the measuring point from all directions with the same probability and in which the (averaged) sound pressure is independent of the location. The exception is the close-up range around the sound source, this range being delimited by the effective **reverberation radius** [3]. A typical reverberation radius would be 0.5 m (or less); the effective reverberation radius is calculated from it via a multiplication with the square root of the beaming factor (e.g. 0.5 m x 3 = 1.5 m).

To be a bit more precise: the free and the diffuse sound field superimpose within the whole of the reverberation chamber (which forms an LTI-system); close to the source, the free sound field is more dominant while further away the diffuse field takes over. Given spherical (non-beaming) radiation, the beaming factor is $\gamma = 1$; the boundary between free field and diffuse field is defined by the **reverberation radius**. If beaming occurs, we need to use the effective reverberation radius instead: $r_H^* = r_H \cdot \sqrt{\gamma}$. For broadband excitation, the low-loss sound reflections lead to the creation of countless♣ standing waves, with the density of the eigenmodes rising with the square of the frequency. Exciting the reverberation chamber with a (very slow) sine-sweep, the individual resonances clearly emerge in the low-frequency range whereas for high frequencies, there is merely a tangle of smaller and larger peaks (**Fig. 11.54**). And here we have the fundamental issue of measurements in the RC: these maxima and minimal are strongly dependent on the location – they do not represent room-related constants. While the eigen-frequencies of the room indeed need to be seen as constants (given constant room temperature, humidity and air pressure), it depends on the loudspeaker- and microphone-positions whether the matching oscillation modes are excited and measured.

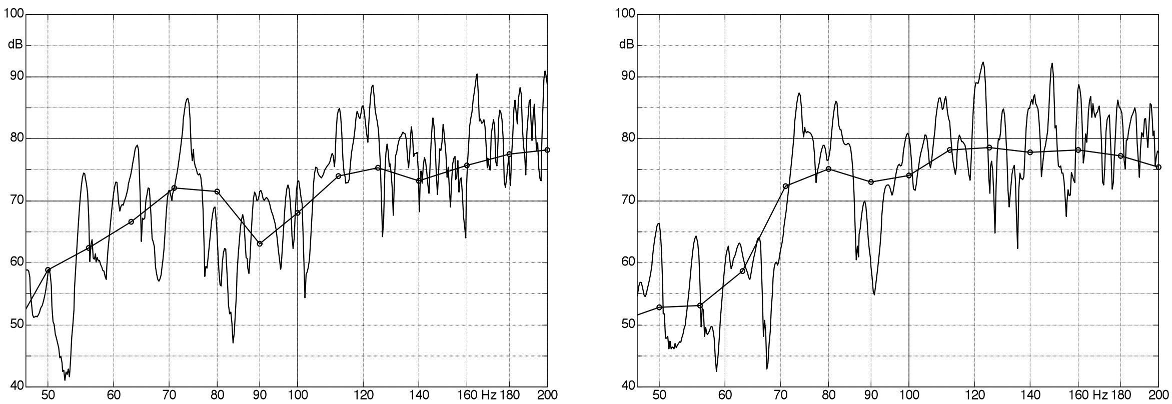


Fig. 11.54: Sweep-measurements in the RC, 2 microphone positions; •—• mean values across a 1/3rd octave.

Since the resonance-peaks found in the reverberation chamber via a sine-sweep may vary by the odd 30 dB or so when changing the microphone position, it is customary not to use sine-tones for the measurement but **noise** of a width an octave or of 1/3rd of an octave. This noise, however, is a stochastic signal and thus requires a special measurement approach. Each noise measurement performed over a period of time represents an average over samples that must be interpreted merely as an estimate of the true value of the basic collective. Therefore two subsequent measurements will not yield the same but merely similar results. For **normally distributed noise** (as mostly used in room acoustics) the squares of the sound pressure (required to calculate the RMS-value) will show a χ^2-scatter. Extending the averaging time of the bandwidth reduces the standard deviation of the measurement errors. [Bendat / Piersol].

♣ Strictly speaking, the reflections may be counted, after all, so: "a lot, a real whole lot".

The lower the center frequency of the 1/3rd-octave to be analyzed, the smaller the absolute bandwidth; the lowest 1/3rd-octave therefore requires the longest averaging time. A 1/3rd-octave bandwidth of 23 Hz corresponds to f_m = 100 Hz, with a standard deviation of the normalized measuring error of about 2%. At a stately 30s averaging time, that is! If we now position the borders of the **confidence interval** at $\mu \pm 3\sigma$, then 99.7% of all measuring results differ by less than ± 0.5 dB from the true value. Thus, the 1/3rd-octave level spectrum of the loudspeaker voltage may be measured with sufficient accuracy with 30 s averaging time. The 1/3rd-octave wide sound pressure spectrum of the reverberation chamber could also be determined with this approach, but the fact the SPL (stochastically) depends on time and additionally on the location♣ needs to be considered. A level that is representative for the diffuse field only results when the number of room resonances per 1/3rd-octave is high enough. Without going into detail too much: that will surely not be the case below 100 Hz (Fig. 11.54), and even above 100 Hz, pronounced level differences are still visible (**Fig. 11.55**). The level measurement was therefore not done at one point in the reverberation chamber but via a rotating microphone.

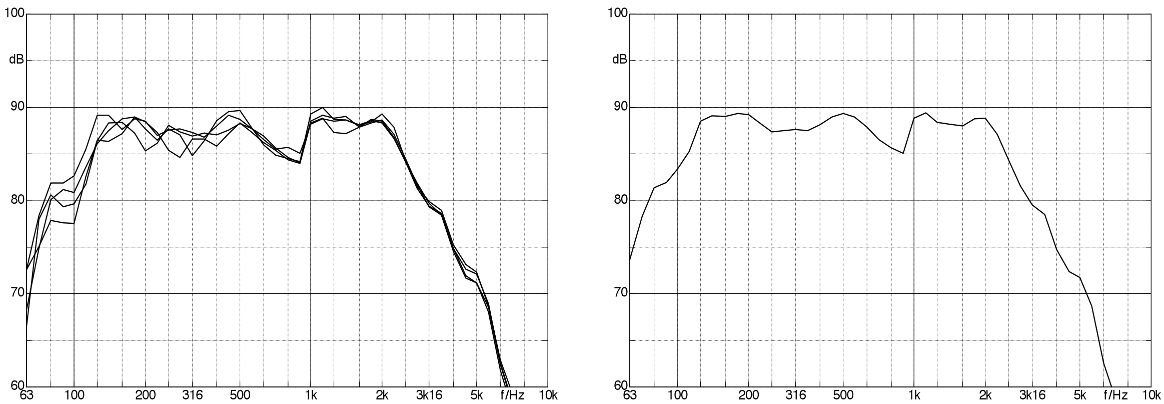


Fig. 11.55: 1/3rd-octave level in the RC, measured at 4 positions with a stationary microphone (left). On the right, an averaging along a circle (not oriented in parallel with the walls) of a diameter of 3 m is shown.

The measurement microphone takes 80 s for one orbit on the 9.4 m long circular track. This makes for an adequate averaging accuracy both in terms of time and location – at least within the framework of the chosen task definition. The sound pressure level L derived from energy-related averaging along the circular orbit first results in the intensity $I = 10^{-12}\ \mathrm{W/m^2} \cdot 10^{L/10\mathrm{dB}}$; from the latter, the sound power P_{ac} may be calculated:

$$P_{ac} = 0.038\mathrm{m}^2 \cdot \left(1 + S\lambda/8V\right) \cdot \frac{V/\mathrm{m}^3}{T_N/\mathrm{s}} \cdot I \qquad \text{Sound power}$$

In this formula, S is the surface area of the room, λ is the wavelength, V is the volume of the room and T_N is the reverberation time. The term within brackets represents the so-called Waterhouse-correction♥ which considers the energy concentration close to walls.

As an example: 100 dB sound pressure level yields (with V = 220 m^3 and T_N = 2 s) a sound power of 42 mW in the high frequency range. The small difference between the intensity level L_I and the sound pressure level L_p ($L_I = L_p - 0.2$ dB) is considered in the pre-factor of 0.038.

♣ The propagation and reflection of each individual wave is subject to a determined process,
♥ Waterhouse R.V., JASA Vol. 27, March 1955.

With the instrumentation for determining the sound pressure levels in both the anechoic chamber (AEC) and the reverberation chamber (RC) ready to go, measurements of the radiated power could start. Two objects came first:

- 8"-loudspeaker (*Eminence α-8*), mounted in an airtight enclosure (22x30x18),
- 12"-loudspeaker (*Celestion Blue*) in the open VOX AD60-VT (Fig. 11.52).

Fig. 11.56 shows the results. The AEC-measurements were taken at a distance of 3 m but recalculated for 1 m (L + 9.5 dB). The RC-measurements were obtained from averaging over a circular path as described above; the level measured in the diffuse field was recalculated for 1 m. Pink noise served as test signal, it was filtered to a width of a $1/3^{rd}$-octave (IEC 1260 class 0), with the $1/3^{rd}$-ovtave-voltage fed to the loudspeaker amounting to 2.8 V_{eff} for both measurements.

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Fig. 11.56: Comparison of AEC- and RC-measurements. AEC: 2.8 V per $1/3^{rd}$-oct., 1m. RC: 2.8 V per $1/3^{rd}$-oct., $r_H \rightarrow$ 1m. In the frequency range below 125 Hz, the sound fields in both rooms show “acceptable” artifacts. “RAR” = AEC, “HR” = RC

For both loudspeakers we see clear differences between the frequency responses measured in the AEC and the RC. The main reason of the deviations is the beaming increasing with rising frequency, but the different radiation impedance also plays a role. The Eminence-speaker mounted in a relatively small, airtight enclosure acts, *at low frequencies*, approximately as a spherical source – in the AEC, its radiation impedance is mainly formed by a mass [3]. In the RC, we find a much more complicated radiation impedance depending on the individual RC-data and the position of the loudspeaker. The small number of room-modes per $1/3^{rd}$-octave has the effect that the speaker can feed sound energy only into a few narrow frequency bands with a relatively high efficiency, and therefore the RC-level (recalculated to 1 m) is somewhat smaller than the AEC-level. The VOX-enclosure has a rear opening of 49x21 cm^2 and consequently beaming may be expected already in the low-frequency range (rising with increasing frequency) – but in a different manner than with the Eminence-speaker. The VOX was measured freestanding in the AEC, and set on a 50-cm-high stool in the RC. The latter, stage-typical mode of operation causes differences in the radiation impedance up to about 600 Hz – these will have to be discussed below in connection to Fig. 11.61. In addition, there is the special location- and mode-dependent loading in the RC. The question regarding the efficiency therefore needs to be discussed specifically for the given room – there are systematic differences between the efficiencies determined in the AEC and the RC. These differences are on the one hand typical for the respective sound field, but on the other hand represent effects of the individual room parameters.

To be able to more precisely quantify the beaming behavior, horizontal directional diagrams (i.e. the directional gain) were taken for both loudspeakers in the AEC using $1/3^{rd}$-octave-noise (**Fig. 11.57**).

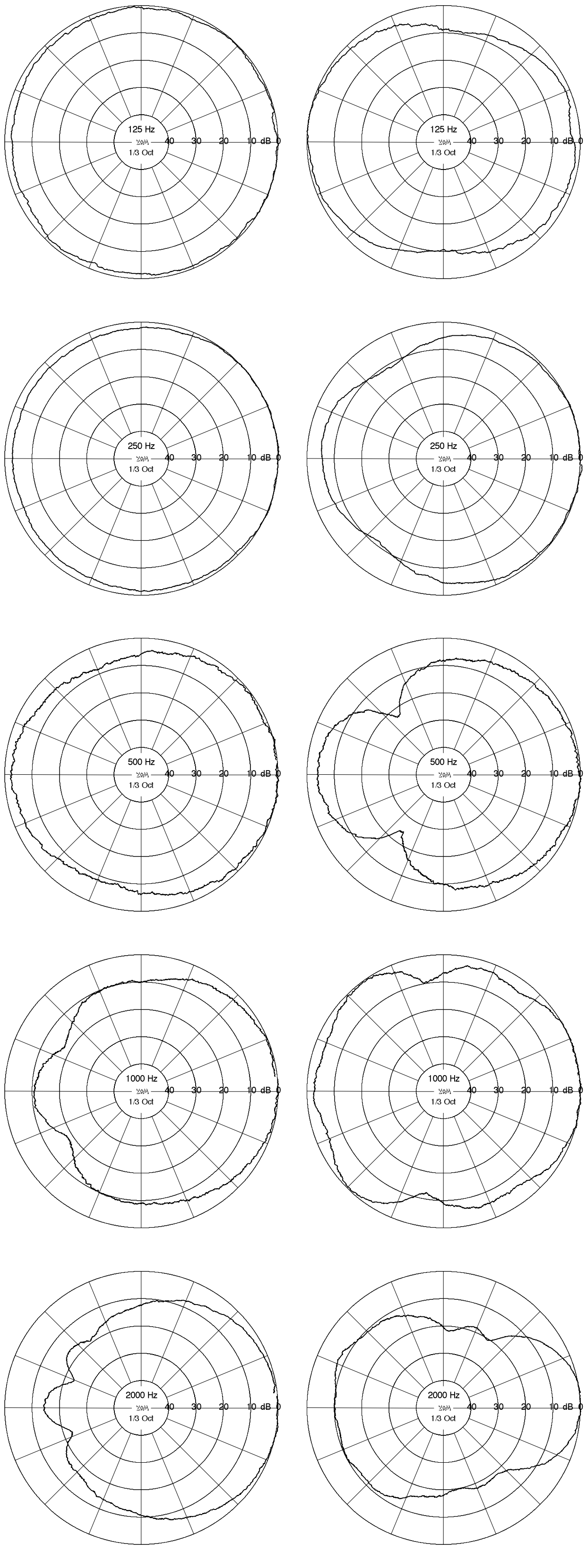

Fig. 11.57: Horizontal directional diagrams.
Eminence Alpha-8 (left), VOX AD60-VT (right).
All directional diagrams are normalized to the maximum.

In **Fig. 11.57** we see, in the left column, the directional diagrams of the Eminence speaker, and on the right those for the VOX-Celestion. The Eminence was mounted in an airtight enclosure, and the Celestion in the AD60-VT-housing open to the rear (Fig. 11.52).

In the Eminence, we find almost textbook-grade beaming increasing with rising frequency, while the VOX shows a much more complex behavior. There is no frequency range in which the latter acts as a pure dipole because the air within the enclosure forms, in cooperation with the complex impedance of the “compensation opening”, a phase shifting filter. The characteristic of this filter is remotely reminiscent of a bass-reflex box with a rather special tuning – certainly not one following the Thiele/Small-approach. That is not required anyway: this enclosure is supposed to radiate the tone of the guitar optimally and may (or even should) be shaping the sound – something rather not desired in a HiFi-loudspeaker.

Not all guitar loudspeakers are mounted in open enclosures: the probably most famous representative of the closed box may be the one by Marshall. However, Fender – more known for open enclosures in their smaller combos – early on offered a closed speaker housing for the Showman and Bandmaster setups. These included classical bass-reflex enclosures with sometimes quite ingenious co-axial bass-reflex openings. It appears that in the upper power-range, the 2- or 3-part “piggyback”-solutions are a bit more dominant compared to combos reigning in the lower-power range – but that must not be seen as a dogma. In the end, each guitarist decides according to sonority and radiation characteristic – or simply grabs "same as Jimi had".

Fig. 11.56 already reveals much about the radiation but does not directly represent the efficiency. The latter may be determined in the AEC via integrating over the squared sound pressure along the enveloping surface, or in the RC using intensity and spherical surface of the reverberation radius. For the AEC-measurement, either a large number of measuring points (or paths) are required, or a rotationally symmetric radiation; for the measurement in the RC we need merely the SPL in the diffuse field, volume (cubic capacity) and reverberation time. In order to limit the effort, the **efficiency** was determined not in the AEC but in the RC – starting with nominal conditions, i.e. $P_{el} = U^2 / R_N$. This specification is physically still not entirely correct but does deliver purposeful comparative values for the operation from a stiff voltage source. Guitar amplifiers do not generally feature low output impedance but approach this mode of operation as the rather typical clipping occurs. Supplementary measurements regarding the physically exactly defined efficiency will follow.

Fig. 11.58 shows the nominal efficiency of the Celestion "Blue", established in the RC and with the speaker mounted in the VOX AD60-VT enclosure. Certainly impressive but not at all unique: the thin lines in the figure belong to the competition issued by the same manufacturer and behaving similarly efficient. The new neodymium speaker ("Neodog", uppermost curve) even steps up the game. The figure on the right, however, shoes that the efficiency may be smaller, as well: only the JBL-box with its 12"-speaker weighing in at 9 kg can reasonably keep up – the other two speaker boxes were obviously were optimized using other criteria.

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Fig. 11.58: Left: Nominal efficiency of the Celestion "Blue". Thin lines: 4 further Celestion 12" speakers for comparison: Neodog, Vintage-30, G12-80, G12-30H. Right: Full-range speaker-boxes. The "nominal efficiency" was established for the specified nominal impedance, irrespective of the actual speaker impedance.

Let us quickly discuss, using two examples (**Fig. 11.59**), the question whether speakers using **Alnico-magnets** are "louder" or "deliver more treble" compared to speakers deploying ceramic magnets. P12-R and L-122 (both featuring Alnico magnets) have a smaller efficiency than the Vitage-30 (ceramic magnet). The Celestin "Blue" (Alnico), however, shows a higher efficiency than its ceramic-fitter competitor Eminence L-125. Besides the magnet material, mainly the magnet size and the membrane are of importance – the "inspired Alnico sound characteristics" are nothing but vapid advertisement.

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Fig. 11.59: Nominal efficiency as in Fig. 11.58, comparison Alnico- vs. ceramic-magnets.

We now turn to the correctly defined **efficiency**, i.e. the ratio between emitted and received active power. Again the RC is used, with its special characteristics. As depicted in **Fig. 11.60**, the *real part* of the electrical impedance differs from the *nominal impedance* in particular at the resonances points 95 Hz and 190 Hz, and in the high-frequency region. Hence in these areas the loudspeaker efficiency is higher than the "nominal efficiency" determined relative to the nominal impedance (8 Ω). The differences are clearly visible but may be ignored when aiming for a rough orientation. This approach may be allowable even more so because all 12"-speakers investigated here showed similar frequency responses of the impedance. Merely at the main resonance (around 95 Hz), the behavior may be substantially different. If this range is of particular interest, exact impedance measurements are required.

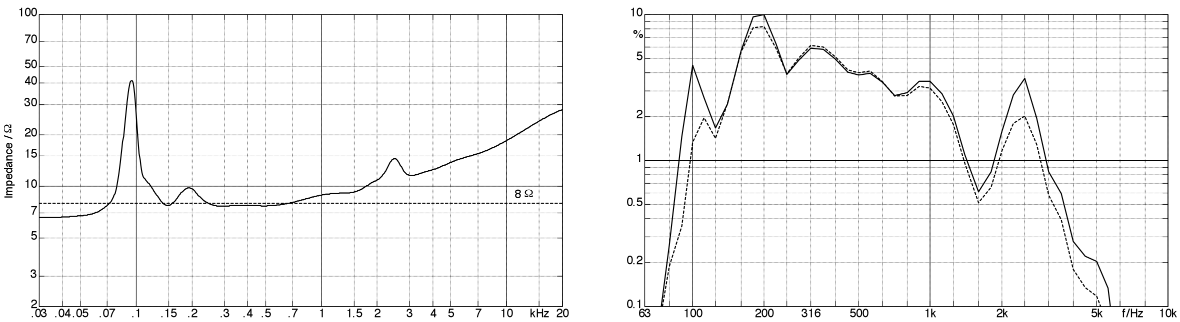

Fig. 11.60: Real part of the electrical impedance (left), comparison between nominal efficiency (---) and actual efficiency (1/3rd-octave average). VOX AD60-VT with original loudspeaker.

It has already been mentioned that the loudspeaker efficiency is not a constant but depends on the acoustical environment. The VOX AD60-VT, a small combo, finds itself often placed on a stool in its everyday stage work. The controls are better accessible that way, and the guitarist can better hear him/herself. On the other hand, one could leave the VOX on the floor, as well – the stored sound settings could be called up via a footswitch. How does the sound radiation of these two modes of operation compare? Since the load impedance rises as the speaker approaches a reflecting (floor-) surface, the level radiated at low frequencies will increase up to 3 dB (**Fig. 11.61**). Closing the rear of the amp will have the opposite effect: the level decreases across a wide frequency range, and only at very low frequencies there is a gain. The latter is not generally desirable, because many guitarists will rather leave this frequency range to the electric bass.

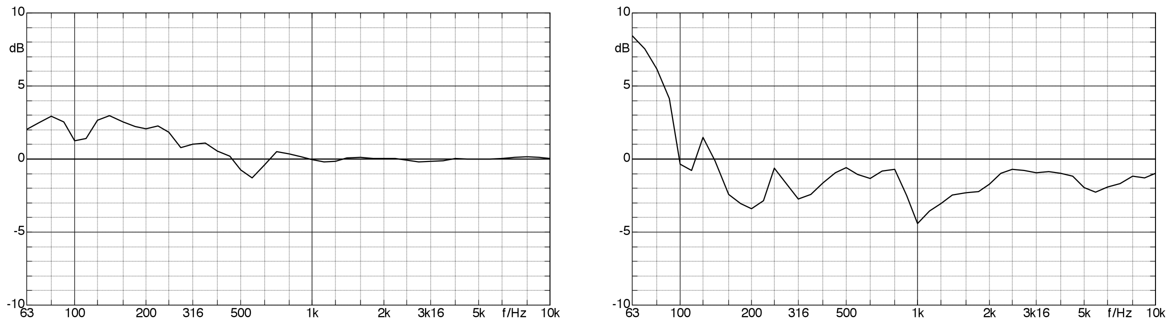

Fig. 11.61: Left: level gain when placing the VOX AD60-VT on the floor (compared to placement on a stool). Right: level loss when closing up the rear of amplifier. Both measurements taken in the reverberation chamber.

The following page compares measurements in the AEC and the RC for several loudspeakers. All 12"-speakers were measured mounted in the AD60-VT-enclosure.

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Fig. 11.62: Comparison between measurements in the AEC (——) and the RC (----), recalculated for 1W / 1m. 1/3rd-oct. analysis w/50% overlap (main/side 1/3rd octave), pink noise. Ordinate: sound pressure level dB_{SPL}.
The measurements for the first 5 lines of figures were done using the AD60-VT-enclosure.
The thin angled lines in the figures are not target-curves but serve for orientation only.

The frequency responses shown in **Fig. 11.62** indicate common characteristics due to the enclosure (and possibly due to similar constructional details in the speakers), and also show differences that have their reason in the different membrane designs. The difference between the AEC- and RC-measurements is of particular interest since here the **directivity** manifests itself (Chapter 11.4). Two peculiarities need to be considered: 1) the speaker remained at the same location in the RC, its radiation impedance therefore is highly room-specific, 2) the reference direction in the AEC was always 0° even if more sound power was radiated in other directions – therefore, negative directivity is possible. It has already been elaborated that a directivity of $d = 0$ dB does not always imply spherical radiation.

Given the measurement curves in Fig. 11.62, it should be mentioned once again that for guitar loudspeakers, different optimization-guidelines are valid compared to e.g. a studio monitor. With slight exaggeration, we could state: *if the efficiency is high enough, the rest comes together by itself.* A Canton Quinto will not make the hard-rocking player happy at all – it just ain't no box for guitar. A single Vintage.30 will generate at its maximum permitted power (60 W) up to 123 dB at 1 m distance, while the Quinto will manage only 102 dB at maximum power. In absolute terms, we would have to feed the Quinto with 100 times its maximum power to make it compete with the Vintage-30. Conversely, the Vintage-30 would be utterly out of its depths as a studio monitor, with a way-too-unbalanced frequency response. None of the instrument speakers analyzed in Fig. 11.62 could be attributed a "bad" frequency response – the peaks and dips are typical for the genre, with one guitar player preferring this and the other player preferring that.

Measuring frequency responses aids in objectively determining differences and similarities – but it can not replace a listening test. From the measured curves we can derive general statements about efficiency and therefore about loudness; and we may obtain some very general ideas about the sound: the pronounced 1.5-kHz-dip combined with the 3-kHz-peak of a G12-H will clearly shape the sound. However, whether the Celestion "Blue" also entered in the diagram for comparison will sound better or worse – the diagrams cannot say anything about that. The trade business has masterfully understood how to fuel the flames of "tuning" and "retrofitting": the guitarist who is unhappy with sound of his rig will find so many clever statements suggesting that changing the pickups or the pots or the loudspeaker will push him/her into the professional realm. The swapping of components may be purposeful, it the original parts were truly substandard. On the other hand, whether swapping a G12–H (at 119 €) for a "Heritage" G12-H (195 €) will transform scrap into Hendrix-like sound – that is more than just doubtful. This author had the exciting pleasure and privilege of ear-witnessing (in the front row at a perceived 150 dB) the Guv'nor JH letting loose heaven and hell with two Marshall stacks in the Congress Hall in Munich – but had the master decided to present the encore via a wall of AC-30's … that would have been (very) fair enough, as well. It's in the fingers – we need to be reminded of that fact again and again.

11.6 Non-linear Distortion

In communication engineering we very carefully distinguish between linear and non-linear signal distortions: a linear system generates linear distortion exclusively, and a non-linear system (as far as it is free of memory) generates only non-linear distortion. Generally, one seeks to avoid mixing up linear and non-linear effects by defining sub-systems that individually generate purely linear or purely non-linear distortion.

In a **linear system** (e.g. an amplifier) the principles of proportionality, superposition and absence of sources hold. The latter characteristic is easily explained: where there's no input signal, there's no output signal. For a (non-zero) output signal $\tilde{y}$, a matching (non-zero) input signal $\tilde{x}$ must exist. If $\tilde{x}$ is doubled, $\tilde{y}$ must double, as well – that's proportionality. We quickly realize that the "linear function", as used in mathematics and defined by the linear equation $y = k \cdot x + m$, will fulfill the requirements of proportionality and freedom of sources only if m is zero. The law of superposition requires that the mapping of a sum must equal the sum of the mapped summands. Thus: $y = T\{x\}$ represents the mapping of the input signal x onto the output signal y. If the sum of two signals is fed to the system input, the following must hold in a linear system:

$$y = T\{x_1 + x_2\} = T\{x_1\} + T\{x_2\}$$ Superposition in the linear system

Proportionality and absence of sources alone do not suffice to specify linear behavior, as shown by the example of an **ideal full-wave rectifier** (that reverses the sign of a negative input signal): this device is source-free, and an n-fold input signal is matched with an n-fold output signal – but as soon as a further signal (e.g. a DC voltage) is added at the input, the waveform of the output changes … the rectifier is non-linear.

It is tempting to go and reduce the linear system to the matching-formula $y = k \cdot x$; however, this would unduly exclude the group of differential equations. A system that maps the speed of a mass onto its acceleration is a (time-related) **differentiator**♣. This system meets the requirements of absence of sources [$d/dt(0) = 0$], of proportionality [$d/dt(k \cdot x) = k \cdot dx/dt$], and of superposition: $d/dt(\xi + \mu) = d\xi/dt + d\mu/dt$. The differentiator is a linear system in spite of the fact that its sinus-transfer characteristic is not a straight line but an ellipse. Typically, the **linear distortion** generated by a linear system is specified for sinusoidal drive-signals as amplitude- and phase-distortion (or delay-time distortion), and is graphically represented as amplitude-frequency-response and phase-frequency-response. The bass-cut generated by an RC-highpass is a linear signal distortion, as is the presence boost of an equalizer (that of course must not be overdriven). Reacting to an impulse-like excitation, the resonant circuit of an equalizer will ring (theoretically for an indefinite time). Without a doubt, this is a signal distortion – but a linear one. Unfortunately, there is often a lack of distinction between linear and non-linear distortion, especially when it comes to loudspeaker characteristics discussed in popular "science". **Non-linear distortion** results if a system fails to fulfill one of the above mentioned linearity criteria – this system is then non-linear. Whenever possible, we try to separate linear and non-linear distortion into subsystem (possibly only existing as a model): a linear subsystem described by its "straight" characteristic, and a (memory-free) non-linear subsystem defined by its "curved" transmission characteristic.

♣ The formula-representation is meant to save space: it may not meet the expectations of all mathematicians.

