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.

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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!

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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 $1/3^{rd}$-octave measurements, the voltage per $1/3^{rd}$-octave was kept constant (pink noise + $1/3^{rd}$-octave-filtering). Polar diagrams were taken in the AEC with $1/3^{rd}$-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 $1/3^{rd}$ -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.

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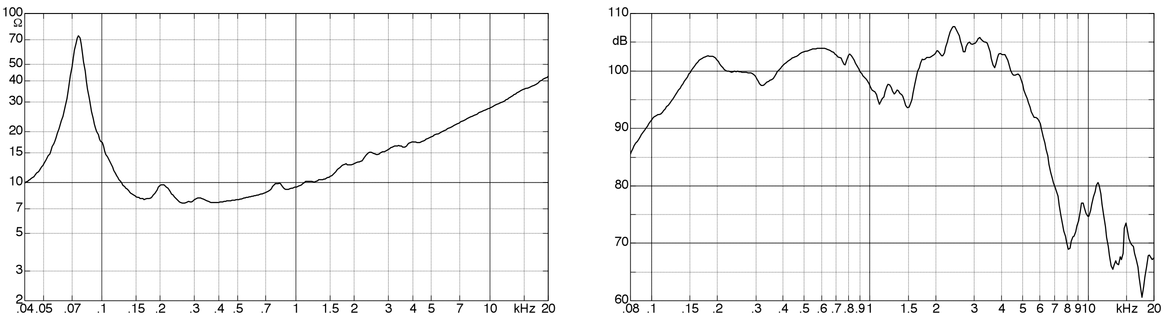

Fig. 11.51: Magnitude of impedance (left), SPL in the AEC (speaker mounted in VOX AD60-VT enclosure).

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.

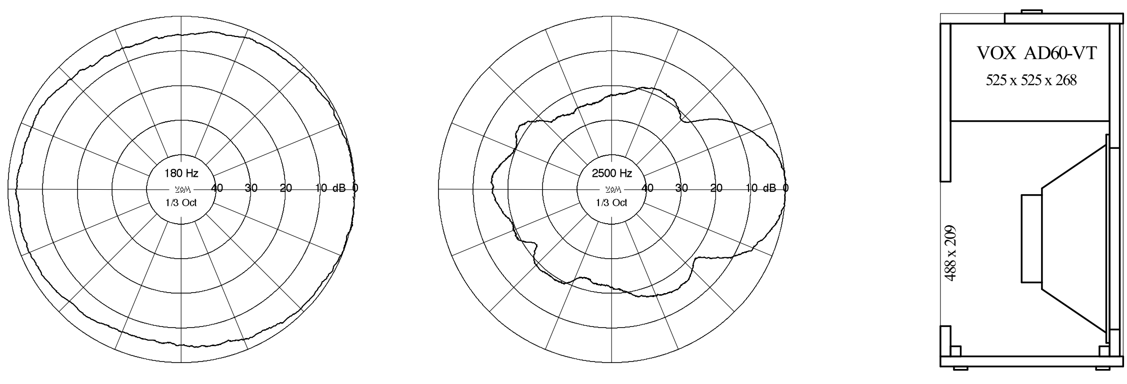

Fig. 11.52: Directional diagram in the horizontal at two different frequencies; VOX AD-60VT-cabinet.

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 $P_{el} = U^2$ / 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.

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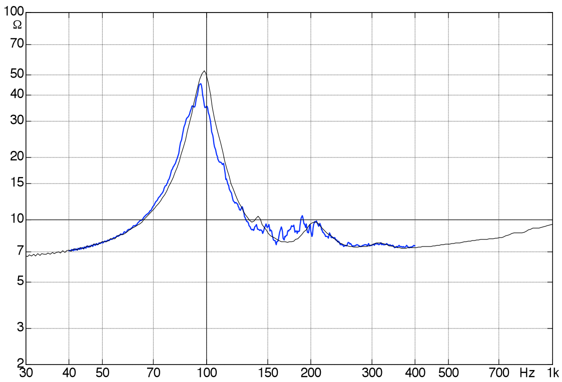
For **measurements in the AEC**, we assume that the sound wave emitted by the loudspeaker is not (or almost not) reflected anywhere; as it hits the glass-fiber wedges that constitute the borders of the room, the energy of the wave is (almost) completely transformed into heat. In this mode of operation, 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.



