

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.

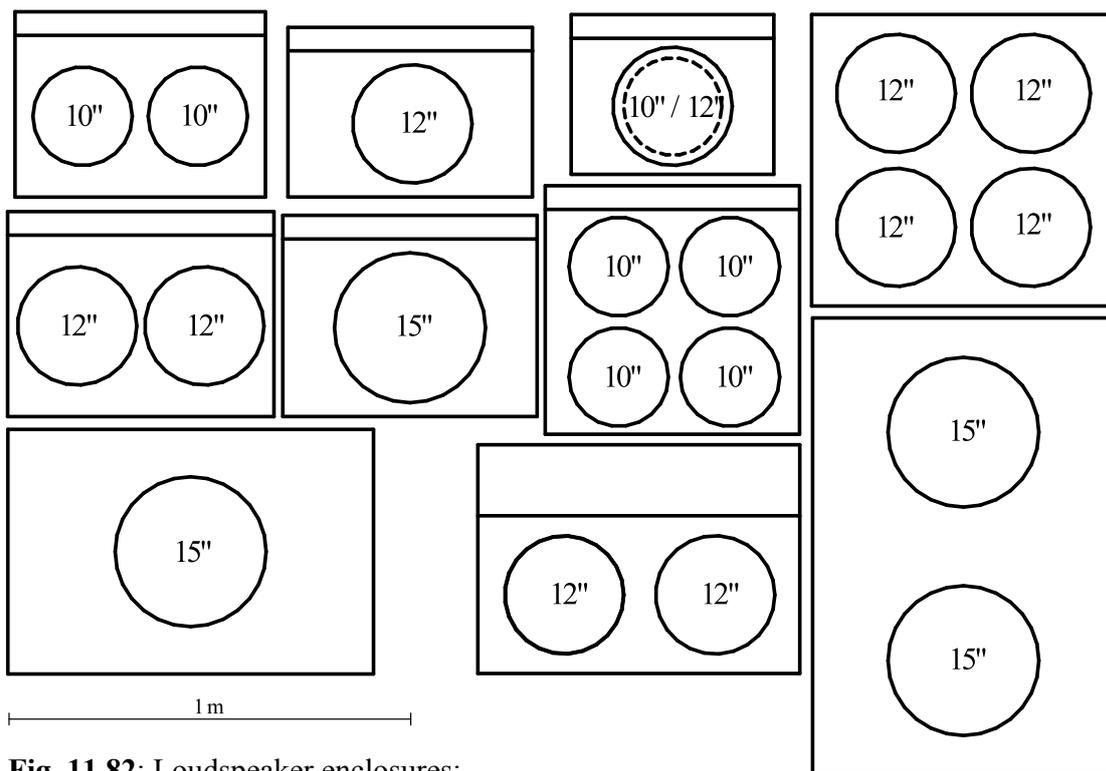


Fig. 11.82: Loudspeaker enclosures;
Membrane-diameters in inches

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 inversely 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, $p \cdot V^{1.4} = \text{const}$ 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/m². This causes a rear wall with an area of 0.18 m² (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 be 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.

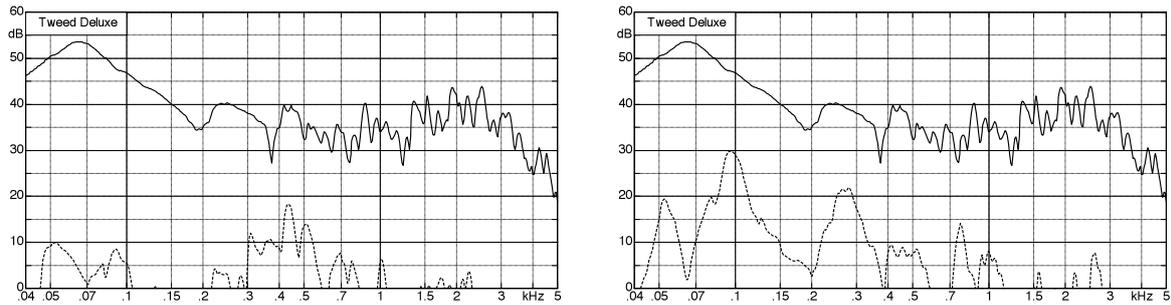


Fig. 11.83: Membrane-velocity in comparison to the velocities of cabinet sidewall (left), and baffle (right).