It is, however, in many cases not possible to divide a real system into *one* linear and *one* non-linear subsystem: since in non-linear systems the (commutative) exchangeability is not there anymore, it may be that a plurality of subsystems is required, and the corresponding description may become highly complicated. The dynamic **loudspeaker**, as well, includes several non-linearities that do not allow themselves to be modeled in one and the same subsystem: the displacement-dependent force-factor (aka. transducer coefficient *Bl*), the displacement-dependent stiffness of the membrane-suspension, and the inductance that is also displacement-dependent. If the loudspeaker is mounted in an airtight enclosure, the non-liner stiffness of the air-suspension caused by the enclosure weighs in, as well. With the speaker mounted in a ported enclosure, the airflow within the port-tunnel introduces non-linearity. All these non-linearities generate a non-linear transmission characteristic but also cause a reaction on the electrical side and make for a strongly non-linear loudspeaker-impedance. In addition, we have non-linearity generated in the amplifier and the output transformer (if one is present). All in all we get a complex system with coupled non-linearities – and one that produces pronounced linear distortion to top it all.

For a loudspeaker mounted in a cabinet that is open to the rear, we may neglect the non-linearity of the air. At low frequencies, we may – to start with – dispense with a consideration of the inductance so that as a first approximation, a non-linear mechanical subsystem and a non-linear magnetic subsystem remain. The mechanical non-linearity is found in the stiffness of the membrane-suspension, i.e. the inner centering (spider) and the outer fastening (surround). As the membrane is deflected slowly, force is directed against a progressive spring with its stiffness increasing as the displacement increases. The stiffness is a system-variable while the displacement is a signal-variable. If a system-variable is dependent on the signal, we always have a non-linear system. The non-linearity in the magnetic system clearly is the transducer coefficient (the force-factor): as system-variable *Bl*, it takes care of the proportionality between current and Lorentz force: $F = Bl \cdot I$. However, this proportionality requires that the system variable *Bl* is independent of the signal – specifically independent of the displacement. That is not the case here: with increasing displacement, the coil moves out of the magnetic field and therefore *Bl* decreases. A further effect may play a role in this scenario: two magnetic fields superimpose as current flows. One is constituted by the permanent field generated by the permanent magnet, the other is the AC field surrounding the voice-coil wire. Because the ferro-magnetic parts located in the magnetic circuit all show a non-linear characteristic (the specific magnetic conductance μ is field-dependent), "modulations of the magnetic field" may result. Some manufacturers seek to decrease the effect via short-circuit rings while others do not do anything about it, regarding it as typical. It is here where the peculiarities of the **guitar loudspeaker** begin: while for HiFi-speakers there is consensus that non-linearity must be as small as possible, opinions diverge considerably when it comes to guitar speakers. You may hear (or read) on the one hand that the guitar speaker is, after all, a loudspeaker too (correct), and thus what has been taught in the HiFi-domain for decades cannot be wrong (??). On the other hand, (positive) reviews including evaluations such as "dirty midrange" give rise to some hope that at least a few designers have recognized that sound-shaping function of the guitar loudspeaker.

We must not fail to mention here though, that not only among the manufacturers, but also among the players multiple opinions abound. You get the Jazz-dude who brutally chokes the hard-won brilliance of the guitar by bottoming out the tone control, the Country-picker with his piercing treble, the crunching-along Blues-man, the chainsaw-ing Metal-ist, the Jack-of-all-Trades cover-guy, and the folksy oom-pah-strummer. A consolidated drive towards standardized loudspeaker distortion may not be expected given such a heterogeneous population and mix of opinions.

Here's an example taken from loudspeaker history: **JBL**, the renowned American speaker manufacturer, looks back on a long tradition as supplier for cinemas, recording studios, and living rooms. Nothing but High-Fidelity – non-linear distortion is marginal as a matter of course: ... *"low distortion which has been always associated with JBL products"*. In the early 60's the demand for instrument speakers grows, and at JBL, a tried and trusted workhorse, the D-130, is modified to become the **D-130F**. The changes mostly relate to the air gap that was – according to statements by the designer (Harvey Gerst) – slightly enlarged in order to obviate damage. And then there's the designation: F is for Fender, the largest customer. Years later, the **K-130** follows with double the power capacity compared to its predecessor but still *"clean at any volume level"* (a quality that probably would not have always applied to the associated musicians). Both the D-130F and the K-130 were fitted with Alnico magnets, but the next generation – the **E-Series** – received ceramic magnets. This prompted JBL-mastermind John Eargle to state that Alnico was known for its *"inherently low distortion performance"*. However, according to him, the new E-Series is even better: *"The improvement has been in reducing second harmonic distortion",* obtained with the "*symmetrical field geometry"*. Given this upgrade, the loudspeaker is eminently suitable *"for vocals – and guitar"*. Presumably, this further added to the already long list of JBL-users shown in the adverts. Due to space-restrictions, this list cannot be commemorated in its full extent here, but the following may serve as an excerpt: Count Basie, Harry Belafonte, Tony Curtis, Sammy Davis jr., Doris Day, the unforgotten Carmen Dragon, Duke Ellington, Ella Fitzgerald, Hugh Hefner (!), Dean Martin, Frank Sinatra, not to forget Richard Nixon and "The Duke" John Wayne[♣]. Global super-stars, all of them – and all of them JBL-users. Such a feat of course calls the competition into the arena. And thus it was that **Electro-Voice** retaliates with a big swing, proclaiming: *"Symmetrical magnet gap structures have been promoted as desirable in a guitar speaker. We have found this to be a fallacy"*. Because: *"A coil moving in an asymmetrical magnetic gap will generate a mixture of odd and even harmonics, resulting in a more complex, richer sound."* To each his own … there's no accounting for taste.

So, let's not begrudge The Duke the undistorted JBL-sound of his electric guitar (hm … still thinking about that one …), and Joe Bonamassa his EV-sound chirping from the 4x12's – beauty is in the eye of the beholder. What can be said about **magnetic non-linearity** from a scientific angle? If you leave the pole-core (the cylinder in the interior of the voice coil) formed as a cylinder over its full length, as shown in **Fig. 11.63**, then an asymmetric scatter-field will result: the shape of the field above the coil is different from the shape below it. Reducing the core-diameter in the lower section, though – as it is shown in exaggerated fashion in the figure – will render the two stray-fields more symmetric. The result is that a symmetric Lorentz-force acts on the voice coil for both positive and negative displacement. As already mentioned, this force depends on current and displacement. While the current-dependency is desired, the displacement-dependency is not, because it generates non-linear distortion. For a symmetric field, the distortion is of even-order (even function) – given asymmetry distortion, odd-order also weighs in.

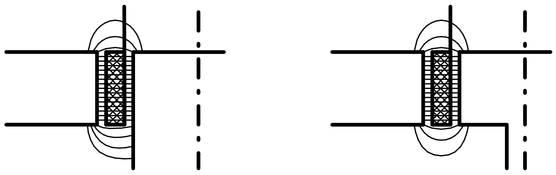

Fig. 11.63: Different designs of the pole-core. On the left it is purely cylindrical, on the right offset. The scatter-field generated outside of the air gap depends on the geometry of the pole-core.

[♣] From JBL's 1968 loudspeaker brochure.

The field-limiting at both ends leads to a degressive, clipped current/force-characteristic that leads some people to conclude that the course of the oscillation would now also be limited – similar to an overdriven amplifier. This assumption, however, overlooks that the force is mapped onto the displacement (via Hooke's law) only in the range below the resonance. At resonance the membrane acts (in the simple model) as a damper, and above the resonance it acts as a mass [3]. If, in the range above the resonance, the motive force becomes weaker at strong displacement, this reduces the *acceleration* primarily. Of course there will be an effect on the displacement, as well, but: displacement and acceleration are in opposite phase. Or, to be somewhat more precise: the acceleration is the second derivative of the displacement. Analytically, this leads to a non-linear differential equation that can be solved approximately – with a big effort, however (the system is not just weakly but extremely non-linear).

The qualitative effects of the inhomogeneity may be studied rather well using the **simple membrane model** [3]. If we reduce the membrane to spring, mass and damper, and the electric side to the ohmic voice-coil resistance, we have a 2nd-order system that may be described via the **frequency of the pole** (resonance frequency) and the **Q-factor of the pole**. The pole-frequency unambiguously results from the stiffness and the mass, while two limiting cases are of importance for the Q-factor of the pole: open circuit and shot circuit on the electrical side. Given the open circuit, no current can flow and consequently the magnetic field will exert no force onto the membrane. The Q-factor of the pole depends solely on the mechanical parameters: $Q = \sqrt{sm}/W$. The purely mechanic dampening of the membrane is relatively small such that the Q-factor of the pole is considerably larger than 1 (5 is not uncommon). As the voice coil is shorted (or as an amplifier with a very small internal impedance is connected), the voice-coil resistance that is transformed from the electrical to the mechanical side acts as an additional dampening♣, and the Q-factor of the pole drops below 1. Since the electromechanical coupling becomes smaller at large displacement (due to the inhomogenities in the field), the membrane-dampening decreases with increasing drive levels – the displacement tends to become too large, and not too small as it would with degressive limiting [3, Chapter 6.2.3].

In the asymmetrical magnetic field, the reset-forces acting at the extremes of the membrane-displacement are unequal (in terms of magnitude), and therefore the average force is not zero. A steady force of the frequency 0 Hz results that pushes the membrane out of its neutral position in the direction of the weaker of the two fringe-fields. Because in reality the two fringe-fields are never exactly identical, this effect always occurs: a small asymmetry suffices to make the membrane wander slightly from the idle position. This enhances the lack of symmetry, and the membrane continues to wander off – it is only stabilized by the onset of the counter-force exerted by the membrane-suspension. Therefore, 2nd-order distortion is to be expected in the range above the resonance – even if there is a symmetrical layout of the magnetic field. “Field-modulation” is a further source of 2nd-order distortion: part of the magnetic field generated by the flowing current superimposes onto the steady field of the permanent magnet, i.e. the flux density therefore fluctuates in sync to the excitation current. Because of this, the force obtains a share that is dependent on the square of the current – and that implies 2nd-order distortion. Another way of explaining the effect: the flowing current generates attracting forces between neighboring ferromagnetic parts (through which the field flows). These attracting forces are <u>in</u>dependent of the sign and therefore generate even-order distortion (just like a rectifier). Relief, if at all sought, could come in the form of a short-circuit ring. It forces the AC field out of the magnetic circuit, and the 2nd-order distortion decreases.

♣ The eddy-current brake in lorries and trains works based on a similar principle.

Analyzing the current fed from a stiff voltage source will give a first impression regarding the linearity (or non-linearity) of the loudspeaker. The mechanical membrane-impedance *F/v* is transformed to the electrical side with the square of the transducer coefficient. If we disregard the inductance, the electrical impedance consists of two components: the voice-coil resistance (e.g. 6 Ω), and the transformed mechanical impedance. Any non-linearity in the electrical impedance (given a stiff voltage source these would be **current distortions**) will consist of two components: a non-linear transducer coefficient (*Bl*), and/or a non-linear membrane impedance. In a loudspeaker, both these components are present: both the stiffness of the membrane-suspension and the transducer coefficient are displacement-dependent. The current-curves for the operation close to resonance are shown in **Fig. 11.64**, with the distortion being very significant. It should be noted that the voltage amounts to merely 10 V_{eff}, i.e. nominally only 12.5 W for this 8-Ω-speaker (deployed in a 60-W-amplifier). Moreover, since the impedance will be at its maximum at resonance, the power taken by the speaker will be even (much) less – we are far away from any undue overload situations. The curves reveal a strong 2^{nd}-order distortion. The amplitude of the 2^{nd} harmonic rises to up to 67% of the 1^{st} harmonic; this would correspond to a harmonic distortion of k_2 = 56% (the approximation U_2 / U_1 should not be used anymore at such high distortion levels). It is beyond the aim of this chapter to localize or separate the individual roots of these distortions – the effort would grow too big. Rather, we will present comparative distortion measurements; these will consistently show that all investigated loudspeakers are strongly non-linear systems even at very moderate drive levels. Not that guitar players would generally be adverse to such characteristics …

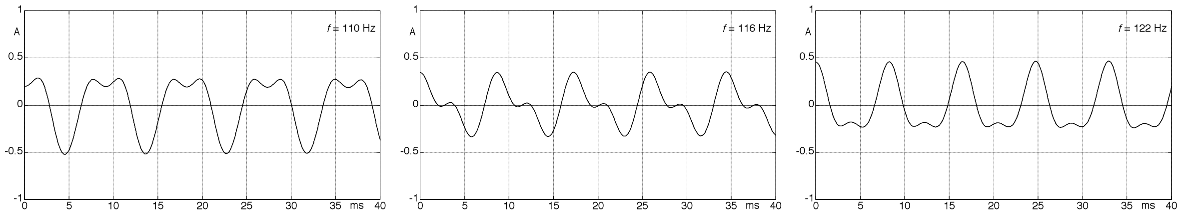

Fig. 11.64: Time-functions of the loudspeaker-current fed from a stiff voltage source, U = 10V; VOX AD60-VT.

The frequency responses of the distortion (**Fig. 11.65**) show that the maximum distortion of the current happens at the main resonance (116 Hz); it is here that the displacement is at its maximum. There are two reasons that 2^{nd}-order distortion can rise to such heights: at resonance, the 2^{nd}-order harmonic of the current is highest (due to the mentioned non-linearity), and at the same time the overall current becomes smallest (due to the rise in the impedance. The difference of the two levels (the distortion dampening) therefore has a pronounced maximum here. However, the current-distortion describes predominantly the electrical behavior – non-linearity in the sound radiation needs to be analyzed separately.

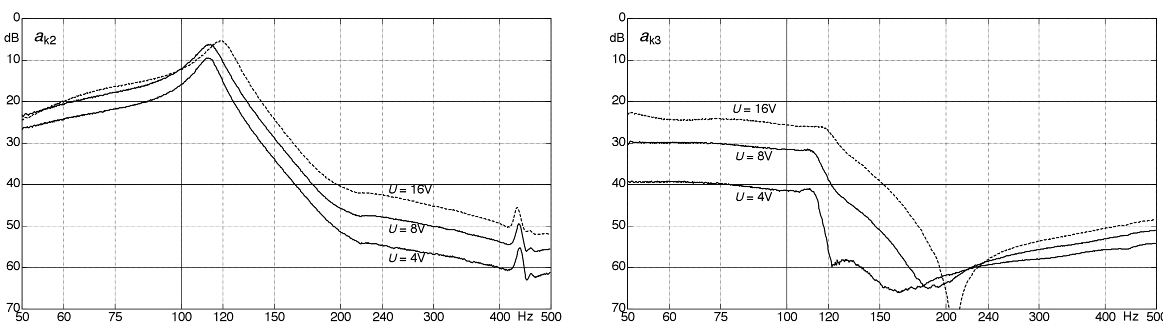

Fig. 11.65: Frequency-dependence of harmonic distortion in the current when fed from a stiff voltage source (as in Fig.. 11.64). The drive-level-dependent shift of the maximum is a result of the strong non-linearity.

In order to measure the (non-linear) distortion in the sound pressure, the speaker was operated from a stiff voltage source and mounted in the VOX AD-60VT enclosure. Measurements were taken in the AEC, with the microphone on axis at a distance of 3 m (**Fig. 11.66**). The results are not untypical for a dynamic woofer: at low frequencies very strong distortion is generated. Then, from 70 Hz up, the 3rd-order distortion drops off faster than the 2nd-order distortion, and above 150 Hz, the THD remains below about 1%. Compared to the analysis of the current (Fig. 11.65), the distortion has mostly increased.

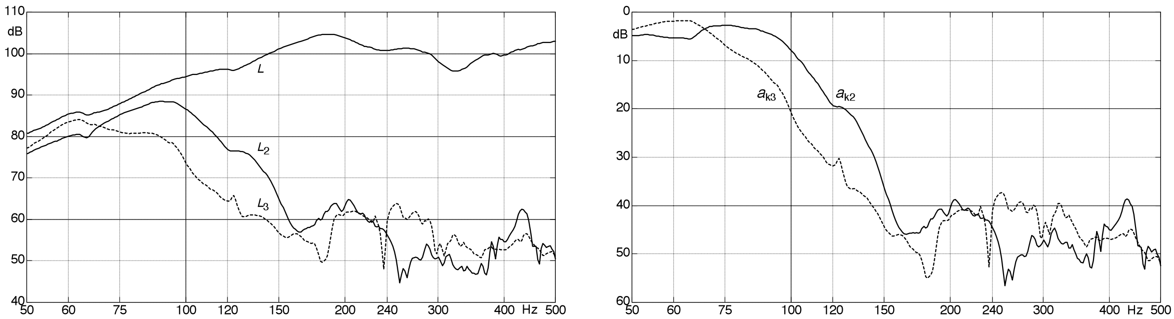

Fig. 11.66: Frequency responses of SPL (left) und distortion level. VOX AD60-VT, U = 10V.

However, a THD of 1% is not the professed aim for a guitar loudspeaker – some speakers easily reach ten times that distortion: **Fig. 11.67** illustrates measurements with another Celestion loudspeaker, operated with the same voltage and in the AD-60VT-enclosure: the Celestion “Blue”. This speaker is not broken – far from it; it’s just that the input voltage of 10 V already pushes it close to the borders of its power capacity (15 W). On the other hand, this should not be taken as evidence that a THD of 10% would be typical for getting near to the power limits: the Vintage-30 speaker (depicted below the “Blue”) is specified at 60 W and, at 10 V, distorts similarly to the “Blue”. Since the concept behind the Vintage-30 is that it should be a descendant of the “Blue”, it is only sequacious that it should distort like the latter.

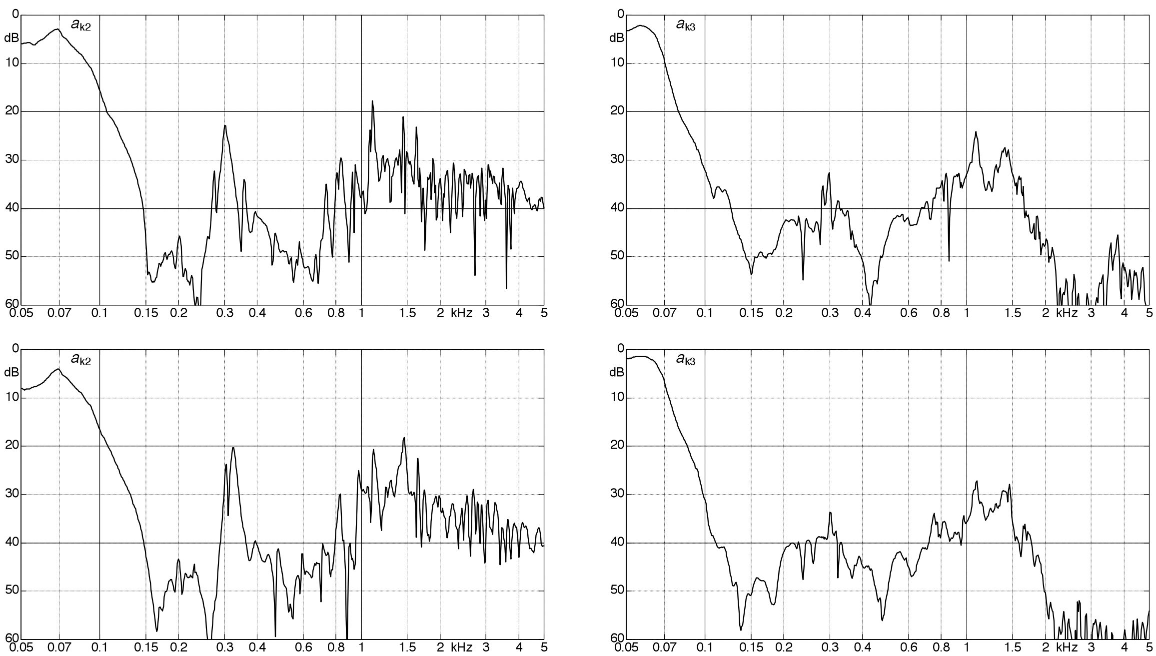

Fig. 11.67: 2nd- and 3rd-order distortion. Enclosure: VOX AD-60VT, U = 10V, d = 3m.
Upper row: Celestion "Blue" (P_{max} = 15W); lower row: Celestion Vintage-30 (P_{max} = 60W).

Documentations on loudspeaker non-linearity are often limited to measurements of the harmonic distortion; this may have to do with the fact that such measurements are a standard tool in systems analysis. Since Brüel&Kjaer has released its legendary instrumentation-combination of the 2010/1902-devices, difference-frequency measurements are also common for band-limited systems – but there is a further distortion-mechanism that is found especially in loudspeakers: **sub-harmonics**. This term means to describe the generation of distortion tones that have a frequency lower than the excitation frequency, e.g. $f/2$ or $f/4$. **Fig. 11.68** depicts, accordingly, two spectra derived from the sound pressure. A sinusoidal voltage (10 V) was imprinted at the loudspeaker connectors, with f = 1.6 and 1.5 kHz. Highlighted in grey are those spectral lines (broadened by leakage) that could be expected as regular "harmonic distortion"; in addition, however, we see a sub-harmonic developing, and frequency-multiples of it. The double-peaks in the right-hand diagram point to fast time-variant processes: the spectrum resulted from a sweep, and the "sub-harmonic distortions" change their level very fast.

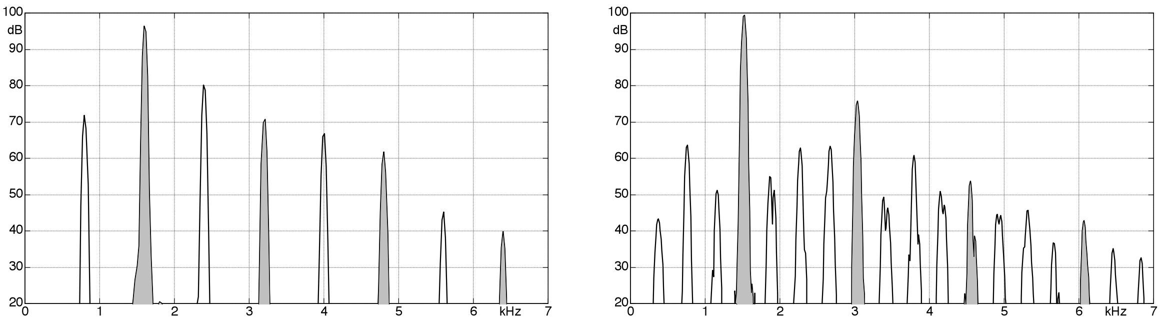

Fig. 11.68: Sub-harmonics at half (left), and a quarter (right) of the excitation frequency.

Such sub-harmonics appear – if at all – only in small frequency ranges. **Fig. 11.69** shows two spectrograms representing the level as grey value across the f/t-plane. The bottom rising curve belongs to the level of the first harmonic, the levels of the higher harmonics follow above. The grey dots or groups of dots appearing in the right half of the diagram point to sub-harmonics (or their frequency-multiples). The speaker analyzed on the left (Jensen P12-N) shows sub-harmonic distortion only at an excitation frequency of about 1760 Hz, while the speaker on the right (Celestion G12-Century) features it in several ranges from 920 Hz up. Both speakers have an impedance of 8 Ω and both were measured at 10 V. The C12-N is specified with a power capacity of 50 W, and the G12 at 80 W – neither speaker is therefore operated close to any power limit.

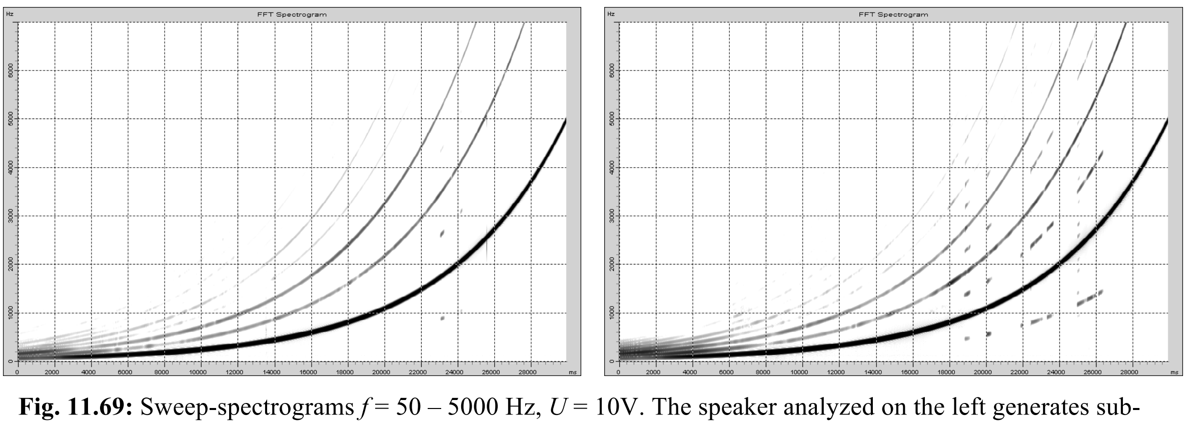

Fig. 11.69: Sweep-spectrograms f = 50 – 5000 Hz, U = 10V. The speaker analyzed on the left generates sub-harmonics only at about 1760 Hz while the speaker on the right does so in several frequency ranges. Abscissa-scaling: sweep-time = 0 – 30 s; ordinate-scaling: frequency = 0 – 7 kHz.

The generation of sub-harmonic distortion cannot be explained merely via a curved transmission characteristic. As **Fig. 11.70** shows, a tilting vibration of half the frequency is superimposed. Mathematics laconically (and correctly) explains such phenomena with “solution of a non-linear/time-variant differential equation”, physics offer “parametrically excited eigen-oscillations of a system with time-variant system-variables”. Time-variant quantities are indeed easily imaginable: the membrane deforms, and the location-dependent stiffness of the membrane is certain to be dependent on load – and therefore on time. Moreover, the oscillation of the membrane by no means needs to be one-dimensional: tilting- and tumbling-movements are possible, and the overall system is of a complicated, non-linear nature. We may expect sections of the membrane oscillating with the same phase – but of course not the whole area; there will be phase shifts, and since the system parameters are time-variant, most probably there will be time-variant phase shifts, as well. Simple models fail here, e.g. since already the superposition principle may not be applied anymore.

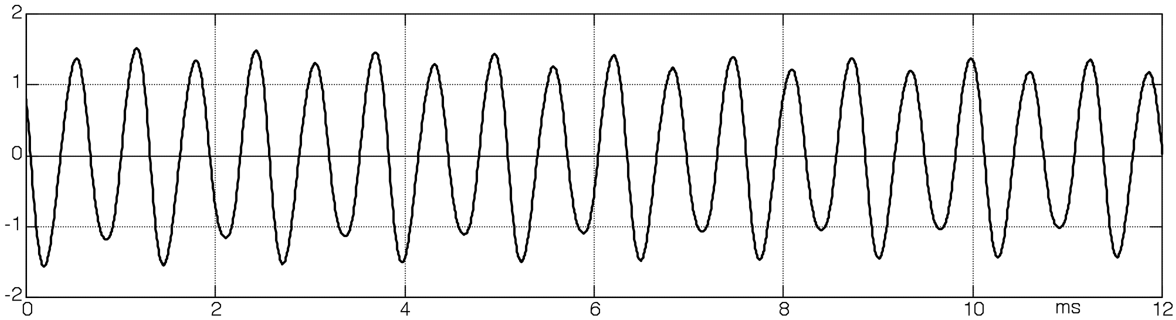

Fig. 11.70: Time function of sound pressure in a sub-harmonically distorted sinusoidal tone: f = 1,6 kHz.

Fig. 11.71 is targeted at showing at showing another example of the complexity of sub-harmonic distortion: above 1.5 kHz, this loudspeaker generates sub-harmonics not only at half the excitation frequency, but – among others – also at $f/4$ and $f/5$ (and at the multiples).

