10.9 General operating characteristic

The topic of the previous chapters was the performance of individual tube stages or individual circuit sections; in the following the focus shall be shifted to higher-level considerations. The operating characteristic of a guitar amplifier can be looked at from two different angles: from the point of view of the circuit design (i.e. how does the circuit function?), or from the point of view of auditory acoustics (i.e. how does the amp sound?). Of particular interest is to causally interconnect the two approaches – that, however, is also the most difficult task.

10.9.1 Tube sound vs. transistor sound

In fact, transistors seem to have only advantages over tubes: they are smaller, cheaper, have no fragile glass containers, do not need heating. Apparently, they have the single disadvantage that guitar amps designed using them do not sound good. Of course, this is a highly subjective judgment, and of course there are other opinions – however, in particular the early transistor amps had few advocates, and aside from all mysticism there are without a doubt differences from the point of view of systems theory. Still, there is no "tube sound" *as such*, just as there is no "transistor sound" *as such*. A guitar amplifier does not sound better merely because it is fitted with tubes, and a transistor amp does not need to inherently sound bad. It might, though. Fender's Solid-State-Series – the one advertised in 1968 with *'superb sound'* letting you *skim the waters of musical greatness* – was rather unsuccessful. *'Curious refrigerators'* was the term used by the German Gitarre&Bass-magazine in their Fender special edition. To vindicate the name "Fender", one could argue that "this wasn't Fender, this was CBS"; however, after the sale of his company to CBS, the same Leo Fender designed and produced (with his new company Music-Man) hybrid amplifiers featuring a transistor pre-amp and a tube power-amp. These amps – at least today – by far fail to achieve the fame and glory of Fender's black-faced heroes. The same with VOX: the company did not become famous with the transistorized Defiant, but with the all-tube AC30. Guitarists and transistor amps: not a love at first sight.

Transistor amps sound sterile, impersonal, lifeless, they buzz, crackle, sound scratchy, and on top of everything, at the same wattage they are not as loud as tube amps. These subjective judgments elude any circuit analysis. Who wants to stipulate how a guitarist should perceive his guitar sound? Plus: even if this is pure imagination, it is easily conceivable that this kind of imagination has repercussions on the virtuosity. Electrical engineering with its many disciplines is actually only challenged when causal links are brought in: *'hot tubes for a warmer sound'*, or: *'tubes do a rounder limiting and thus sounds less sharp'*, or: *'tube amps sound better because 2^{nd}-order-distortion dominates in them'*. Still, it is not that simple. If the audible differences in sound could be traced to a single reason, we would probably see exclusively transistor amps today. As is the case for public address systems: who would make the effort to stage several hundred tube power amps? But nine tube amps lined-up behind a guitar player: even today, that is not very strange. "Very loud" in the case of nine AC30, and "VERY LOUD" if six Super-Twins are stacked to a pyramid. Why do they do that – what is the secret of the tube? With such a presumptuous question, the answer can only end up in hybris … anyway: the secret i.e. the undiscovered country is in the diversity, in the interaction of a multitude of non-trivial components and characteristics, respectively.

Harmonic distortion, slew-rate, frequency response, input- and output-impedance, shifts of operating points … and all that in combination. What is the effect of the level-dependency of the 4^{th}-order harmonic-distortion on the sound? Does the 4^{th}-order distortion have to be even considered at all, and if yes, up to which order are distortions relevant? To measure the distortion is simple, but to determine its effect on the sound is difficult. For one, comprehensive auditory tests are required, and then each judgment is dependent on many boundary conditions: on the setting of the tone controls, on the loudspeaker, on the listening location, on the guitar, and of course on the generated tones. Because this variety of parameters is vast, the developers of transistor amplifiers do not only have to develop circuits but also “survival strategies”.

One of these strategies is: *as long as the frequency response is identical, the sound has to correspond, as well*. That thought is too simple. Here’s another: *since it is unknown how the characteristics of each individual tube stage affects the sound, every detail of the tube circuit needs to be modeled*. Sounds as if it would be on the safe side – and it would be if indeed every detail were known. And another variant: *we tune and retune until everybody is satisfied, even if the new circuit has no relations to anymore to tube circuits*. Possible, but easily subject to diffuse criticism: something is missing! No one knows exactly what it is that’s missing, but everybody is convinced: that is not the ideal tube-sound. Or the opposite approach: the designers are happy (sounds quite good and doesn’t go up in smoke anymore), the management as well (remained even 2% below the pre-calculation), and the sales department agrees (finally they finished it). It’s juts that not only the calculation but also the turnover is below plan. It is a difficult market: the original Bassman is a legend but the Music-Man is not. Despite the fact that behind both the mastermind was Leo Fender.

This book with its focus more on guitars will not answer the question which type of distortion will render the sound sparkling-creamy-wooden-throaty – that topic belongs to a book exclusively dedicated to amplifiers. Still, there is room for a few basic thoughts. The previous chapters dealt with the non-linear behavior of the amplifier; from the point of view of the author, this is a main theme. In tube amplifiers, several linear and non-linear systems interact: *high-passes* in the coupling-capacitors, in the output transformer and in the loudspeaker, *low-passes* in every tube, in the output transformer, in the loudspeaker. In every tube, in the output transformer, and in the loudspeaker we also find non-linearity. It all makes for an almost unfathomable system – even without negative feedback. It’s not that we couldn’t describe the individual sections of the system – it is the overall judgment that is so difficult. Something that is routine in the LTI-system develops into a vast problem for coupled non-linear systems. For example, it is only in the linear system that it makes no difference whether filter-poles are realized in the electrical of the mechanical domain – here we can compensate e.g. a treble-loss of a loudspeaker by an electrical filter. If indeed linear behavior is desired, equal-sounding amplifiers can easily be built with both tubes and transistors. With non-linearity entering the picture it gets very complicated, though.

After several decades of searching for the right sound, transistorized guitar amps today have matured to the point where the acceptance can to said to be good. Nevertheless there are still innumerable tube amps on the market, and many guitarists are likely not to buy anything else for decades to come, ready to invest those 200 € now and again in a quartet of tubes. The manufacturers have learned from the mistakes of the early years, and offer well-sounding circuits and amps. However, the guitarists have also educated themselves and now can hear details that 50 years ago would have been classified as unsubstantial.

What adds to the problems that non-linear amplifier circuits can pose are **psychometric** issues: how do we measure auditory perceptions? Here, the range starts with "plug in, turn up, listen" and ends with round-robin-tests carried out on a global scale. For guitar amps, we mostly find the former experimental method: listening test in the store, in the rehearsal room, in the editorial office. The results of such test are often ignored or contested on the side of the acousticians (who typically have a scientific background) – not so much because of the involved jurors (normally not that wee known) but because of the unscientific approach lacking objectivity and reproducibility. Is the hand-wired boutique-amp praised primarily because it hails from California and sets you back 5000.- €, as every person testing the amp is made to know first of all? Would the amp be as convincing if it would remain unknown and hidden behind a curtain? We find a nice example for such a scenario described by Uli Emskötter in SOUND-CHECK magazine [issue May, 2000]: in a test of guitar cables, all involved perceive "*pronounced differences in sound*". A week later, in a repetition – this time as a blind-test – shows: "*bewildering result – the judgment came out as entirely different*". It is nothing new to psychologists that the type of presentation procedure will influence the result of perceptional tests. The insights these experts have regarding **experimental methodology** are beneficial for listening experiments, as well (see Chapter 10.9.4).

Guitarists not only perceive the sound of their guitar but they also evaluate it. For the **perception process**, a relatively small inter-individual variance may be assumed, however the **evaluation process** always depends on various boundary conditions. We all know this: first the new CD by the latest superstar is lauded to be among the 10 best releases of the year, next thing we know it appears in a TV-show listed among the most embarrassing oeuvres. Although the auditory event remains exactly the same, the evaluation of it changes. Another example that every studio musician is familiar with: you do a mix-down, find a suitable mixer-setting, everybody is delighted and calls it a day. The next day you listen again – without any changes to any setting – and everybody is disappointed: the vocals are too loud, the drums are trebly, the bass too fat … or the other way round. The reasons for this change are rarely found the technical issues (loudspeakers cooled down, humidity different) but with all likelihood the difference is found in the changed judgment-standard. The underlying processes may develop over minutes or even hours, but time-invariant, systematic differences (bias, offset) are also known: there is a tendency to set a value controlled by the subject to high [12, loudness scaling].

Here is an episode showing how much our value judgments are affected by cognitive processes. It may be a singular case, but is likely to happen quite often in a similar way. After a gig, a young musician addressed me speaking in highest terms of the *"super-sound"* my guitar-rig had: *"You can't beat the good old AC-30 – that's pure tube-sound."* I am sure the people at VOX would have loved to hear that – although I wasn't using an AC-30. At the beginning of the 21st century, this legendary amp is not as ubiquitous as it used to be, and the younger generation apparently is not as familiar with it. Indeed it was a VOX I was playing through, as the gilded badge on the front of the amp confirms, however it rather was a AD-60-VT. That amp features a transistor-preamp and a transistor-supported 1-W-tube-output-amp. It doesn't sound that horrible, either, in fact it sounds pretty darn good, and its **AC-30TB** model cooperates most harmoniously with the Historic Les Paul. In any case, associating the terms *VOX = AC-30 = tube amp = super sound* appears to be hard-wired into many (though not all) a musician's brain. Had the amp-label not read VOX but Solid-State-MOSFET, the judgment could easily have been "*doesn't sound too bad – for a transistor amp*". Indeed the psyche plays an important partner-role in the wide and colorful world of psychophysics. The psyche's counterpart in this area, i.e. physics, and more specifically circuit technology, will now get some attention as well.

From the many amplifier circuits we chose a Fender- and a Music-Man-circuit (**Fig. 10.9.1**) because Leo Fender was involved both. He probably was not the only designer but at the very least the responsible patriarch. Starting out from the RCA-application-notes, the circuit of the **Twin-Reverb** – as it is presented here – was developed over the years into an internationally recognized standard that inspired competitors, as well. Once the era of the octal-tubes had passed, noval-tubes entered service at Fender in the mid-1950's, and in particular the high-gain 7025 (and its colleagues, the 12AX7 and the ECC83) won the pole-position that they never relinquished again.

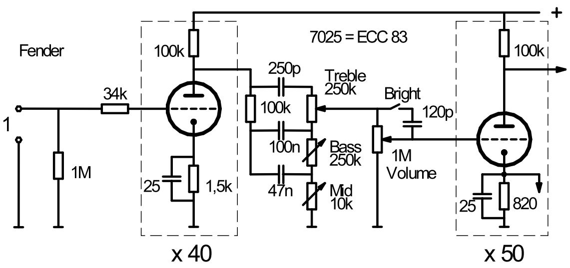


Fig. 10.9.1a: Fender AA763 (Twin-Reverb).

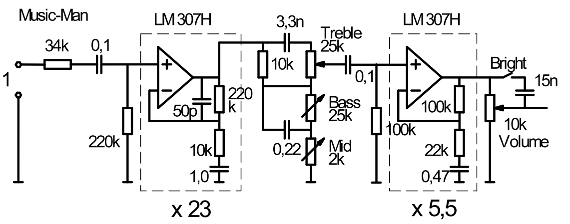


Fig. 10.9.1b: Music-Man 2100

The Fender-circuit starts with a triode-input-stage of a gain factor of about 40. Tone-filter, volume control and intermediate amplifier follow. The Music-Man circuit shows considerable similarity although the input OP-amp has only a gain factor of 23 – a concession to its lower supply voltage that leads to earlier limiting. Despite small differences, the effects of the tone-filters are comparable. The volume pot, however, is connected not directly after the filter but inserted after the intermediate amp. First, these circuits will be compared regarding their linear behavior.

Differences show up already at the input: the MM has lower impedance than the TR: the pickup-resonance receives a stronger dampening. On the other hand, the input capacity is lower in the MM (do consider the Miller effect!). The series-capacitance in the MM-input has barely any effect on the signal, and no shift of the operating point need to be feared, either (due to the symmetrical limiting in the OP-amp input). The 50-pF-capacitor is there to reduce the gain at high frequencies, it has an effect from about 10 kHz The 1-µF-cap reduces the gain at very low frequencies (below 20 Hz). Compared to the TR, the impedance level in the MM-tone-filter is lowered by a factor of 10 to normal OP-amp-typical values. Disregarding this change, the two tone-filters are indeed very similar, despite one missing capacitor in the MM – however such variants with only two capacitors did exist at Fender, as well (e.g. the Super-Amp, see Chapter 10.3). These circuits were modified again and again.

The small-signal transmission factors of both circuits are shown in **Fig. 10.9.2** (referenced to 1 kHz for both). This similarity is not likely to have been an accident; rather the Fender circuit will have been the given objective. The only significant difference in the small-signal behavior is the different input impedance; we can only surmise that the design process was possibly checked with a low-impedance generator such that this aspect did not become apparent. Large differences are apparent, however in the behavior for strong signals i.e. at high drive-levels.

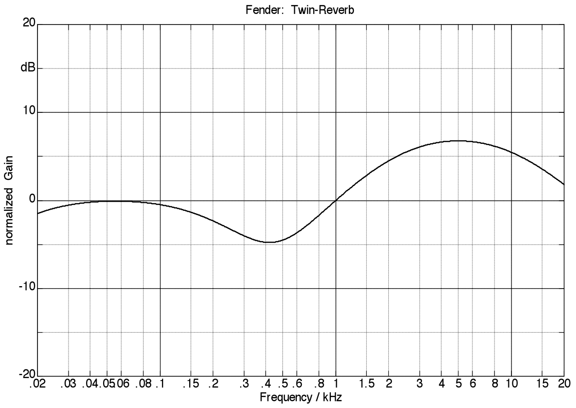


Fig. 10.9.2: Frequency responses referenced to 1 kHz. Bright-switch in “off”-position for both cases.