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 ...*

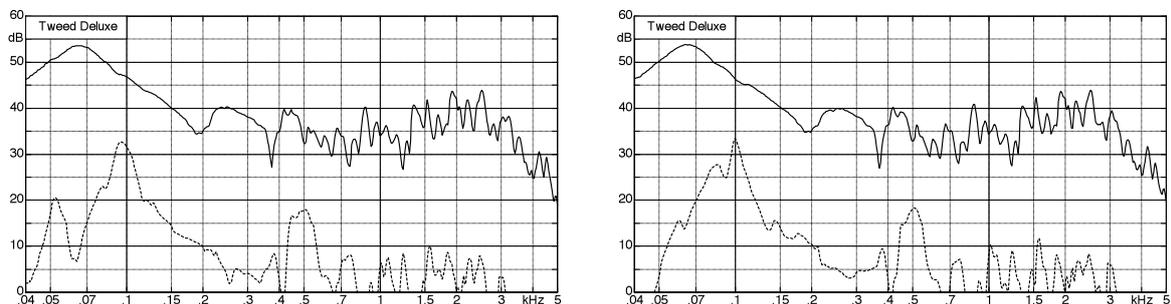


Fig. 11.84: Membrane-velocity in comparison to the frame-velocity for two different cabinet foundations.

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 airtightness (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.

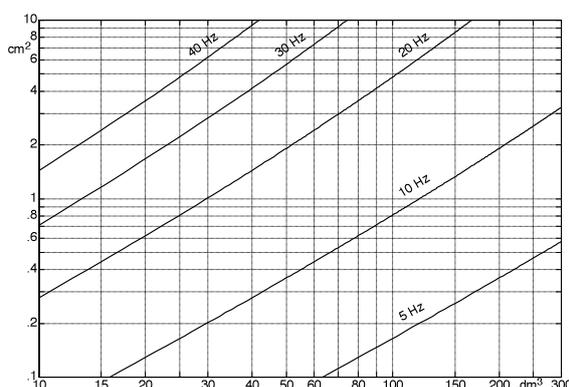


Fig. 11.85: correspondence between volume of the enclosure (abscissa, $\text{dm}^3 = \text{liter}$), area of gap (cm^2) and resulting resonance frequency.

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).

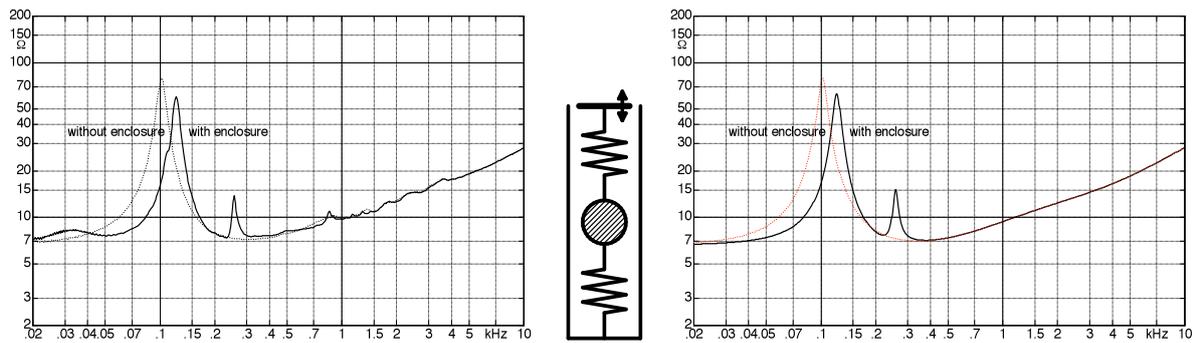


Fig. 11.86: Frequency response of the G12-M impedance (see also Fig. 11.17); measurement (left), model (right). Middle: simplified longitudinal-oscillation model of the $\lambda/2$ -resonance.

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.

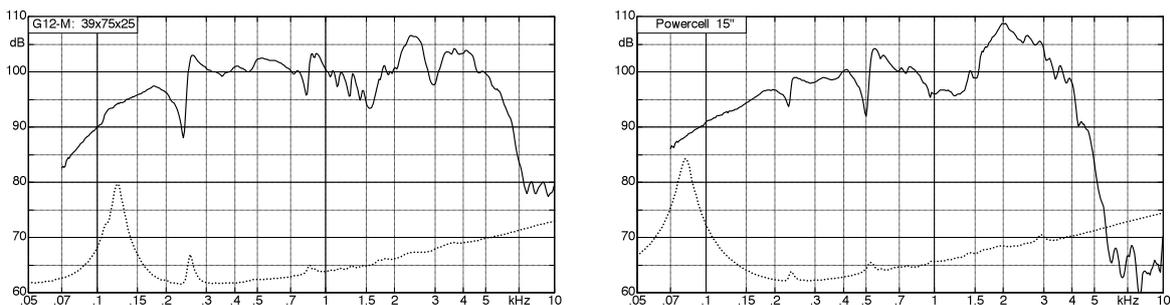


Fig. 11.87: Transmission frequ. response: enclosure with cavity resonances. 39x75x25 (l.) and 40x74x39 (r.).