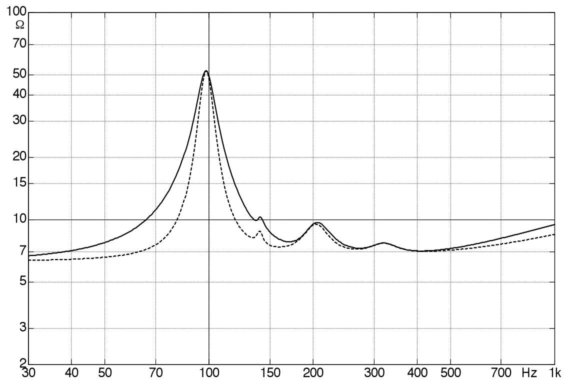


Fig. 11.53: Left: magnitude of the electrical loudspeaker impedance (AEC ——, RC ——). On the right, the comparison between magnitude (——) and real part (-----) of the electrical impedance is shown (AEC).

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.

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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.

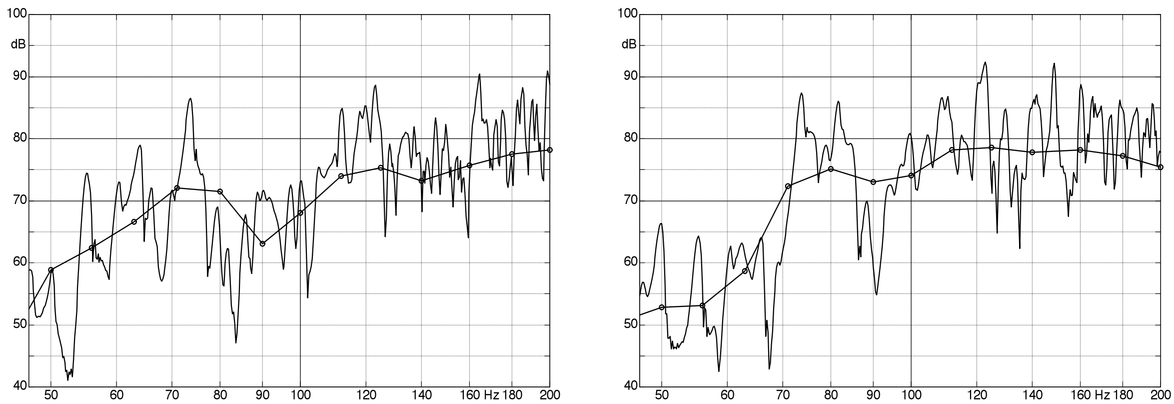

Fig. 11.54: Sweep-measurements in the RC, 2 microphone positions; •—• mean values across a 1/3rd octave.

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".

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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.