The drive limit of the amplifying element (tube, OP-amp) is the significant factor for the behavior at high drive-levels. Differences pop-up right at the input: in the TR we have grid-current distortion that is not there in the MM. From about 5 kHz, the MM shows slew-rate distortion but the TR does not. Highly significant: the TR has the volume control positioned after the first tube while the MM has it only after the second OP-amp. With the treble-control turned up fully, the gain factor in the MM from input of the first OP-amp to output of the second OP-amp is about 126. To maintain distortion-free operation, the input voltage must not increase beyond 70 mV. For the TR, this is quite different: assuming 35 V as limit of the first tube for hard clipping, the permissible input voltage would be about 900 mV. Having said that: as already mentioned in Chapter 10.1.4, it is difficult to compare tube- and OP-amp-distortion. Below the clipping-limit, the OP-amp works practically distortion-free, while for a tube, distortion rises continuously across the drive-level-range. **Fig. 10.9.3** shows, for the MM, the maximum input level for undistorted operation. Especially in the brilliance-range (3 – 5 kHz) that is so important for Fender guitars, distortion can very easily occur even though the volume pot may turned up only a bit. In Fender amps, the **Bright-switch** most often is located at the volume control, but for some MM-amps this is included into the negative feedback of the first OP-amp – possibly to reduce OP-amp-noise. Switching-on the Bright switch in the latter case further decreases the treble headroom (right picture), independently of the position of all tone pots and of the volume control. This marks a difference to the Twin-Reverb and to similar Fender amps. The MM-amps therefore do show clear differences in their behavior at high drive-levels compared to typical tube amps.

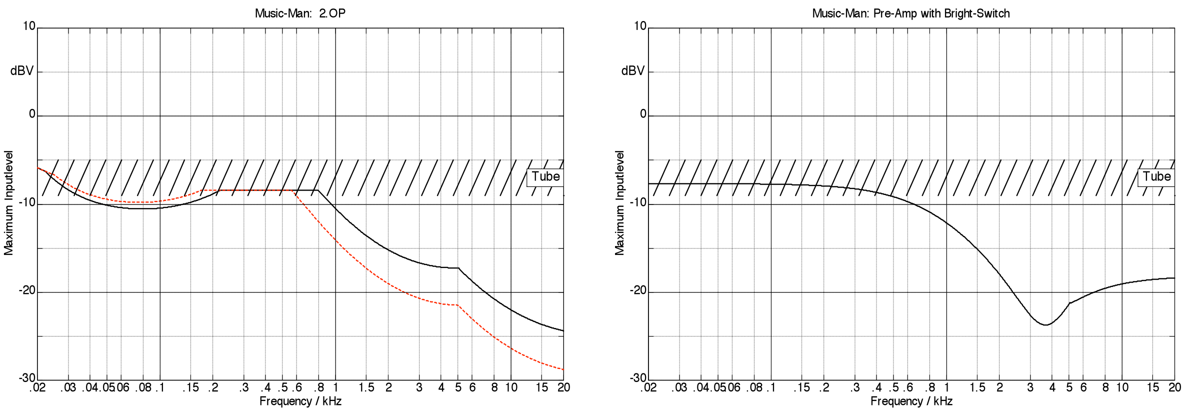


Fig. 10.9.3: Music-Man: maximum input level for undistorted operation. **Left:** solid line = tone controls as in Fig. 10.9.2, dashed line = treble control turned up fully. **Right:** amplifier in which the Bright-switch changes the gain of the input-OP-amp. Hatched area: tube input-stage.

10.9.2 Tube-Watt vs. Transistor-Watt

Allegedly, tube amps are louder than transistor amps rated at the same output power. Before we discuss this topic on a scientific level, we first need to establish what exactly is meant with "this is a 50-W-amp". *Not* meant is the power consumption i.e. the power drawn from the mains. Rather, such a specification always refers to the output power fed to the loudspeaker. If the speaker impedance were frequency-independent and real, this power could be stated without any issue. However, the loudspeaker impedance is frequency-dependent and complex, despite the simple 8-Ω-label. In order to still be able to specify a number of watts, an ohmic resistor replaces the speaker, and it is for this resistor that the indicated number of watts holds. In other words, the manufacturer specifies that an amp generates e.g. 50 W at 8 Ω. This does not in fact tell us how much power this amplifier can deliver into an 8-Ω-loudpseaker, because an 8-Ω-speaker does not have 8 Ω at all frequencies (Chapter 11.2).

In an 8-Ω-resistor, an alternating current of the RMS-value of I = 2 A generates the RMS-voltage U = 16 V; the product of current and voltage yields the power: P = 32 W. In order to clarify that these are RMS-values, a tilde is often put over the formula symbols:

$$P = \tilde{U} \cdot \tilde{I} = R \cdot \tilde{I}^2 = \tilde{U}^2 / R\,; \qquad \tilde{U} = \sqrt{P \cdot R}\,; \qquad \tilde{I} = \sqrt{P / R}$$

For an RMS-current of 2 A, the matching current amplitude is $\sqrt{2} \cdot 2$ A (i.e. 2,8 A), and correspondingly the voltage amplitude of 22,6 V matches an RMS-voltage of 16 V. Multiplying the amplitude-values rather than the RMS-values yields double the power: a 32-W-amp turns into a 64-W-amp. This (higher) wattage specification is not in use in the professional audio technology – rather, the nominal power calculated from the RMS values is specified; in the example this is P = 32 W. What does this power depend on? Its factors are e.g. the squared RMS-voltage, and a more or less arbitrarily defined nominal resistor R that initially replaces the loudspeaker. The resistor is defined as fixed quantity in the data sheet; the voltage is, however, variable. So, for which drive-levels do we specify the nominal power? For studio- or HiFi-equipment, the largest voltage just below distortion level is used, or the voltage at which a certain total harmonic distortion (THD, to be specified) occurs: e.g. 32 W at 8 Ω and for k = 1%. For a guitar amplifier, such a THD-specification is not possible, and therefore – for a sine-shaped drive signal – the output voltage is visually judged to specify at which level clipping occurs. Again: for the calculation this limiting voltage may not be substituted into the formula because it represents the amplitude (i.e. the peak voltage). Rather, this limiting voltage needs to be divided by $\sqrt{2}$. Alternatively, the amplitude is used, and the calculated power is then divided by 2. As an example: for an 8-Ω-resistor, clipping occurs at 40 V. The resulting RMS-voltage is 28,3 V, and the power is calculated to P = 100 W. Alternatively: $40^2 / 8 / 2 = 100$.

Incidentally, it is not sufficient that a loudspeaker box connected to a 100-W-amplifier can withstand merely 100 W. Since guitar amps are typically overdriven, they generate more than the specified nominal power. Given that the nominal power mentioned in the above example is independent of the load, for a square-shaped signal the power would be double, i.e. 200 W! This is because square- and sine-shaped signals of the same peak-value differ by a factor of $\sqrt{2}$ in their RMS-values.

The limiting-voltage (i.e. the voltage at which the output voltage starts to clip) is, however, not entirely independent of the load because the internal impedance of the power supply is not zero. The **power supply** furnishes the operating voltage to the power amplifier – for a tube amp e.g. 450 V. This is a dc voltage the value of which depends on several parameters: on the mains voltage, on the power transformer, on the rectifier, and on the load. In the unloaded state, the operating voltage has its maximum value but buckles ("sags") under load, i.e. as the amplifier feeds power to the loudspeaker. This has a very simple reason: the current flowing to the power amp first needs to pass through the mains-transformer and the rectifier – and either of them causes a voltage drop. The exact voltage and current time-curves are anything but simple to describe (these are coupled non-linear systems), but we do not need to examine this very precisely here. With a load connected, the operating voltage buckles and decreases, e.g. from 450 V down to 400 V, or even down to as low as 360 V. Given a large mains-transformer and an efficient silicon-rectifier, the voltage drops only little; with a small transformer and a tube-rectifier the drop is larger – this is another genre-typical difference. Massive 100-µF-capacitors make the "sagging" (as well as the subsequent recovery) slower than the (from today's perspective) puny little 16-µF-caps. Here we actually may have a difference between tube-Watts and transistor-Watts: modern transistor amps often have very "stiff" power supplies, i.e. power supplies with a small internal impedance the voltage of which decreases only little as a load is connected. Tube amps (especially if they are from back in the day and carry a tube rectifier) have power supplies with comparatively larger internal impedance (see Chapter 10.1.6). Of course the two aspects are not necessarily connected to each other: a tube power amp could just as well include a power supply with low internal impedance – but in particular the legendary amps do not. For a guitar-note played after pause, the full charge of the power-supply-cap is available during the first instant. The limiting voltage may e.g. be 40 V yielding 100 W of nominal power into 8 Ω. However, the voltage buckles after a few milliseconds and the limiting voltage drops to e.g. 35 V. With the power being in a square-dependency to the voltage, the power decreases to 77 W. Measuring the nominal power with a continuous sine-tone yields the second value, i.e. 77 W. For a transistor amp fitted with a "stiff" power supply, the limiting voltage may decrease e.g. only from 37 V to 35 V, so that both amps have the same nominal power. For an impulse, i.e. as a string is struck, the tube amp does however have a higher power; in the example it is 100 W rather than 85 W. In case the limiting voltage of a tube amplifier does not only decrease by 12,5% but by 15% or 20%, these differences become substantially larger.

Thus, one difference in the power yielded by tube- and transistor-amps relates to the temporal behavior: the "attack" is delivered with more power in a tube amp. This holds for the generic circuits – of course it could be designed exactly the other way round. Consequently, the theorist is of course right as he states: "there is no difference between tube-watts and transistor-watts; watt as the unit for power is universally standardized". However, in just the same way the musician is correct in perceiving his or her tube amp as louder. It is not the unit of measurement that is different but the measurement process. A second difference is found in the resistance of the loudspeaker that is not constant, but frequency-dependent and complex. The magnitude of this complex resistance, the **impedance**, may easily reach 20 Ω or 30 Ω at certain frequencies although the loudspeaker is specified at 8 Ω. Not only the copper-resistance of the voice coil contributes to the impedance but also the inductance of the voice coil and the moving mechanic component as they are transformed into the electrical domain (Chapter 11). At the resonance frequency, the loudspeaker assumes high impedance, and the same happens at high frequencies.

Fig. 10.9.4 shows the frequency responses of some guitar speaker boxes: the changes with frequency are most obvious. It is rather up to the manufacturer which impedance value he specifies for the respective box. There are indeed standards for this, however the musician and the manufacturer do no actually shake hands over a sales-deal based on specific DIN- or ANSI-norms. For the following consideration, we simply assume the loudspeaker impedance to be 8 Ω at *one* frequency, and 16 Ω at *another* frequency. If the amplifier has a transistor-typical "stiff" power supply and features an also transistor-typical strong negative feedback, the output voltage will be impressed i.e. almost independent of the load. With a 16-Ω-load, the amp will merely be able to feed half the power that it can generate in an 8-Ω-load. The situation is very different for a tube amplifier: operating it without a speaker could even cause flashover at the power tubes – the voltages that may occur are that high. The tube amp is not actually a true current source, but it does feature higher internal impedance compared to a transistor amp. This has consequences on the power delivery. For example: an amplifier with 8 Ω internal impedance feeds P_1 = 50 W into 8 Ω and P_2 = 44 W into 16 Ω. A (transistor-) amp with 0 Ω internal impedance would generate 50 W and 25 W, respectively. As the loudspeaker impedance increases, the power delivered by a transistor amp will decrease more strongly than for a tube amp. Again, the exact calculation is rather complicated because linear behavior (internal impedance) and non-linear behavior (limiting voltage) interact, and also because not a sine-tone but a guitar-signal drives the amp. Still, the statement remains: your typical tube amplifier will generate on average more power into a loudspeaker than a transistor amp having the same nominal power rating.

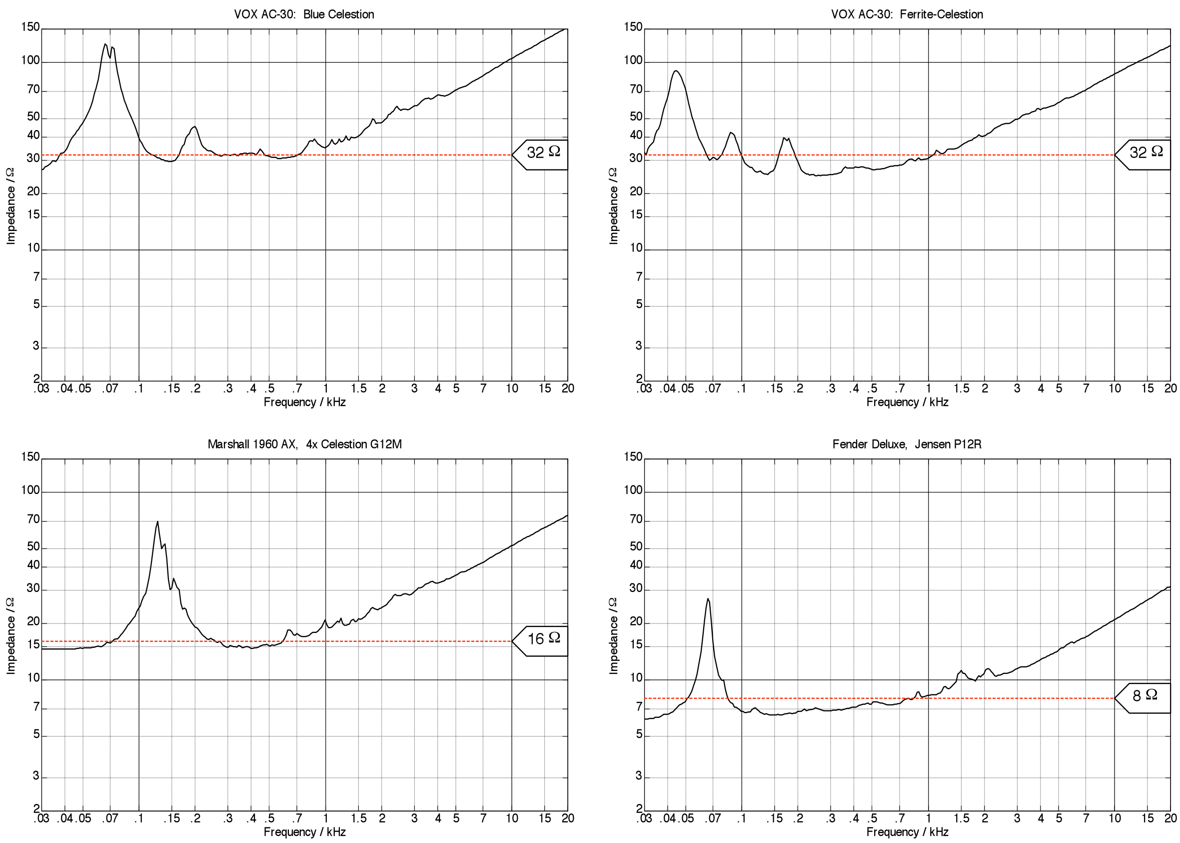


Fig. 10.9.4: Frequency responses (impedance) of typical guitar speakers; measured in a reflecting environment.