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.

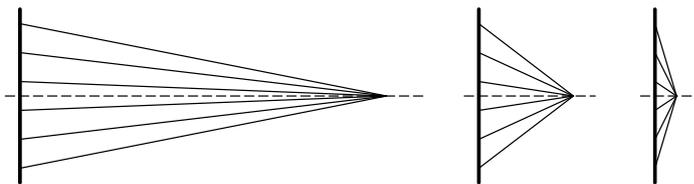


Fig. 11.88: The closer the microphone gets to the membrane, the more the individual sound paths differ in length.

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.

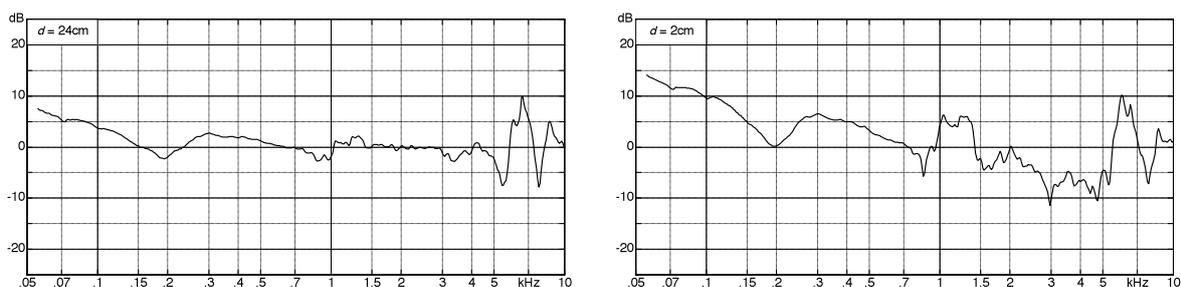


Fig. 11.89: Tweed Deluxe, axial near-field measurements; normalized level changes relative to $d = 1$ m.

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 be 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.

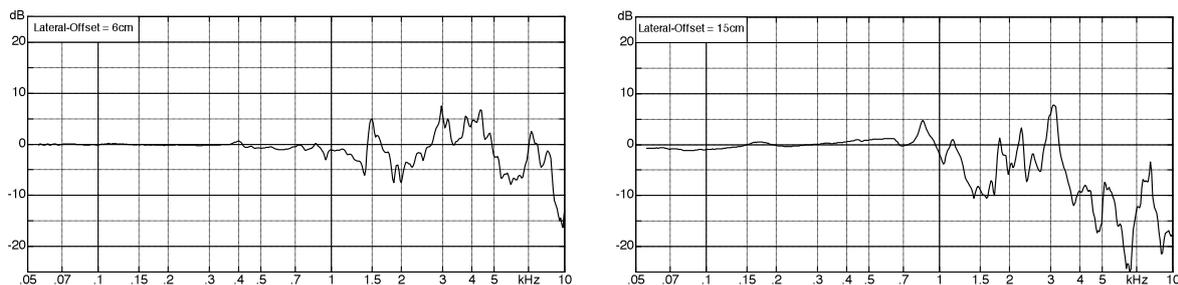


Fig. 11.90: Level changes with repositioned microphone; const. distance to the baffle. Left: 1x12”; right: 2x12”

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.