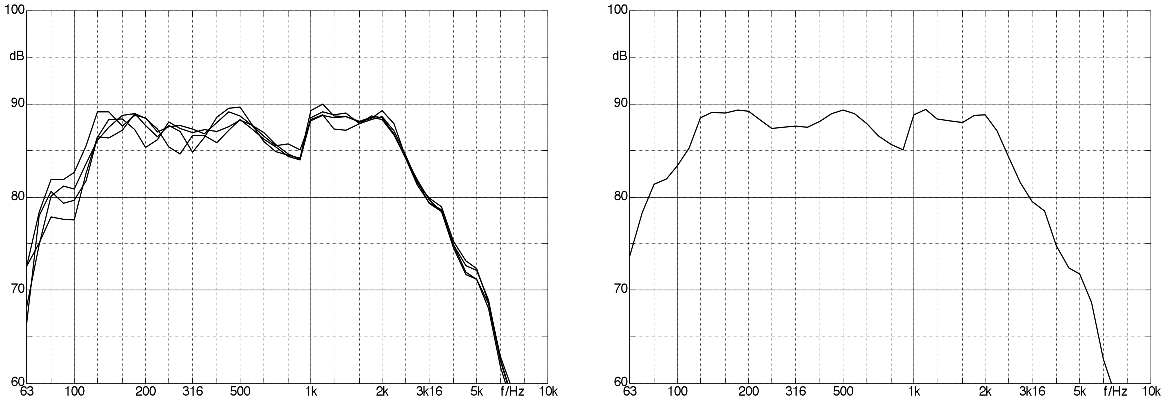

Fig. 11.55: 1/3rd-octave level in the RC, measured at 4 positions with a stationary microphone (left). On the right, an averaging along a circle (not oriented in parallel with the walls) of a diameter of 3 m is shown.

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.

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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 1/3rd-octave (IEC 1260 class 0), with the 1/3rd-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 1/3rd-oct., 1m. RC: 2.8 V per 1/3rd-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 1/3rd-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 1/3rd-octave-noise (**Fig. 11.57**).

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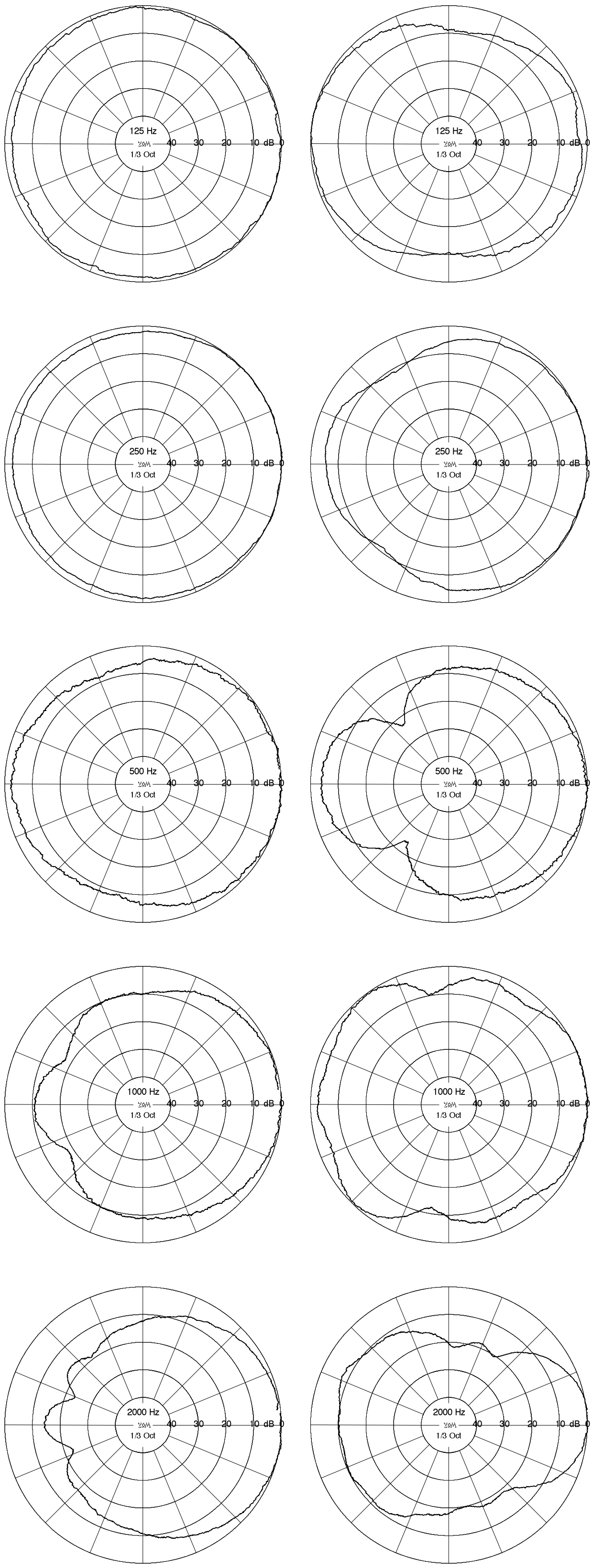