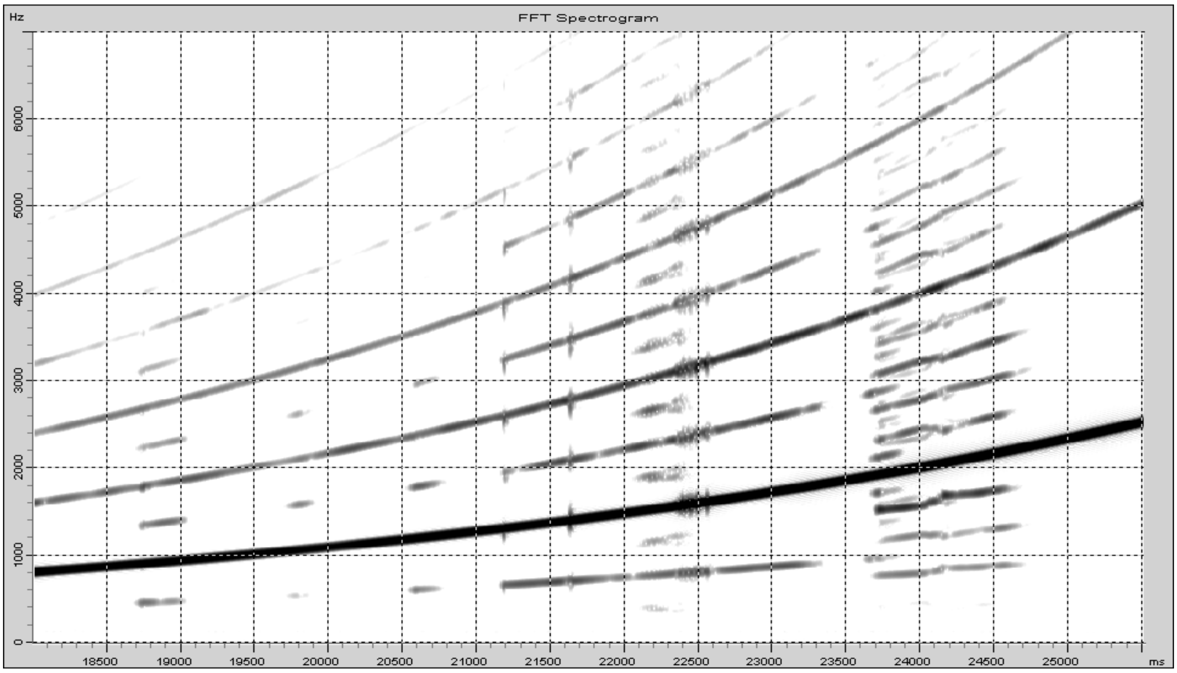

Fig. 11.71: Jensen P12-R, sweep-spectrogram, f = 800 – 2500 Hz, U = 10V.

The **levels of the sub-harmonics** follow their own laws, not even showing the power laws to be expected for standard models. IN **Fig. 11.72**, the level of sub-harmonic (*f/2*) is shown dependent on the level of the primary tone (*f*). For primary tones below a certain threshold (here at just under 108 dB), there is no sub-harmonic at all. Going across that threshold, the sub-harmonic builds up. As we reduce the level of the primary tone below the threshold value, the level of the sub-harmonic remains constant first – only as the primary tone falls below about 104 dB, the sub-harmonic disappears again.

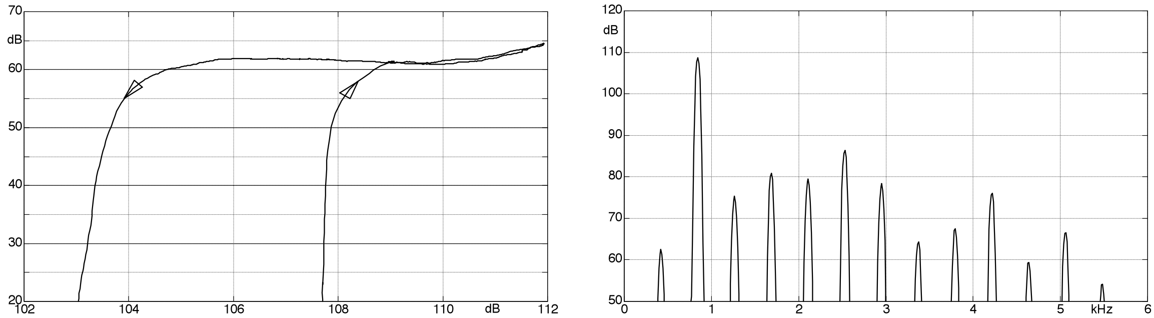

Fig. 11.72: Level-hysteresis (left), distortion spectrum (right). Celestion Vintage-30, f = 844 Hz.

Fig. 11.73 more precisely depicts the evolution of the level for three frequencies. The generator level rises by 25 dB during 30 s: the corresponding measured sound pressure levels are shown. At 1081 Hz, no sub-harmonic is created and the levels grow monotonously. At about 1.3 kHz, however, a sub-harmonic appears around -11 dB (50W / 12.5 = 4 W), and this has effects on all measured sub-harmonics. From -9 dB, there are audible beats.

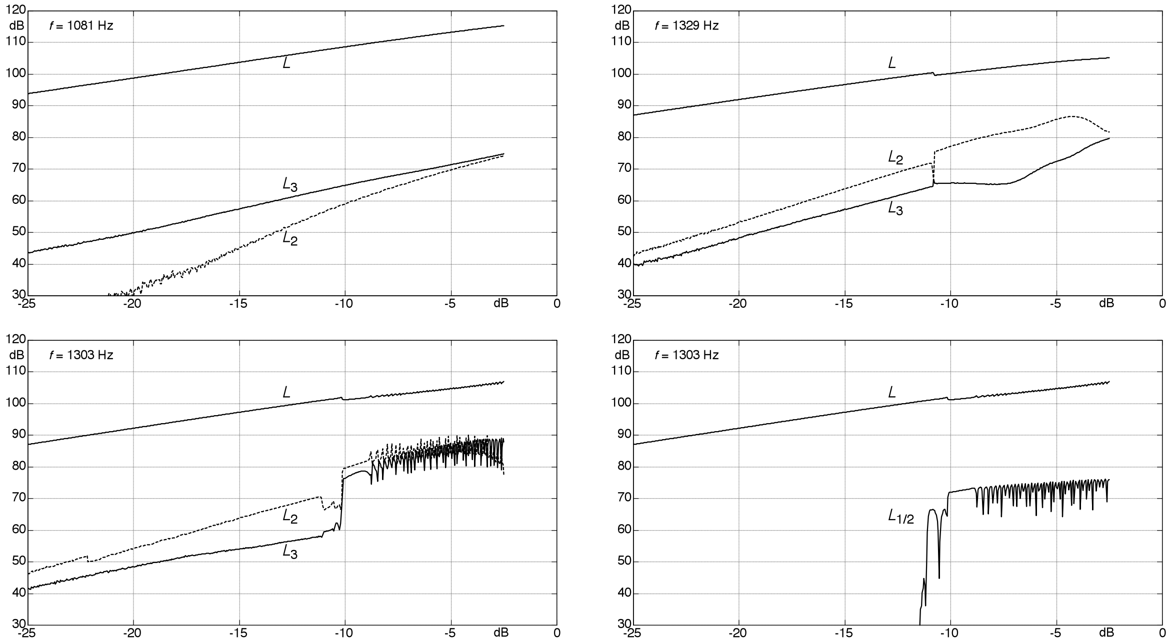

Fig. 11.73: Sum SPL L and distortion level, Eminence L-105; 0dB = maximum power.
Lower left: for f = 1303 Hz, the level development of the sub-harmonic (f/2) is shown.

The following **Fig. 11.74** depicts the non-linear behavior of several loudspeakers in an overview; supplementary measurement data are added in the last of the three diagrams.

Fig. 11.74a: Distortion suppression (harmonic distortion attenuation) a_{k2}, a_{k3} of various loudspeakers. The distortion suppression of the sub-harmonic is, respectively, included at the upper right in the right-hand.

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Fig. 11.74b: Distortion suppression (harmonic distortion attenuation) a_{k2}, a_{k3} of various loudspeakers. The distortion suppression of the sub-harmonic is, respectively, included at the upper right in the right-hand.

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Fig. 11.74c: Distortion suppression (harmonic distortion attenuation) a_{k2}, a_{k3} of various loudspeakers. The distortion suppression of the sub-harmonic is, respectively, included at the upper right in the right-hand. Given the nominal impedance of 8 Ω, the voltage (10 V_{eff} from a stiff voltage source) results in a power of 12.5 W. All measurements were taken in the AEC, with the 12”-speakers mounted in the VOX AD-60VT enclosure, the 15”-speakers mounted in an air-tight enclosure measuring 36x74x40 cm^3, and the 10”-speakers in an air-tight enclosure measuring 39x39x25 cm^3.

These figures are reserved for the printed version of this book.

11.7 Alnico vs. ceramic magnets

Alnico! Guitar players feel that this word connects them to the innermost circle of magic. Pickups? Only those fitted with Alnico. Loudspeakers? Same! For ceramic magnets „don't sound right" … sometime, somewhere, an enlightened shaman has expressed this, and his disciples keep repeating it all over the world. The **Celestion Blue**, "*the world's first dedicated guitar loudspeaker"*, sported – of course! – an Alnico-magnet. As the manufacturer that produces "*the finest guitar loudspeakers that money can buy*", you owe that much to yourself. Give me Alnico or give me death! However, as we take off the rose-colored glasses of the ad-writer, things become less euphemistic: Alnico-magnets were the industry-standard to generate strong magnetic fields. Carbon-steel magnets were produced until 1910 [21], from 1917 there were cobalt-steel magnets, and from the mid-1930's we see magnetic alloys that contain, besides steel, also aluminum (Al), nickel (Ni) and cobalt (Co): AlNiCo-magnets. They appear in many compositions designated with numbers and letters, and even according to prescription, if more precision is required: 8% Al, 14% Ni, 24% Co, 3% Cu, the rest Fe. However, the effect of a magnet is not only the result of the formula – it's the crystalline structure that does it. So if the label says Alnico-5 on two magnets, the impact may still be different. For this reason, there are subgroups such as e.g. Alnico 5-A, or 5-B, or 5-C, 5-7, 5-BDG, 5-ABDG, or whatever their designation may be. To trust the conjecture that there would be a magnet-material named Alnico-5 that generates that wonderful "vintage sound" – that's believing in a fairytale. In fact, there is a multitude of Alnico-5 materials featuring rather different characteristics. We must also not forget that, because of competition amongst manufacturers, we also have Ticonal, Nialco, and Coalnimax. All these materials have a very high remanent flux density of between 1.2 – 1.35 T, and therefore are of excellent suitability for loudspeakers. However, as a side effect of WW II, supply bottlenecks and restrictions on "metals needed for the war effort" with corresponding cost-explosions happened, and so the manufacturers were very happy that low-cost **ceramic magnets** became available as replacements. Guitarists were less happy, because "ceramic does not have the sound of Alnico". Well then, what makes ceramic magnets so distinct over their ceramic imitators?

Assuming the same volume, Alnico-magnets are stronger than ceramic magnets. That is no knock-out criterion, though, because it only pushes up the weight of ceramic-magnet loudspeakers. The flux density in the air-gap is not limited by the magnet (that could be enlarged almost at will) but by the saturation of the field-guiding pole-plates. Considering that, for operation in the optimum operating point, Alnico magnets need to be oblong while ceramic magnets need to be disc-shaped, both materials can serve equally well to reach similarly high flux densities (and flux). Hearsay has it, however, that, as the material became warm during operation of the first speakers fitted with ceramic magnets, the flux density dropped. Indeed, the flux density decreases with almost 0.2% per °C, and depending on the material, 100 °C are not out of the question when pushing the speaker. For voice-coil carriers made of paper, this was kind of a maximum allowable temperature, anyway. However, as high-temperature-resistant materials (Nomex, Kapton, glass fibre) were introduced the maximum temperature for the coil rose to above 250 °C, and at that point it is conceivable that some ceramic magnets had problems. Corresponding difficulties have been largely overcome by now, and industry offers ceramic magnets that tolerate loudspeaker-typical temperatures. Also, the heat generated in the voice-coil does not directly flow fully into the magnet material, and the magnet does not become as hot as the voice-coil. And incidentally, the main allegation towards the ceramic-faction is not that it's weak-kneed but that there's a sound-deficit, somehow, kinda. Alnico has that "vintage" sound, and thus sounds good. Vintage, that's more treble, or (depending on the source) less treble; either way: simply better.

Eminence, *the world's largest loudspeaker manufacturing company, with the finest voice coils in the industry*, explains the Alnico-sound as: *"warm, bluesy tone"*. **Jensen**, on the other hand, *the inventor of the loudspeaker*, sees *"their sparkling trebles"* as the Alnico-characteristic. **JBL**, *world's leading loudspeaker manufacturer*, defines Alnico via "*it's low distortion performance*", **Jensen** does so via *"their dirty midrange"*. Everybody can find his/her own thing with Alnico.

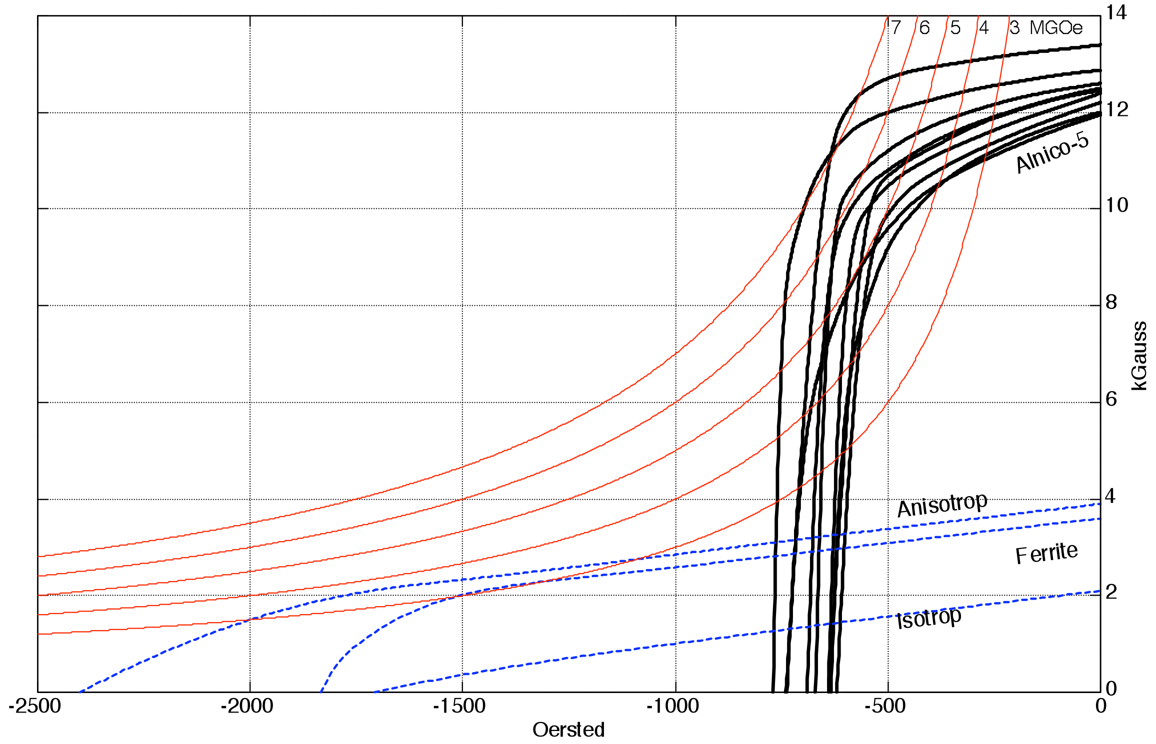

Fig. 11.75: *B/H*-characteristics of various Alnico-5 magnets [22, 23]. 1Oe = 80A/m, 10kG = 1T.

In **Fig. 11.75** we see hysteresis curves of various Anico-5 magnets in comparison to three ceramic magnets. The remanent flux density of regular Alnico-5 magnets is just short of four times that of anisotropic ceramic magnets – conversely, the coercitive field strength of the latter is 3 times larger that that of the Alnico-5 magnets. Again, just to be clear: there is no *one* Alnico-5 magnet nor is there *the* ceramic magnet, and moreover the data on remanence or coercitivity allow merely for approximate conclusions regarding the operating point. A **comparison** can highlight the fundamental differences of the two material groups. A 100-W-lamp (10V/10A) is to be lit up; batteries of 1V/10A and 10V/1A are at our disposal. Whether we connect 10 of the 10-V-batteries in parallel, or 10 of the 1-V-batteries in series does not make any difference to start with – both variants enable the lamp to receive 10V/10A. That does not mean that there are no differences at all anymore: the 10-V-batteries might be a bit more expensive, or larger, or come from a country to which (despite unbelievable successes in sports) economic relations are for the time being uncalled for … anyway, the normative force of facts will call for the 1-V-batteries. The clever businessman will however still try to give the 10-V-batteries a chance, and may for example advertise the source of their energy as “directly from the sun” (sustainability is “in”). He might give 0,5% of the unfortunately 50%-higher sales price back to the estranged country (with the imperative condition that an English-language crèche is financed with the money). This diversification increases the market share, makes for a nicer company-car, and enriches the world by another crèche.

To transfer this scenario to loudspeakers: the volume-specific energy of the magnetic field (the energy density) w corresponds to half the product of flux density B and field strength H. Alnico facilitates higher B-values than ceramic but does not reach the high field strengths of the latter. To compensate, Alnico magnets need to be long and slender, ceramic magnets need to be short and wide. That's just like a series circuit compares to a parallel circuit. Both magnet materials enable the realization of the required specific energy of the magnetic field: ceramic corresponds to the standard, and Alnico sort of corresponds the crèche in the above example.

Why exactly does the ***BH*-product** of a magnetic material have that kind of importance when the definition of the transducer coefficient includes only B but not H? It is true, indeed: the Lorentz-force depends – except for the length of the wire – only on the flux density B. However, in air (as in air gap) the flux density is connected with H via μ_0 such that inevitably a specific H is connected to a correspondingly specific B. The Cu- or Al-winding also located within the air gap does practically not change anything about that – because these materials are not ferromagnetic. The product of the field strength in the air gap and the flux density in the air gap happens to correspond exactly to double the **energy density** w_L of the field within the air gap. With the air-gap volume V_L, the energy in the air gap computes to $W_L = w_L \cdot V_L$. The energy must be made available by the magnet; for the ideal magnetic circuit, this holds: $W_L = w_L \cdot V_L = W_M = w_M \cdot V_M$. Spelled out: magnet-energy = air-gap-energy. Within the magnet, the formula $w_M = 0.5 \cdot B_M \cdot H_M$ holds; consequently for a small volume of the magnet, the BH-product of the magnet needs to be as large as possible. As an **example**: for an air gap of an area of 10 cm^2 and a width of 1 mm, the air gap volume is to 1 cm^3. Given B = 1.5 T, the energy in the air gap is 0.9 J = 0.9 Ws. This value is not directly connected to the sound power to be generated: one may imagine the magnetic field as a kind of catalyst that is necessary but will not be used up. The radiated sound energy is not sourced from the magnetic field but from the electrical energy (fed from the power amplifier). Assuming the BH-product characterizing the magnet to be 45 kJ/m^3 (not untypical for Alnico-5), a volume of the magnet of 40 cm^3 (or a magnet mass of 286 g) results. A ferrite magnet generating only as little as BH_{max} = 22 kJ/m^3 would require 81 cm^3 (or 390 g). This would be the situation for the ideal (i.e. loss-free) magnetic circuit. Alas, this idealization is not even approximately realistic, and so the magnet needs to be bigger: for Alnico 2 – 3 times, for ferrite 3 - 4 times … or still bigger, depending on the individual realization. To achieve comparable air-gap energy, ceramic magnets are therefore larger and heavier than Alnico magnets. Still, any differences in sound or efficiency cannot be substantiated that way.

The energy within the air gap is, however, only a first parameter in the electro-acoustical transducer process. As already shown by Fig. 11.1, the **shape of the magnet** (long/slender vs. short/wide) causes different geometries in the magnetic circuit, and from this shape, two different behaviors may result in dynamic operation (i.e. given current-flow and displacement). It is therefore not sufficient to merely check the static magnitudes in the air-gap – the membrane is to move, after all. Indeed, there is a dynamic magnet parameter showing differences: the so-called **permanent permeability**. In a permanent magnet, it characterizes the B/H-relationship for small shifts in the operating point. For a field-change forced by a current, the operating point does not move along the limit-curve of the hysteresis but within it on a smaller slope. This slope is the permanent permeability, also called reversible permeability. It is about 5 in Alnico-5 and about 1 in ceramic. These data are relative permeabilities, i.e. for *small field changes* the ceramic magnet behaves like air while Alnico is already perceivably ferromagnetic. Globally seen (for large changes in the field), both magnets are of course ferromagnetic, but for differential considerations material-specific differences emerge.

It is, however, insufficient to regard merely a differential magnet-parameter (the permanent permeability) und predict differences in the operational behavior merely based on this. In the respective operating point, not only the slopes of the hysteresis characteristics differ, but the coordinate values, as well. Since, compared to the Alnico magnet, the ceramic magnet features smaller *B* and larger *H*, a kind of transformation needs to be done via an area-reduction: from the wide magnet cross-section to the comparably small air-gap cross-section. This **transformation** will not only adapt *B* and *H* correspondingly, but also the slope of the hysteresis such that the effective permanent permeabilities become closer to each other. Whether they in fact become equal or whether differences still remain, depends on the individual design, and on the all-decisive stray-flux.

This holds for both magnet materials: parameter-variations that may already result from smallish construction-changes are so considerable that is of no purpose to generally speculate about type-specific idiosyncrasies. Rather, **measurements** are to reveal typical differences – if such exist to begin with. One quantity that is easy to measure and that gives indications about differential field changes is the electrical impedance. Its high-frequency increase is determined by the loudspeaker-**inductance**, and thus by the magnetic field. In **Fig. 11.76** we see the frequency responses of the impedances of several 12”-loudspeakers. On the left, only marginal differences show up – although two loudspeakers with different magnets were measured (Celestion “Blue” vs. G12-H). In the right-hand section, however, three Alnico-loudspeakers were analyzed – and specifically here clear differences emerge. Conclusion: there is no special “Alnico-impedance”.

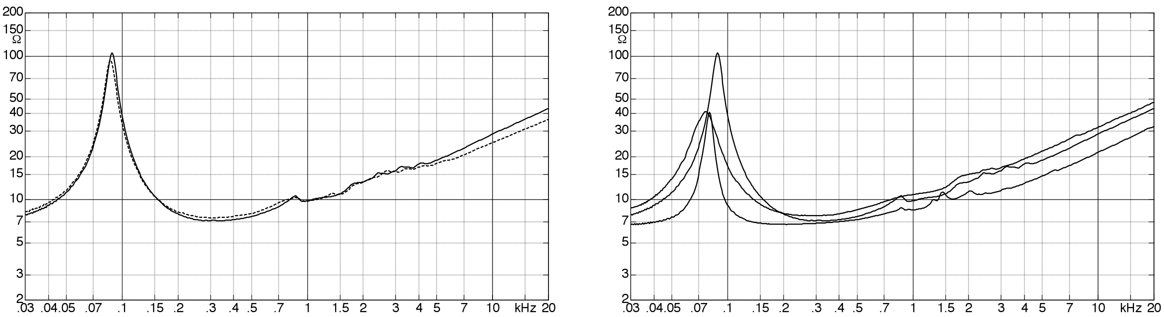

Fig. 11.76: Frequency responses of the impedances of various 12”-loudspeakers (w/out enclosure).
Left: Alnico (——) vs. ceramic (----). Right: 3 different Alnico-speakers (Celestion "Blue", Jensen P12-R, P12-N).

Impedance-measurements will only expose a small-signal characteristic. Loudspeakers will, however, predominantly be operated at large signal levels, with high currents and often close to the power limit. As already shown in Chapter 11.6, non-linear processes step into the foreground here: the voice coil pushes into the fringe-regions of the magnetic field rendering the transducer constant (the force-factor) dependent on the displacement. The membrane-stiffness becomes displacement-dependent, as well, and the inductance shows non-linearity, too. It is certainly possible that the secret of the dearly bought Alnicos lies in the specific non-linearity, and that their **harmonic distortion** shows magnet-typical idiosyncrasies. The magnet should however not be held responsible for any non-linearity of the membrane: that the centering (the spider) becomes progressively stiffer with increasing displacement really has nothing at all to do with the magnetic material. The displacement-dependent inductance, on the other hand, is connected to the magnet, and the signal-dependent transducer constant is, too. Both these non-linearities result from the magnetic field penetrating the voice coil, and because this field is displacement-dependent, the transducer constant becomes signal-dependent. The component of the electrical impedance that stems from the mechanics (that would be everything except the Cu-resistance), consequently becomes non-linear.

A non-linear impedance can be measured by either feeding a sinusoidal current to it from a stiff current source ("imprinting" the current) and measuring the voltage, or by feeding it with a sinusoidal voltage from a stiff-voltage source ("imprinting" the voltage), and measuring the current. The two principles lead to different results because there is no proportionality anymore for non-linear systems. For the following measurements, the voltage was imprinted. Mostly, 10 V were applied, corresponding to a nominal 12.5-W-load for an 8-Ω-speaker. The loudspeakers were not mounted in any enclosure, this leading to larger membrane displacements compared to installation within an enclosure. The harmonic distortion of the loudspeaker current was analyzed, in particular the 2nd- and 3rd- order distortion. It is shown as distortion dampening a_{k2} and a_{k3} (**Fig. 11.77**). 60 dB $\hat{=}$ 0.1%, 40 dB $\hat{=}$ 1%, 20 dB $\hat{=}$ 10%.

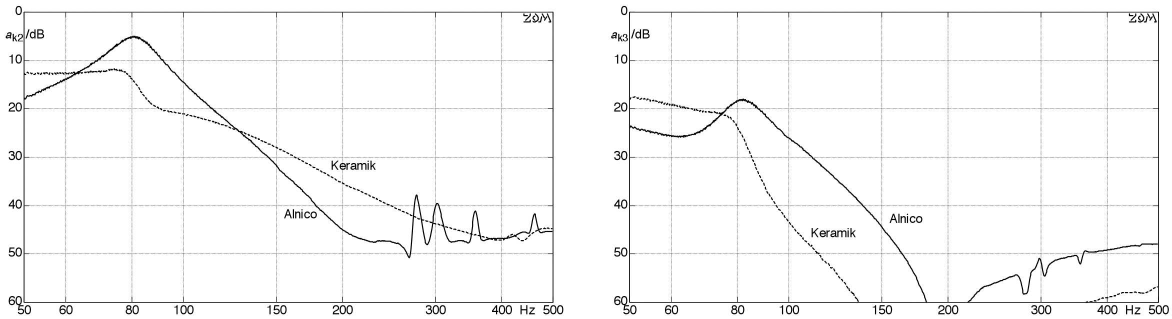

Fig. 11.77: Non-linear distortion of the loudspeaker current for sinusoidal imprinted voltage (10V). Alnico = Celestion "Blue", ceramic ("Keramik") = Celestion G12-H.

In this figure we clearly see significant differences: the 2nd-order harmonic distortion differs by a factor of 3, the 3rd-order distortion even up to a factor of 10! The 2nd-order distortion generally dominates over the 3rd-order distortion, but their frequency dependency differs specifically depending on the loudspeaker. In the range of the main resonance, the Alnico-speaker distorts more than the ceramic-speaker; however, at higher frequencies the differences should be treated with caution. Also, distortion generated by the guitar amplifier – as a rule rather significant – should be considered.

It is only a small step from the measurements shown in Fig. 11.77 to statements such as: **Alnico-loudspeakers distort more than ceramic-loudspeakers**. That is, however, not really entirely correct since from 124 Hz upwards we see the 2nd-order distortion dominating in the ceramic speaker. So, what catchy message should we take home from these measurements? Best would be none – the comparison between two loudspeakers cannot be taken as a significant sample. **Fig. 11.78** offers supplemental analyses: a Jensen C12-N (Alnico) is compared to a Jensen C12-N (ceramic). Now, suddenly, the situation is reversed: the k_2 of the Alnico-speaker is smaller than that of the ceramic speaker.

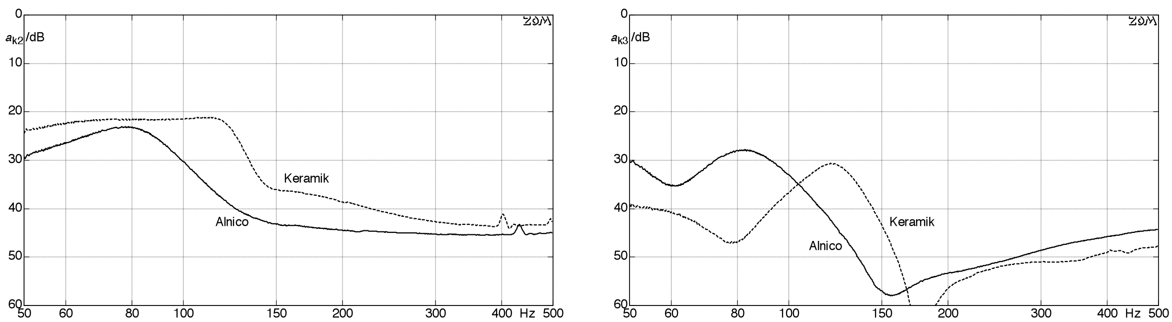

Fig. 11.78: Non-linear distortion of the loudspeaker-current given sinusoidal imprinted voltage (10V). Alnico = Jensen P12-N, ceramic ("Keramik") = Jensen C12-N.