As an example we will look more closely at the frequency response of the speaker impedance of a Marshall 1960 AX speaker. It is specified at 16 Ω, its minimum impedance is 15 Ω.

Z reaches its maximum (70 Ω) at 130 Hz. A transistor amplifier rated for 16 Ω and fitted with a (ideally) stiff power supply will feed into 70 Ω merely 23% of the power that it could feed into 16 Ω. In reality, the power reduction will not be as pronounced because the supply voltage will sag less at increasing load-impedance – a reduction to "only" 30% is nevertheless quite drastic. A tube amplifier will behave quite differently: if it is specific for operation with 16-Ω-load, as well, we could expect 60% of the power at a 70-Ω-load, after all – double of what the transistor amp could generate. With a tube amp, the Marshall box will emphasize the frequency range around the resonance frequency, and it will reproduce the treble range more strongly. This tendency cannot be compensated for in the transistor amp by increasing the gain at high frequencies (e.g. by turning up the treble control) because that measure does not influence the maximum power delivery.

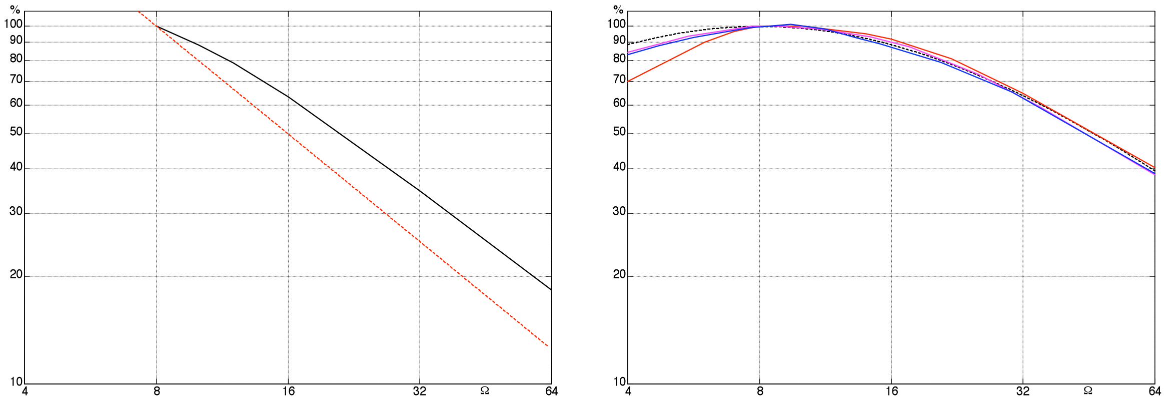
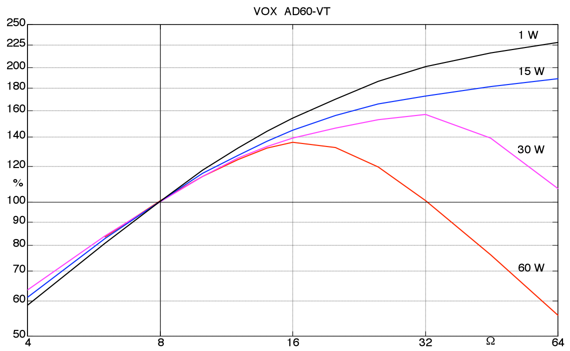


Fig. 10.9.5: Maximum available power dependent on the (ohmic) load resistance. Nominal impedance: 8 Ω. Left: typical transistor power-amp. Right: typical tube power-amp. Dashed: model-calculation.

In **Fig. 10.9.5**, the maximum available power is shown for a typical transistor power-amp and for three tube power-amps, respectively. "Maximum power" means total overdrive. The transistor amp is specified for 8 Ω; for lower loads the amp shuts down. The tube amps are also specified for 8 Ω but can deal with lower as well as with higher load impedances. For the transistor amp, a load-<u>in</u>dependent imprinted voltage was used as idealized **model**, while for the tube amps a constant internal **impedance** of R_i = 8 Ω was assumed. When discussing the internal impedance of a power amplifier, we need to distinguish between linear and non-linear behavior: during linear operation (no overdrive), the typical transistor amplifier features a very small internal impedance (e.g. 0,1 Ω or even less), while a tube power amp without any negative feedback (such as the VOX AC30) possesses e.g. 80 Ω (there are several variants). The AC30 therefore emphasizes already in its linear operational mode those frequency ranges where the loudspeaker features high impedance. In non-linear operation, the internal impedance can only be defined using special model laws; the dashed line in Fig. 10.9.5 was calculated for tube amplifiers and R_i = 8 Ω. Again, the frequencies of higher speaker impedance are emphasized although not as much.

The **VOX AD60-VT** realizes an interesting concept: this guitar amp uses a weak double triode (ECC83) as push-pull power amp and supplements the missing power via transistor-support. The peculiarity here is that the speaker impedance influences the power that the power amp is able to muster. The lone tube is not included as an alibi, as both power-measurement and listening tests prove (Fig. 10.9.6). What is not advertised as loudly, is that in the AC-30-power-amp, pentodes (EL84) do the work while in the AD-50 VT, triodes are on the job. They do, however, this job with very good success.



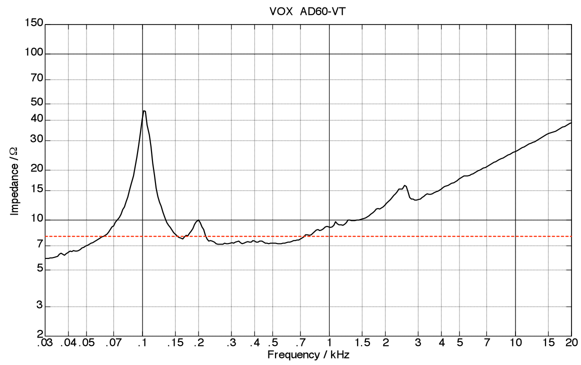


Fig. 10.9.6: VOX AD60-VT: standardized maximum power (left), frequency response of the impedance (right).

Fig. 10.9.6 depicts the maximum offered power dependent on the load resistor. The power-range selectable via a switch is the parameter. The power-characteristic is not really identical to that of a tube amp but the result is quite easy on the ears. Relative to the 8-Ω-reference, a boost can be seen and heard in the frequency ranges with higher speaker impedance – this boost is even stronger than that found with a "true" tube amp. In the 60-W-mode, the power maximum is at 16 Ω – this supports an operation with a (serially connected) second 8-Ω-speaker: more power is available although the treble boost effect is now getting a raw deal.

To summarize: both the impulse power (also termed peak power) and the power delivered in the higher-impedance frequency ranges of the speaker is higher for a typical tube amplifier than for a typical transistor amplifier – with are both rated at the same nominal power for the same nominal load resistance. A percentage value of the difference that would be generally valid can, however, not be given, since the individual circuit concepts are too different.

In closing we should quickly also visit the issue of **loudness** – which in the end is the main aspect of interest to the musician. It is well known that doubling the amplifier power will not always double the loudness. On the other hand, the rule taught in psychoacoustics that for doubling the loudness the 10-fold amplifier power is required, only holds for a 1-kHz-tone. The guitar generates a broadband sound that does not share much with a pure tone, and this naturally needs to be considered. However, of even more practical importance is the fact that the musician judges the loudness of his or her instrument based on how well it can (sonically) hold its own relative to other instruments. In this scenario, the absolute loudness is not as important as the so-called **partial masked loudness** [12]. For example, we may think of a keyboard player sounding a loud chord, and of a guitar player remaining unheard[1] although his amp generates 10 W into the loudspeaker. The latter is not broken at all, but the sound it radiates is fully masked by the sound of the keyboard. As the power of the guitar is increased (e.g. to 20 W), the guitar becomes audible. However, as long as the keyboard is sounded, the loudness of the guitar remains a partial masked loudness and the guitar will be perceived softer compared the loudness it would have when played on its own. For the increase of the partial masked loudness the simple 10-dB-per-doubling-of-loudness law does not hold; a smaller dB-value is valid, e.g. merely 3-dB per loudness doubling. That way, relatively small power-differences gain a bit more significance than basic psycho-acoustical know-how would acknowledge. We shouldn't overdo it, though. The difference between a 50-W-amp and a 55-W-amp remains insignificant. The exact location of the perception-threshold can only be established for each case individually because the masking effects are dependent on the temporal and spectral structure of the involved sounds.

[1] *Translator's comment: the guitar not loud enough - as if that ever happened! Not a realistic example, it seems. Maybe the other way 'round ..... the Leslie for the Hammond won't ever match the Marshall stack, anyway ...*

10.9.3 Coupling capacitors

Coupling capacitors? For some just a cheap run-of-the-mill product about as portentous as a tapping screw, for others a true object of desire worth investing the occasional 50 €. The coupling capacitor separates the AC-part of the plate voltage from the DC-part – it "couples" the AC-signal to the next stage in the circuit. A plate voltage varying between e.g. 200 V and 300 V may alternatively be seen as 250 V DC voltage with a superimposed AC voltage (of an amplitude of 50 V) – the coupling capacitor passes the ac component and blocks the dc component. Let's ignore, to start with, that the coupling cap in fact doesn't let anything through because there's an insulator inside, and let's also disregard that this insulator, on closer inspection, does not perfectly insulate, either. A capacitor lets through AC and blocks DC – that is a good first working hypothesis. We may – or may need to – modify it when necessary. If it is that simple, why do "those in the know", the self-proclaimed amp-gurus, report with a conspiratorial vibe that the long-desired sound presented itself only after swapping the coupling caps? And why is it that – depending on which camp one seeks association with – the original Fender-sound allegedly can only be achieved with the ABC-Orange-Drops, while the opposing camp warns of using exactly these ABC-Orange-Drops (mid-rangy sound, totally unsuitable), recommending rather the yellow Mustard-caps? Wait – not the ABC-Mustards: these are inadequate copies (clones, so to say), you should use the others, the original copies. Even better: use silver-foil capacitors, or, if that much budget can be committed, copper-foil-caps. … "committed" …, no let's not go there, and rather focus, with the naïve curiosity of the researcher/scientist, on the task at hand: trying to find for a grain of truth in that pile of rubbish.

In the framework of the present considerations, a capacitor is a component in an electrical circuit, and as such is subject to the rules and standards of electrical engineering. Whether it has an aura, whether it holds spiritual energy or ethereal psi – that will not be investigated here. The very powerful instrument of **Maxwell's equations** describes, to general satisfaction, the processes in electromagnetic fields. It has made wireless transmission, space exploration and EMP computable; so why not use it as well on the triode-preamp-stage of a guitar amplifier? These equations are the big guns pressed into action whenever starting-from-scratch is called for – but fear not, they will serve here as a mere launch pad that is left behind as quickly as possible. With the limitation to the audible frequency range and to concentrated components (i.e. components smaller than about 1 km in size), Maxwell's second equation may be simplified, resulting in Kirchhoff's second law (the **loop rule**):

$$\oint_K \vec{E} \cdot d\vec{k} = 0 \quad \Rightarrow \quad \sum_{i=1}^{n} U_i = 0$$ Maxwell's and Kirchhoff's 2nd law, resp.

The line integral of the field strength $\vec{E}$ along the closed curve K is zero, as is the sum of the n branch-voltages U_i along the closed loop. Correspondingly, the (slightly modified) 1st equation of Maxwell is:

$$\oiint_A \vec{S} \cdot d\vec{a} = 0 \quad \Rightarrow \quad \sum_{k=1}^{n} I_i = 0$$ Maxwell's and Kirchhoff's 1st law, resp.

The enveloping-surface-integral over the current density $\vec{S}$ is zero, as is the sum of all n node currents I_i (**nodal rule**, rule of charge conservation).

In an electromagnetic field, there are just three quantities characterizing the material: the specific resistance ρ, the permittivity (dielectricity) ε, and the permeability μ. Limiting ourselves to the audible frequency range, we may disregard the magnetic properties of typical materials used for capacitors. Thus, for the formal description merely the loop rule, the nodal rule, and two equations relating to the material remain. Despite this simple analysis, a number of misunderstandings exist that have their roots in the inappropriate application of actually appropriate insights. The area of electrical engineering was in fact not developed to build guitar amplifier but it is, with all due respect, quite a bit older. Besides other important technical fields such as power engineering, one main focus was communication technology, and here in particular the question how to achieve long-distance **wireless** transmission of speech or Morse-code. Heinrich Hertz [1886] and Guglielmo Marconi [1901] are to be mentioned, amongst many other pioneers. The first radio transmission across the Atlantic succeeded a generation before the start of the Rickenbacker/Gibson/Fender-age, and already then capacitors were in use. The so-called "Leyden jar" was invented even much earlier in 1745. Radio transmission, however, does expressly <u>not</u> work in the *audio* frequency range. High frequencies (or *radio* frequencies – sic) are required: 1 MHz for the AM-range and about 10 MHz for the short-wave range. Considering this, and also the fact that the "bibles" from back in the day were titled "The <u>Radio</u> Engineers' Handbook", and not "The Guitar Amp Designers' Handbook", it is easy to imagine what can happen: someone (from the guitar community) reads that caps may have their problems at higher frequencies, and immediately fears for the silvery highs of the famous Fender-sound. As if they had been built for just that situation: there are indeed **Silver-Mica-Caps**! They will make that Fender-treble bounce right back, won't they! The echoism "silvery highs" should not be criticized – that can very well remain here as a term of art. It is also correct that capacitors become **inductive**, but at which frequency does that occur? Even for a wound capacitor (very remotely related to a coil), effects of this inductance happen only above 1 MHz, i.e. at "higher frequencies". However, that does not mean the "higher *audio* frequencies" are affected – there is a factor of about 50 separating these ranges! In just the same way, the **loss-factors** known to "rise towards higher frequencies" will worry only the RF-technician or HF-filter-designer. Typical guitar amplifiers, however, include neither a short-wave-input nor a 12th-order elliptic filter, and the low-loss styroflex capacitor is rather misplaced here (it's too hot an environment for it!).