Fig. 11.57: Horizontal directional diagrams.
Eminence Alpha-8 (left), VOX AD60-VT (right).
All directional diagrams are normalized to the maximum.

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".

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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.

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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.

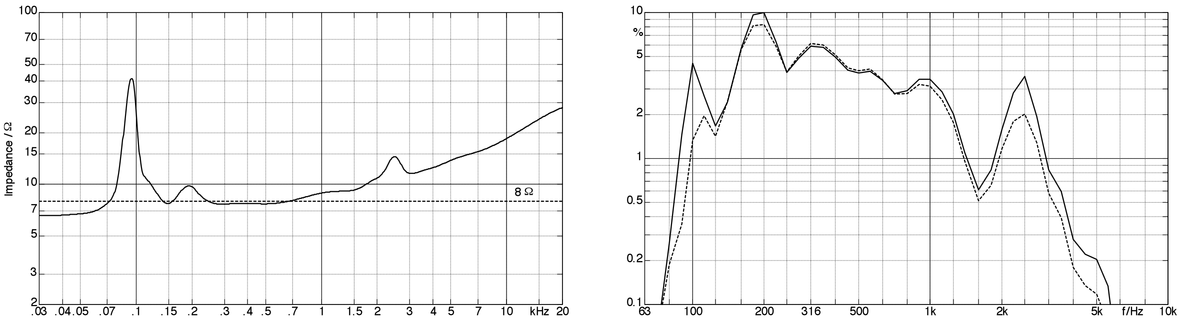

Fig. 11.60: Real part of the electrical impedance (left), comparison between nominal efficiency (---) and actual efficiency (1/3rd-octave average). VOX AD60-VT with original loudspeaker.

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.

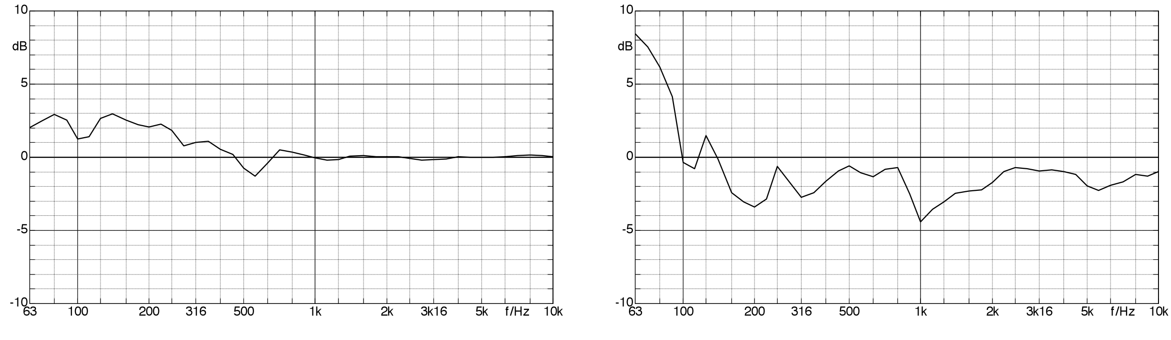

Fig. 11.61: Left: level gain when placing the VOX AD60-VT on the floor (compared to placement on a stool). Right: level loss when closing up the rear of amplifier. Both measurements taken in the reverberation chamber.

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.

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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.

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