In fact, the two **Jensen-loudspeakers** (Fig. 11.78) represent an ideal pair: both are sourced from the same manufacturer, both have a 12”-diameter, both play in the same power-league: 50 W, 1,5"-voice-coil. Only the material of the magnets is different: ceramic (C12-N) vs. Alnico (P12-N). O.k. – the price is also different … we understand: the expensive cobalt. What some of us do not understand: why do the resonance frequencies of these two speakers (bought at the same time) differ, too? By 56%, after all – specifically 120 Hz (C12-N) vs. 77 Hz (P12-N). Don’t start with “the magnet change might retune the resonance” – the mechanics do not require any magnetic field for that. At least the stiffness of the membranes is very different, as a simple push with the fingertip confirms. So, there’s not just another magnet included, but the membranes are entirely different, as well! One can imagine the kind of “wisdom” that results if, after comparing these two loudspeakers, musicians post their findings on the internet. Without a doubt, there are type-specific differences in the non-linear behavior of the loudspeakers, but it is not possible to derive any Alnico-specific characteristic from these.

Checking out two **Celestion loudspeakers** may serve as a counter-example to the above comparison: Vintage-30 (ceramic magnet) vs. “Blue” (Alnico-magnet). **Fig. 11.79** indicates the corresponding comparison-analyses. Up to 250 Hz, we in fact recognize merely a slightly different resonance frequency in the k_2, and in the frequency range above the effects of the modes of partial oscillations can be seen. In the k_3, the differences are somewhat larger but by no means classifiable as a characteristic. But now it gets really interesting: in the second row of the figure, two Alnico loudspeakers, specifically two Celestion “Blue” bought at the same time, are compared. The differences between these two speaker-specimen (both Alnico, both of the same construction!) are, as a whole, larger than the differences between the differences found between the Alnico- and the ceramic-speaker shown in the upper row in the figure!

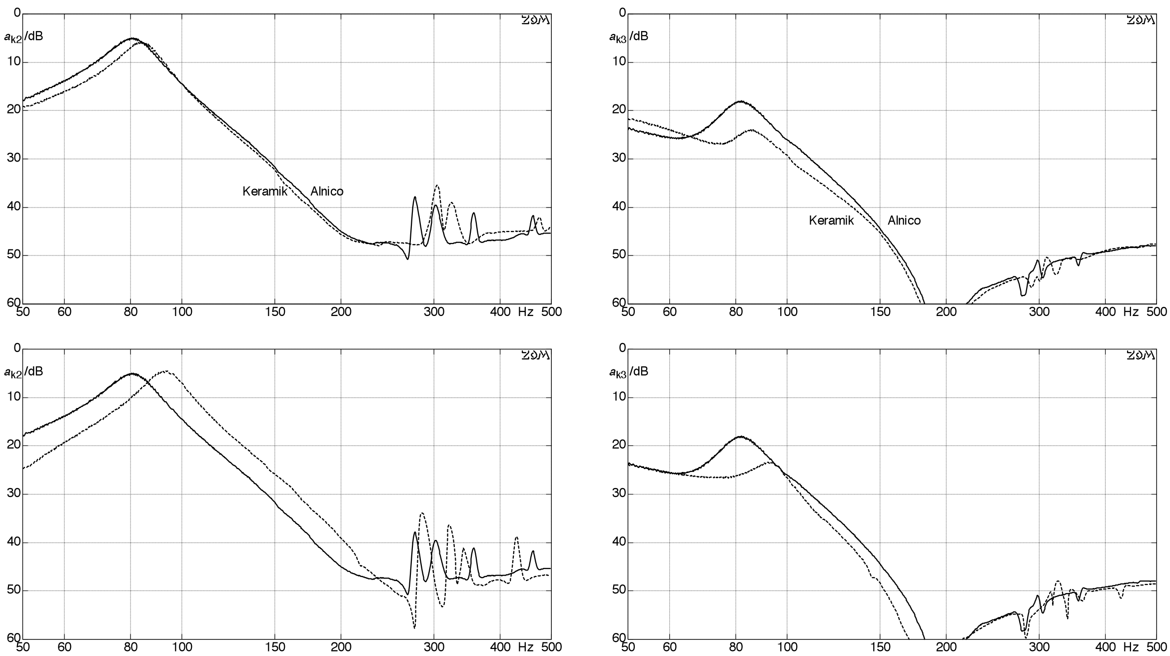

Fig. 11.79: Non-linear distortion of the loudspeaker-current given sinusoidal imprinted voltage (10 V).
Alnico = Celestion "Blue", ceramic (“Keramik”) = Celestion Vintage-30.
Lower row: two Celestion "Blue" specimen.

To conclude these measurements, **Fig. 11.80** shows comparisons across 4 Alnico- and 5 ceramic-loudspeakers. Again, the effects of different membrane-suspensions dominate, while an “Alnico-characteristic” is nowhere to be found.

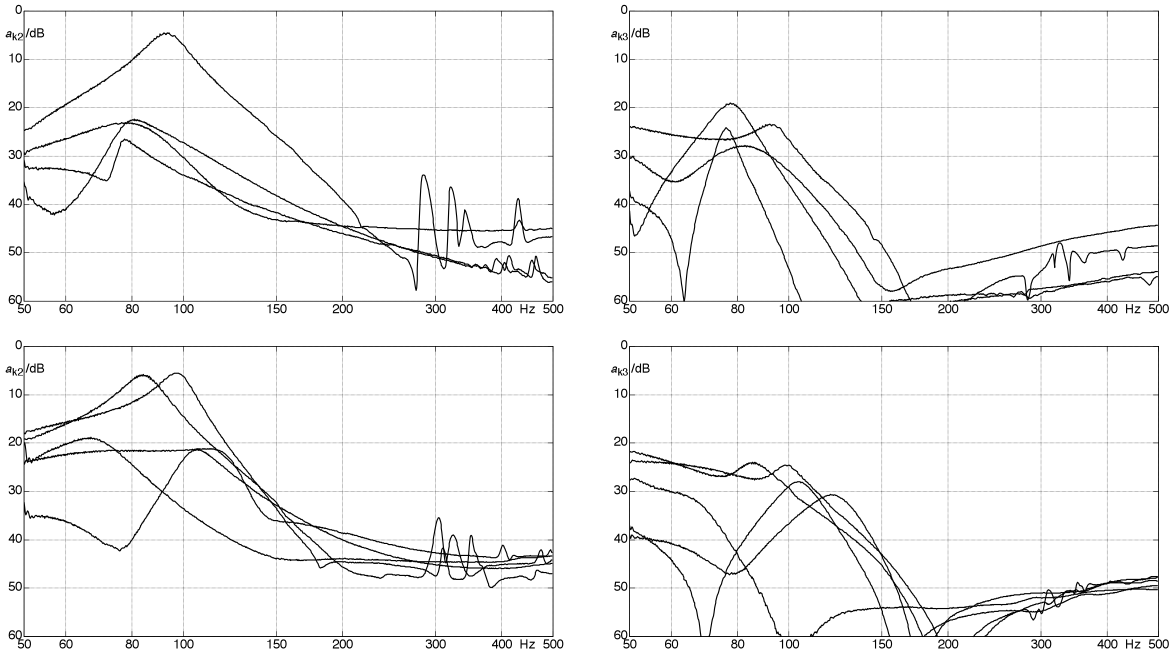

Fig. 11.80: Non-linear distortion of the loudspeaker-current given sinusoidal imprinted voltage (10 V).

After this analysis of the electrical two-pole parameters we of course need to pay tribute to the transmission parameters. After all, the meaning of life for the loudspeaker is not just to offer a load to the amplifier – it is supposed to radiate sound. Still, the trend found in the distortion measurements continues here (**Fig. 11.81**). The differences between Alnico-speakers of the same type are similar to the differences between Alnico-and ceramic-speakers – there is nothing whatsoever to be found that could be interpreted as a magnet-specific sound. That does not mean at all that using Alnico-speakers is pointless. Jensen and Eminence, for example, do not offer an immediate ceramic-alternative to the P12-N and the "Legend 122", respectively. If you want to have the sound of these legends, you will have to buy them – the C-12N and the “Legend 125”, respectively, differ in more than just the magnet. With Celestion, the situation is different: a serious alternative to the Celestion “Blue” stands ready in the form of the Vintage-30, with the latter having four times the power capacity but still costing only one third of the former – or even only one twelfth, if you calculate per watt. However, the flair surrounding “the Blue one” is so attractive that there is no cure for its lure. And so there will always be true-Blue devotees to the brilliant (or soft) and the dirty-distorted (or distortion-free) sound of the Alnicos.

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Fig. 11.81: Differences in SPL between two specimens of the Celestion “Blue”, and between the “Blue” and the Vintage-30. Measured in the AEC; 1W @ 1m; speaker mounted in the enclosure of an AD60-VT.

11.8 Loudspeaker enclosures

11.8.1 Basics

Often, cases guitar-amplifier and -speaker are mounted within the same enclosure (combo); alternatively, there is also the two-part piggy-back or stack design. From the multitude of sizes available on the market, **Fig. 11.82** shows a small selection: predominantly, 10”- and 12”-speakers are found, occasionally also 15” (with 1” = 2.54 cm). The small combos almost always have a large opening in the rear while the larger enclosures are either of closed design or realized as ported box (bass-reflex). In the widest sense of the word, the enclosures open to the rear also represent a kind of bass-reflex system – albeit a very special one.

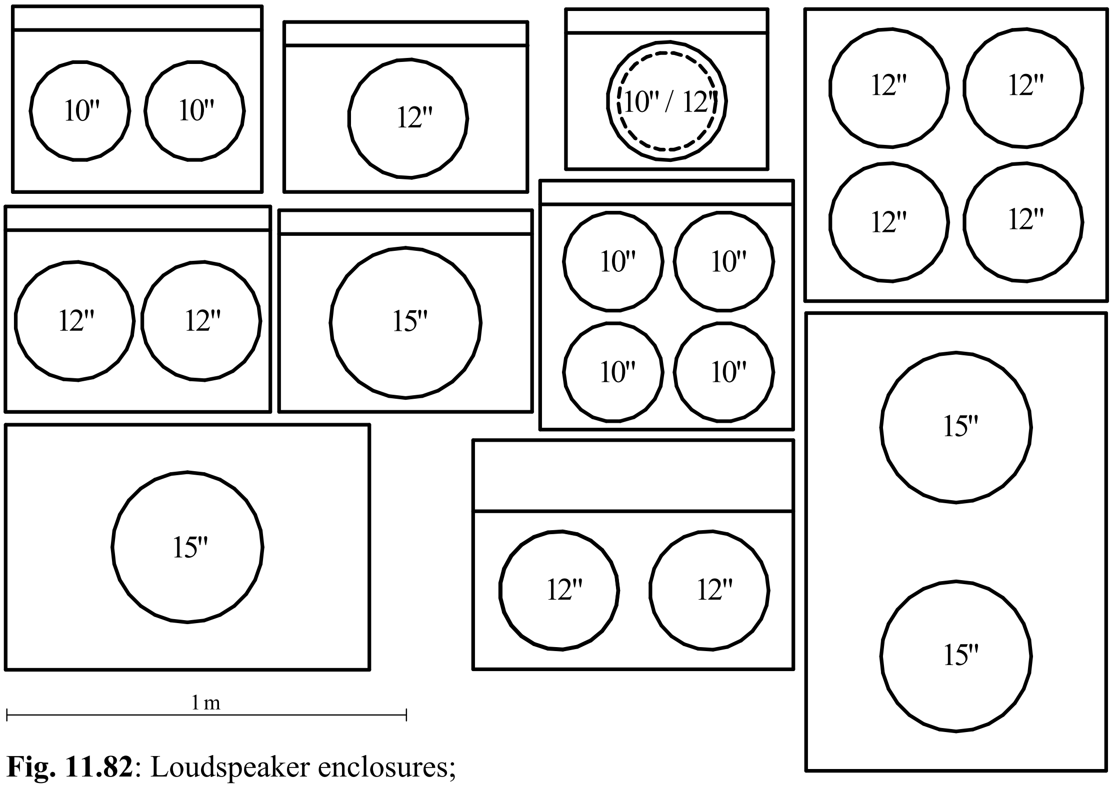

Fig. 11.82: Loudspeaker enclosures; Membrane-diameters in inches

The enclosure (or cabinet) makes a significant contribution to the sound generation. If it is airtight, it predominantly has the effect of an air-suspension to the membrane that increases the resonance frequency. Since this air-stiffness grows invers to the volume, a small enclosure would strongly increase the resonance frequency – it is presumably for this reason that small cabinets mostly have an open back. The **stiffness of air** is $s_L = 1.4 \cdot 10^5 \text{ Pa} \cdot S^2 / V$ for adiabatic changes. In this formula, S is the effective membrane surface, and V is the net volume of the enclosure. For a 12”-speaker and a 50-litre-box, we calculate 9179 N/m – this approximately corresponds to the stiffness of the membrane. As an example: with such a mounting, the resonance frequency for a Celestion Blue would rise by 50%. However, the enclosure acts as an air-suspension only at low frequencies; from about 300 Hz, **standing waves** establish themselves in the interior – these represent a complex, frequency-dependent load for the membrane. The effects of such cavity-resonances could easily be mitigated via a porous absorber loosely placed into the enclosure, but this approach is not normally taken with instrument loudspeakers. For one, these resonances liven up the sound, if they are the correct ones, and second, because any absorption kills off sound energy (transforming it into heat). Since guitar-speakers by their nature need to be loud, absorbers are normally eschewed.

Very fundamentally, loudspeaker enclosures may be divided into *open* and *closed* cabinets. The **closed cabinet** acts as an air-(suspension)-spring towards the backside of the membrane for low frequencies: as the membrane moves into the interior, the pressure therein rises. Taking adiabatic changes as a base, $p \cdot V^{1.4} = const$ holds. An **example with numbers**: as the 12”-membrane moves 2.5 mm into the interior of a 38-liter-cabinet, the volume decreases by 0.5%, resulting in an excess pressure of 670 N/m^2. This causes a rear wall with an area of 0.18 m^2 (38cm x 48cm) to receive a force of 121 N. In layman’s terms: about 12 kg push against the rear wall. That corresponds to the weight (or the mass) of no less than three Celestion “Blue”! It is immediately clear that such a wall needs to be attached firmly and must not be too thin. If, however, the rear wall has an opening (or consist of two sections with a gap in between), as is the normal case for small cabinets, then the “excess pressure can be vented”, and the forces acting onto the cabinet walls are significantly smaller (at most a tenth). The open cabinet is barely strained by the sound pressure and therefore the material used makes (acoustically) no difference.

Sure: there are Leo Fender’s pine-crates and their unique sound. *Though this be madness, yet there is method in it …* so teaches us Shakespeare. The probably impossible-to-silence legend tells us that that a guitar combo needs to by crafted from finger-jointed **pinewood** with glued-on “tweed”. No, it needn’t. Of course, there are sound sources the sound quality of which depends on the utilized wood – the acoustic guitar is a good example. However: would you use pinewood? Never. The HD-28 made of pine, or the big Guild? No way, definitely NO way, at all! The Stradivari? Come on! Maple, that’s a tone-wood, spruce as well – cedar, too. Not pine, though. Pinewood was available on location, it was inexpensive, it was easy to process. Moreover, Leo Fender was not a luthier – he was trained as bookkeeper. In an acoustic guitar, the body needs to vibrate in order to radiate sound. That may be another reason why it is not plastered with tweed or Tolex. Also, the walls of an acoustic guitar are not half an inch thick –indeed there seem to be fundamental differences. In a guitar combo, it is the loudspeaker membrane that vibrates – it does generate the sound. Without a doubt, the cabinet acts as an acoustic filter; that, however, is due to the dimensions and not due to the material. While the cabinet is made to vibrate by the sound the speaker generates, the corresponding effects are, for the most part, entirely negligible relative to the membrane vibrations.

To list the most important impacts of the cabinet: it operates as a conduit to the sound, it makes for the formation of cavity resonances, and it (mechanically) supports the loudspeaker. To the latter characteristic we may attribute a **mechanical impedance** against which the loudspeaker braces itself. If this impedance is infinitely high (huge mass), the loudspeaker frame mounted to it cannot vibrate. Of course, the cabinet does not have an infinitely high mass, and therefore there will be a small movement at the interface between baffle and speaker-frame. Mechanics teach us: *Actio = Reactio*: the force acting on the membrane is just as big as the counter-force acting on the speaker frame. But let’s think for a moment: the membrane has a weight of maybe around 30 g, while the speaker weighs in 3 kg – or up to 10 kg for some US-made muscle. Doesn’t the tail wag the dog here? Even if the speaker remains *un-mounted*, the sheer mass of it will prohibit any significant movement of the speaker frame. Okay, there may be some resonances where a small cause may escalate to have a big effect. As is so often the case, measurements clarify the situation: with a laser-vibrometer, it is easy to target a point on the loudspeaker frame or on the cabinet, and to measure the speed of oscillation – also termed (particle-) velocity. Reference for these measurements is the velocity of the membrane: we find it to move with 1 m/s, while the cabinet wall shows 0.01 m/s. We have thus verified that the cabinet does vibrate – but there is not relevance to this vibration when considering the sound of the amp.

Fig. 11.83 depicts cabinet-vibrations (----) in comparison to membrane-vibrations. The sidewall of a Tweed Deluxe (fitted with P12-R speaker) vibrates with considerably smaller amplitude compared to the membrane; merely at 440 Hz there is a noteworthy maximum – but even here, the vibration of the wall is merely one tenth of the membrane-vibration. It therefore remains negligible. The baffle (right hand part of the figure) vibrates more strongly than the sidewall – which is not surprising given its spartan mounting and a thickness of merely 9 mm. Still, big effects may not be expected: the radiated sound power is based on the *square* of the velocity. Assuming an equal area, a difference in the velocity-level of 20 dB results in a power difference of 1:100 in favor of the membrane.

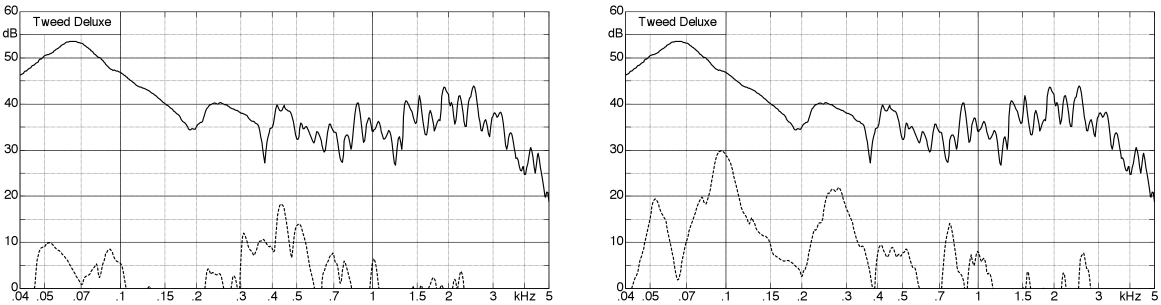

Fig. 11.83: Membrane-velocity in comparison to the velocities of cabinet sidewall (left), and baffle (right).

These results are supplemented via the velocity levels for membrane and loudspeaker frame as shown in **Fig. 11.84**; the measurement point was at the mounting ring between two screw-holes. The speaker frame does vibrate, no contest there, and is influenced by the equipment-feet of the combo. For the measurement shown on the left, the combo was set onto a stone-table without feet, while for the one on the right, it sat on its factory-fitted rubber-feet. The impact on the speaker-frame is clear, while that on the membrane is just about visible. If we would attribute any significance to such small effects, we would also need to specify the mechanical point-impedance of the combo-base. For the musician’s everyday life, however, it does not play a big role whether the combo is placed on a stool or on a beer-crate. In case we would consider that, we first would have to specify the height above ground: whether it is 45 or 50 cm makes a huge difference due to the resulting comb filtering. In theory, anyway: most guitarists don’t really care as long as the thing won’t topple over. Still, knee-deep in all this scientific stuff we almost forgot: *any combo made of pine can’t be kept from sounding fine …*

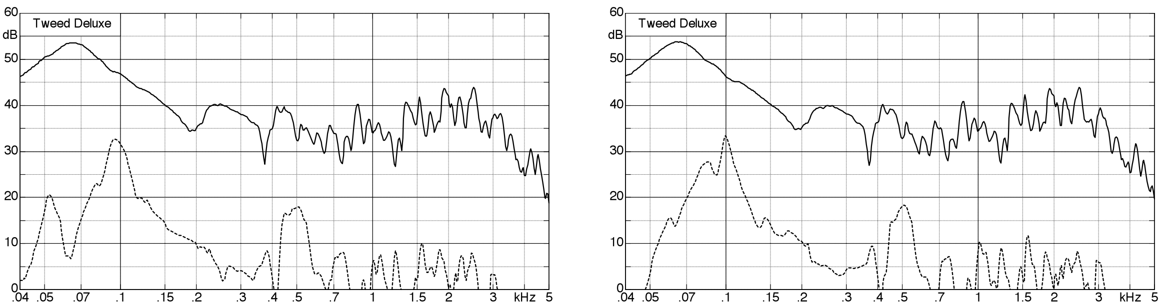

Fig. 11.84: Membrane-velocity in comparison to the frame-velocity for two different cabinet foundations.

Conclusion: Only the super-thin baffles of early Tweed amps could be seen as having a marginal influence on the sound, if any. For enclosures with regular wall-thickness (2 cm), only the dimensions and the resulting cavity resonances are of significance. It is irrelevant which wood is used – the cabinet-vibrations are of secondary importance compared to the membrane vibrations (see also Chapter 11.8.2).

Let us now consider the **closed cabinets**, with their probably most prominent representative being Marshall's 4x12"-box. The possible amount of the sound pressure inside a closed cabinet has already been elaborated, and also which forces can act onto the cabinet walls. It is good practice to build such enclosures to be very stable and maintain their shape, and to bolt down the back panel using a substantial number of screws. It is also wise to insert one or even two internal braces. A thin, strongly vibrating back panel will deprive the loudspeaker of vibration energy, without re-radiating much of that energy but converting most of it into heat. Such a back panel is, indeed, not a membrane suspended in a flexible surround, but needs to bend to vibrate – this generates much inner friction i.e. useless heat. Not so much that the box would burst into flames – the heat energy does not reach that kind of level. However, it is energy that is lost to the generated sound. Of course, it is conceivable to design special cabinets with a back panel that will dissipate exactly those sound energies that would lead to atrocious sound ... but that would lead us astray from the beaten track that the sacred cows travel on …

A small detail that keeps on being discussed when it comes to sealed enclosures is the air-tightness (or lack of it). How leak-proof is the cabinet without a leak, actually? You get the full bandwidth from "seal it all off with silicone" to " leave a clearance of 1 mm all around – otherwise it gets jammed". In the simple model, a leak (a gap) is an acoustic filter: in conjunction with the radiation impedance, the air within the gap forms a mass, and the air within the enclosure acts as spring. That's your ready-made **2nd-order low-pass**: spring and mass combined generate a resonance; for excitations below the resonance frequency, the gap is open, and for frequencies above the resonance it is closed because here the inertia of the air prohibits stronger movements. The **ported-box (bass-reflex) enclosure** takes advantage of the same principle; it belongs to the "leaky" cabinets [3]. As an example: connecting a 1.5-V-battery to the loudspeaker will (almost) abruptly change the air pressure in the enclosure. However, depending on the polarity, air immediately starts to flow through the gap into or out of the enclosure, and the pressure balances itself out. When exciting the membrane with higher frequencies (e.g. with 1 kHz), the pressure cannot even out quickly enough due to the mass inertia, and the enclosure operates as if no gap at all were present. If a cabinet has little leakage, this can manifest itself as an effect only at low frequencies. The smaller the area of the gap, the lower is the frequency range in question. Since speaker-boxes for guitar do not have to reproduce frequencies down to 20 Hz, the requirements regarding their air-tightness are not very stringent. We get some orientation-values from **Fig. 11.85**. This diagram does not consider that the stiffness of the membrane-suspension and of the enclosure-walls can have an effect on the resonance, and that a considerable flow-resistance occurs in particular in narrow gaps (slits). Still, the figure is useful to approximately estimate the effects – in practice, a simple impedance measurement will deliver data about the actual resonance. An example is shown in Fig. 11.86: a weak leakage resonance occurs at 33 Hz.

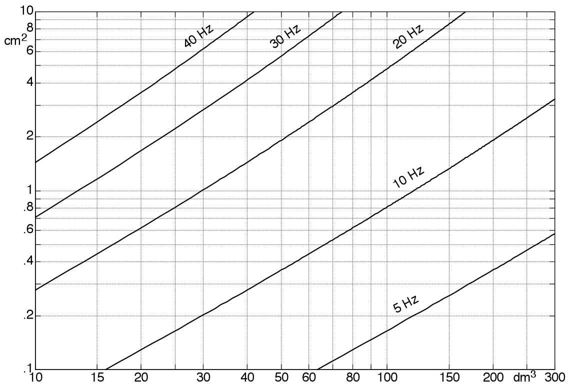

Fig. 11.85: correspondence between volume of the enclosure (abscissa, dm^3 = liter), area of gap (cm^2) and resulting resonance frequency.
Example: a 100-liter-cabinet has gaps with a total surface of 5 cm^2 – this results in a leakage-resonance at about 20 Hz.

The shift in the resonance caused by the air-stiffness can be clearly seen in **Fig. 11.86**. The impedance plot furthermore reveals the primary cavity resonance (260 Hz) that forms as longitudinal $\lambda/2$-oscillation within the enclosure (70 cm length). From this frequency, the enclosure volume does not act anymore as concentrated stiffness but as a continuum. A mass (about 12 g) and two springs of approximately equal stiffness can form a simplified model of this lowest cavity resonance. Transformed to the electrical side, the impedance of this analogy corresponds well with the measurement. The differences occurring above 300 Hz are mainly due to membrane-resonances (partial oscillations).

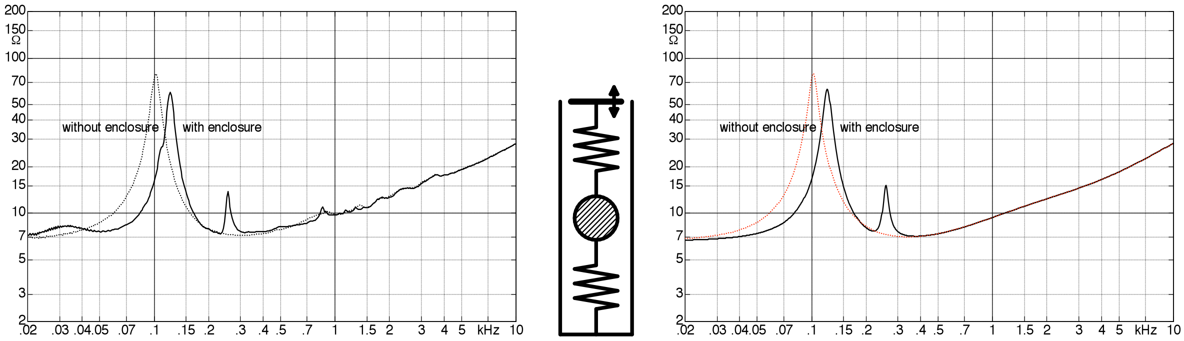

Fig. 11.86: Frequency response of the G12-M impedance (see also Fig. 11.17); measurement (left), model (right). Middle: simplified longitudinal-oscillation model of the $\lambda/2$-resonance.

The cavity resonance makes itself felt in the transmission frequency-response via an S-curve (**Fig. 11.87**). Just below this resonance, the efficiency deteriorates, and just above it there is an improvement. The loading from the rear of the membrane has effects onto the radiation occurring on the front, as well – this is easily explained by the relationship between source- and load-impedance. Driving the speaker from a stiff voltage-source, the electrical source impedance of the FI-transducer [3] is the ohmic voice-coil resistance (below 1 kHz we may disregard the inductance). The source impedance of the mechanical side of the transducer therefore is a purely ohmic resistor. This resistive source is loaded by several mechanical components: the membrane, the (inner) cavity resonance, and the (outer) radiation impedance. The three impedances need to be added up resulting in an overall impedance, and therefore each of the three will influence the matching. Bold and simple: even if you secure the membrane only from the *inside*, it still cannot radiate any sound on the *outside* anymore. The frequency-selective change of the matching lies at the core of the function of any reactance-filter – as such the S-curve is no surprise. That the effect onto the electrical impedance frequency response partially is rather small may be traced to the relatively low efficiency: the ohmic voice-coil resistance dominates (possibly together with the voice coil inductance) the electrical impedance. Of course, resonances pronounced to that degree are indeed audible; whether they sound good or bad is – as always – a matter of personal taste.