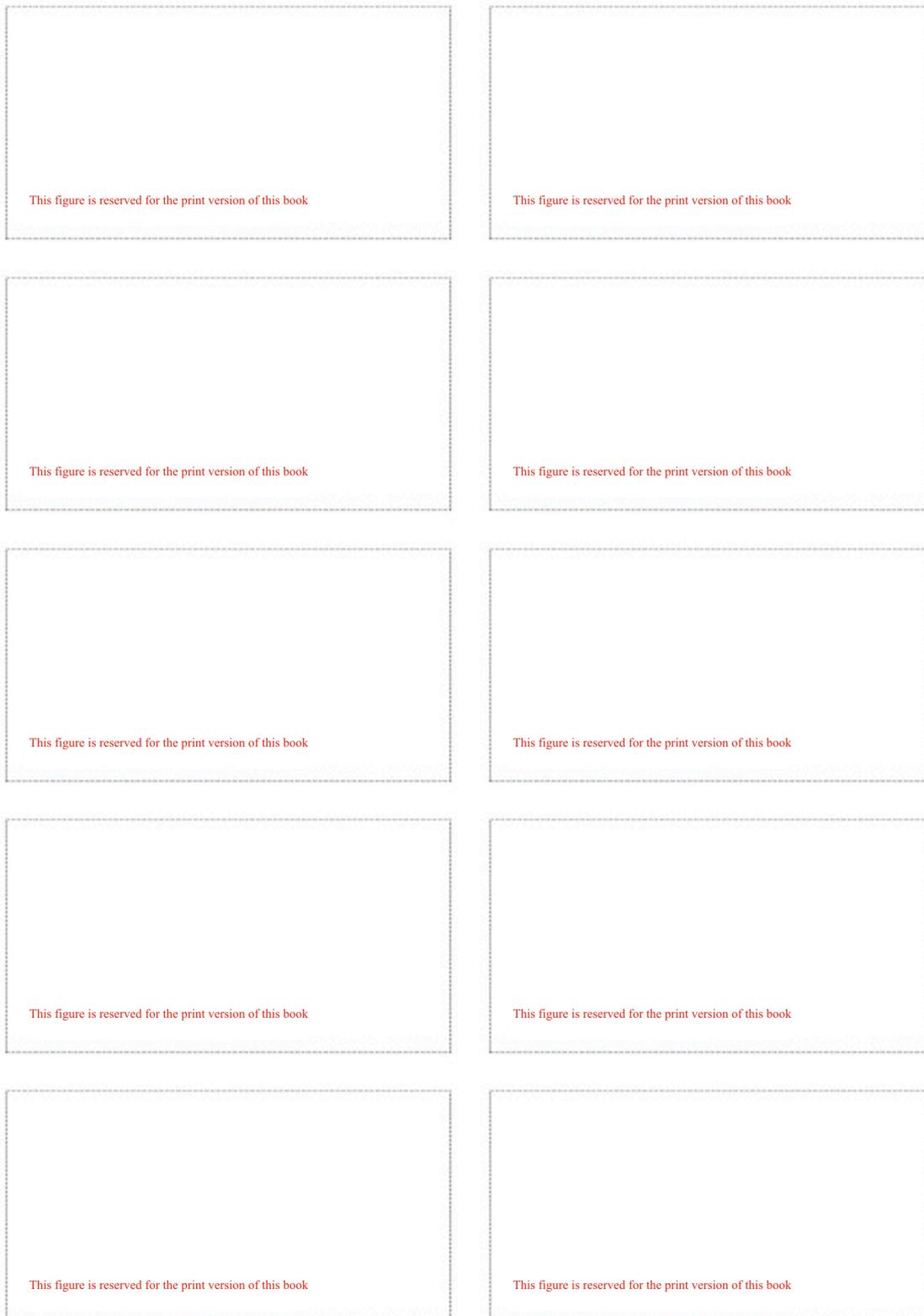


Fig. 11.91: Loudspeaker-emulation in the Line-6 "POD". The absolute ordinate-scaling is arbitrary.

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.

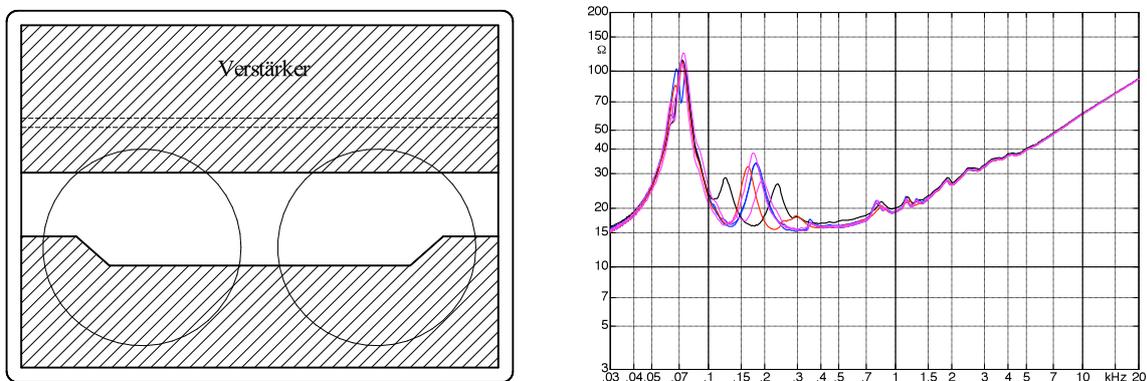


Fig. 11.92: Left: 2-part rear-panel of the VOX AC-30 (“Verstärker” = amplifier). Right: frequency responses of the impedance (series connection of 2 speakers). The different impedance curves correspond to modifications in separation-board and air vents.