On top of the difficulties to interpret books on RF-technology in the correct way, we also encounter the problems found in **psychometrics**: how do we measure audio-perceptions? In Chapter 10.9.4 (sound event, auditory event), some suggestions were made, and in particular the need for blind-tests was emphasized. Daily practice, however, looks quite different. For example, a manufacturer states that his capacitors require a run-in time of **100 hours** until they sound good. The guitarist having fitted his (sound-wise not convincing) Marshall with these caps hears exactly that, and reports to all colleagues: "it was only after 100 h that the treble was how it should be." Indeed, it is generally known that technical devices require a run-in time: that relates to the car-engine (1000 km run-in) as it does to the charcoal on your grill (no smoke should remain after 10 minutes. And if a newly elected politician gets a 100-day grace period: why not the capacitor, as well? Let's consider this: an amateur musician playing for 10 h a week will after 2,5 months arrive at the point where he remembers how the caps sounded when they were just freshly soldered-in: it was atrocious, the treble just wasn't there. Now, however, after almost a quarter of a year, it suddenly has appeared. And the reason for this improvement must not be attributed to the tube-aging♣, not to the (possibly new) loudspeaker and certainly not to the strings changed multiple times. No, it must be the caps – in fact the guitar mag has found the same, and even recommends 200 h of run-in time

♣ the manufacturer says (1952 A.D.) that a run-in time is required. Why are we not surprised: it's 100 h.

(i.e. … wait another quarter of a year). And why wouldn't it be the caps – for tubes, nobody doubts time-variant sound changes, either, do they? Whether copper-foil capacitors sound better than aluminum-foil capacitors ('cause copper has lower resistance) – somebody must be able to verify that beyond reasonably doubt! Skeptics pointing to missing data regarding the thickness of the foil could possibly be convinced, if indeed the listening experiments would not show such dramatic deficits. If such an experiment includes the announcement: " I will now solder-in the copper-foil-caps; that will give a much more forceful sound", it immediately becomes futile since all participants will be biased. A much better approach can be found with **Tone-Lizard.com**: here it may happen that the same capacitor is connected to 5 taps of a 6-position switch – not 5 equal caps but in fact one and the same capacitor connected to the 5 positions, and without the judging guitarist knowing this. The latter declares: '*the orange-drop in position 1 sounds much better than the one in position 4*'. No further comment necessary. Obviously, this is just a single case, but just as obviously, the judgment ' *I cannot for the love of it hear any difference*' is not likely to occur a lot because no sound-expert will want to damage his/her reputation. And so capacitors all have their own sound, as does every bolt and every rivet.

It is not possible to find out who first introduced the myth of the 100-h-run-in-time for capacitors. The following might be an explanation, though. In data sheets for capacitors we find a **time-constant** of 100 000 s for high-grade builds – this equals just under 28 h. If now someone has some lingering remains of memory that for a full charging process, 5 time-constants waiting time is a good measure, then we arrive at 140 h … there it is – the run-in time? No, that's not it, at all, of course – these are all data relating to the insulation (!!) behavior (which is not unimportant but an entirely different issue, so let's postpone the discussion of it a bit). The above idle-state time-constant would only play any role for the capacitor by itself i.e. completely disconnected (which you wouldn't want in a guitar amp, would you!?). Connected in its regular habitat, the capacitor is loaded on one end with the grid resistor (e.g. 1 MΩ), and at its other end with the tube and the plate-resistor (e.g. 50 kΩ). Multiplying these resistances with the capacity yields the actual time constant in operation, and that is, at e.g. 22nF · 1,05 MΩ = 0,023 s, significantly smaller than the idle-state time-constant. If you really could talk about a "run-in time" here, and multiplied (according to the above approach) that time by a factor of 5, you would arrive at a "run-in time" of a full 0,12 s.

Alternatively, there is reasoning based on material changes that start only after the onset of a voltage and need to stabilize first. It appears that someone has looked too long over the shoulder of the colleagues dealing with **electrolytic capacitors**. This type of cap uses, similar to the foil-capacitor, a rolled-up aluminum foil – however the dielectric is not formed by plastic foil between the aluminum layers but by a thin Al_2O_3-layer (aluminum-oxide) that grows onto the anode of the capacitor during a **formation process**. It is only this oxide-layer that has the insulating effect, and therefore an unformed electrolytic cap must never be subjected to a voltage. However: what does this have to do with the coupling capacitors that are never, ever, of the electrolytic sort in a tube guitar amp? In the latter, polyester or polypropylene capacitors are used that do not have to – and cannot – be 'formed' in the first place. Still, couldn't there be any other slow-moving and possibly unknown processes at work on the metal surface or in the dielectric? Yes, actually, that is highly likely – the field-strength is high and the temperature, too. Let's assume that a capacitor somehow changes during the first 100 d. What parameters could in fact change? In any case the **impedance**, i.e. the complex ac-resistance, would have to change. Only variations that would have any effect at all on the impedance could change the grid-bias-voltage – hopefully nobody will claim that the sound might change while all tube voltages remain the same. Now, here's the thing about impedance: it is only defined in the linear model; there are no impedances in the **non-linear**

model. Could a coupling capacitor be non-linear? Sure, every capacitor is non-linear – it increases its capacity when a voltage is applied (charged electrodes attract each other, decreasing the distance between them and increasing the capacitance). However, already simple math shows that the resulting pressure (some kPa) remains (together with the elasticity module of around 1.000.000 kPa) below the critical deformation limits by several orders of magnitude. Measuring the THD confirms: as long as a HDK-capacitor (highly uncommon in guitar amps) is avoided, the THD remains below 0,01 %. Thus: we do have linearity in caps – with good approximation.

So: which parameters could change? To start with the **insulation resistance**: it could e.g. increase from 50 GΩ to 100 GΩ (☺), or it could decrease to e.g. 25 GΩ (☹?). Taking 1 MΩ as the input impedance of the following stage, a leakage current of 4 nA (or 2 nA, or 8 nA) would flow at an operating voltage of 200 V, yielding a shift in the operating point of 4 mV (or 2 mV or 8 mV). None of these values represents any reason for concern, plus (most importantly): at an insulation resistance of 25 GΩ, the manufacturer would have pulled the plug because this would be outside of the specification at least for branded products. Incidentally: will you wait for 100 h for the insulation resistance to deteriorate? That could be achieved in a much simpler way. In any case, the insulation resistance may be a shambles (as it actually is for the very long serving caps in vintage amps – sic!), but for new name products it will be good enough, and will not have any audible effects on the sound.

And on to the **capacitance**. Does that change during the first 100 h, right? Of course it does - panta rhei – everything changes all the time. Data books for high-quality capacitors indeed specify a change: e.g. <± 1%, at 70°C , for the first two years. No cigar, then, either. And again we need to note: if the sound were indeed better only after the capacitance has in- or decreased by x% - the improvement could much simpler be achieved than by waiting for all that time.

The **loss factor** remains. It could drop – that sounds desirable even without any exact knowledge (lower loss of whatever) – or it could rise. Hold on … the amp sounds better after 100 h, so presumably it will be *lower* losses? From "smeared sound to clear sound", as the guru elucidates. Has anyone actually posed the question why we would solder a capacitor – no: several capacitors for serious money into our amp, if it sounds “smeared” for 100 or 200 or even more hours? Someone probably did ask, and the stunning answer would be: “because afterwards a undreamt-of sound experience will set in that off-the-shelf products can never impart.” Per Aspera ad Astra – we know this, as well. If the loss-factor were crucial to the sound (and indeed who wants any “loss” in their sound), then the lowest-loss dielectric would have to be used in coupling capacitors, correct? Consequently, and after excluding the les temperature-resistant polystyrene, **polypropylene** would be the right choice. And indeed, it is exactly this material that is found in the Orange-Drop-caps already mentioned. However: the guru from the other side of the fence strongly advises not to include those, but only the ones with **polyester** as dielectric, just as in the original in the 1960’s. The manufacturer adds: “because of its deeper tonal quality" – whatever that may be. Polyester, however, has a loss-factor about 100 times higher than of that of polypropylene – so now the going gets tough. Could the solution be: *more* losses for a better sound? A short calculation comes to the rescue: polyester will give a loss-factor of a about 1% at 10 kHz which – in the high-frequency equivalent circuit (to be discussed later) corresponds to a 7-Ω-series-resistor (for C = 22 nF). Imagine this connected in series with the source impedance of the preceding tube stage (e.g. 50000 Ω): it will be 50007 Ω instead of 50000 Ω – so much then for that approach. Clear? Or smeared?

No, the losses do not help us here. In short-wave resonant circuits, there would possibly have been trouble, but nobody will include a 22-nF-capacitor is such a design. Nope, losses do not explain any sound difference in coupling-C's.

Don't stop now – here comes the **slew-rate**: the cap-manufacturer still has some aces up his sleeves. The slew-rate is the differential quotient over time of the voltage across the capacitor; it is also called the speed of voltage-change, given in V/µs. Again, a glance into the data books is helpful and indeed they show limit-values that must not be exceeded. Examples are 500 V/µs for a polyester capacitor, or a mere 30 V/µs; or 750 V/µs for the polypropylene cap, or even more than 1000 V/µs. Not to forget: the *mica capacitors that considerably improve the sound* excelling at 100.000 V/µs. These are considerable differences, so what is the secret? Let's do the math: a tube that is required to generate an AC-voltage of $30V_{eff}$ at the plate reaches 0,8 V/µs. (N.B.: we work with 3 kHz here since the pickup will never manage full drive-levels at 20 kHz). Or maybe a bit more which brings the slew-rate to just about 1 V/µs. Given the above data, this would not cause any problems. However, the tube may be dramatically overdriven, e.g. by a factor of 30 – and now we would arrive in the range of the above limit-values. Strike!!?? No, still not: the slew-rate happens at the plate but not across the capacitor! The voltage across the capacitor is in fact much smaller, and it even decreases with increasing frequency. That's why the slew-rate is not a meaningful parameter – the **maximum current load** that can equivalently defined for the cap would be more suitable. The data books specify values of 0,1 – 1 A, and even *more than 1000 A for the mica capacitor* – is that good enough? Well, yes – your typical noval-triode is able to supply only a few mA. So again: no issue. But here's a hunch based on this line of thought: the originator of the buzz around the cap might have built a loudspeaker crossover at some point. That scenario provides an entirely different picture – we are confronted with the big-boy-currents: $100\ W = (5A)^2 \cdot 4\Omega$, i.e. 7 A peak current. Could it be that someone has put to (mis-) use the **frequency-crossover** design-rules in the context of coupling capacitors? Sure: the manufacturer of the expensive copper-foil-caps is also known for his crossovers, isn't he? Directly quoting the guru: "it's an unbelievably fat sound …" Unbelievable?

An on we go: the **inductance** of a coupling capacitor now surfaces. It may *today be reduced to the point where 0,8 nH/mm CS can be reached.* CS stands for "contact spacing" and not for "compact size" … but the latter would anyway not be appropriate given the sheer size of these super-coupling-caps (45 mm x 26 mm). In the Roederstein (ERO) data sheets we find much larger Nanohenry-values (12 nH) – but STOP: that brochure dates back to 1980. What was the size of that super-cap again? 45 mm in length? So it's 45x0,8 nH = 36 nH? And ERO was also per CS? No, that was the overall inductance. My, so much has changed since the 80's! Just to mention it: even 36 nH would be o.k. – at 10 kHz a reactance of a full 0,0023 Ω would result. That would have to be added to the 50000 Ω mentioned before. Pythagorean addition to be used, of course.

So, what remains? Not much … maybe the skin effect: that is also an object of adverts. "*At high frequencies* (ah, the old story …) *the current flows merely along the outside of the wire, such that the conducting cross-section is reduced and the resistance grows.*" That is entirely correct: *at high frequencies*. H. H. Meinke always started his famous lectures with just this issue, and noted: "*… and that can start already at 10 kHz*". It is from around that frequency that the resistance of connecting wires increases noticeably – maybe from 0.002 Ω to 0,004 Ω. Nothing could be added here that was not already covered above when we looked at the loss factor. No, the skin effect doesn't contribute anything either, at audio frequencies. And another thing: someone advertises that the connecting wires of his caps are made of tin-coated copper, and not of copper-coated steel reaching merely 30% of the conductivity of copper.

 Translated into English by Tilmann Zwicker

Here's a suggestion for improvement: "*our replica caps come including the connecting wires and not without – the corresponding enhancement in terms of conductivity is beyond anything that could be expressed as a percentage.*"

Lest our commentary turns into mere satire, let's try to arrive at a summary. First, however, a look into the internet: word is that there are rationales that are so misguided that it is not possible to even get close to them on the basis of normal training in electrical engineering. Our capacitor-guru provides some inspiration: *every capacitor brings* ***unwanted effects*** *in the form of additional inductances and resistances (ESR = equivalent series resistance). While the inductance practically has next to no influence, the ESR determines – in conjunction with frequency and capacitance - the loss-angle tan(∂).* O.k. – it is possible to express the issue that way – all text-books on electrical engineering do it in a similarly. Let's move on: *tan(δ) does not remain the same for all frequencies but is frequency-dependent* (that is correct). *If tan(δ) rises strongly with frequency, the frequencies are not all treated in the same way; if there is a smaller change between 1 kHz and 100 kHz at least the frequency spectrum of an impulse mixture is transmitted more time-correctly. The highest frequencies are shifted in terms of amplitude and phase = differential phase-error. For the ideal capacitor, tan(δ) needs to be as frequency-independent as possible implying an ESR that drops towards high frequency. That way, time-distorted frequencies are shifted further towards the MHz-Range (=smallest possible error within the audio-range).* Phew – now we have arrived at a hodge-podge of desire & reality, of science & sales – this is now really incorrect.