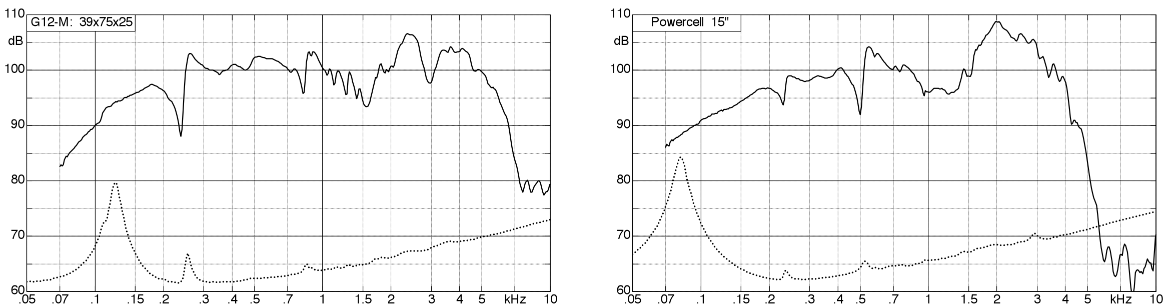

Fig. 11.87: Transmission frequ. response: enclosure with cavity resonances. 39x75x25 (l.) and 40x74x39 (r.).

As the preceding chapters have shown, loudspeaker and cabinet act as two filters connected to the output of the amplifier. A signal tapped from the input of the power amplifier misses just that filtering. That is why we most often find microphones in front of the loudspeakers – to record or mix instrument-specifically. However, in the **near field** of a sound source with a relatively large area there may be transfer functions that depart significantly from the far-field characteristic. Strictly speaking, the border between near-field and far-field is assessed according to the size of the loudspeaker enclosure, but as a simplification to start with, we may use the size of the membrane as a criterion: if the distance between microphone and speaker is only about as big (or even smaller) than the diameter of the membrane, then the microphone is located within the near-field. If more than one speaker is mounted in the enclosure, the diameter of the equivalent membrane must be considered. For a typical 4x12”-box that would mean not merely 28 cm but already almost 1 m. Customarily, microphones are positioned more closely to the box, i.e. within the near-field.

To model sound-radiating surfaces (e.g. membranes), they are divided up into small partial areas each radiating spherical waves (according to Huygens’ principle). **Fig. 11.88** points this out using the example of a plane membrane (on the left side of the figure). For a point infinitely far away, the outgoing sound rays are travelling in parallel and the individual sound-paths are of equal length. The closer the measuring point gets to the membrane, the more unequal the sound-paths become, resulting in different delay times between the sound rays. This has no bearing for low frequencies, but for higher frequencies the different path-lengths may be equal to half a wavelength, and interference cancellations will then happen.

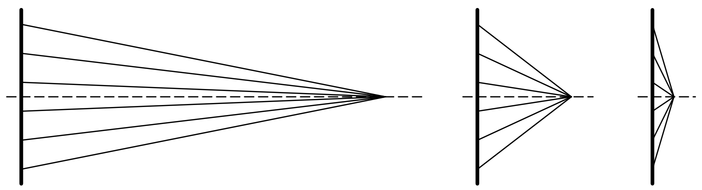

Fig. 11.88: The closer the microphone gets to the membrane, the more the individual sound paths differ in length.

Fig. 11.89 shows the effects of such interferences, measured in the AEC with a Tweed Deluxe. As the microphone approaches the centre of the membrane axially from a larger distance, the SPL increases. This does not happen for all frequencies in the same way, however! Since the absolute sound pressure levels are not as relevant here, a constant was subtracted in the diagrams such that values floating around 0 dB result. As the speaker approaches, predominantly the low frequencies are emphasized, and furthermore other frequency-selective filtering occurs. A condenser microphone with an omni-directional characteristic was employed; for directional microphones, an additional proximity effect needs to be considered [3]. If the loudspeaker is not positioned in the AEC but on a reflecting surface, environment-dependent comb-filter effects weigh in as well.

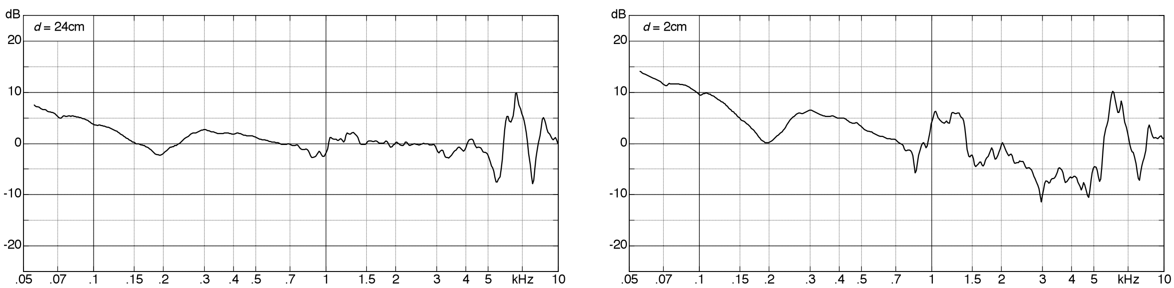

Fig. 11.89: Tweed Deluxe, axial near-field measurements; normalized level changes relative to d = 1m.

Even when keeping the distance between microphone and loudspeaker constant and "merely" adjusting the lateral location in front of the speaker, the transmission frequency response changes, as **Fig.11.90** strikingly proves – it shows the difference in SPL as the microphone is repositioned by the given offset**.** The left-hand diagram was recorded from a Tweed Deluxe, the right-hand one from a 2x12"-box. It is understood that such significant differences have a dramatic effect on the sound. Consequently, the choice of the microphone position may possibly by more crucial than the choice of loudspeaker! The directionality of the microphone will also influence the sound: if the mike is positioned directly at the cloth protecting the speaker, sound waves from different membrane areas arrive from different directions. However, if the microphone is located (in the recording studio) at a greater distance from the loudspeaker, it will record sound reflected by the room in addition to the direct sound.

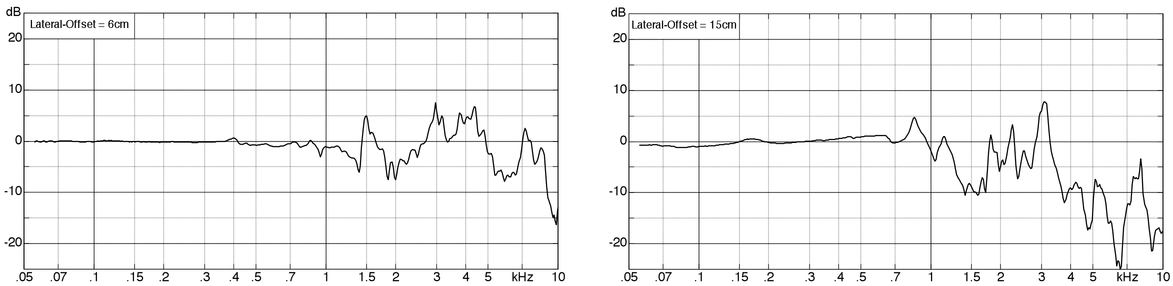

Fig. 11.90: Level changes with repositioned microphone; const. distance to the baffle. Left: 1x12"; right: 2x12"

Given the results of these measurements it is understandable that musicians and studio-experts pay the utmost attention to the choice of the microphone and to its position. Often even two or three microphones are deployed to record a guitar speaker, with small markers on the speaker cloth supposedly guaranteeing the retrieval of the magic spot. Global rules such as "microphone distance = membrane diameter", or "microphone distance = 3 x membrane diameter", or "don't point the mike to the center of the speaker but to a point half the distance between dust cap and rim", are well meant but must never be generalized. What sounds good with one speaker can be utterly unsatisfactory with another – individual tuning is required.

In order to avoid the not insignificant effort of bringing (besides the guitar) the whole amplification equipment, setting it up, and painstakingly finding the right microphone position, many musicians (and producers ☺) often opt for a radically simpler approach. The guitar is plugged into a "**modeling amp**" that takes care of all necessary linear and non-linear filtering. By now word has spread that this also includes the filtering contributed by the loudspeaker. Daily studio-practice shows that this route makes it possible to generate nightmarishly artificial guitar sounds, but it also proves that impressively wonderful results can be achieved – which afterwards need to be camouflaged with fake-evidence ("*even in the tile-covered bathroom, we had a '64 Blackface fitted with NOS-tubes and miked up with three condensers"*) to be able to survive in a world of vintage-craziness. That such a modeling amp, good or bad, cannot emulate the directional characteristics of its paragon has been already noted on Chapter 11.4. Also, if an amp with a high-gain-sound is to be emulated: the feedback onto the guitar (that can support the ringing of the strings or even generate ringing by itself) is missing if the guitarist only uses headphones to hear himself play. Of the multitude of modeling amplifiers (whether with or without power amplifier), we selected the **POD 2.0** made by Line-6. This is not meant to give a rating in terms of *particularly good* or *particularly bad* – the device simply was easily available.

Fig. 11.91 shows the frequency responses of the loudspeaker emulation of the modeling amp (Line-6 POD 2.0) Unfortunately, the manual does not give any information about the virtual microphone position.

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Fig. 11.91: Loudspeaker-emulation in the Line-6 "POD". The absolute ordinate-scaling is arbitrary.

The **VOX AC-30** will serve to find out which effects small details in the enclosure can have on the sound. In this combo, the amp is located together with the two 12”-loudspeaker in one cabinet; merely a lateral board separates the two sections (**Fig. 11.92**). However, there is a special variant, the “AC-30 Super Twin” in which the amp is afforded a separate cabinet (i.e. it’s a piggy-back design), and therefore the separation-board in the loudspeaker enclosure (with otherwise identical dimensions) is omitted. At first glance, there are thus two different enclosures: with and without amplifier. However, the separation-board in the regular AC-30 features a rather big opening in order to allow, via a kind of chimney effect, cooling air to get to the amplifier positioned above the board. This air escapes through vents on top of the enclosure – accompanied by sound, of course: because where there is an airflow, sound will also pass. The dimensions of the vents have definitely changed over the years – whether the opening in the separation-board is subjected to the same “time-variance” was not investigated.

For the transmission behavior this implies that not only do we need to pay attention to the fitted loudspeakers but also to the cabinet-design. The electrical loudspeaker–impedance changes as the separation-board is removed, or as the air vents are changed. Fig. 11.92 shows that in the range of 100 – 300 Hz the impedance may vary by a factor of 2. And since the tube output stage of the AC-30 lacks any negative feedback and therefore has a high output impedance, this impedance change makes itself felt in practically the same magnitude in the transmission frequency response. The impedance maximum at 170 Hz, for example, has the same effect as if we had boosted a narrow band around this frequency by 6 dB with an equalizer. Here, we find an interesting parallel in the area of acoustic guitars: measurements performed by Fletcher and Rossing [1] with a Martin D-28 show a strongly pronounced resonance at 200 Hz in the sound spectrum. Possibly, the selective emphasis of this frequency range positively influences the sound of acoustic and electric guitars.

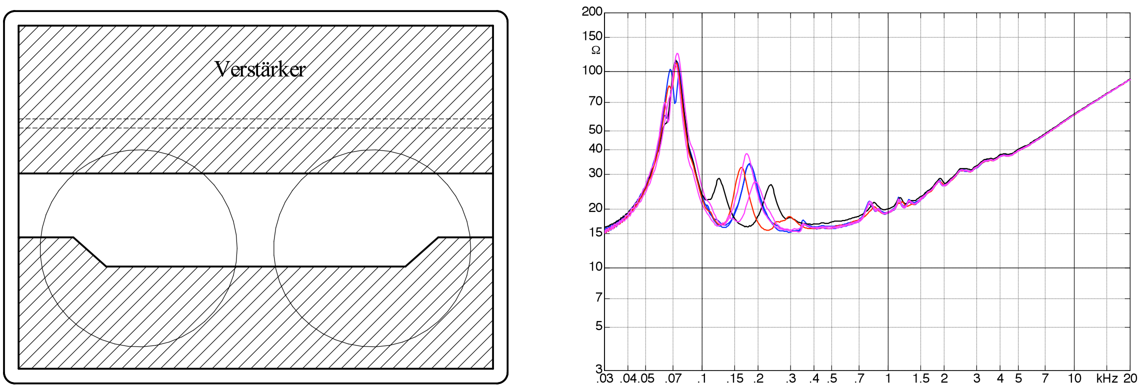


Fig. 11.92: Left: 2-part rear-panel of the VOX AC-30 (“Verstärker” = amplifier). Right: frequency responses of the impedance (series connection of 2 speakers). The different impedance curves correspond to modifications in separation-board and air vents.

11.8.2 Comparison of various enclosure-materials

An often-asked question: what is the contribution that the type of wood used for the enclosure makes to the sound (i.e. to the transmission function) of the loudspeaker? Dealers point to historic models, and attribute to the wood a significance similar to that it would have for Italian master-built violins – and the musician believes it and shells out the money. To go beyond assumptions and obtain some objective data, we analyzed a number of cabinets of identical dimensions but made out of different woods: pinewood (18 mm), poplar (14 mm), and medium-density fiberboard (MDF, 14 mm). The enclosures were carefully assembled by Tube-Town (www.Tube-Town.de) and were measured with the same loudspeaker installed (Eminence MOD-12). The external dimensions were 50 cm x 41 cm x 30 cm. The *sealed enclosures* were closed off to the rear with a non-reinforced panel while the *open cabinets* featured two boards to the rear that had a gap of 13 cm between them.

All measurements were done in the anechoic chamber at 3 m distance on axis. The speaker was fed from a stiff voltage-source (2.83 V at first, later more); a B&K 4190 served as measurement microphone,. The resulting frequency responses of the SPL (recalculated for 1 m distance) are shown in **Fig. 11.93**. There are visible differences between the wood-types, but they are so small that they will be insignificant for everyday stage-use. In fact, our hearing does not recognize such small sound differences in music performances. Moreover, production tolerances will have a similar magnitude. However, the differences caused by changing the back panel (open vs. closed) are of significance – the sound does change.

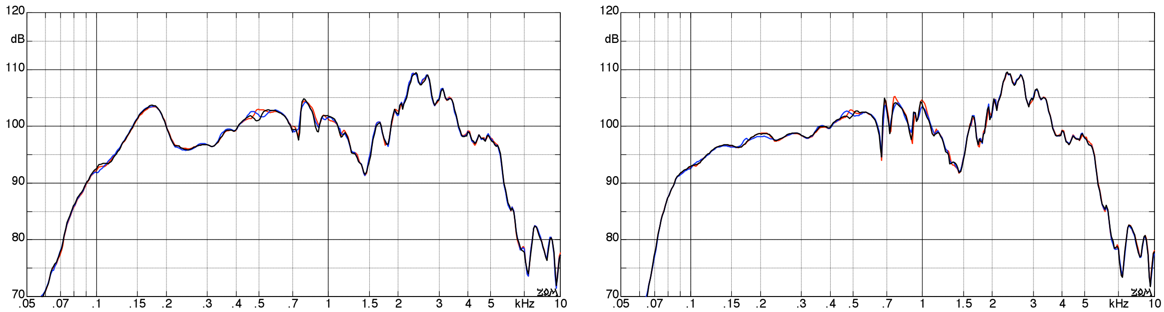


Fig. 11.93: SPL (1W/1m); enclosures: pine (black), poplar (red), MDF (blue).
Left: open rear-panel (gap of 13 cm). Right: closed rear-panel.

Fig. 11.94 shows the corresponding frequency responses of the impedance; again there are no peculiarities. The pronounced similarities guarantee practically the same behavior when driving the speaker from a high-impedance source (tube amplifier) – independent of the wood type. However, the changes in the rear-wall have in a particularly strong effect for operation from a high-impedance amp because transmission behavior <u>and</u> voltage at the speaker change.

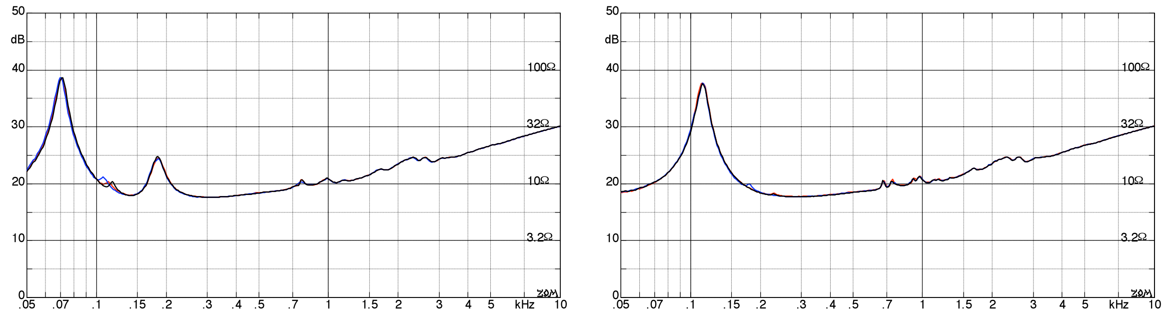


Abb. 11.94: Frequency responses of the impedance, loudspeaker and enclosure as in Fig. 11.93.

In **Fig. 11.95** we find supplementary measurements with pinewood-enclosures that either had no back-panel at all, or a partial one consisting of one board of a given size, or a partial one consisting of two boards of the same (given) size each (i.e. the latter corresponded to the *open cabinet* of the previous measurement). Again, there are no unexpected peculiarities.

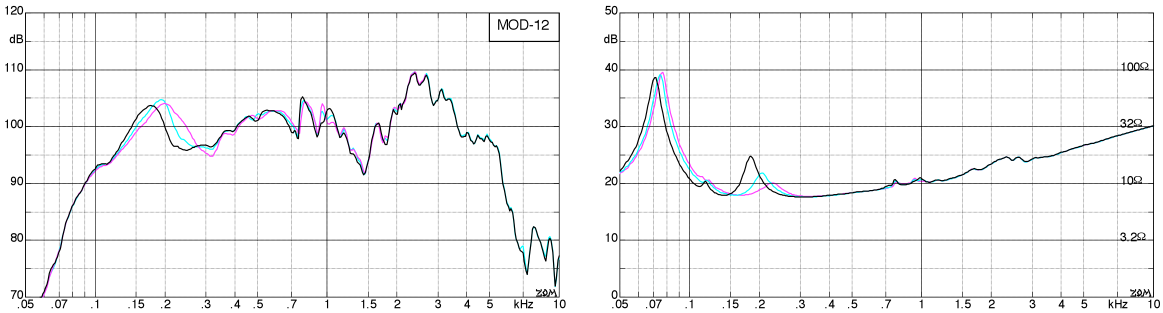


Fig. 11.95: SPL and impedance; completely w/out back panel (magenta), 1 board (cyan), 2 boards (black).

The operation with 2.83 V (resulting in 1 W at 8 Ω) is typical for loudspeaker measurements but does not correspond to customary power loading. As long as the speaker works in a reasonably linear fashion, the transfer function may be taken at any voltage. However, since loudspeakers can generate significant non-linear distortion, we opted to include measurements at a higher power level: at 2.83 V, 8.94 V, and 17.9 V, corresponding (at the nominal 8-Ω-impedance) to a **power of 1 W, 10 W, and 40 W**, respectively. Upping the power from 1 W to 10 W and 40 W, the level rises by 10 and 16 dB, respectively. This is shown in **Fig. 11.96** – merely in the bass-range we see deviations due to very strong distortion. To facilitate comparing the curves, **Fig. 11.97** depicts a representation normalized to 1 W: the 10-W-curves was lowered by 10 dB, and the 40-W-curve was lowered by 16 dB. Overall, the 40-W-curve is low visibly at too low a level; this is, however, not wood-specific, but simply caused by the heating up of the voice coil (all measurements were done with stiff *voltage*-source).

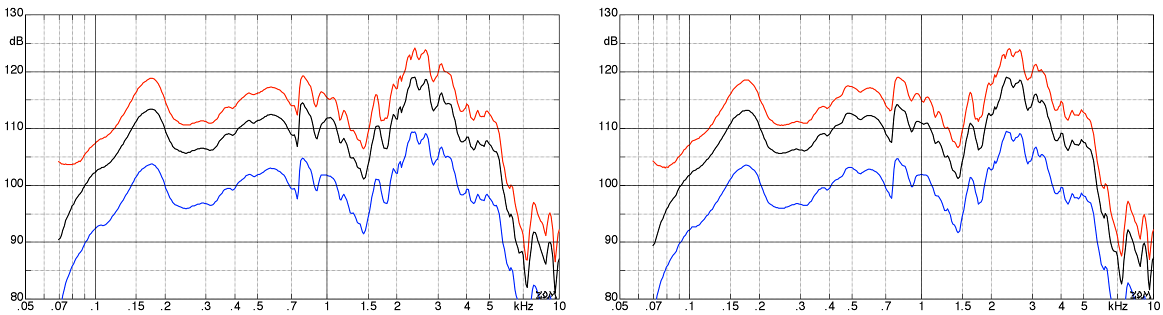


Fig. 11.96: SPL at 1 W (blue), 10 W (black), 40 W (red). Enclosures: pine (left), poplar (right).

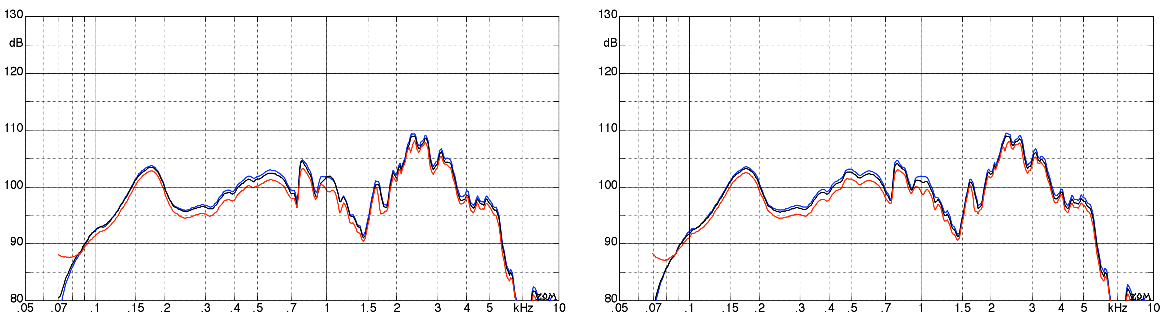


Fig. 11.97: as in Fig. 11.96 but representation normalized to 1 W.

The wood of the enclosure might influence the transmission function in two ways: via changes in the mechanical impedance of the bearing of the loudspeaker, and via sound radiation of co-vibrating enclosure walls. In order to obtain quantitative data relating to **enclosure vibrations**, we carried out measurements with a **laser-vibrometer** (Polytec). The loudspeaker was again fed from a stiff voltage source (2.83 V, 8.94 V, and 17.9 V). The laser-vibrometer measures the velocity; from this the displacement can be derived via integration, and the acceleration via differentiation. For the *ideal* loudspeaker (given a stiff current source) the acceleration is imprinted at $f > f_{Res}$; in the *real* speaker, resonances of the membrane cause selective frequency dependencies. Acceleration values corresponding to up to the 100-fold of the gravitational acceleration may be expected: 30 N at 0.03 kg yields 102 g (1 g = 9.81 m/s^2). Only the membrane experiences such strong acceleration, however; the side-panel vibrations are markedly weaker relative to the membrane-vibrations (**Fig. 11.98**).

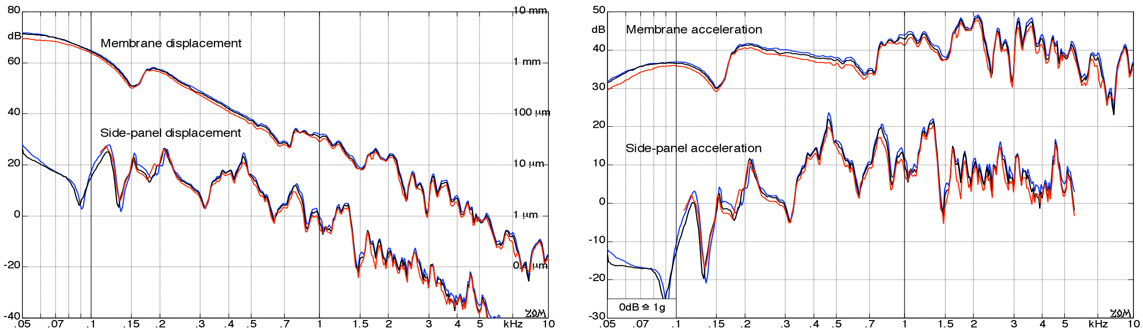


Abb. 11.98: Displacement (left) and acceleration (right) of the middle of the side panel; poplar; 1 W, 10 W, 40 W. Curves were normalized to 17.9 V (40 W), i.e. the 1-W-curve was elevated by 16 dB.

As is well known, the radiated sound power depends on the *square* of the velocity, on the size of the vibrating area, and on the radiation impedance [3]. The latter, and the effectively radiating area as well, can only be determined with much effort; therefore here just an approximate estimate: if the velocity of the membrane is, at 460 Hz, about 7 times as high as the velocity of the side panel (ΔL = 17 dB), the membrane will radiate about the 49-fold sound power at this frequency compared to the side panel. The other 'round: the side-panel contributes merely 2% to the sound radiation. Even if it were 5%: that's still rather insignificant. The contribution of the baffle is similarly small; only the **back-panel** weighs in with two relatively strong vibration-maxima (**Fig. 11.99**).

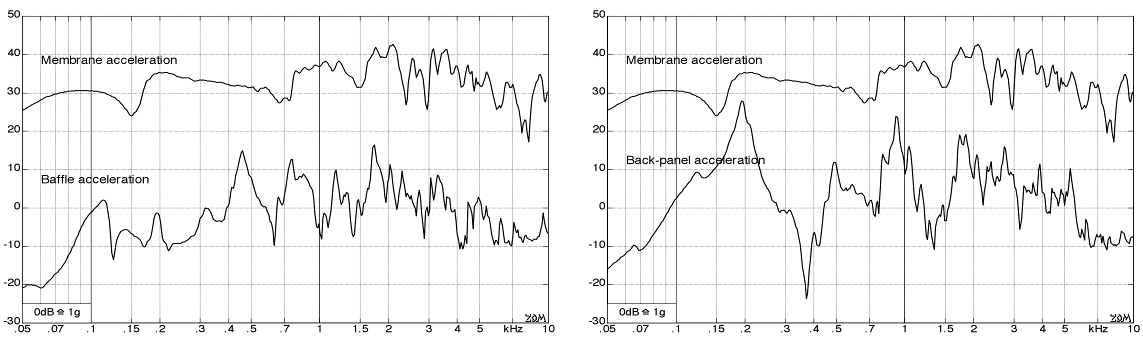


Abb. 11.99: Acceleration, poplar-enclosure, P = 10 W. Left: baffle, right: edge of back-panel.

Each of the two rear-panels is bolted on only three of its sides so it's understandable that larger vibration amplitudes are possible. Of course, the one-point-measurements presented here cannot provide exact data of the sound radiation – due to the lack of a scanning vibrometer, an exact sampling of all enclosure surfaces was not possible, and the selected measuring points can only give a first impression. The comparison of the three enclosures shows that all their walls vibrate in a similar manner (**Fig. 11.100**). The maxima of the rear-panel vibrations are a bit stronger in the poplar-made cabinet. For the sidewalls and the baffle, we see clear differences in the resonance frequencies but the maximum levels are similar. There are several reasons why the measured enclosure vibrations contribute so little to the SPL. The vibration amplitude of the cabinet walls, for example, is never larger than that of the **membrane vibration**. At low frequencies, the whole **membrane surface** vibrates with the same amplitude, which is not possible for a board bolted down at its rims even at its resonance. As regards the **radiation impedance**: the rear panel vibrates (at 200 Hz) strongest with its free rim, similar to a **dipole**. With an outward movement, the outer surface of the rear-panel generates excess pressure while the inner surface generates low pressure – both balance themselves out momentarily around the rim of the panel. Regarding the sound radiation, this is a most inefficient movement that is termed "operation with acoustical shortcut". At higher frequencies, lines of nodes appear in all enclosure walls, separating areas of the panels that vibrate in opposite phase: as one point of the panel moves outward, a neighboring point moves inward at the same moment. With the two movements being in opposite phase, only little sound is radiated. In Fig. 11.100, the SPL-measurement is again included for comparison: as different as the enclosure vibrations may be, they all have very little bearing on the sound pressure level.

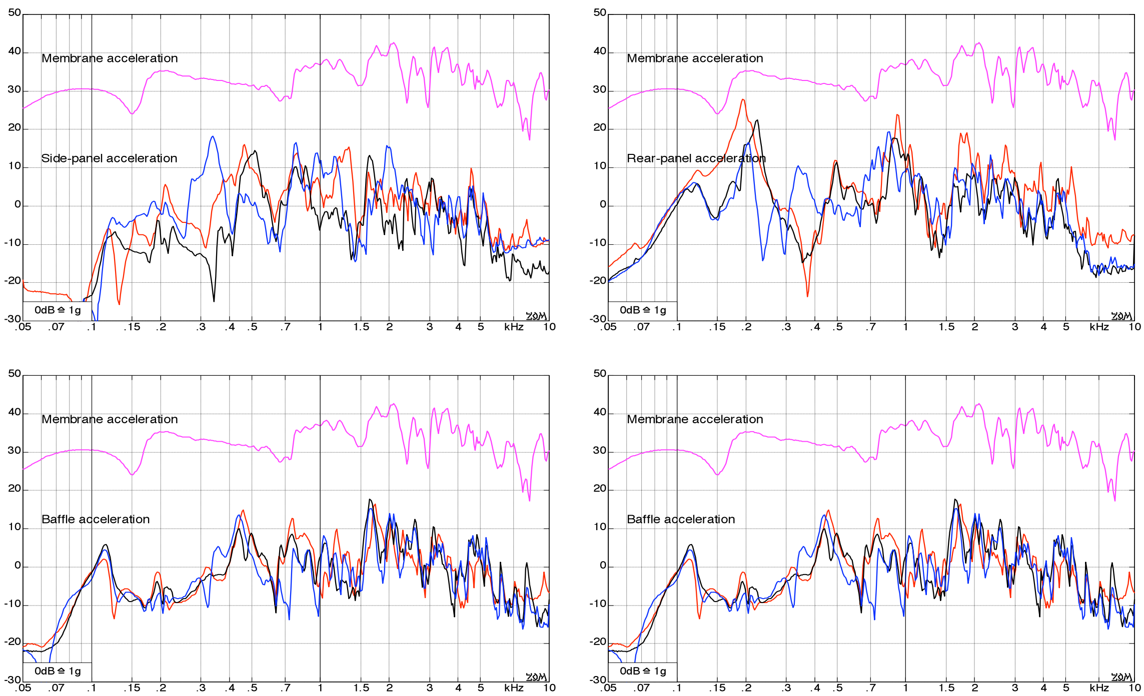


Abb. 11.100: Left: acceleration of the enclosure wall. P = 10 W; poplar (red), pine (black), MDF (blue). Right: SPL-measurement axially, 3 m in front of the enclosure, P = 1 W; color-coding as above.

Abb. 11.100 depicts the frequency response of the SPL in front of the membrane; however the loudspeaker radiates in all directions. **Fig. 11.101** shows the SPL frequency-responses for two further measuring points: 3 m behind the enclosure and 0.5 m above it.

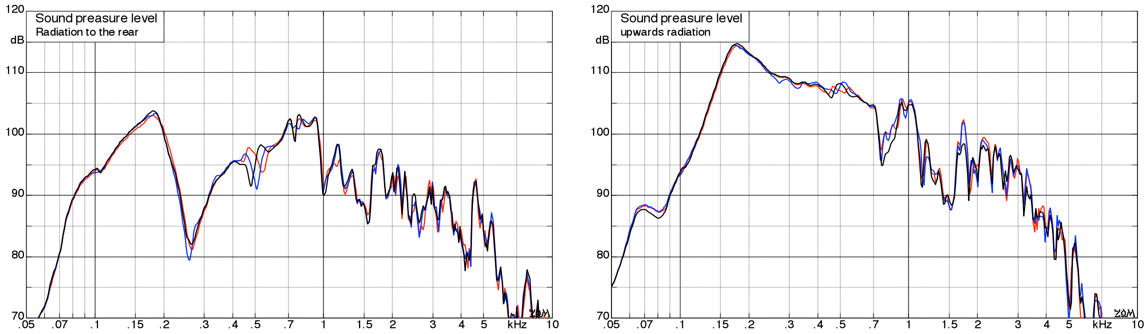


Fig. 11.101: Left: SPL 3 m behind the enclosure, P = 1 W; poplar (red), pine (black), MDF (blue). Right: SPL 0.5 m above the leading edge of the enclosure, P = 1 W; color-coding as above.

All SPL- and vibration measurements were done with one and the same loudspeaker, an Eminence MOD-12. To mount it, the rear panels had to be disassembled and reassembled each time. Repeat-measurements carried out to investigate the reproducibility showed SPL-differences that can be traced to the mounting of the rear panels (**Fig. 11.102**). Measuring the rear-panel acceleration showed a very strong dependency on the torque with which the mounting screws were tightened. This torque had not been checked when re-mounting the loudspeaker♣; consequently it can be assumed that enclosure-specific differences found in the SPL are in part due to differences in the attachment of the rear panel. **Therefore the differences purely due to the wood turn out to be even smaller.**

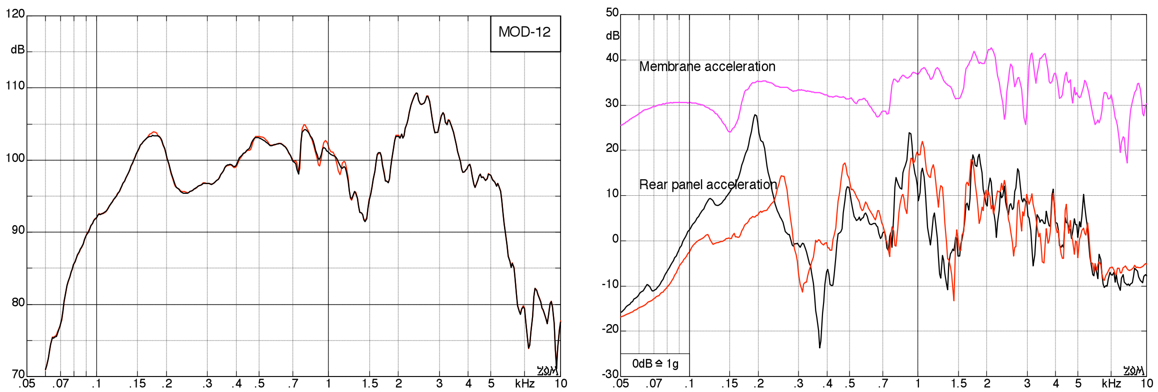


Fig. 11.102: Rear-panel screws tightened with different torque. Left: SPL; right: rear panel acceleration

Given these measurement results, the question poses itself why the dealers put so much emphasis on the wood used for the construction of instrument-loudspeaker cabinets, i.e. why it is imperative that the guitar box is made of "Baltic birch" or "solid pine". Simple answer: because it has always been that way – there's no connection to vibration-engineering. You can't build a Fender "Woody" using MDF-panels, because you will want to see only the most

♣ The screws fastening the rear-panels had been tightened „strongly" by hand each time.

strikingly beautiful wood grain. With a Tweed Deluxe, you could use MDF – as long as nobody looks inside. Leaving aside cosmetics, we have sound, weight, and durability remaining. Without further testimony, let's believe in the higher on-the-road resilience of the precise finger-joint construction. The question regarding the maximum weight we shall delegate to those tattooed knights-of-the-long-braid who will willingly schlep all that stuff back and forth every night. That leaves us with the sound. Is the wood in fact supposed to vibrate or not? Fortunately, our much-shepherded problem-child cannot be bothered about that question and just vibrates, as soon as it receives the invitation to do so from the membrane – irrespective of whether it is pine, poplar, birch, particle-board or MDF. Not in the identical manner for each of those materials, but so little that any influence on the sound radiated by the membrane remains marginal. We would not be adverse to the wood contributing some resonances (this being a sharp contrast to the world of HiFi) since the electric guitar has to offer little in that area. However, such contributions would have to be product-specific, and that would require a disproportionate effort – in a number of ways, not just in the tightening-torque for the mounting-bolts. The dimensions are crucial, as well: if the rear-panel rests on a slightly convex bar, it will vibrate differently compared to it sitting on a concave bar. Minute tolerances would be of importance here – one reason why acoustic guitars are not bolted together from planks. Speaker boxes, on the other hand, receive just that treatment – there appear to be differences to your D-28 or J-200, after all.

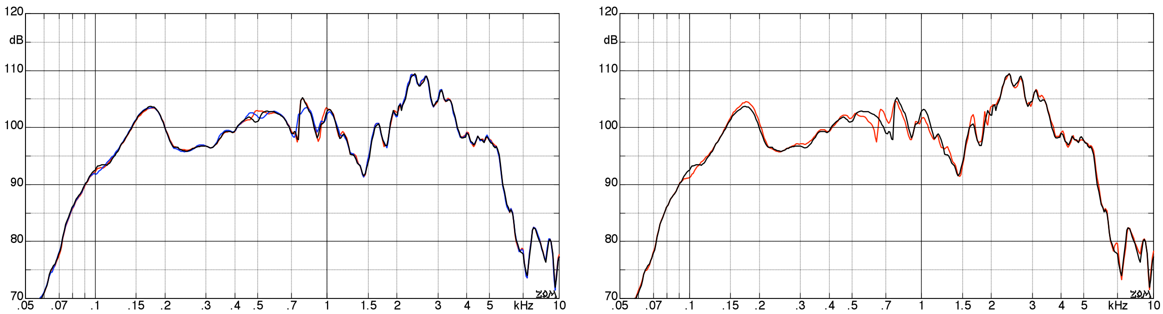


Fig. 11.103: Left: SPL (1W/1m), cabinets made of pine (black), poplar (red), MDF (blue).
Right: SPL (1W/1m), pine cabinet; 60x40x29 cm^3 (red), 50x41x30 cm^3 (black).

In **Fig. 11.103**, the differences caused by the type of wood are contrasted with those caused by changing the enclosure dimensions. The latter are varied by only a few centimeters – but hat is enough to result in greater differences that swapping poplar for pine.

Whether the loudspeaker is **front- or rear-mounted** onto the baffle-board also makes for small differences in the frequency response: The rear-mounting results in slight advantages: a gain of 1 – 2 dB in the range between 0.2 and 1 kHz, and a loss of about 2 dB around 3 kHz. The exact values depend on the given chassis and the dimensions of the enclosure.

The **speaker-cloth on the baffle** can have a two-fold effect: comb-filtering because sound is reflected back to the membrane, and – in particular at high frequencies – absorption. Some cloths, for example the material used in Fenders "Silverface"-amps, have next to no effect at all. Others, such as e.g. the thick material used by Marshall, cause an attenuation of about 1 dB at 1 to 5 kHz … which can certainly be measured but will be audible only when listening VERY closely.

11.9 Beam blockers, diffusers, and such

In a loudspeaker, beaming effects increase with increasing frequency (Chapter 11.4). The treble, i.e. frequency range upwards of about 1 kHz, is predominantly radiated on-axis, while the lows propagate spherically in all directions. If the loudspeaker (e.g. a 1x12”) is set on the floor, the guitarist standing right in front of it or next to it gets to hear too little treble. If the guitar player positions the speaker at the level of his head, the treble will be unbearably shrill (and dangerously loud, potentially damaging the hearing system). Therefore, beam blockers are available that are supposed to distribute the treble within the room, working similar to a diffuser lens.

The concept of the acoustical lens has in fact been around for quite a while – it is already mentioned by Olson [1957]. Similarly to an optical lens, the peripheral sections of a wave need to be delayed if divergence is called for (**Fig. 11.104**). To achieve that, the peripheral sound rays are run through an array of slanted sheets bent in serpentine fashion, creating a longer, indirect path and therefore a phase-shift. JBL has introduced these acoustical lenses in the early 1970’s, but they vanished again from the market as horns were further developed.

Fig. 11.104: Acoustical diffuser lenses; pictures from: www.jblpro.com

Today, not lenses but massive scattering bodies are deployed in order to reduce beaming effects in guitar loudspeakers. The **Weber Beam Blocker** (**Fig. 11.105**) is supposed to scatter the treble coming from the speaker-center via a spherical cap of convex shape. However, theoretical acoustics teach that beaming will occur the stronger, the larger the (uniformly) radiating source is – a ring-shaped emitter therefore does not have less beaming compared to the membrane centre thought to be the source of the treble. Reality is even more complex because it’s not only the centre of the membrane that can radiate treble but the fringe areas as well, and because the beam blocker will reflect sound back to the membrane, too.

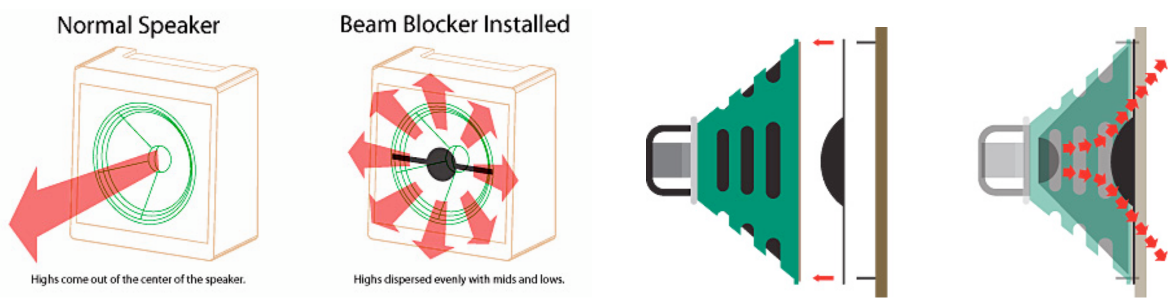


Fig. 11.105: Weber Beam Blocker, www.webervst.com/blocker.html

To obtain quantitative data, a **Tubetown-diffuser** was fitted to a 12"-loudspeaker mounted in the cabinet of an 18-W-Marshall. Measurements were done in the anechoic chamber with the loudspeaker box positioned on a turntable (B&K 3922), and the microphone (B&K 4165) located at the elevation of the speaker axis. Distance was 3 m. To obtain the colored directional spectrograms, pink noise was analyzed using overlapping 1/3rd-octave filters. The results are shown in **Fig. 11.107** (level dynamic = 40 dB) as a function of turn-angle (abscissa) and frequency (ordinate). The directional diagrams pictured below this are horizontal cuts through the color-diagrams.

As can be seen without much difficulty, this diffuser has practically no effect at low and middle frequencies – this indeed being purposeful. Around 3.5 kHz for the G12H, and around 5 kHz for the P12R, a slight broadening of the radiation is achieved. The effect is moderate – as is the price. “I’d rather invest those 15 Euro in a few beers – that will change my sound, too” … this assessment would not seem unreasonable. For those who want to experiment themselves (with the diffuser, not with beer): fasten a cardboard disc (∅ 8 cm) to the outside of the speaker cloth, and if you like what you hear, then buy the professional diffuser and mount inside of the cloth to the loudspeaker frame. Or make one yourself from cardboard.

Jay Mitchell proposes another solution in the "*Manufacturers' and Retailers' Forum*": a doughnut of foamed plastic is positioned within the circular cutout in the baffle board that however must not touch the membrane. The thickness of the doughnut is just under the thickness of the baffle board (about 15 mm), its outer diameter corresponds to the speaker-cutout in the baffle (about 28 cm for a 12”-speaker). The hole in the centre of the foam doughnut measures about 7 cm. Supposedly this arrangement will also distribute the treble better within the room. Our measurements cannot confirm this assumption: the main effect is a dampening of the treble. Which may in fact be a solution for the original problem, too.

Hoovi offers a rather more expensive solution: a *handsomely* styled reflector panel that is intended to deflect the sound to the side and to the top. Indeed, this works, and you can join the fun for the stately sum of around 350 Euro per speaker. Don’t stumble over the thing, though, and make sure you don’t leave it behind during the load-out. That would be rather aggravating considering the price. Also, you will not want the precious device to be scratched – but then you won’t let your roadie throw your prewar-Adirondack on the truck without a case, either; so: take along a tailor-made transport case. And don’t you dare set, instead of the Hoovi, a slantwise cut detergent-drum in front of the amp! That does work as well – but looks decidedly less noble♣.

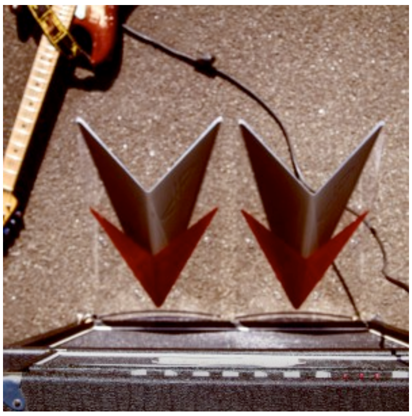


Fig. 11.106: Deeflexx, Donar's missile. [www.hoovi.at] Particularly interesting is the solution for 2 speakers: the sound deflected towards the right from the speaker on the left … where does that in fact go? Yep, exactly – that’s where it goes!

♣ Cited from the depths of the www: “if the guitar player doesn’t cut it, at least his rig should look cool…” Opposing view: “no way I’m going to let such a shitty-looking thing ruin my vintage AC-30-appeal”.

 Translated by Tilmann Zwicker

The following measurements were done, at a distance of 3 m in the AEC, with a Tube-Town diffuser attached to a G12H that was mounted in a Marshall-18-W-cabinet.

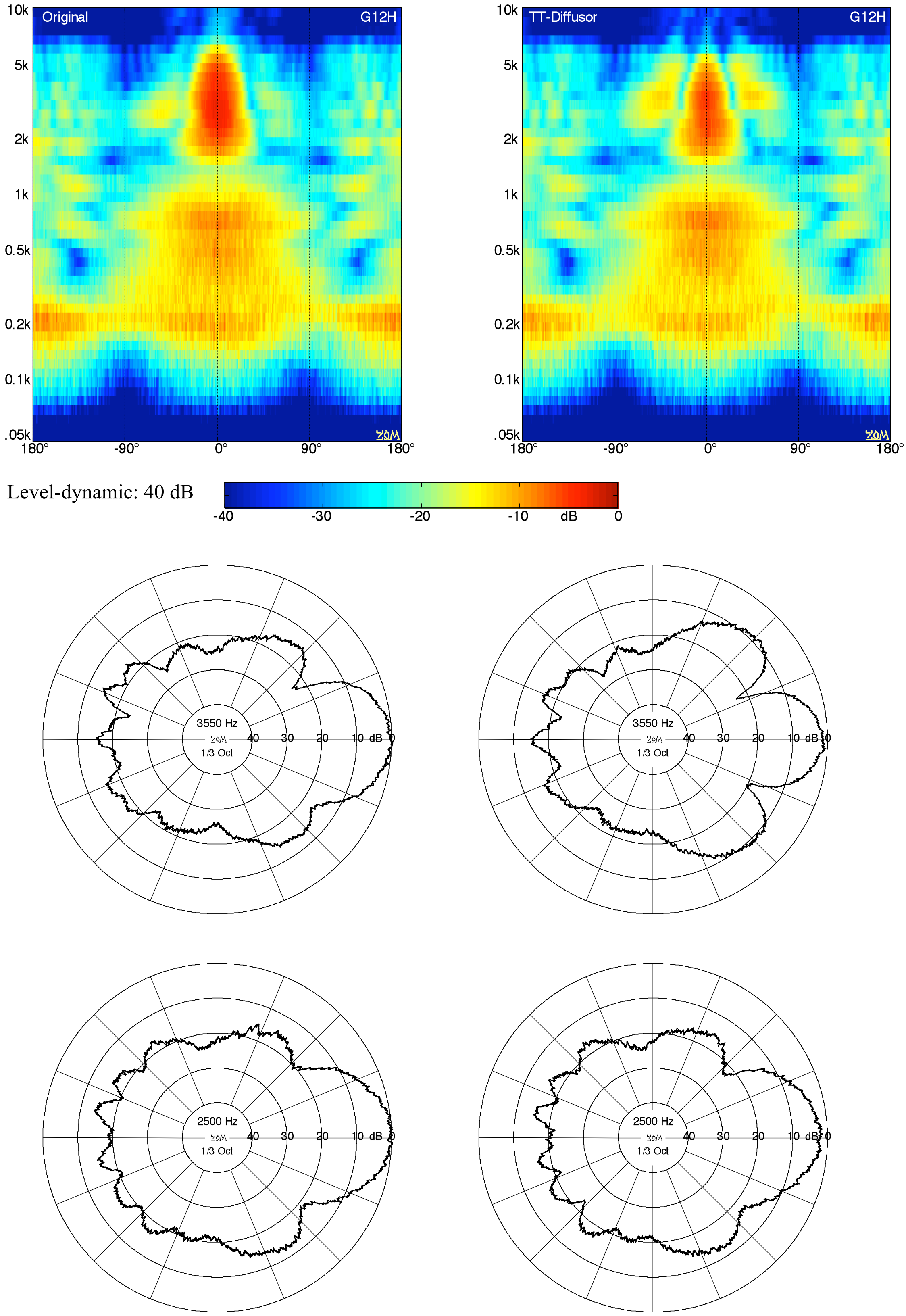
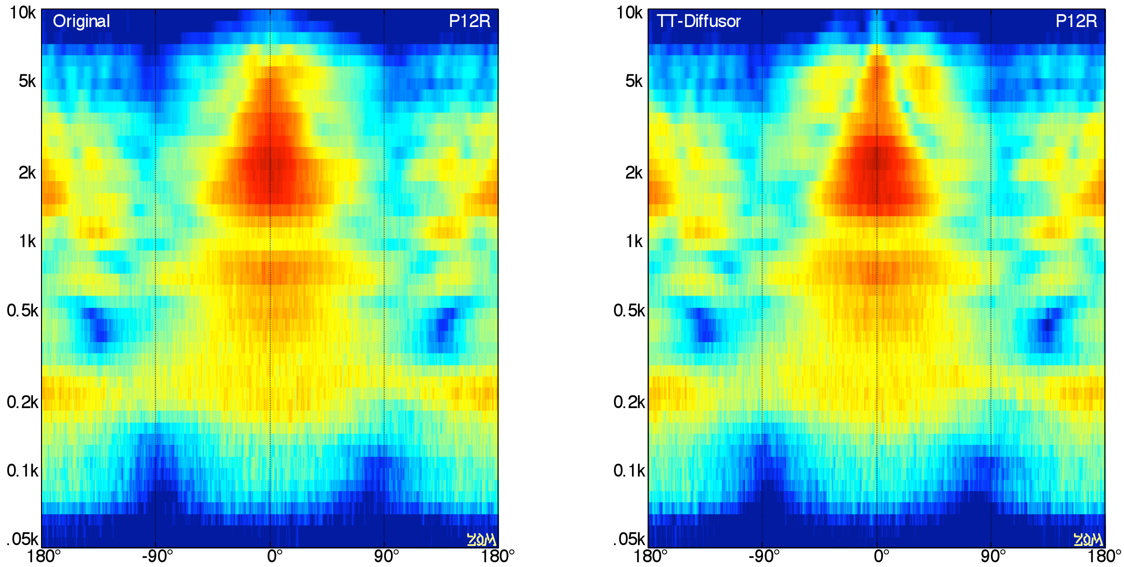
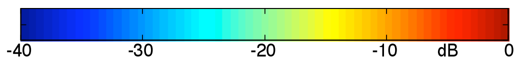


Fig. 11.107a: Celestion G12H; without (left) and with (right) Tubetown diffuser.

The effect of the diffuser shifts to higher frequencies for the P12R (which radiates somewhat more treble than the G12H).



Level-dynamic: 40 dB



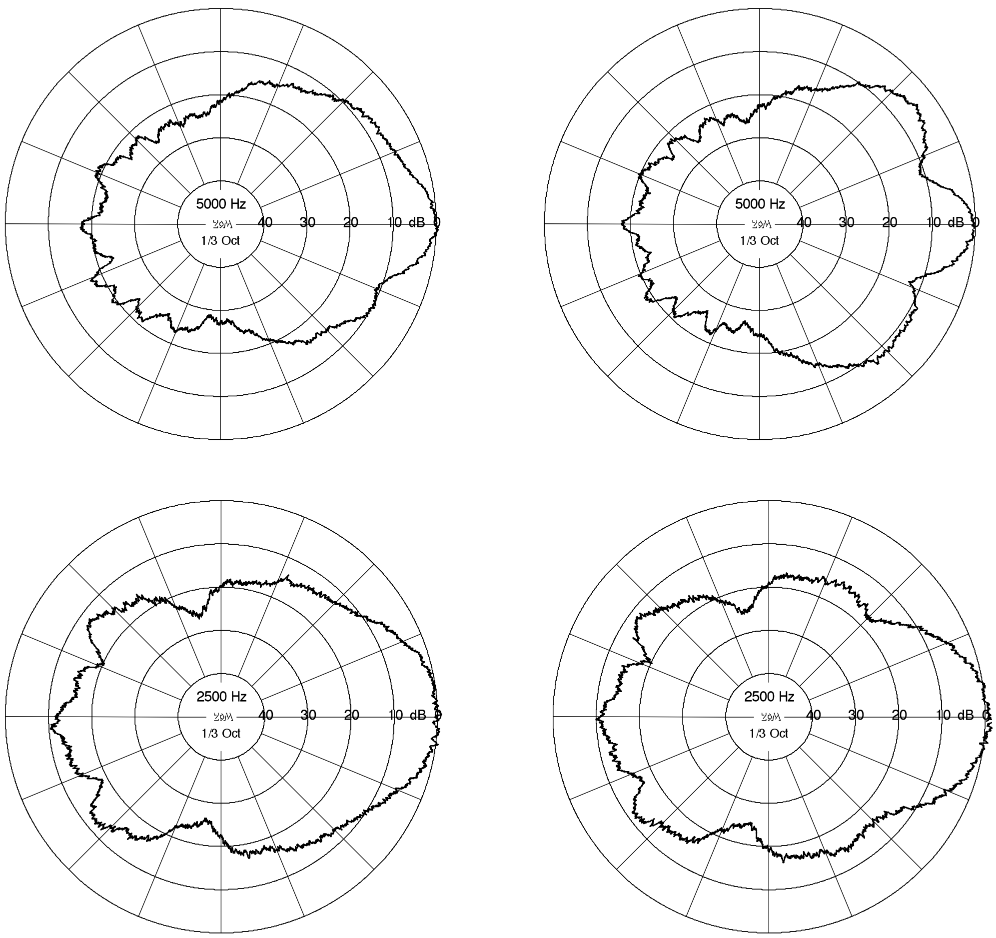


Fig. 11.107b: Jensen P12R; without (left) and with (right) Tubetown diffuser (left).

For the angled diffuser (construction similar to the Deeflexx), the effect is more brute, the distribution is broader, and there is a total treble-loss on axis.

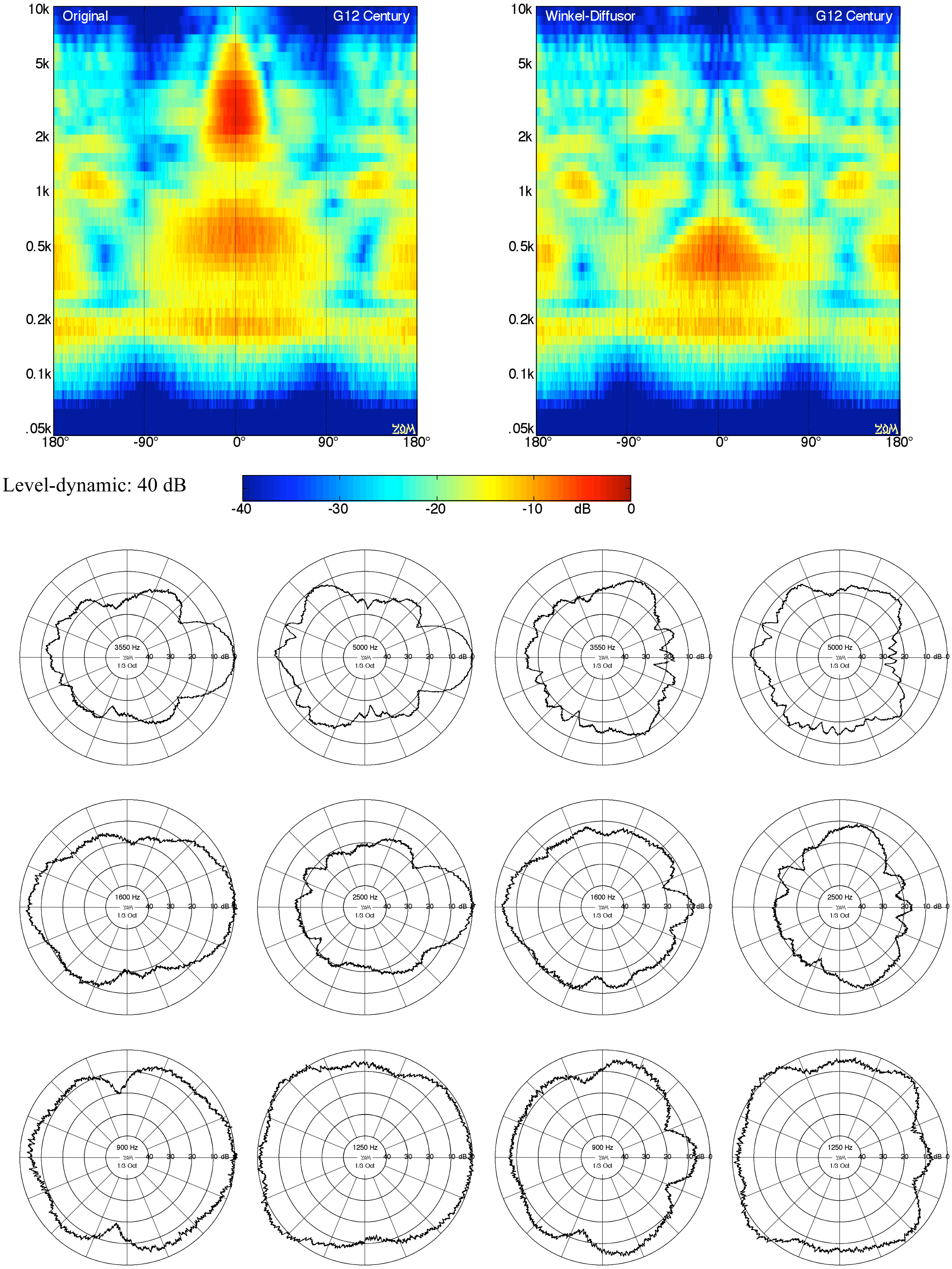
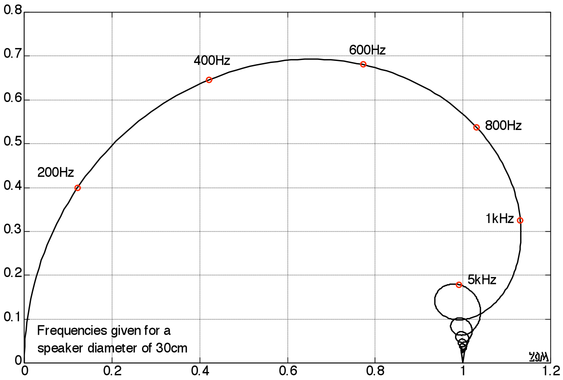


Fig. 11.107c: Angled diffuser in front of a VOX AD60-VT, Celestion G12 Century. It was not the DeeFlexx that was measured but a replica of equal dimensions.