The above "sales-supporting comment" could be shortened to: *every signal-carrying capacitor deteriorates the reproduction of impulses, and it does this the stronger, the more the loss-factor depends on the frequency.* Whether there are actually any time-distorted frequencies does not need to be investigated yet, but the reproduction of impulses needs to be looked into. Let's start with an equivalent circuit for a simple high-pass, as it is also presented by our capacitor-guru for a coupling capacitor (**Fig. 10.9.7**):

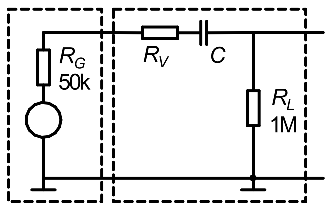


Fig. 10.9.7: The coupling capacitor in the high-pass circuit.

The source (this could be the plate-circuit of the preceding tube) is drawn as a voltage source with a series resistance R_G, R_V = ESR, the load (this could be the grid circuit of the subsequent tube) is shown as a simple real resistor R_L. If needed, this simple schematic could be extended, but for basic considerations, it is adequate. R_V is frequency-dependent – our guru states this, as well. He does not state that in this equivalent circuit diagram, C needs to be frequency-dependent, too – however, since this dependency is small, it may be ignored for now (just as L may be ignored for now). We assume that R_V is a simple, real, frequency-independent resistor, for example a short length of copper wire. Copper? No, why don't we better turn to silver wire as it is offered likewise by the capacitor manufacturer: with a choice of Sterling silver (92,5%), or pure silver (99,97%). For our thought experiment, we naturally take only the purest silver because of the higher conductivity (silver conducts about 5 – 10% better than ordinary copper). Such a short piece of wire will measure maybe 2 mΩ, and this we have to think as connected in series with our source impedance. The two resistors maybe added up yielding a full 50.000,002 Ω. This is one of the undesired side-effects: the source resistance is increased.

What does the loss-factor of this model-capacitor look like? Since the ESR was assumed to be frequency-independent, tan(δ) rises proportionally with frequency. Big mistake, because: *for the ideal capacitor tan(δ) needs to be as frequency-independent as possible,* i.e. the ESR should decrease with $1/f$: $R_V \sim 1/f$. The guru states: *the smaller the increase of tan(δ), the higher the voltage steepness dV/dt and the smaller the differential phase-errors in the audio range – especially for higher* voltages. Voltage-dependent errors? The model of our guru is a linear one and thus cannot include any voltage dependent components. It isn't that capacitors would show no non-linear effects at all, but if you want to describe non-linearity, you have to have a non-linear model. Let's stick to the linear model, though, in order not to further increase the already considerable confusion; for small voltages things are linear, anyway.

Let's go back: we have the allegation that *highest frequencies are amplitude- and time-shifted.* Just to be precise: a frequency cannot be shifted – neither in amplitude nor in time. Frequency is the inverse of the cycle duration: a cycle duration of 10 ms results in a frequency of 100 Hz. What is presumably meant: highest-frequency <u>signals</u> are time-shifted and changed in their amplitude. We could also say: in the highest frequency range, amplitude- and phase-changes occur. That, indeed, is how this would be expressed in systems theory. *Differential phase-errors* – no, you wouldn't really say that. What could be meant here? Maybe it is the spectral derivative, $\mathrm{d}\varphi / \mathrm{d}f$, that usually is supplemented with -2π and designated the **group delay** $\tau_g = -\mathrm{d}\varphi / \mathrm{d}(2\pi f)$. This is in systems theory, usually. It is not problematic to use uncommon terms, but for misinterpretations, the liability needs to go to the party responsible. Again, in a nutshell: *the ideal capacitor does not generate any amplitude- or delay-distortion*♣. Is that what was meant? O.k. then – let's move on.

Not any amplitude-distortion. At another passage it is even more drastic: *the ideal capacitor should not influence the audio signal at all.* So obviously, there must be signals that are not audio signals, and these are clobbered by the capacitor? Correct? No – wrong! In a loudspeaker-crossover, the capacitor connected ahead of the tweeter is supposed to attenuate low- and mid-frequency signals i.e. it is there to reduce their amplitude. Are these signal also audio signals? Of course they are! But lets set aside the crossover – it is indeed possible to talk, in the context of a coupling capacitor, of *small influences on the audio signal, or none at all*. The capacitor operates (together with the resistors) as a high-pass filter and therefore attenuates the (very-) low-frequency components. Our guru, however, seems not interested at all in the low-frequency range, since he localizes – even several times – the undesired effects in the range above 1 kHz: *a capacitor would let high frequencies pass without limitation if it weren't for the losses.* Herein lies a grain of truth: the impedance of a capacitor would continuously drop with increasing frequency if there would be no losses. But wait a moment – is this really about the **impedance**? At 1 kHz, the losses (10^{-4}) increase the impedance of a 22-nF-cap (polypropylene) by 0,0000005%. That's no joke, it's covered in the basic course in material science and components => $\sqrt{1+\tan^2(\delta)}$! Even at 10 kHz (and further up in the audio range), the impedance is not an issue. So, then, on to the **phase**, or rather its spectral derivative. We read that it would be ideal if *voltage and current would occur at the capacitor with an exact 90°-phase-shift between them*. Indeed, that is correct: only the ideal capacitor can achieve that.

♣ In systems theory, the term *distortion* is employed in two ways: non-linear distortion (harmonic distortion) and linear distortion (amplitude-, phase- and delay-distortion).

The real, lossy capacitor shows a phase-shift that is different from 90°. Is that bad? Apparently so, because *differential phase errors in the audio range* result. If the phase shift is frequency-independent, then “all frequencies arrive at the same time” – to stick with the terminology of our capacitor-guru. In other words: signals of different frequencies all are subject to the same delay. However, if the phase-shift is frequency-dependent, the “high frequencies are time-shifted”. Or better: higher-frequency signals are time-shifted (or delay-shifted, or, alas, phase-shifted) relative to lower-frequency signals. Possibly, an impulse will be stretched out, or “smeared”, due to the phase-shift. We do recall that term: “smeared sound”. The absolute phase difference between current and voltage (which for the ideal capacitor will be 90°) can be accepted; the *differential phase-error* (that, in the absence of any explanations, needs to be interpreted as the group-delay) causes impulse distortion. So, our guru had a convoluted and sibylline way of saying it – but what he actually wanted to express is this: **"The loss-factor of a capacitor causes group-delay distortions that smear the sound (at high frequencies)”**. As a remedy – direct quote guru – the loss factor needs to be as independent of frequency as possible, or, in other words, the group-delay needs to be as frequency-independent as possible. At first glance, this sounds familiar: systems theory says such a system is a **linear-phase system**, and certifies a distortion-free behavior.

And with this, we have arrived at the core of this grandiose misunderstanding: the group delay is a transmission quantity (a quadripole-quantity), while the phase-shift between current and voltage is a two-pole quantity. Differentiating the wrong phase will yield a wrong result. More specific: a **quadripole** is a system with four terminals (also called two-port network), with a two-terminal input and a two-terminal output – the high-pass shown in Fig. 10.9.7 would be an example. That input and output in this example have the same ground connection does not make the system a tri-pole – it still is designated a quadripole. Between the input signal (input voltage) and the output signal (output voltage), a complex transmission function is defined from which the frequency response of the phase and of the group delay can be derived. A capacitor, on the other hand, is a **two-pole** because it has merely two terminals. A complex impedance is defined between the voltage and the current, and a phase frequency-response can be derived from this impedance. But try and deduce a group-delay frequency-response from this – that is nonsense. There are merely two special scenarios in which it is purposeful to see a two-pole as a quadripole: if voltage is the input quantity and current is the output quantity, or vice versa. Now, one could argue that every quadripole is on fact constructed from two-poles and that therefore any deficiencies of these two-poles must also be a deficiency of the quadripole. This, however, is not the case. In the present framework, we cannot present the systems theory in its full scope but have to refer the reader to special literature [5, 6, 7]. Very briefly: in the high-pass mentioned above (and equally for an RC-low-pass), the input voltage is divided between R and C – it does not fully span across the capacitor.

The simple formula $d = \tan(\delta) = 2\pi f \cdot R_V \cdot C$ yields the loss-factor d as tangent of the loss-angle ∂. In the ideal capacitor, a sine-shaped voltage precedes the current by exactly 90°; for the real capacitor this angle is smaller than 90°. For a loss-angle of δ = 0,01° (polypropylene at 1 kHz) the phase-shift between current and voltage therefore does not amount to 90° but to 89,99°. If the loss-angle were frequency-independent, as curiously demanded by the statements of our guru, then the phase-shift differentiated with respect to the frequency would result in a constant value of zero (the derivative of a constant is indeed zero). That would seem to be the ideal case: no smearing of impulses.

Now, the phase-shift between the current through and the voltage across a capacitor is one thing, and the phase-shift between the input and output of the coupling circuit is something different – in fact something entirely different.

The phase-shift φ appearing in Fig. 10.9.7 between the generator voltage and the output voltage (across R_L) can easily be calculated:

$$\varphi = \arctan\left(\frac{1}{(R_G + R_V + R_L)\cdot 2\pi f \cdot C}\right) \approx \frac{1}{(R_G + R_V + R_L)\cdot 2\pi f \cdot C}$$ [6, 7, 17, 18, 20].

Fig. 10.9.8 shows the frequency response of the phase in a presentation with both axes in logarithmic scaling – for the ideal, loss-less capacitor! In reality, the group delay♣ depends on the frequency, and dispersive impulse-distortion results. Again: the case calculated here is the best-possible one with the phase-shift between voltage across and current through the capacitor being exactly 90° at all frequencies. However, for a real polyester-capacitor, practically the same figures would emerge – the difference would be entirely insignificant: e.g. for the group delay it would be as little as 0,0004% (1 kHz). Using a capacitor with a constant loss-angle across the frequency range would deliver differences of a similar magnitude. Relative to a load resistor of 1000000 Ω it does indeed not make any difference whether the ESR is 7 Ω or 0 Ω. What does make a difference is a change in the capacitance – shown in the figure for a 30%-increase. It will be discussed later whether such huge **tolerances** can occur at all in a high-grade capacitor, and, if yes, whether they are significant.

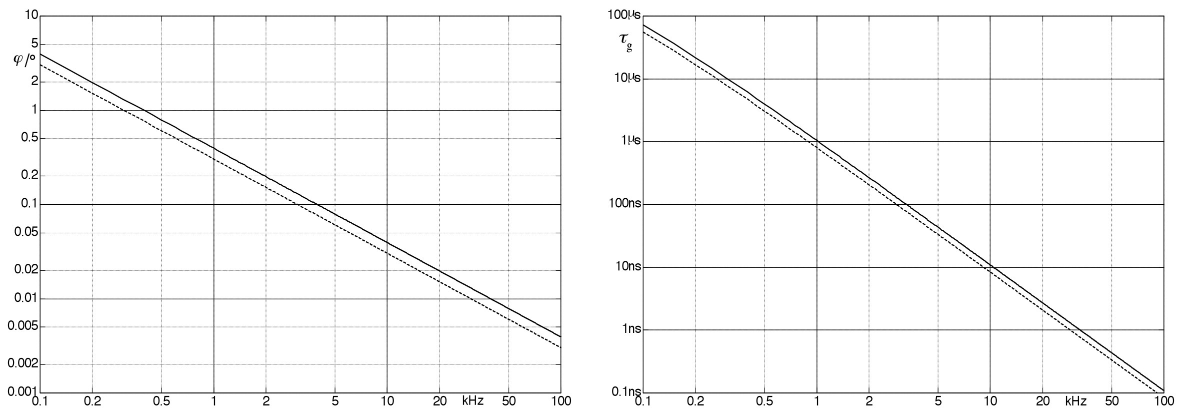


Fig. 10.9.8: Phase frequency-response and group-delay frequency-response of the circuit acc. to Fig. 10.9.7 R_G = 50kΩ, R_V = 0, R_L = 1MΩ, C = 22 nF. The dashed line is valid C = 28,6 nF (i.e. +30% tolerance).

In summary: the conjecture that the loss-angle would have to be as frequency-independent as possible leads to incorrect conclusions, since it is derived from an entirely unsuitable two-pole phase-angle. For a typical coupling circuit with tubes, all capacitors (including theoretical, ideal capacitors) generate practically the same group-delay distortion ("impulse smearing"). This delay distortion is, however, so small, that it remains far below the threshold of audibility.

♣ In case the group-delay is to be derived from the phase frequency response via graphical differentiation (gradient), a representation with linear scale on both axes needs to be used, not with double logarithmic scale.

 Translated into English by Tilmann Zwicker

The last statement needs to be commented on. First, we give the word to our guru: *the value of tangent ∂ does not always tell the full story. If it rises drastically, not all frequencies are treated in the same way; if it changes less between 1 kHz and 100 kHz (as it is usually the case for metal-foil capacitors), the frequency spectrum of an impulse-mixture should be reproduced more accurately with respect to time. In this context, a shallow rise of tangent-∂ between 1 kHz and 100 kHz is desirable, combined with a simultaneously stronger dropping ESR. For the same reasons, developers aim for rise-times of below 1 µs (about 1 MHz bandwidth).* That is plain wrong. The term "rise-time" is used in circuit technology to define the time period during which an impulse response rises from 10% to 90%. The rise-time is 2,2 times the length of the time-constant that in turn is the inverse of the angular cutoff-frequency. From this, the cutoff-frequency calculates as 2,2 / (2π·1µs) = 350 kHz, i.e. not 1 MHz. Even if not rise-time but settling-time were meant, the bandwidth-specification is still incorrect: for 1 µs settling-time, a bandwidth of 500 kHz results for steep-slope bandwidth limiting (since the term is not really used much for a first-order low-pass). Anyway, rise-times of below 1 µs, since: *the superposition of room-reverb onto the original signal needs to be correct to the microsecond, so the ear can pick out the exact location in the space.*

This we need to think about – that is not entirely wrong. The threshold for detecting an inter-aural delay (localization blurring) can be determined to be as low as 10 µs under laboratory conditions, and specialist literature reports even lower values than that. And if you really want to achieve a "safety zone" of a factor of 10, the result is indeed 1 µs. However, to conclude from this the requirement of a bandwidth of 350, or 500, or even 1000 kHz – that would be nonsense. A pure 1-kHz-Tone can perfectly be shifted by 1 µs without tapping into the RF-range. The hearing system can perform the delay-resolution of 10 µs (as mentioned above and if indeed it does that well at all) in the mid-frequency-range, i.e. at around 1 or 2 kHz. At 10 kHz, this just noticeable difference has grown quite a bit (100 µs as a rough guideline – the data depend highly on the experimental conditions), and beyond 20 kHz there is no hearing. Or is there?