11.10 Horn-loudspeakers

The classic guitar speaker arrangement does not feature a horn – but in a PA, or in the control-room of a recording studio, horn-loudspeakers may well be deployed. This is to increase the scarily low efficiency, and to modify the directionality. In a studio-speaker at most only about 3% of the power generated by the amplifier is converted to sound (Chapter 11.5), so experiments to improve the matching were done early on [Olsen]. The source impedance of the membrane is relatively large, and that of the load is small: in such a scenario, we would call for a transformer in electrical engineering; in acoustics, we would call for – right: a horn.



$$\underline{Z}_{\text{Ko}} = \underline{p}/\underline{v} = Z_0 \cdot \left[R(2ka) + j\,X(2ka)\right]$$

$$R(2ka) = 1 - 2\frac{J_1(2ka)}{2ka}\,, \quad J_1 = \text{Bessel-function}$$

$$X(2ka) = 2\frac{H_1(2ka)}{2ka}\,, \quad H_1 = \text{Struve's function}$$

$$2ka = 4\pi af/c\,, \qquad a = \text{membrane-radius}$$

Fig. 11.108: Normalized radiation-impedance for a piston-membrane [3].

Fig. 11.108 shows the complex radiation impedance $R + jX$. Multiplied by Z_0 = 414 Ns/m^3, it gives us the sound-field impedance $\underline{p}/\underline{v}$ for a circular, plane membrane: a first approximation for the loudspeaker-loading by the adjacent air [3]. Below 450 Hz, the membrane loading is predominantly imaginary; the membrane shoves air back and forth without actually sending off a lot of effective power in the form of a wave. Above 450 Hz, the real part does dominate, but at the same time, the membrane starts to have beaming effects. Positioning a horn in front of the membrane increases the real part of the loading at low frequency, and therefore improves the efficiency. However, in the bass-range this solution would require horns of enormous size, and therefore horn-systems operate mainly in the middle and treble range.

For first considerations it is purposeful to assume the cross-section of the horn to be circular (calculations may be done using cylinder coordinates). **Hyperbolic horns** give advantageous dimensions, with the radius $r(z)$ of the cross-section growing with z from the horn-“throat”:

$$r(z) = r_{TH} \cdot \left(\cosh\frac{\varepsilon \cdot z}{2} + M \cdot \sinh\frac{\varepsilon \cdot z}{2}\right)$$

Here, z is the distance to the throat (radius r_{TH}), M is a form-factor, and the horn-constant ε represents how fast the radius grows with increasing z. Given M = 1, the area increases according to an exponential function (exponential horn); given M = 0, the increase happens along a chain-line (catenary horn). For an exponential horn, the (plane) cross-sectional area grows exponentially: $S(z) = S_{\text{TH}} \cdot \exp(\varepsilon z)$, with the area of the throat being S_{TH}. Towards lower frequencies, a cutoff-frequency $f > \varepsilon c / 4\pi$ (for the wave-propagation within the horn) results from the flare-rate ε. The “mouth” of the horn (mouth-radius R) yields a further cutoff-frequency $f > c / \pi R$ for optimal matching. If the cutoff-frequency of the mouth is too high, disruptive reflections may occur within the horn.

Fig. 11.109 shows, for three different horns, the cross-section as it develops with the increasing value of z, and also the logarithm of the real part of the acoustical load impedance (0 dB = Z_0); on the right, the impedance without the horn (red curve) is included for comparison. For calculating the load, the length of the horn was assumed to be infinite so that reflections and standing waves could be ruled out. In horns of finite length, part of the wave running towards the mouth is reflected; the smaller the opening at the mouth is, the stronger the reflection. **Fig. 11.110** depicts two cases of identical wave-cutoff frequency but different mouth-cutoff frequency. The optimum angle at the mouth is about 90°.

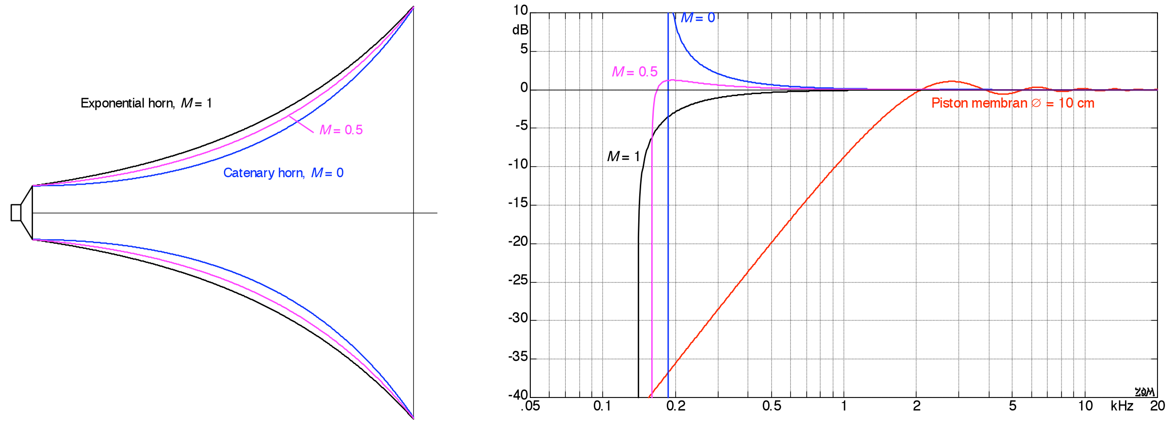


Fig. 11.109: Cross section (links) and membrane-loading for various horns. Throat-radius = 5 cm.

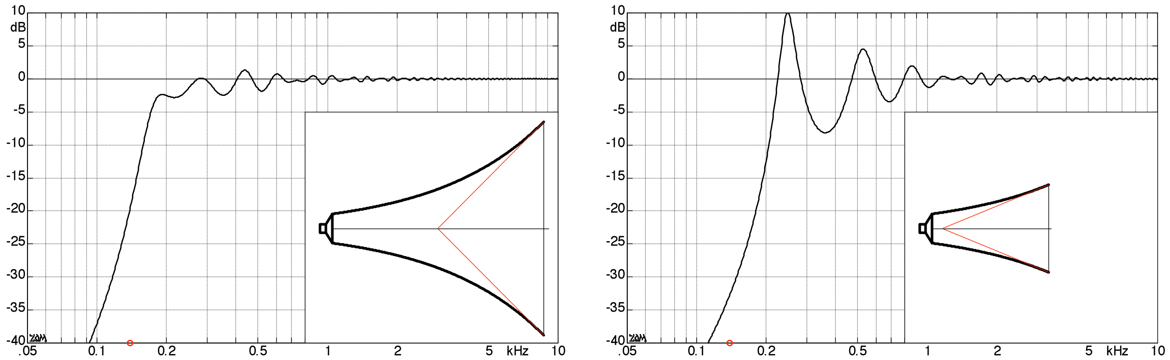


Fig. 11.110: Logarithm of the real part of the membrane loading for two different horn lengths; equal ε .

The circular cross-sectional area is a first approach towards calculation. In reality the cross-section develops from a round throat-area to a rectangular mouth-area, allowing for different directionality in the vertical plane compared to the horizontal plane. The **beam-width *Φ*** is a measure for the radiation but still remains rather limited in its meaningfulness, as it is seen in **Fig. 11.111**: despite equal angle the directivity of two loudspeakers may differ significantly.

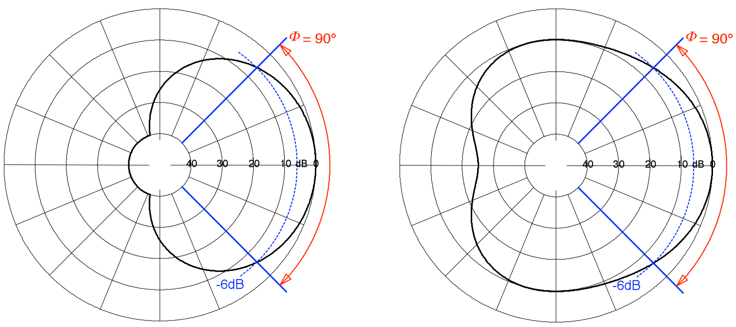
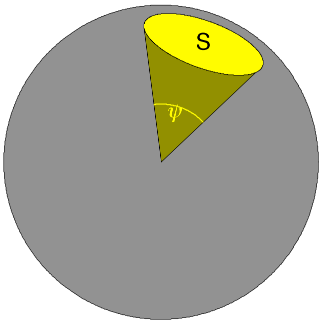


Fig. 11.111: Directionality; differing directionality factor despite equal aperture angle.

Besides the beam-width that gives a single number for the beaming behavior in one plane, the **directivity** yields an average value for all directions. Formally, the squared directional gain Γ needs to be integrated in space along an enveloping surface surrounding the source, and reciprocally be referring to this surface [3]. The logarithm of the resulting **beaming index** γ becomes the directivity d. In the simplest case, a sphere with the surface area of $HF = 4\pi r^2$ serves as the enveloping surface.

$$\gamma(\omega) = \frac{HF}{\oint\!\!\oint_{HF} \Gamma^2(\omega) \cdot dHF} \qquad d = 10\lg(\gamma)\,\mathrm{dB} \qquad HF = \text{enveloping area}$$

Let us assume as an example that the source radiates conically into the sector of a sphere, with a center-angle ψ (**Fig. 11.112**) and a spherical-cap surface S. For $\psi = 180°$ (half-space) we get from this a directivity of $d = 3$ dB, and $\psi = 60°$ would yield $d = 11.7$ dB.



$$\gamma = \frac{4\pi \cdot r^2}{S} = \frac{2}{1 - \cos(\psi/2)}$$

Fig. 11.112: Beaming for a conically radiating emitter.

The first rectangular horns produced in large quantities were **radial horns**. For this type, the horizontal dimension grows in linear fashion such that the vertical dimension needs to take care of the progressive increase required for the exponential growth of the area (**Fig. 11.113**). This geometry achieves a reasonably frequency-independent aperture angle – at the expense of the vertical directionality. Later developments (such as the so-called Mantaray horn by Altec Lansing, Fig. 11.113) allowed for a frequency-independent patterning of the directivity index (rather than of the horizontal aperture angle). The result was not perfect nor did it extend over the whole frequency range, but worked to a passable extent from a recommended cutoff frequency. Behind Altec Lansing, other manufacturers (JBL, Electro Voice, et al.) followed suit and developed horns with an approximately frequency-independent beaming index. At low frequencies (where the wavelength is large relative to the dimensions of the horn), all horns exhibit little beaming – only in the mid/high frequency-range, the specified beaming occurs.

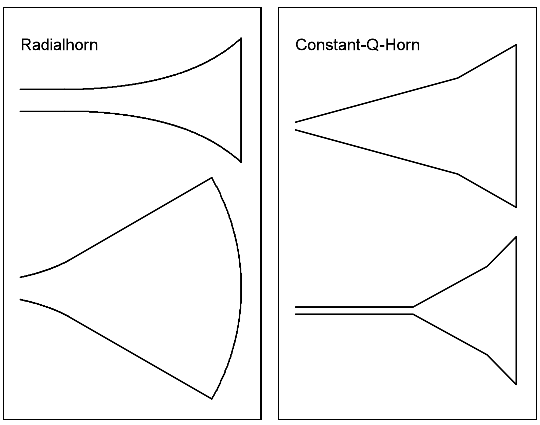


Fig. 11.113: Radial horn and Constant-Q-horn; vertical (top) und horizontal lateral dimensions.

Fig. 11.114 shows the beaming of a radial horn [D.B. Keele, AES Prep. 1083] compared to a constant-Q-horn [JBL 2356A]. The directivity index (i.e. *DI* and *d*, respectively) increases between 500 Hz and 15 kHz by 10 dB for the radial horn. On one hand, this is helpful because the power-frequency-response of typical horn-drivers decreases from about 2 kHz up, but on the other hand it is unattractive: only the direct sound, but not the diffuse sound, profits [3].

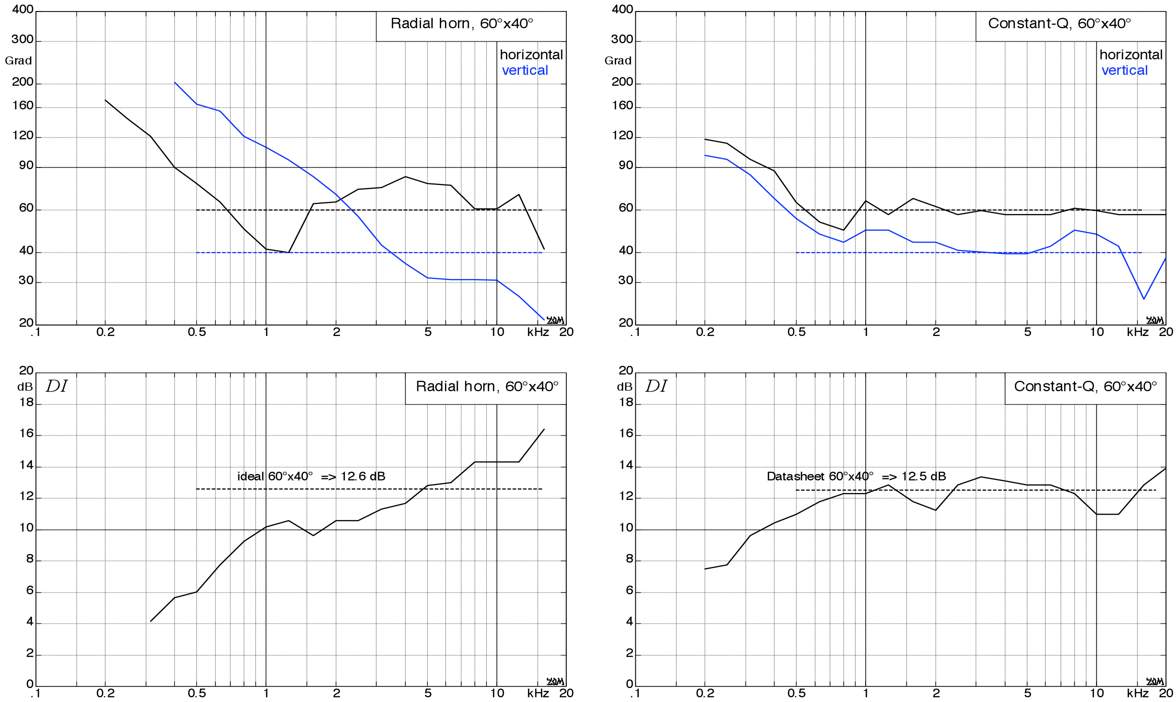
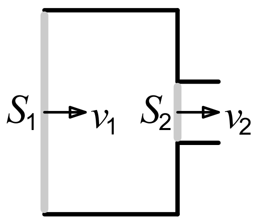


Fig. 11.114: Aperture angle and directivity index (*DI*) for two different horns. The radial horn was marketed as 60°x40°-horn (according to the datasheet) – rather courageous given the vertical beaming.

In midrange- and treble-horns, the horn does not directly sit on the membrane but connects to it via a **compression chamber** (**Fig. 11.115**). Assuming a location-independent pressure, the continuity requirement ($q_1 = q_2$) yields the relationship between membrane (Index 1) and the starting point of the horn (throat, Index 2): the load impedance rises by the ratio of the areas. In practice, the compression chamber is not of cylindrical shape, though, but forms a so-called **phase plug** that supports avoiding path-dependent interferences. Drive and membrane combine into the **driver**, to which horns with varying beaming behaviors can be fitted. In order to be able to specify driver-data independently of the horn, the former is mounted to a **plane wave tube** (PTW) – a tube with a length up to 6 m in which the waves can travel without reflections. The input impedance of the tube is approximately real: $p/v = 414$ Ns/m^3.



$$p = const$$

$$q = v_1 \cdot S_1 = v_2 \cdot S_2$$

$$Z_1 = \frac{F_1}{v_1} = \frac{S_1}{S_2} \cdot Z_2$$

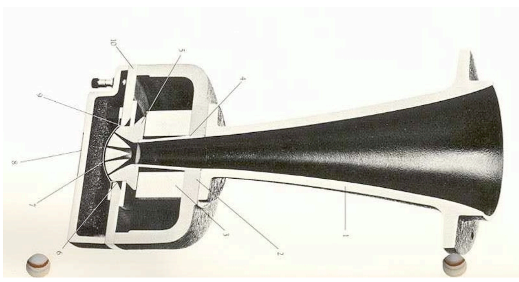


Fig. 11.115: Compression chamber. Photo: Lansing-Heritage

In **Fig. 11.116** we see a typical frequency response of a PWT. For the measurement, the driver (JBL 2451) was mounted to a 1.5”-PWT (although in the datasheet a 1"-PWT is noted). At middle frequencies, this driver reaches an **electro-acoustic efficiency** of 30%. From 3 kHz, the coupling deteriorates such that at 10 kHz, only 3% remain – which is not bad either. As this driver is mounted to a radial horn, the beaming of the latter (increasing with frequency) makes for a partial compensation of the treble loss, at least for the direct sound in front of the speaker. According to the rules of simple room acoustics, the beaming has no effect on the diffuse sound. The manufacturers recommend compensating the weak treble via filters (equalizer, EQ), but that only works up to a point: a 10-dB-boost requires the 10-fold power!

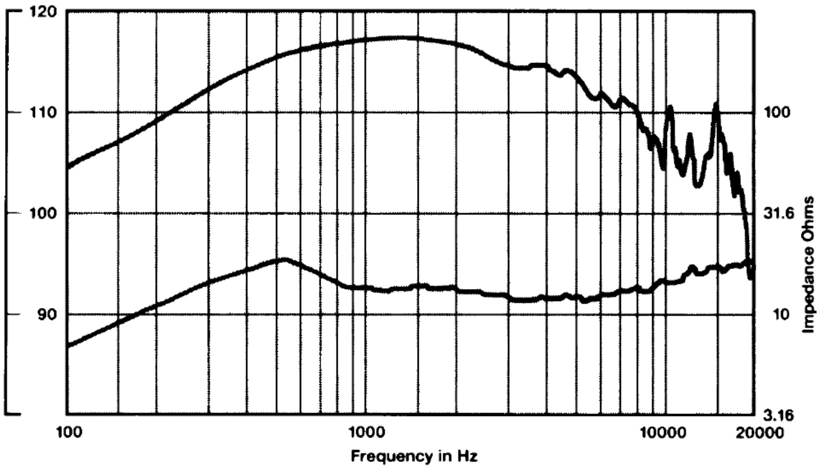


Fig. 11.116: Power-frequency-response and impedance of a driver [JBL 2451] mounted to a PWT. The ordinate specifies the SPL (in dB) obtainable with P_{ak} = 1mW. Since the acoustical loading is real, the sound pressure can directly be recalculated into the sound power: $P = p^2 \cdot S/Z_0$.

Fig. 11.117 shows two further horns: with acoustical lens, and with separating strips within the horn. The extreme vertical beaming of the lens is probably not entirely unrelated to its becoming extinct. The Smith-horn is a kind of multicell-horn but includes a closed bottom and a closed lid.

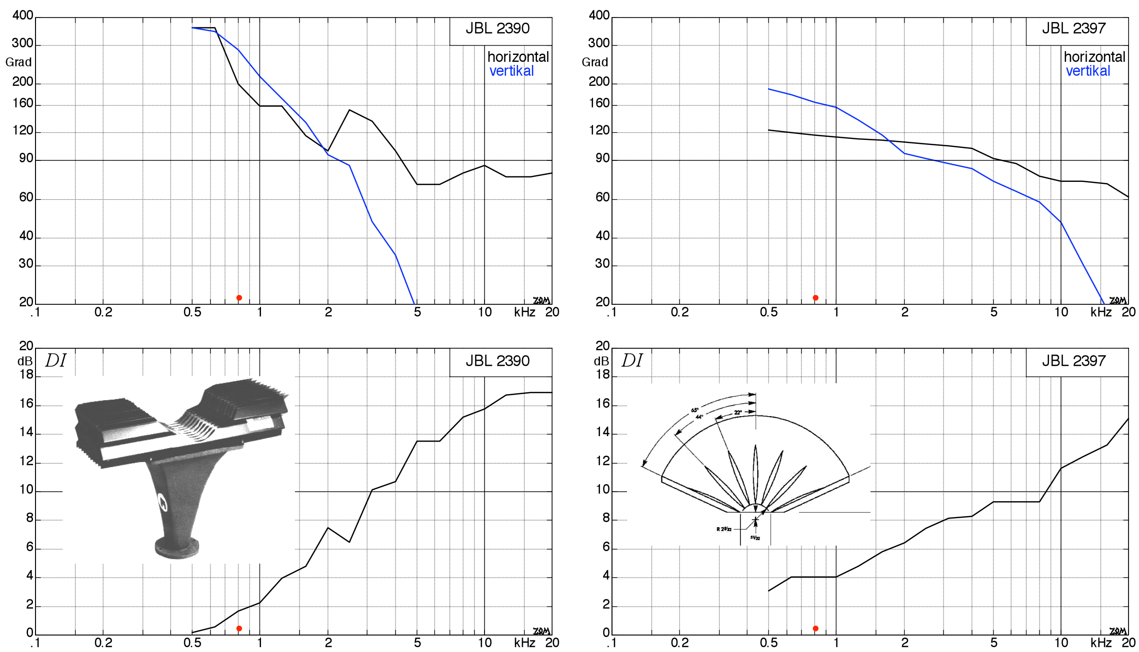


Fig. 11.117: Beaming for a JBL-lens (left) and for a so-called Smith-horn (right). “vertikal” = vertical

11.11 Studio-monitors

In the control room of a recording studio, high-grade multipath loudspeakers are deployed. Especially during the 1950's to 1970's, they were often fitted with mid- and treble-range horns. Contrary to widespread opinion, the frequency response measured on-axis is not the most important criterion. The frequency dependency of the "free-field transfer function" is not unimportant, but premium loudspeakers handle this aspect so well that other criteria move to the focus, for example the beaming, or (at high monitoring volume) the distortion (THD, difference tones, sub-harmonics). Because satisfactorily handling the whole audible frequency range with a single loudspeaker is not possible, filters (crossovers) take care of a subdivision into several frequency bands fed to corresponding speakers. Shown in **Fig. 11.118** is a simple circuit, as it is found (with slight modifications) in many DIY-guides. For the corresponding calculation it is assumed that the loudspeaker impedance is real, and that impedance and transmission-factors are frequency-independent. These assumptions are far from reality: the impedance is complex and dependent on frequency (Fig. 11.9), as are the transmission factors (in particular the phase). However, let us follow for a moment the idealized train of thought: the 2nd-order low-pass shifts the phase from 0° to -180°, and the 2nd-order high-pass generates a shift from 180° to 0° – such that across the whole frequency range the speaker voltages are in opposite phase. Only connecting the speakers 'out-of-phase' will avoid a complete cancellation at the crossover frequency (600 Hz in our example). That an all-pass filter results is, on the other hand, not critical: out hearing system does not take notice of that [3].

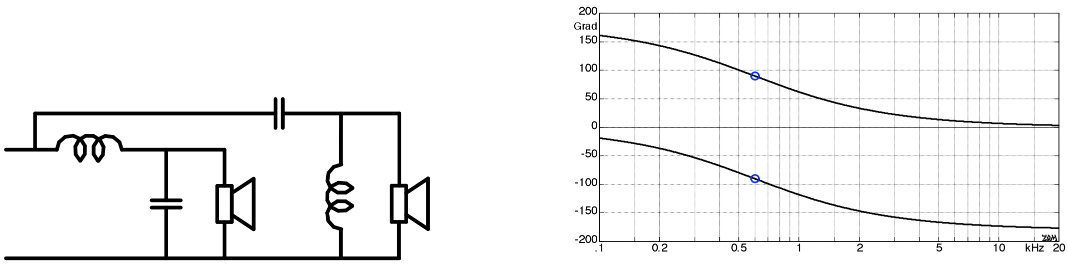


Fig. 11.118: 2nd-order two-way crossover: circuit (left), frequency response of the phase (right).

The big problem is created at the crossover frequency, if both speakers radiate the sound with the same amplitude. Even if the two partial sounds sum up perfectly on-axis – the radiation towards the sides always involves a phase shift, creating a destructive interference. If the difference in the path-length corresponds to half the wavelength ($\lambda = c/f$), the partial sounds cancel each other out (**Fig. 11.119**). An improvement is possible via so-called coaxial systems with the woofer being positioned behind the mid-range-speaker on the same axis; however here the speakers may get in each other's way. The argument that we should simply listen only exactly in front of the speaker does not hold water, either: the reflections arriving from the side do influence the hearing perception, as well.

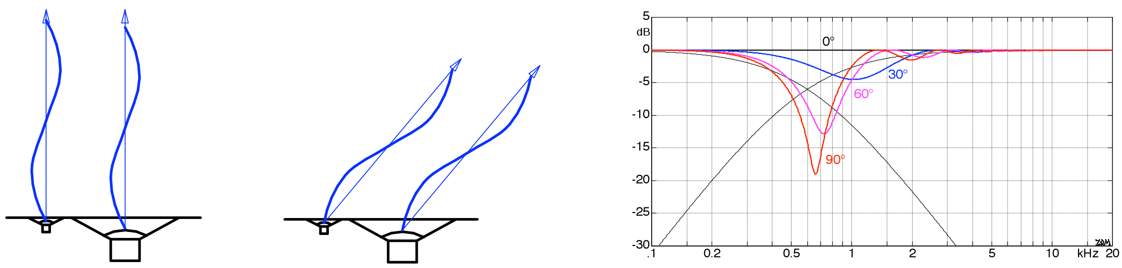


Fig. 11.119: Interference in the crossover-frequency-range: cancellation for radiation towards the side.

The following examples show the beaming behavior of different 3-way-speakers. We see from **Fig. 11.120** how even well known manufacturers struggle: the often requested “monotonous increase” of the directivity is nowhere in sight.