Now, every audiophile has gathered (from wherever) that the stimulation with pure sine-signals is something quite different than real sounds because the latter contain tons of impulses. And so one of our gurus manages to demand, on his webpage, on the one hand a bandwidth of 1 MHz, and to refer, on the other hand, to a thesis that very accurately takes the upper limit of hearing to be at around 19 kHz. How does that fit together? We are not talking about 19 or 20 or, even better, 22 kHz – here very casually a factor of 50 is built in, as a reserve. To voice, in one and the same sentence, an opinion and simultaneously the counter-opinion – that is normally only achieved by certain politicians (or showbiz-people).

This mixing-of-what-must-not-be-mixed-up is done – for audio signals – in the following way: *every signal – and that means indeed EVERY signal – is in fact the sum of an infinite number of sine signals*. Yessss!! You can't maneuver around good ol' Baron Fourier. In principle, this statement is correct but we must not take the "in fact" too literally. Mind you: the Fourier analysis is a model consideration, and every signal could just as well be segmented into many other (not even necessarily orthogonal) functions rather than sine functions. What is valid for signals is also valid for systems (as long as they are liner and time-invariant): the consideration of processes in the spectral domain is equivalent to the consideration of the processes in the time-domain [6, 7, 17, 18, 20]. If the hearing system cannot hear continuous tones with a frequency of above about 20 kHz (and moderate SPL), it cannot hear, for impulses, any of their spectral components that lie above about 20 kHz, either.

Now, this finding must intricately be reformulated in such a way that the capacitor manufacturer will get a sales boost. That is done in the following way (under the same web-address): *music is not just composed from pure sine-tones but from a very broad spectrum of different impulses that in part lie very far outside the hearing range but which strongly influence the hearing perception*⊗. *For this reason we use – for audio amplifiers – a capacitor that falters only above 100 MHz. In the coupling branch of a pre-amplifier, this capacitor gives us an impression of how a piece of wire would sound.* No, this is not a printing error – it indeed is supposed to read not 100 kHz but 100 MHz. And a few lines further we find: *mica is the ideal dielectric for capacitors, yielding the following properties: ... applicable up to very high frequencies into the* <u>*GHz-range*</u>. *Mica capacitors are highly favored for filter circuits where, due to their properties, they can bring a considerable increase in sound-quality.* Gigahertz! Is there no end to this?! Is the Terahertz range next? And yet Schöne et al. have already proven in 1979, that a reproduction of the ultrasound range adds nothing whatsoever to the perception♣. That was an investigation carried out by the Institut für Rundfunktechnik (the internationally renowned German broadcast technology institute), though, and in some audiophile circles the preference is not to take note of research done there. Any self-appointed guru who pushes the requirement a further few MHz into the RF range is seen as the new messiah. Skeptics, however, are branded as "*infidel physicists whom one should give a wide berth*". "C'est la gare" is the only congenial answer to that.

Let us revisit the example used in Chapter 8: a bed of a length of 1,5 m will be judged as too short for most grown-ups, while a length of 2 m is quite comfortable. Now, there are a few people who are taller than 2 m, and to accommodate these cubo-philes, a bed should, for good measure, be a bit longer. Taking the above approach used by our capacitor manufacturer for the 100-MHz-capacitor, the bed should be about 10 km long, just to stay on the safe side. Has anybody thought of "indulging" our other senses that way? Our visual sense would lend itself as a candidate: the limitation to the frequency range generally as "visible" (380 – 770 THz) seems overly restrictive, and why not give the TV a correspondingly enlarged bandwidth (i.e. X-ray radiation)? And, of course, that should extend into the lower range, as well: the microwave oven would stand ready to be a splendid "optical subwoofer".

But back to the audio amplifier: the frequency range up to 20 kHz needs to be reproduced precisely, and since no amplifier will shut down abruptly above this limit, a few more tens of kHz are purposeful to let the amp taper off. In case listening experiments result in other numbers, any conclusions may be put under scrutiny on the test bench. Ill-considered phase responses and listening experiments with biased subjects are, however, not conducive. And one more thing about the 10-µs-delay-distortion mentioned above: even smaller values may be audible <u>inter-aurally</u>. Given that, and the fact that most people have evolved beyond mono into stereo-territory, wouldn't it be desirable that the capacitance-tolerance of the wonder caps would be of matching dwarfishness? From this point of view, it is peculiar that one of the manufacturers specifies tolerances of -20/+30%. Sure, hand-made, every capacitor is one of a kind. Or maybe the manufacturer is aware how strongly the group delays of any two headphone systems or of two loudspeakers (of the same type, respectively) can differ? Maybe he knows all this and just doesn't tell? And continues to jumble and confuse things while feverishly searching for the ideal capacitor that blocks and at the same time passes DC.

⊗ That is why they are called "lying outside of the hearing range" (sic).
♣ P. Schöne et al.: Genügt eine Bandbreite von 15kHz... (Rundfunktechnische Mitteilungen, 1/1979).

 Translated into English by Tilmann Zwicker

From the world of the space-worthy 100-MHz-capacitors that can deal with 1000 A now back to the guitar amp. Don't panic: here we don't find either. Even the 10 µs that may be crucial of the inter-aural delay do not give us a headache. For all guitar players that do not carry a stereo-system to the stage: the threshold for diotically♣ perceivable group delays is about 2 ms [3, 12]; one may get to somewhat smaller values via special training, but this is not an issue in practice. So, for sure, there is no "smeared sound" due to the coupling cap with its group delay of τ_g = 0,0001 ms. Things may be entirely different in loudspeaker crossovers where we have large currents on the way – maybe not 1000 A and 100 MHz but still: this is the power-engineering-league. The coupling capacitor plays in Little League: it's communication engineering and a few microamperes on this playing field.

Let's acknowledge the difference (whatever it may be) between power- and communication-engineering, and between research and marketing. After we have (full-monty?-) scientifically shown that coupling capacitors cannot contribute anything – really not anything – at all to the sound, we could conclude with a real bombshell and note that these caps do in practice influence the treble, after all. In fact, that is easily explained and we will get to its in a bit. First, the relation to the equivalent circuit needs to be covered, in more detail. Gotta do it.

In the daily routine in the lab, a coupling capacitor is described via two quantities: capacitance (e.g. 22 nF) and dielectric strength (i.e. proof voltage, e.g. 400 V). The third parameter (the loss factor, is of significance only if the capacitor is connected to inductances. This would be the official position, and according to it all capacitors of equal capacitance would have to sound the same. The teachings of electrical engineering do however also state that the function of a capacitor is of such infinite complexity that only rigorous simplification makes the above analytical description possible. The series connection of an ideal capacitor and an ideal resistor is just about the simplest approximation: more sophisticated models consider special polarization effects, as well, and they arrive at more complex equivalent circuits (Chapter 9.4). However, at middle and high frequencies coupling capacitors are of such low impedance that only a very small AC-voltage is created across them. Of the 30 V_{eff} plate-voltage, a mere 0,2 V_{eff} are found across a 22nF-high-pass capacitor (22nF/1MΩ) at 1 kHz, and therefore the divergence of that cap from the ideal cap is not that significant. At low frequencies, however, the AC-voltage across the capacitor rises: half the frequency – double the cap-ac-voltage, until the cutoff frequency is reached at 7 Hz. Somehow, though, the lows never turn up in reports about the sound of coupling caps; it's always the highs that are smeared, that sound "mushy" or "hollow", and only "open up" after 100 h. Still, we could – for once – consider the lows as well … the real deep lows:

Let us consider once more the high **DC-voltage** across coupling capacitor: depending on the circuit this will be 150 – 300 V, in special cases even more (beware: mortal danger!). If the insulation resistance of a coupling capacitor is e.g. 1 GΩ, about 200 mV are measured across the following 1-MΩ-resistor (at 200 V plate voltage) – for an ECC83, this is already quite a lot (Fig. 10.1.14) and may cause audible effects. However, whether the sound is improved or damaged by this offset-shift cannot be generally predicted. There is always the same rationale: we encounter too many sound-determining parameters. This lack of a general prediction may not be really necessary, any way: for new high-grade capacitors, the insulation resistance is far higher than the one used in the above example, but for decade-old capacitors it many be much lower.

♣ diotic presentation: both ears receive the same signal (mono, both ears listening)

In data sheets we find, for polypropylene-caps, insulation resistances specified to 20000 GΩ, and of 1000 GΩ for MKT-capacitors (each for 22 nF). These values are given for room temperature – which is not the normal internal condition in a tube amplifier where we easily find 70°C. This reduces the insulation resistance by a factor of 5. Still, even then, for MKT-caps the insulation resistance will remain as high as 200 GΩ; for our above example that would lead to an offset shift of 1 mV. That is a value that will not by any stretch of the imagination have an impact on the sound. Measuring various mostly old (but unused i.e. N.O.S. – new old stock) capacitors yielded values between 5 and 100 GΩ – that is clearly worse and could possibly classified as borderline regarding audible effects. However, really bad were the 0,15-µF-caps taken from an old VOX AC30 (of 1965 vintage): one still measured 2 GΩ but the other had dropped off to merely a 100 MΩ insulation resistance. The capacitance-values were still within the 20%-tolerance, but the leakage current shifted the operating point to an extent that in fact should have been designated a catastrophic failure. The amp, however, still worked, and whether the sound generated with this capacitor is judged as good or bad, as broken or as vintage, and the caps *therefore* are judged as junk or holy grails – that must be dealt with in the subjective domain.

Looking at things in a very fundamental way, it is possible that besides the purely electrical parameters, electro-mechanical parameters may also play a role. Indeed the coupling capacitor is charged via a high-impedance resistor, and if the capacitance changes over time, the capacitor acts as an AC-voltage-source – even without a guitar connected to the amp. The same principle as the one for a condenser microphone holds [3]: the high-impedance resistor (of e.g. 1 MΩ) prevents a quick charge transfer, and for an approximately constant charge, any small relative change in charge superimposed on top of this approximately constant charge corresponds to the change in voltage. Specifically: as the capacitance changes by 1‰, an AC-voltage of $U_{DC}/1000$ results. With a capacitor charged to 200 V, this would be 200 mV. Whether the capacitance can really vary by 1‰ is a different question. In a combo amp with speaker and amplifier in one and the same enclosure, we do find high sound pressure levels reaching 100 Pa and more. The resulting forces acting onto the capacitor housing will change the capacitance – but not normally by as much as 1‰. A simple consideration will help to estimate the order of magnitude: as a solid object is submerged in water, it is subject to a water pressure mounting with the submerge-depth. This pressure will crush even submarines made of steel if they dare to dive too deep – a capacitor however is much more fragile than a submarine. The higher the pressure, the more the capacitor electrodes will be pressed together. So, which dive depth might be equivalent to the above mentioned 100 Pa? Which special laboratory could be entrusted with finding this out? 100 Pa makes for 100 N per square meter … that corresponds to merely 1 cm dive depth! So: no special lab – the bathtub is good enough. Although: 200 V in the bathtub … no, better not. Dear music magazine journalists (if you at all accept advice from a scientist): do not try to do this at home! Danger to life! Only as a model experiment: the SPL generated in a combo is about as big as the water pressure at a depth of 1 cm. That should not deform a foil-capacitor to any substantial degree. For an orientating measurement, some brand-new 22-nF-capacitors were charged to 200 V and checked for microphonics: for SPL-values of 130 dB, the AC-voltage generated remained below 0,03 mV. Assuming 30 V to be actual ac-voltage at the plate, this microphonics-induced voltage would be smaller by factor of one million – for sure fully insignificant. Given the multitude of capacitor constructions that have found their way into guitar amps we cannot generally exclude that some capacitors would be among this crowd that exhibit much stronger microphonics – but the likelihood has to be seen as extremely small.

 Translated into English by Tilmann Zwicker

What remains to be looked at? How about reports such as: "*foil capacitors sound somewhat different than polypropylene*"? This statement opens up similar dimensions as "at night it is colder than outside". Very basically, capacitors may construction-wise be categorized into foil-, electrolytic-, sinter- and air-capacitors; for the dielectric, polystrol, polyester, or polycarbonate are in use, or the cited **polypropylene**. The typical polypropylene capacitor is a foil-capacitor consisting – in the KP build – of two foils on top of each other (metal and polypropylene, respectively), or – in the MKP build – of a metalized polypropylene foil. "If you hold two fingernails at a distance of one mm, you get a capacitance of 1 pF" – H.H. Meinke, unforgotten, r.i.p. To keep the in-between space in shape and enhance the insulation, we insert a thin foil in there, e.g. a foil of polypropylene. This also increases the capacitance by the relative permittivity (the relative dielectric constant) which for polypropylene amounts to about ε_r = 2,2, while for polyester it is 3,3. Both plastics belong to the group of **dielectrics** and therefore are insulating materials. The term "insulating" does however not imply that there are no charge carriers within them – the difference is that they are not as easily relocated. Current is nothing else than relocated charge: $I = \mathrm{d}Q \,/\, \mathrm{d}t$; i.e. no movement of charge, no current. In a copper wire the electrons can be very easily moved around (at an astonishingly low speed but in huge quantities), while in a dielectric there are next to no *freely movable* charge carriers present. Still, there are charges: positive atom cores, negative electrons, positive cations and negative anions. As a voltage is applied to the capacitor electrodes, forces are exerted onto the charge carriers, trying to shift and bend them; this is called the **polarization**. Since there are different kinds of charge carriers, there are also different kinds of polarization mechanisms. They are the cause of the **capacitor-losses**.

All materials are "built" of atom cores and atomic shells (model of the atom according to Bohr), and as an electric voltage is applied, an **electron-polarization** will occur in every material: the electric filed-strength shifts the electron shell relative to the atom core. This happens very quickly and is effective up into the THz region. In polar materials (e.g. polyester), the permanent molecular dipoles rotate under the influence of the external electrical field – this is called **orientation-polarization**. In materials containing ions, a counter-shifting of anions and cations occurs: this is the **ion-polarization**. Finally, it can happen in highly inhomogeneous materials that free charge carriers accumulate at insulating grain boundaries – here we have the **space-charge polarization**. All these polarization effects draw their actuation-energy from the electrical field and since none of these processes is reversible, part of the electrical energy is irreversible converted into heat. This caloric energy is not available to the electric circuit anymore (i.e. it is lost) – this is why we have "**losse**s". **Fig. 10.9.9** shows typical values.

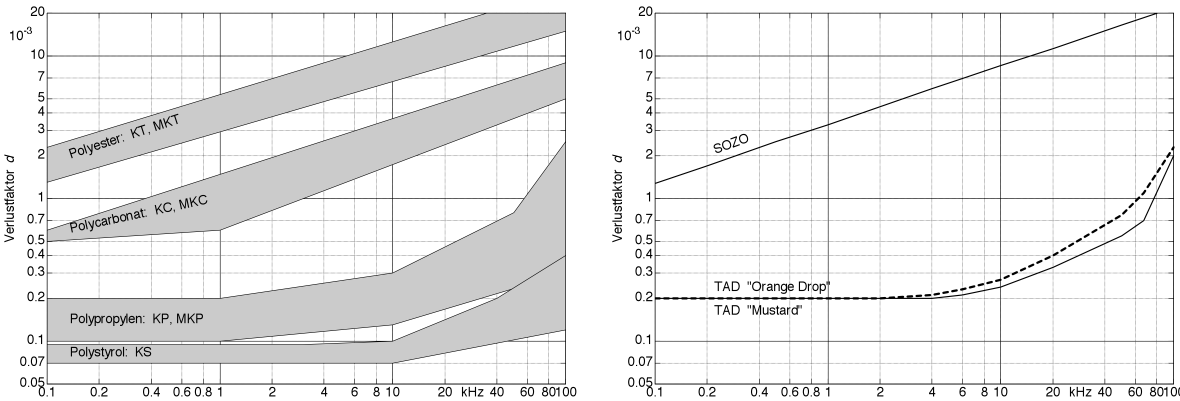
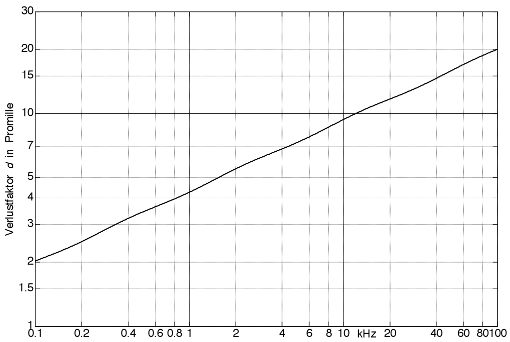


Fig. 10.9.9: loss-factors in typical coupling capacitors. Data-book information (left), measurements (right).

Borrowing from macroscopic effects, we could say: the microscopic polarization movements generate friction and corresponding losses; the latter are modeled as resistor(s) in the equivalent circuit. It was noted already at the beginning of this chapter that dielectric material properties are described ‘merely’ via ε and ρ. These material parameters are, however, frequency- and temperature-dependent (to bring up the most important influences), and in the general case also defined as direction-dependent tensors. Even using a simplified approach, conclusions based on the infinitesimal small cube and applied to the volume of the real capacitor do not result in merely *one single* resistor and *one single* capacitor but in a complicated network with actually an infinite number of components. It is, however, possible to recalculate this structure with good approximation into an impedance-equivalent (or impedance-like) circuit. This **equivalent circuit** has a big advantage over the capacitor model consisting of the two frequency-dependent components $R(f)$ and $C(f)$: it can be used to describe processes in time. The latter would be not easily handled with frequency-dependent components.



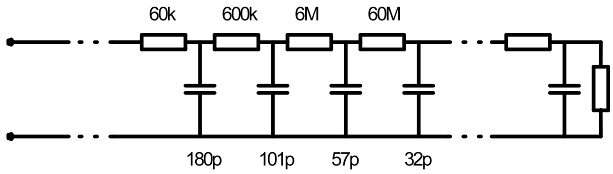


Fig. 10.9.10: Impedance-equivalent-circuit for a 22nF polyester capacitor (continued-fraction expansion).

Such an equivalent circuit is shown in **Fig. 10.9.10**. It is not the only possible one – depending on the desired accuracy, there are in fact myriad variants. In the diagram, we can see the slightly rippled approximation that could easily be improved at the expense of the number of components used. The chosen continued-fraction series expansion includes series-resistors the value of which rises, from left to right, by a factor of 10 each, and parallel-capacitors the value of which decreases, from left to right, by a factor of 1,78 each. (For a reduction of the ripple, both these factors need to be reduced). To the right, the “ladder” continues until we arrive at resistor values that correspond to the insulation resistance (lower cutoff frequency). The continuation to the left determines the high-frequency trend of the loss-factor. For the frequency range shown in the figure, the ladder does not need to be continued to the right at all if the given component values are used. To the left, the continuation needs to happen up to 60 Ω / 1 nF; a parallel capacitor (20 nF) and a series resistor (0,17 Ω) conclude the circuit. We can imagine that, for an insulation resistance of e.g. 1TΩ, the ladder is to be elongated further to the right, but it then becomes also clear how small the additionally included capacitances are (relative to 22 nF). For a capacitor terminated with extremely high impedance, this extension might be required, but for a typical tube circuit (1 MΩ), an equivalent circuit with a largest-resistor-value of 60 MΩ suffices as a good compromise.

Granted, this equivalent circuit is not that simple, either, but with today’s computer-support, “impulse smearing” (group-delay distortion) can easily be determined. However, since a change in dielectric (e.g. to polypropylene) has no audible effect for the typical tube-coupling (as elaborated above at length), we will do without further explanations towards this.

 Translated into English by Tilmann Zwicker

Let us now turn to the question whether there could not be another reason why guitar magazines again and again report of **sound-changes** due to capacitors. Of course, we immediately remember the -20/+30% capacitance tolerances of the super-capacitors as mentioned above. However, first there are components boasting narrower tolerance ranges (the 60-€-mica-caps, for example, have a tolerance of *merely* 10%, at the most ☺), and second, the reports related mostly to changes in the *high*-frequency range. If we insinuate that this judgment does not relate to the MHz-range, but to the upper audio-range, what could be a technically supported reason? The size of the capacitors! Not the capacitance but the physical geometric dimensions. A coupling capacitor may come in axial build of the size ∅6x14, or the size ∅22x35 (mm each). Does size matter? Depending on circumstance, maybe yes. Since this capacitor is (relative to the resistors surrounding it) of low impedance at higher audio frequencies, the plate-ac-voltage is connected across it – independently of the capacitor's polarity. Between this electrode-surface of several square-centimeters and all conducting amplifier components, stray-capacitances result. In many guitar amps, the wire connected to the grid of the tube in question is of the un-shielded kind, and this will create a small capacitance between the coupling capacitor (plate) and the grid. This will not be a big capacitance, maybe 1 pF or 2 pF. Although every amplifier is put together a bit differently, with a big likelihood this capacitance will *increase* if a larger-volume-cap is incorporated. A mere 2 pF – that doesn't sound like much. However, we now need to consider the **Miller-effect** that increases (e.g. for an ECC83) the grid-input-capacitance by 100 pF (or even more) for any added 2-pF-grid-anode-capacitance. The tube itself has, according to the data sheet, C_{ga} = 1,6 pF, which yields (subject to the voltage gain) about C_E = 80 pF. Since the circuit build will not be totally free of capacitances, let us assume in the example C_E = 120 pF. This value would now be increased by the coupling capacitor to 220 pF. In conjunction with the source impedance we now arrive at a

low-pass with 7,2 kHz cutoff frequency.

Do compare this number with the Megahertzes cited in the capacitor adverts and do consider how big the reactance values could be here. Sure, not every amp has to be like that, indeed there are countless variants: Fender- and VOX-amps the insides of which deservedly have been called "birds-nests" of "cable jumble" already by other authors. Then there are boutique amps with wires bent at exactly 90° angles, fiber-boards, turret-boards, PC- and PTP-boards, source impedances of only 50 kΩ, but also of 250 kΩ, plus many more anomalies and peculiarities. And, indeed, stray-capacitances. So, as our guru introduces a hand-wound cap into the circuit with his heated iron while the circle of disciples holds devout silence, and as he calls for a listening test: maybe the sound of the amp has actually changed. This is because some wires were bent, because the plate-capacitors are moved closer to the grid wire by 1 cm, because the performing guitarist doesn't dare to dig into the strings as much in view of the horrendous price, or because the loss factor at 100 MHz has suddenly been reduced. There are even more possibilities, more things between heaven and earth, more knowledge, and more BS (not meant as abbreviation for Bachelor of Science). Science is not always welcome in this vicious cycle, and especially not the science of the electric current. Some authors in musician's magazines generally dismiss their perceived enemy ('*studied physicists*') and advise to '*give scientists a wide berth*'. The latter will reciprocate right away, generally disqualifying every non-technician (or non-scientist ☺) as not having any ability to do scientific work.

Science requires reproducibility; the audiophile realm requires reproduction – that is something different. If an admirer of the arts buys a multi-million-€ painting not as an investment but just because he loves it, why would anybody carry out research to find a technical reasoning? Why would the aesthete want to know whether the green used by Gaugin might be greener by 0,221 nm than the green used by Dali? Somehow, this seems to be different in the area of audio-technology. Here, the Strat-player argues with the particularly high (or particularly low) weight of alder/ash, and the owner of a Plexi reasons with the particular group delay distortion of the Yellow Mustards. A 330B-fan will replace the polypropylene-capacitors by oil-paper caps because – as all the enlightened know – polypropylene is a synthetic, or, in lay-man's terms a plastic and so of course these caps will have that horrible, synthetic *plastic-sound* (advertisement). Do not ask whether an oily sound is actually preferable, because the 300B has entirely different problems: these oily comrades come either with aluminum foil, or with copper foil. Copper has better conductivity, and therefore – says the ad – the copper-sound will be better. A hand-wound aluminum cap will set you back 12 € a piece, but that is anyway more within the low-cost segment in these circles, and does not really match the matched triode-pair (at 250 €). And so copper-foil it is, because: the conductivity is 60% better, and the price is 100% higher – that works as a beginners-set. For the next birthday, we will nevertheless rather reward ourselves with the real deal: with **silver-foil capacitors**, because: silver has still better conductivity, says the ad, and who but the webpage of the manufacturer would know better. So: silver. There's a lingering memory from that dreaded latin class: silver – argentum – Argentarius? Sin-offering … no: money business! That fits: big money business, because: there's not just one coupling capacitor in that radio – er: guitar amp, but there are two … no: three. Per channel! O.K., there's the little box on the on-line order-form: enter "6". And stay strong, as in the box on the bottom the sum appears: 1101,00 €. 'Tis the birthday – off into the shopping basket, done. Well … just to be safe, enter "resubmission" for the next but one birthday – at the latest, replacements should be acquired then, because: for Ag-caps, the manufacturer explicitly mentions the minimum life-time: 2 years. That's not difficult: acquire, solder in, wait for the burn-in time to pass, listen, buy replacements, solder in, wait for the burn-in time to pass, and so on. And in case anybody has any doubts at all about these mod(ification)s: data tables from electrical engineering: indeed, the conductivity of silver is better than that of copper by 6%. *Though this be mod-ness yet there is method*, or so Shakespeare notes.

The capacitance-tolerance of these money-capacity-robbing darlings is specified to +30% … o.k., it is what it is, don't get wound up, they are hand-wound. "*Quality has its price (sic)*" You should not take too narrow a view on the fact that the auditory system can muster the cited µs-resolution – if at all – only inter-aurally i.e. "between the channels". The audiophile writes in an internet chat room: *hopefully this tolerance will not have a big impact in front of a tweeter?* No, no worries – tweeters are generally known to be very tolerant towards minorities. Plus, if indeed any uneasiness remains, for sure there will be someone offering – for something like 2022,00 € – a selected version with smaller tolerance. Don't you even think about the 1%-filter-caps! They are down the cheap end, and there's no way they can sound at all. If only the best is good enough: **selected Ag-caps**. Grab them every other year, or every 10 000 km, whatever comes first.

By the way, what would the synthesis of idiographic♠ und diotic♠ be? Audiophile??

♠ idiographic = describing the very special; diotic = listening with the same signal at both ears.

Dielectrics for capacitors

Mica
Up to 125°C (max. 155°C). Relative permittivity ε_r = 5,5 … 7.
Also: Phlogopite mica, Micanite, Micalex, mica foil, Samikanite (with different data).
Mostly electron- and ion-polarization. Losses are frequency-independent (≈GHz).
Highest stability of capacitance over time; smallest temperature-coefficient.

Polystyrene (**KS**, Styroflex)
Thermoplastic, mostly electron-polarization.
Since 1936 up to 60°C, since 1953 up to 70°C (max. 85°C). Relative permittivity ε_r = 2,5.
Very high insulation resistance, very small losses.

Polypropylene (KP)
Thermoplastic, mostly electron-polarization.
Available since 1960; up to 85°C. Relative permittivity ε_r = 2,3.
Very high insulation resistance, very small losses.

Polycarbonate (KC)
Thermoplastic, mostly electron- and orientation-polarization.
Available since 1961; up to 100°C, max. 125°C. Relative permittivity ε_r = 2,8 … 3.
Very high insulation resistance, very small losses.