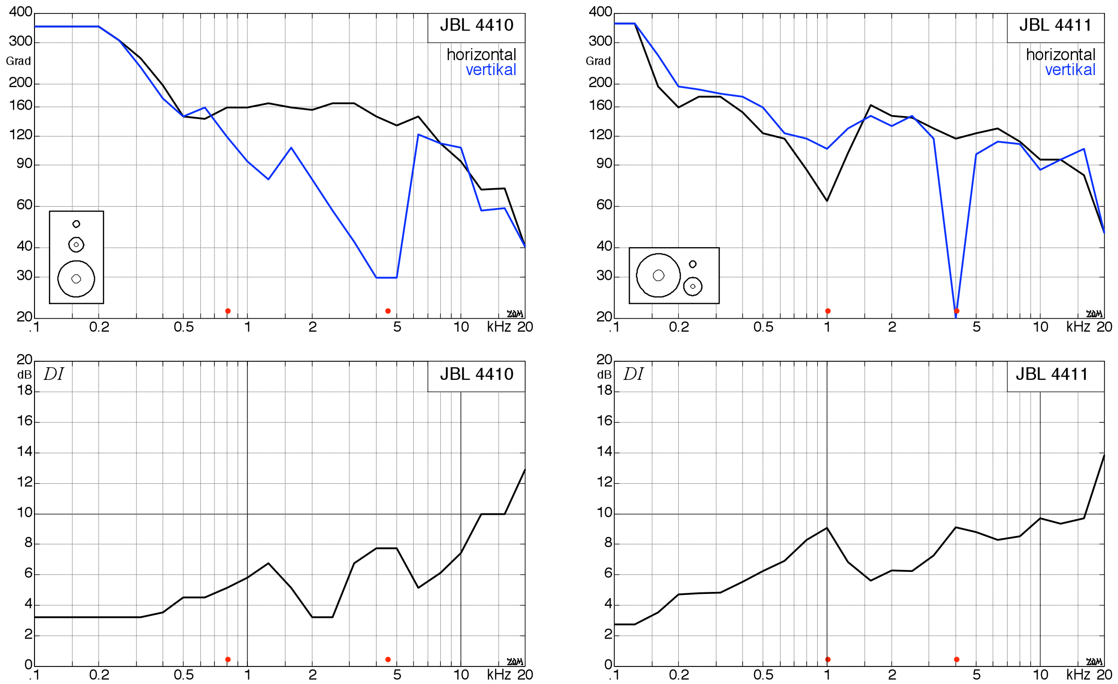


Fig. 11.120: Aperture angle and directivity of two 3-way-speakers (according to the manufacturer’s datasheet). “Vertikal” = vertical

The data of Sentry III are shown in **Fig. 11.121**; this speaker already enjoys a cult-status, and not undeservedly, as the graphs indicate. Still, we need to note that the two frequency responses of the aperture angle are always only simplified representations of a highly complex beaming behavior (Fig. 11.111). Also, hearsay states that there may be manufacturers who will “lend some help” to a less optimal curve and conjure up a characteristic favored by the sales department.

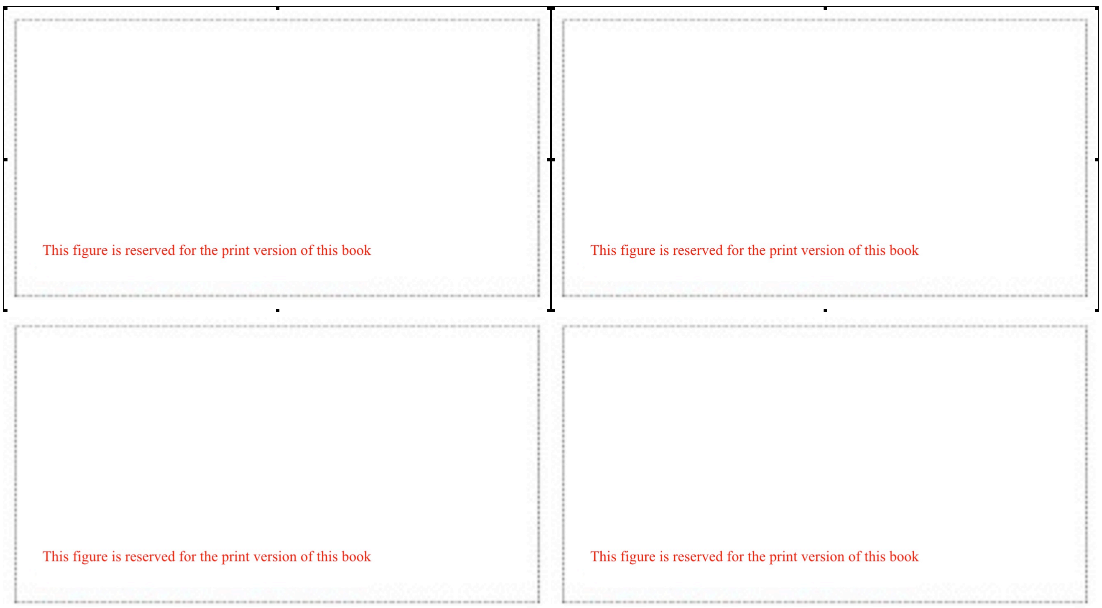


Fig. 11.121: Aperture angle and directivity of two 3-way-speakers (according to the manufacturer’s datasheet).

The following figures (**Fig. 11.122**) belong to two-way speakers. The JBL and the Altec can easily be imagined placed in the studio while the EV-speaker is more intended for PA-use. The 604-8L combines a 15"-woofer with a Mantaray-horn mounted coaxially with the woofer; the two JBL's employ so-called bi-radial horns (100°x100°), and the EV-box sports a 90°x40°-horn. None of the directional characteristics could be designated as particularly good or particularly bad – the quality always depends on the individual deployment-location. In the studio, this will be a relatively strongly absorbent control room where the reverberation time is between 0.2 and 0.4 s resulting in a reverberation radius of about 1.5 m. The effective reverberation radius [3] will then be around 2 – 6 m, and that will give the diffuse sound some significance, after all.

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Fig. 11.122: Aperture angle and directivity of two 3-way-speakers (according to the manufacturer's datasheet).

Fig. 11.123 shows the reverberation time $T_{60}(f)$ of two professional control rooms. For one of them (black curve), the right-hand graph indicates the effective reverberation radii, i.e. the reverberation radii increased by the square root of the directivity factor [3]. An engineer listening back at a distance of 3 – 4 m from the speakers is therefore predominantly located in the diffuse field for low frequencies. Depending on the beaming of the speakers, a very special mixture of direct and diffuse sound results that turns out to be rather … shall we say: "characteristic" for the JBL 4425.

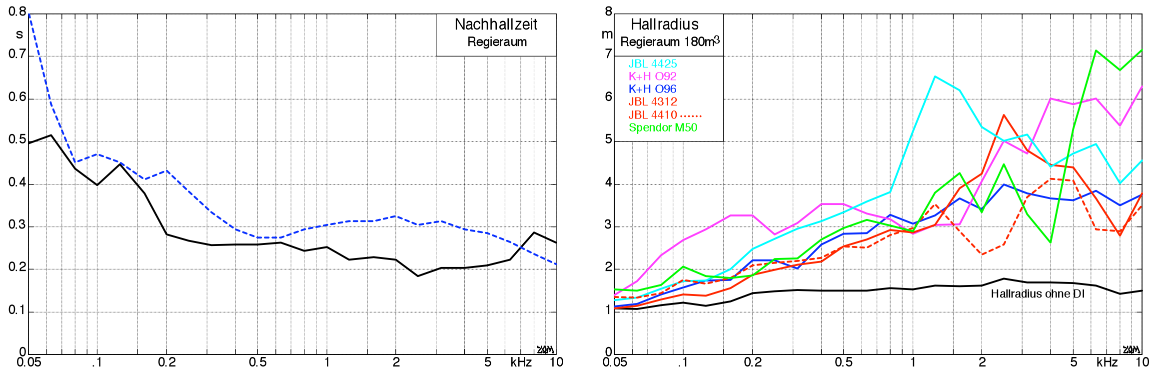

Fig. 11.123: Frequency responses of the reverberation times in two control rooms (left).
The black curve in the right-hand graph indicates the effective reverberation radii for 6 different studio monitors.

As a conclusion, let us look at a few measurements regarding **non-linear distortion** (**Fig. 11.124**). The requirement to be able to generate an SPL of 80 dB at distance of 2 m is not a very challenging one. However, if the maximum harmonic distortion needs to be kept below 0.1%, a few speakers fail, after all. Your classical studio-monitor will be able to rather reach around 1% – that is not all that bad, but more modern, newly developed types are able to remain clearly below the 1%-mark. Of course, the 80-dB-@-2-m is not the maximum required SPL – that would be about 110 dB / 2m. But even at that level, the non-linear distortion should remain "inconspicuous".

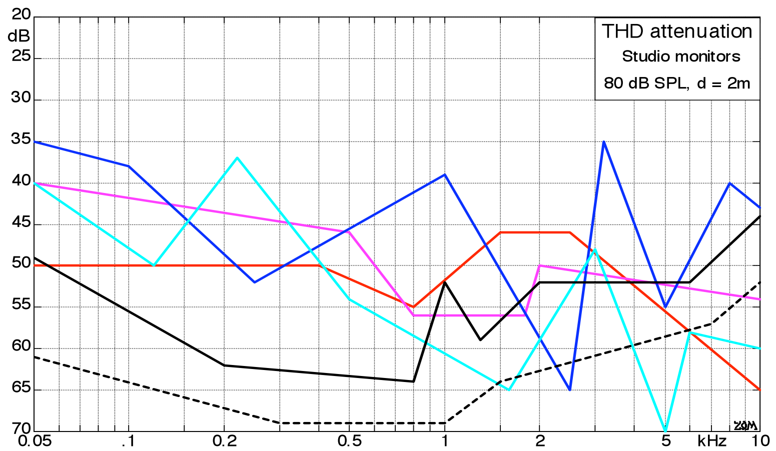

Fig. 11.124: Harmonic distortion suppression of different studio monitors (according to the manufacturer's datasheet).

11.12 Loudspeaker cables

If the cable connecting guitar amplifier and loudspeaker is merely a few meters long, any cable with adequately thick cross-section may be used. “Adequate” is a conductor cross-section of 0.75 mm^2, beyond reproach would be 1.5 mm^2. Regular conductor-copper is perfectly suitable, low-oxygen special copper – or even silver – is not required. It is entirely irrelevant whether 49.4 W or 49.5 W, of an amplifier power of 50 W, arrive at the speaker, and possible sound changes are certainly inaudible at ΔL < 0.05 dB. However, conventional guitar cables are unsuitable because the inner conductor will, as a rule, be too thin. The following table specifies the **percentile power loss** for a loudspeaker cable of a length of 2 m and for a load impedance of 8 Ω:

	Cu	Cu**!**	Ag	Al
2x0.75 mm^2	2,33	2,24	2,10	3,76
2x1.5 mm^2	>> **1,18** <<	1,13	1,06	1,91
2x2.5 mm^2	0,71	0,68	0,64	1,15

Cu = regular cable copper, Cu**!** = high-purity copper, Ag = silver, Al = aluminum♣.

Example: with a 2-m-long 2x1,5mm^2-cable you will experience, given a load of 8 Ω, a power loss of 1.18%; instead of e.g. 50 W, only 49.4 W arrive at the speaker; with a pure-silver cable, that power rises to 49.5 W. For a 16-Ω-load, the losses are even smaller: 49,71 W and 49,74 W, respectively.

In A.D. 2014, robust cables with high-quality collets-plugs are available for less than 10 Euro – these should be good even for professional use. Yes, cables do have a capacitance, and an inductance, and a skin effect may also be observed – but all associated effects are completely irrelevant compared of the loudspeaker impedance. Minimizing such aspects will increase the price but not the relevant quality. As soon as cable comparison tests are carried out under blind-test conditions, the inexpensive cable sounds just as good as the costly designer-cable. “Costly” may indicate 100 Euro – but possibly much more: for 5 m loudspeaker wire, the asking is **21.000.- €**! **Silver, braided.** Here’s a recommendation: cut up in pieces, it makes for a nice (alternative) necklace for the groupies.

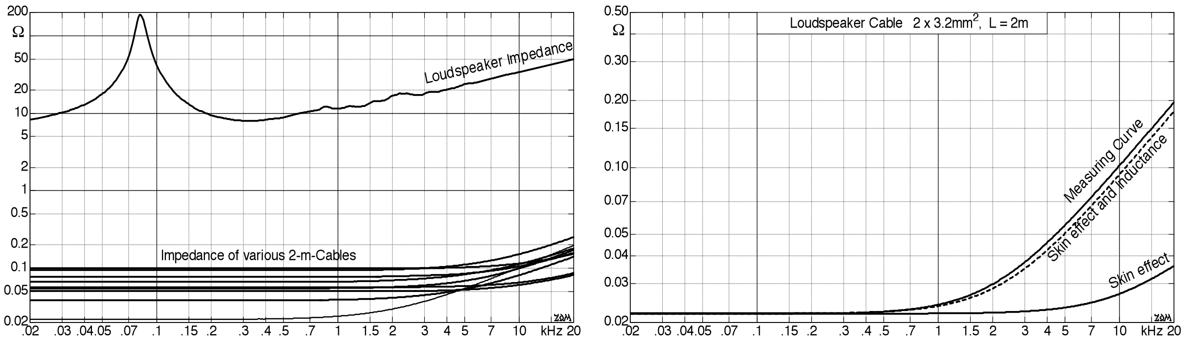

Fig. 11.125: Loudspeaker- and cable-impedance- in comparison (left), calculation vs. measurement (right).

Fig. 11.125 indicates that the measured rise of the impedance of a relatively thick speaker-cable is only in part due to the skin effect. The main share results from the inductance of the two-wire line. Even this increase of the impedance is insignificant because it makes for only less than 1% of the loudspeaker impedance.

♣ A **CCA-cable** (copper clad aluminum) is an <u>aluminum cable</u> with a thin copper coating!

11.A Appendix: metrology & instrumentation

In the following we will give a short overview regarding some of the devices and methods necessary for loudspeaker measurements. More extensive information is available from publications of the instrumentation-manufacturers, in particular from Brüel&Kjaer (Technical Reviews).

11.A.1 Measuring microphones

In contrast to microphones used in the recording studio, the transmission coefficient T_{Up} of measuring microphones should be frequency-independent [3]. G_{Up}, the transmission index of a typical free-field microphone (½" B&K 4190), for example, varies by less that ±0.7 dB in the range from 10 Hz to 15 kHz. This range of tolerance is, however, valid only for axial sound incidence; as soon as the direction of the sound deviates from the microphone axis, beaming effects that increase with rising frequency make themselves felt. In the anechoic chamber (AEC), such **beaming** is of no effect since the microphone is directly pointed towards the source. However, in the reverberation chamber (RC) with its diffuse sound field, attenuation towards the higher frequencies occurs that can easily amount to 5 dB at 15 kHz. For this reason, ¼"-microphones are preferred in the RC, and the fact that they are noisier compared to the ½"-microphone is accepted in exchange. The B&K 4135♣ used for our measurements has a beaming error of 0.5 dB at 5 kHz and of 1 dB at 10 kHz – this we deemed acceptable.

Non-linear **distortion** (harmonic distortion) is far below any relevance in the microphones used, and at the sound pressure levels that occurred. The **intrinsic noise** is insignificant for the 4190 (15 dB_A), and marginal for the 4135 (45 dB_A). Not insignificant are the effects of the **microphone mounting**: clamps and stands reflect waves and lead to comb-filter-like interferences♥. With suitable set-ups, such errors could however be kept below ±0.2 dB.

11.A.2 Reverberation time

The time it takes for the diffuse-field SPL to drop by 60 dB in the reverberation chamber after the sound source is switched off is specified as the reverberation time. In order to mainly measure diffuse sound, the microphone must not be located too close to the sound source, and to catch as many room modes as possible, the microphone should move a long a (slanted) circular path. All measurements in the reverberation chamber were done with 50%-overlapping $1/3^{rd}$-octave analysis (IEC 1260 class 0), with the microphone moving along a circle (∅ = 3m) within 80 s. The microphone boom was mounted to a **turntable** (B&K 3299). The latter transmitted a lot of **structure-borne sound** to begin with (equivalent air-borne SPL 84 dB); however, suitable decoupling reduced this value to some just-about-acceptable 45 dB.

Customarily, the reverberation chamber is excited via broadband noise to determine the reverberation time. After switching off the sound, the slope (dB/s) of the level decay is identified, and the reverberation time T_N results from it. Typical values are 2 – 5 s, and up to 10 s in the low frequency range. Since noise processes are of a stochastic nature, it is necessary to average over several decay processes.

♣ Brüel&Kjaer does offer a special pressure microphone (4136) that would be even more suitable.
♥ M. Zollner: Einfluss von Stativen und Halterungen..., Acoustica, Vol. 51 (1982), 268-272.

Even before this averaging (over several level-decay curves), an **RMS-averaging** is required to make the transition from the sound pressure to the sound pressure level (SPL). **Fig. 11.A1** clarifies this process, first via a decaying sine tone.

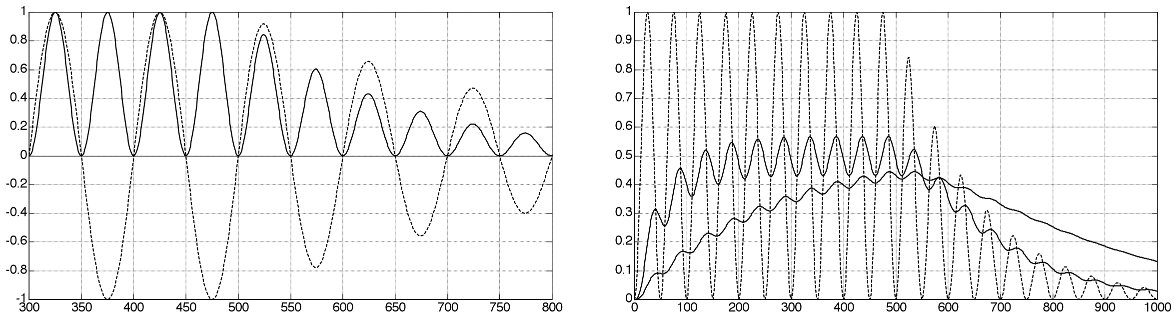

Fig. 11.A1: RMS-averaging. Left: exponentially decaying sine-tone (----), the square of this result (——). Right: exponentially averaged decay process; two different time-constants. Not yet subjected to the square root.

To get to the RMS (Root-Mean-Square) value, the signal needs to be squared (S) in a first step, subsequently averaged (M = mean), and last the square root (R) needs to be applied. While the squaring is an unambiguous step, taking the average is not. In metrology, two averaging methods are predominantly used: the so-called exponential averaging, and the so-called linear averaging – the latter should more appropriately be termed arithmetical averaging. Averaging devices in the above sense are linear low-pass filters that are described unambiguously by their impulse response. To achieve exponential averaging, a straight-forward (1st-order) RC-lowpass is used; its impulse response is a decaying *e*-function. The linear averaging happens in the gap-lowpass that features the unipolar rectangular pulse as its impulse response. Both approaches to averaging may be described by one parameter each: by the time constant τ for the exponential averaging, and by the duration of the rectangular pulse (block length) T. Even for $\tau = T$, the results are not the same.

The general problem of every averaging process becomes clear from Fig. 11.A1: with too short a time constant, the smoothing is insufficient, and with a long time constant, the decay-process representation holds errors. **Fig. 11.A2** shows corresponding level-graphs: it is evident that for the exponential averaging, the slope of the flanks depends on the time constant. With the linear averaging, the decaying slope is merely delayed but its slope remains. For determining the reverberation time, this implies that linear averaging has advantages. It still is not without problems: in particular during the **early decay** phase, the slope is too shallow – this could lead to the calculation of too long a reverberation time. This early decay range is important, since real reverberation curves do not decay as ideally as given in this example but show a degressive decay (i.e. an increasingly shallow curve).

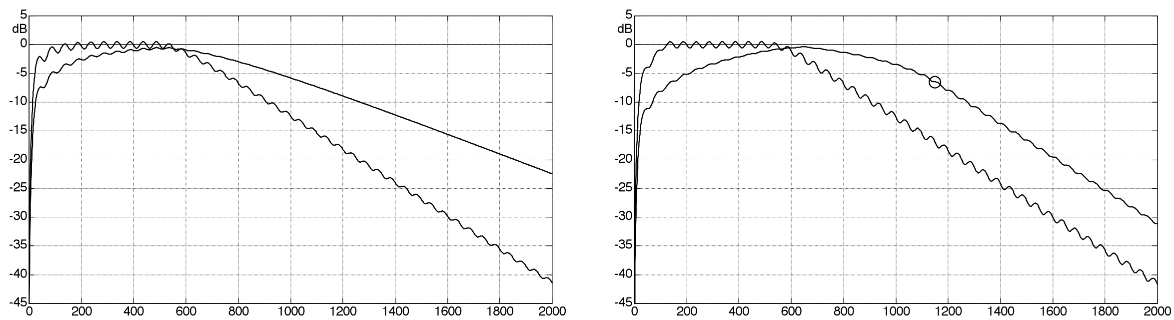

Fig. 11.A2: Level-decay for exponential averaging (left, two different time-constants), and for linear averaging (right, two different time-constants).

From a systems-theory point-of-view, averaging is filtering: the signal to be averaged is convolved with the impulse response of the averager. In other words: every averaging is a weighted (convolution-) integral over a *range*. For the linear averaging (= block-averaging or arithmetic averaging), this integration happens over the block-length T ahead of the center-point (in time) of the averaging. For T = 0.4 s, the linear average indicated at the point in time of t = 2 s specifies the integral over 1.6 … 2 s. Therefore, the linear average value measured at the time when the sound source is switched off (t = 0) does not constitute an averaging over the decay process. Staying with T = 0.4 s: 50 % of the linear average measured at t = 0.2 s is determined from the steady-state process, and the remaining 50% are captured via the decay process. Only the linear average measured at $t = T$ captures 100% of the decay process♣. It is exactly from that point on that the slope of the level-decay is correctly shown when linear averaging is used (Fig. 11.A2, right). Every averaging (over time) needs to happen over a *range* (as elaborated above). If the duration of this range is set too short, the averaging cannot serve its purpose: the convolution with a Dirac pulse results in the unchanged signal. Since the averaging needs to happen over a range (the duration of which needs to be larger than zero), all averages arrive delayed after the signal to be averaged.

If the decay process were an exponentially decaying sinusoidal oscillation as presented in the first figure, and if the travel time in the averager were known, the slope of the process could be precisely determined. However, already an envelope composed of two e-functions (see **Fig. 11.A3**) renders determining the slope problematic: if the slope changes significantly within the block-length T, a linear averaging will not be adequate to reliably detect this.

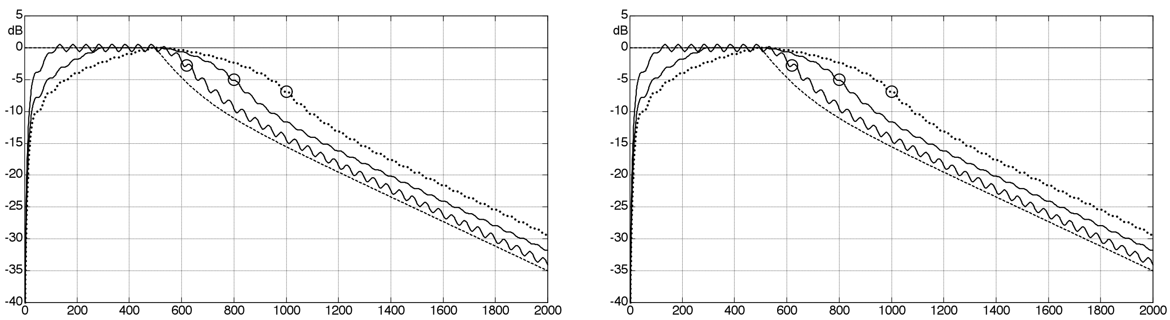

Fig. 11.A3: Level decay for exponential averaging (left, three different time-constants) and for linear averaging (right, three different block-lengths). Degressive decay-curve.

As consequence, the averaging time T, or the time-constant τ, respectively, needs to be short enough not to too much distort the shape of the space-impulse-response (or the space-step-response, respectively), but on the other hand it needs to be long enough to average out the stochastic fluctuations of the noise. This is because, contrary to the figures presented so far, decay curves of reverb are not determined using sine-tones, but with noise (of a bandwidth of an octave or of 1/3rd of an octave). If a linear averaging over e.g. 50 ms is not good enough to average out the noise sufficiently, and if a longer averaging time would prohibit measuring the early decay of the curved decay curve, then a further dimension remains as a way out: the **ensemble-averaging** across different realizations of the noise process. This simply entails averaging over several decay curves – however not with identical noise-excitation but using different excerpts from a noise signal (e.g. 1/3rd-octave noise).

♣ Strictly speaking, the travel time of the sound needs to be considered, as well.

 Translated by Tilmann Zwicker

In **Fig. 11.A4** we see, in the left hand section, 4 decay curves that were determined from the squared SPL-signal via linear averaging over a block-length of 50 ms. For this measurement, microphone and loudspeaker were in fixed locations such that the signal fluctuations are to be attributed predominantly to the stochastic of the noise. In the right-hand graph, a multitude of such curves is included – as is the averaging curve derived from them. For orientation, the dashed line represents the decay for a reverberation time of 1.6 s. We can see that the latter approximately corresponds to the early decay, while the remaining curve is shallower. The fluctuations that still remain in the decay curve are not primarily associated with the noise stochastic with the room. The superposition of many decay processes (with many frequencies and different dampening) does not result in a single decay-time-constant; rather we get a curve of any arbitrary complexity that can normally only be approximated to a straight line in sections. For power measurements in the reverberation chamber, it is not the level range between -5 and -36 dB that is to be captured, but rather the initial slope♣.

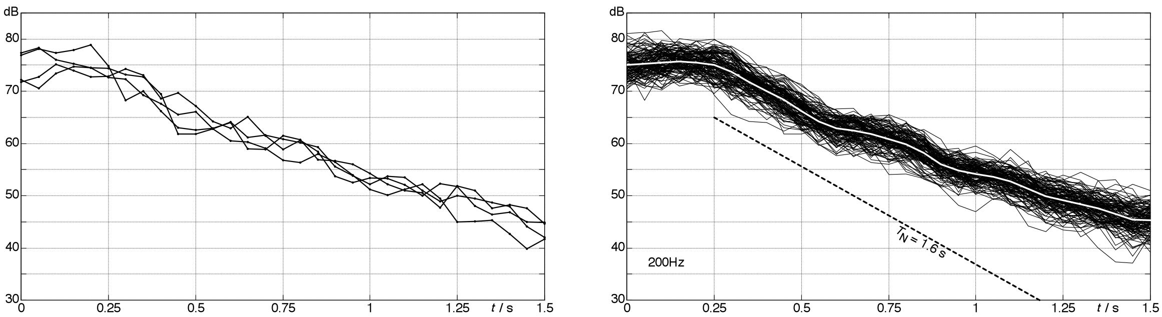

Fig. 11.A4: Decay curves; composed from 1/3[rd]-octave noise (f_m = 200 Hz) via linear averaging (50 ms). In the graph on the right, the white line represents the mean of the ensemble.

In conclusion a short comment regarding the **Hilbert transform**, since it occasionally is accredited the capability to do ideal averaging. For a decaying sine-tone it is indeed possible to derive, from the sound pressure and using the Hilbert transform, the analytical signal, and from this a smooth decay curve. However, given the narrow-band noise commonly used for reverberation measurements, the Hilbert transform is not an option – at least as long as it alone is put to use (**Fig. 11.A5**).

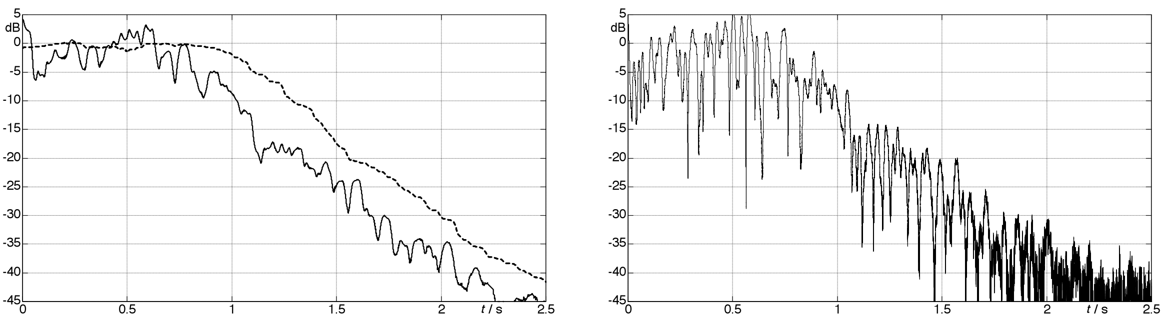

Fig. 11.A5: Left: decay curved filtered with a bandwidth of 1/3[rd]-octave (f_m = 200 Hz), linear average, T = 50 ms and 500 ms, respectively.
Right: Level of the analytical signal belonging to the signal on the left (also termed "magnitude").

♣ H. Larsen, Technical Review Nr. 4, Brüel&Kjaer, !978.