Polyethylene terephtalate (**KT**, Polyester)
Thermoplastic, mostly electron- and orientation-polarization.
Available since 1957; up to 100°C, max. 125°C. Relative permittivity ε_r = 3,3.
High insulation resistance, small losses.

Paper, impregnated (P, MP)
Sulfate cellulose, mostly electron- and orientation-polarization.
Characteristics depend strongly on density, water content and impurities.
Depending on the situation only moderate insulation resistance, small losses. Max. 100°C.

Capacitor-oil (Naphthenic oil etc.)
mostly electron-polarization; however: oxidization products (acids) are polar.
Relative permittivity ε_r = 2,2. Copper will accelerate the oxidization of the oil.
Depending on the situation very limited life-time.

Al_2O_3, Ta_2O_5
For electrolytic capacitors, not used in coupling capacitors.

Ceramics, e.g. TiO_2
For ceramic capacitors; not used in coupling capacitors.

10.9.4 Sound event vs. auditory event

On the one hand, it is possible to document the operational behavior of a guitar amplifier via formula and results of measurements; on the other hand, it may happen by verbal description of sensory perceptions. "*Smells like a goat*" would be a genre-typical choice of words, or "*has one hell of an oomph and creates just the right sizzle*", so stick with auditory perception. If everybody knows what an oomph is, this description indeed does help. However, because scientists often do not know what an oomph is, and because they like to quantify things into interval- and relational scales, there are also numerical specifications such as "*cutoff frequency at 5238 Hz*". So, we have, on one side, physics with its objective sound-event data: 100 W, 8 Ω, 5238 Hz, 10 ms. On the other side we find the auditory event with verbal, subjective judgments; louder, much more authentic, vintage-like, throaty sound, too short sustain, etc. In between there is the magnitude estimation: twice as loud as … , just noticeable reverb amount, 50% longer sustain.

Guitar amps mostly do not play for measuring equipment but for people. Okay, they also play for tables, chairs, the dogs of innkeepers and their fleas, but predominantly for people, after all. Whether a measuring device certifies an increase of the effects-mix from 1% to 2% is insignificant if this remains inaudible in both cases. The physical sound event leads – if it is audible – to an **auditory event**, and it is only the latter that is judged by the listeners. The assessment is anything but objective: whether an amp-sound is judged as being good or bad is a matter of taste and depends on subjective criteria and also on environmental conditions. Everybody knows **optical illusions,** and there is no surprise in the fact that there may also be auditory illusions. Nobody will assume that a car speeding away on a straight road actually decreases in size although the optical angle that it occupies in our visual perception indeed becomes smaller. The brain will correct for the shrinking image on the retina and, in a way, creates an illusion. Is it actually an illusion? The car has not shrunk, after all, just the picture on the retina! Anyway, the term "optical illusion" found its way into everyday language.

What is the reason for such illusions? Is a lion that only then a lion when we see it in full, or is it a lion already as it steps out of the bushes, only half visible? This is a clear-cut case of evolution and/or selection. It was conducive to survival to supplement fragmentarily arriving perceptions, and to correct distorted sensory impressions. The immense flood of data arriving from our sensory receptors needs to be reduced momentarily by many orders of magnitude: the data flow taken from a stereo CD amounts to about 1,4 Mbit/s but at best only 50 bit/s of that arrives at our consciousness. However, the synapses working on our internal signal-processing do not just throw, without discretion, 99,996% of the incoming information into the bin; there are rules – but rules that may change from one second to the next, with our cooperation but also without. Since we perceive our environment exclusively through this information-reducing filter, the philosopher arrives at the conclusion: **nothing is as it seems** – and he seems to be right. The "seems" is attributed to the realm of the perceptions (auditory event), the "is" to the realm of physics (sound event). It must not surprise us if a guitarist perceives sound changes if he is being told that a coupling capacitor has been swapped – although the amp remained in fact untouched, and merely the judgment criteria have undergone a change. The opposite may also happen: a capacitor is indeed swapped but nobody hears a difference. And of course there is the third variant: the swap is clearly audible. There are countless guitar amps, if not more – for the individual case no remote diagnosis can be established. The following explanations can therefore only impart basic knowledge but not offer retrofitting plans for specific amplifiers.

 Translated into English by Tilmann Zwicker

Fig. 10.9.11 shows some optical objects. In the first picture we see two crossing straight lines, in the second two overlapping circles. Or are these in fact other objects? Aren't there two angles with meeting apex in the first picture? We could just as well assume that – but the crossing straight lines are simply more obvious. **Our brain always chooses the interpretation of reality that is more likely**. In this case, this is the crossing of two lines (or two tree branches that have fallen on top of each other). For the same reason we do not recognize, in the second picture, a crescent and a waning moon with a convex lens-type area in between, but two circles. In the third picture, we see two triangles on top of each other that do not at all exist in the drawing. In particular, the "upper", white triangle is predominantly "make-believe" rather than "actually being". The right-hand picture conveys a depth in space that is not at all present in reality. And although this picture does not change, it can "jump" in our perception: one moment we see a cube on the floor, the next we see a cube hanging (fastened with its rear surface to wall) towards the left … or towards the right. Visual perceptions seem not to correlate perfectly with the optical stimuli.

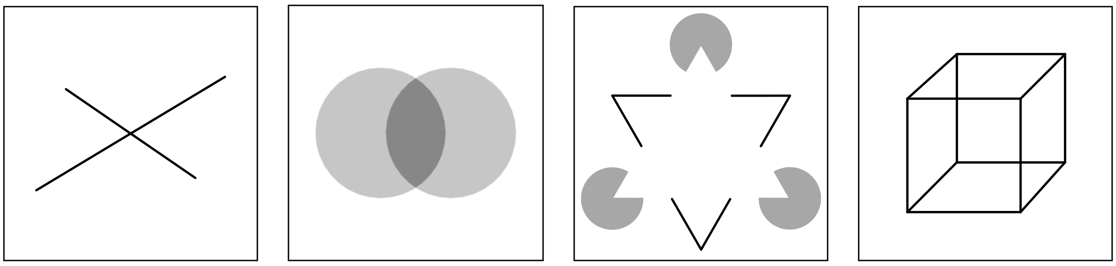


Fig. 10.9.11: Examples regarding the visual perceptions of optical objects. For more examples see D. Picon 2005.

Consequently, we should not be puzzled if auditory perceptions change as well, without any alteration in the acoustical sound event. A special experimental methodology is necessary to establish whether or not there is in fact a **causal correspondence** between a change in our auditory perception and a change in the physical sound event. How would a guitarist who has just swapped a capacitor in his amp (and now plays to check out the result) judge whether any perceived difference in sound is due to the changed capacitor, or due to the (unconsciously) changed way of playing, or due to the (unintended) change in the listening position, or due to changed judgment standards (autosuggestion)? Psychometrics has a few hints here: for example, the sounds to be judged should be presented such that the test person does not know which sound is presented at the given time ("blind"-test). The sounds should have a duration of only a few seconds, and the interval between sounds should be short (about 0,5 s). In a comparison of pairs (A-B-A-B) only a single parameter should be changed at a time. How much does a demo-CD for replacement pickups tell us if there is a different guitar-riff for each pickup, and if possibly different players have recorded the riffs? Not much!!

The first run-through of a listening experiment could, for example, contain simple **nominal verdict**s: the perceived sounds sound **the same or different**. To increase the certainty of the statements, it is necessary to have the subject judge identical sounds without the subject knowing that this pairing is included. A subject that repeatedly hears differences when identical sounds are presented (perceived A-B-A-B is in reality A-A-A-A) will either uncover faults in the experimental setup, or he/she is unsuitable as a test subject. If two sounds are, objectively, not significantly distinguishable, the question about which sounds better is moot.

If A and B are judged as sounding different in the auditory experiment, the second experimental stage can serve to ask about comparative ranking characteristics (**ordinal characteristics**): "*I like B better than A.*", or "*B sounds more distorted than A*", or certainly even "*B has more oomph that A*". In the last stage♣, quantitative **cardinal characteristics** are addressed: "*I would spend 100 € more for B.*"

In order to judge the subjective difference in sounds between A and B, the exact objective difference between these sound needs to be known – that should be a matter of course. For a listening test on the sound of capacitors (Chapter 10.9.3), this implies that the amplifier is always driven by the same signal, i.e. not by a guitarist (with a guitar) playing this now and that then. Rather, the guitar is recorded *once* in an appropriate way, and this recording is fed to the amp in an identical manner for the listening test. Specialist knowledge is indeed required in order not to destroy the sound already by the experimental setup. As a result, the following could be obtained: "*Of 20 subjects only 3 could hear a difference between A and B.*" Or something like: "*15 of 20 subjects judge A as sounding better but would on average accept no more than 10 € additional cost.*" could be the result. Still, even such tests leave questions unanswered: anybody who has not personally participated will not know whether he/ she would belong to a) the 15 or to b) the remaining 5, and if a), then the pecuniary equivalent might be as much as 500 €, as well. In general: if I am asking for the opinion of someone else, then I will receive the opinion of someone else – that is highly trivial. If I want to rely solely on my own opinion, then I need to test everything myself (and why not?). If I do ask another person, I might be i.a. interested in how reliable this person's opinion is. In such a case this approach holds: for a prejudice-free subjective judgment of objective issues, blind tests provide a powerful tool.

But what about those instances when the sound of an amp changes without identifiable objective reason? Those cases when an amp has lost its unique sound after a repair job, although it was – embarrassment city! – accidentally shipped back without having been opened up? The case of the guitar that never sounded right again after it had been kidnapped for a stage-quickie by a pal. Or the case of the capacitor-swap that led to a sound miracle although everybody (or rather all "studied physicists") tirelessly continues to emphasize this to be impossibly? There could be physical reasons (transport, shift of a slightly loose guitar neck, stray capacitances), but we might also see in such cases the impact of judgment benchmarks that are easily influenced. Most people fancy themselves to be superior to the average in many areas, and prefer that their equipment to stand out from the mainstream: alloy wheels … or copper caps. No sooner than a prejudice takes hold, it is pampered and cultivated – the smallest confirming hint is scraped up and blown out of proportion while every counterargument is conveniently ignored. As a rule, every confirmation is trustworthy while every disagreement is questionable. No one is spared this kind of delusion: 94% of all scientists at university deem their research to be above average! *The deeper reason for our biased dealings with information stems from a conflict between the search for truth, and the search for harmony and for agreement with ourselves. To admit that one has been wrong can, after all, chip away at one's self-esteem and one's image.* [R. Degen, Lexikon der Psycho-Irrtümer – *lexicon of psycho-errors*]. This is why an assumed change can lead to a change in perception. If, after 100 h of playing the new capacitors, suddenly the treble comes to life, the underlying mechanism is not necessarily an objective reason – the belief is already sufficient. It is a rather big paradox that training can render our hearing more precise but at the same time more susceptible to influence.

♣ These results may be achieved as well in a single run-through, if a matching evaluation-statistic is employed.

That the brain can be trained is without a doubt. **Practicing** for many years fine-tunes the auditory performance, makes small differences stand out, allows for more comparison patterns to be available, and enlarges the sensory areas in the cortex. From the awareness of above-average hearing-prowess, the idea can easily arise that "the whole hearing" is now perfected and has become the unswayable calibration-standard. In this, it is easily overlooked that numerous auditory functions are not (or only to a very small degree) trainable, after all – they function just as they do for the untrained and are therefore – relatively seen – worse off than at the beginning of the training process.

An example from optical processing: for the cube in Fig. 10.9.11, we can decide whether we want to see it as one whole object (the cube), or as individual lines. Everybody with normal vision can do that; it does not require special training. For acoustical objects, however, different rules apply: in a complex sound made up from partials (harmonics) it is much harder to hear individual partials; often it is even entirely impossible. A simple trick may help: a special (non-masked) partial is switched off (filtered out) for a short time and then switched on again. At the switching-off instant we hear, as expected, a change in sound (thinner, more hollow). As the partial is switched back on, there is a surprising effect: first, the thinner, more hollow remaining sound is joined by an individually audible sine-tone that "melts" into the remaining sound within a few seconds to eventually form the original sound. Something new, especially when appearing abruptly, is deemed important, and the brains switches to "make individual object audible"-mode. After some seconds, the new additional object is categorized as a kind of prodigal son perfectly fitting in with all other objects, and the precedence circuit is switched off again: the partial is not audible per se anymore. No training can change this effect. The auditory perception changes although the sound remains static! On top of such autonomous (endogenous) signal-processing algorithms, other external (exogenous) signals affect the perception process: directional hearing is influenced by visual clues, as well, as is the impression of reverberation and even speech intelligibility. Nothing is, as it appears, and everything appears different

A real-life example shows how difficult listening test can be: in a pretty hefty pickup comparison test (Gitarre&Bass 2/05), there are 10 pages of verbal assessments: *"In comparison almost mushy ... the picking attack substantially softer and brittle ... surprisingly glassy and rich in harmonics ... an entirely different spectrum in the mids ... far less richly colored ... acutely transparent and translucent ... a sound beautifully soft and compressed ... a very creamy tone that however seems a bit dull and lackluster ... although completely covered in wax, the pickups sound open and as airy as un-potted ones.*" These short excerpts indicated that clearly audible differences must exist between the judged pickups. Some 2 years later, the same magazine publishes a flash-back to the same test. This flash-back arrives at the conclusion that "*in fact all models sounded almost the same."* (Gitarre&Bass 5/07) The difference between "*entirely different*" and "*almost the same*" has to be seen – according to the flash-back – in the different recording situation. Mind you: for "*almost the same*", the recordings were not done in a garage but again in the recording studio, and getting "*good and professional results.*" Based on this, every reader can pamper his personal prejudice: one will shell out 400 € for a pair of PAF-clones and enjoy the exclusivity, the other will (because of "*almost the same*") stick with the equipment he already owns, and prefer to perfect his finger vibrato – chacun à son goût. Another one may comment on the published sound examples from the above test with "*You must all be mad! There's nothing to hear but one and the same pickup again and again!*" (Gitarre&Bass 4/08). It does dignify the author of the article that he has not withheld this comment from his readers.

