Measurement lab

How AUDIO tests amplifiers

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An amplifier actually only has to output an amplified input voltage. The amplifier measurements in the AUDIO measurement lab show which errors can occur in the process. We provide an insight into the test lab.

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In the AUDIO measurement lab, every amplifier is put through its paces.

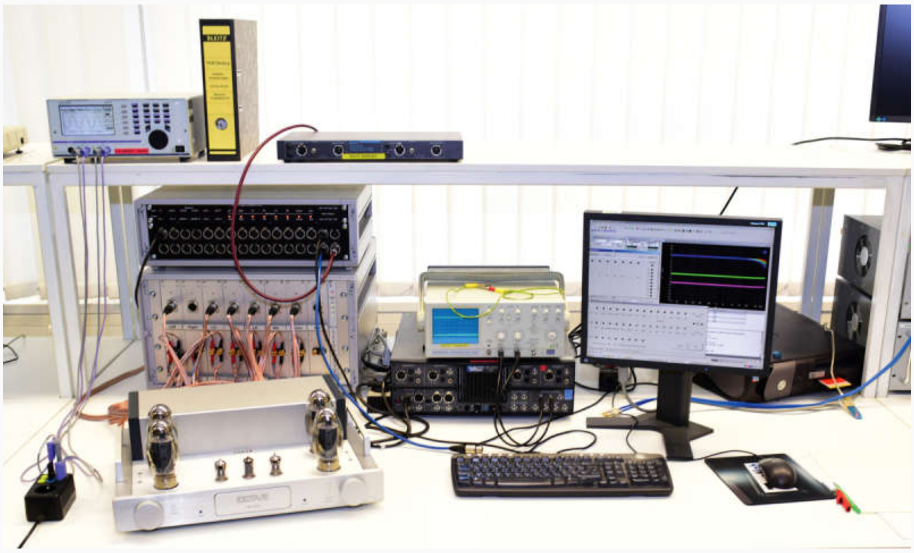
The requirement sounds simple: Take the signal at the input, amplify it twenty times, and output it at the output. An amplifier has an easy job. This is because other components, depending on their task, have to convert mechanical or optical information into electrical information, understand network protocols, or convert electrical energy into acoustic energy. All of this is more complex than simply amplifying an analog signal.

Nevertheless, amplifiers exert a magical fascination. Perhaps also because there are partly contradictory ways to build good amplifiers. Different approaches are reflected in the technical data, which makes the amplifier measurement technique of the test lab working for AUDIO particularly interesting.

Frequency response

When it comes to the amplitude-frequency response, as it should be correctly called, there is a minimum consensus. In the listening range (about 16 Hz to a maximum of 20 kHz) it should be perfectly linear, with deviations of 1dB being too small to be noticed. Many manufacturers exploit this by limiting the frequency response below 20 Hz and above 20 kHz. This can be considered the British school, as companies like Quad and Naim have made it a sound-enhancing practice. The drop at low frequencies keeps the cost of high-quality coupling capacitors in the signal path down, and prevents very low-frequency music or turntable rumble from imposing extreme excursions on woofers - because those also introduce distortion into the listening range. However, limiting low frequencies early on can limit the potential of very wide-range speakers and rob recordings in large halls of their spaciousness.

Limiting high frequencies avoids first of all the influence of ubiquitous high frequency radio fields on the amplifier and secondly the excitation of the amplifier to directly inaudible but extremely power consuming oscillation tendencies. Finally, limiting to high frequencies reduces the approximate short-circuiting effect of capacitive loads, such as those presented by some electrostats. The advocates of frequency responses that extend far beyond the audible range, whose pioneers once included the American specialist Spectral with bandwidths of less than 1 to more than 1 million Hertz (fone MHz), claim an increased resolving power, increased radiance in the highs and more natural breath in the lows. If the amp designer understands his craft, very high bandwidth amplifiers can be used without problems of high frequency interference, oscillation tendency, and capacitive loads. If he doesn't understand it, the test lab has also given a discrete hint at problems at times.



An Audio Precision System 2722 (black, center) records the measured values. The switchers for inputs (top left) and load resistors (below) allow quick measurements.

The frequency response of the phono input is quite interesting: Following the so-called RIAA characteristic, it has to boost the bass and cut the treble, since signals with exactly the opposite curve are cut into the record. This pre-distortion prevents extreme groove excursions at low frequencies. Amplifiers usually succeed in equalizing very well, as MC input measurements prove.

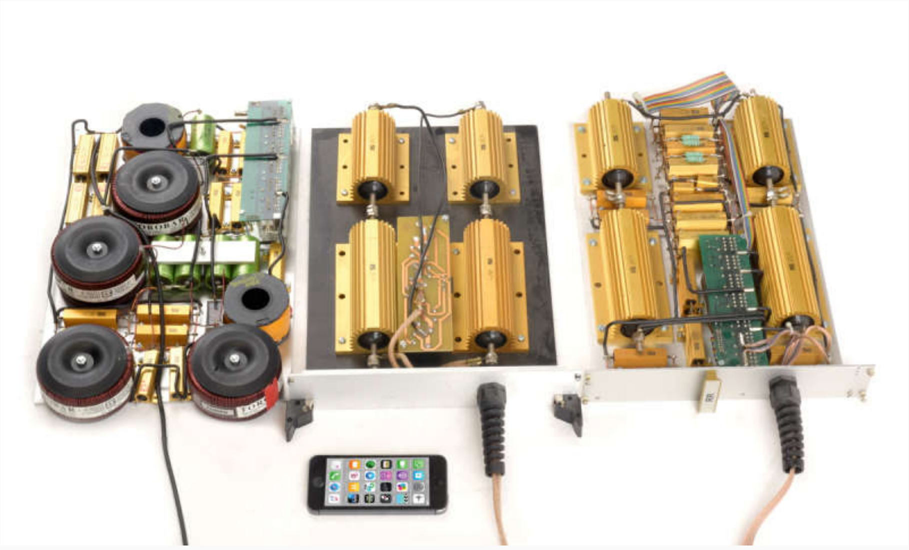
In MM systems, the input capacitance of the MM input forms a natural resonance with the inductance of the pickup, which causes an overshoot at high frequencies - with a subsequent steep drop. This is shown by our measurement with a simulated standard system. So, when combining MM system and phono stage, either the inductance of the system or the capacitance of the MM input should be as low as possible to push the resonance to very high frequencies. AUDIO publishes both values measured by the test lab.

Distortion factor

Non-linear distortions are considered to be the main sound-determining factor in amplifiers: They are a measure for the harmonic content when reproducing a pure sine tone. Since harmonics occur with double, triple or generally integer multiple fundamental frequency, they are called harmonics. Instruments also produce harmonics, some of which are even stronger than the fundamental; the distribution in terms of their levels is responsible for the timbre of the sound. If an amplifier changes this distribution, part of this ability to differentiate is lost. The Testlab measures distortion as a function of output power, with the second harmonic shown in red, and the following ones in green, blue and gray.

A moderately present second harmonic (also referred to as k2) subjectively increases the dynamics. If it is dominant, it can also mask the effect of higher harmonics, which are often perceived as negative in terms of sound, and make them inaudible. This is what makes tube amplifiers, and even more so single-ended tubes, with their high proportion of a second harmonic that rises continuously with the level, so appealing in terms of sound.

Amplifier measurement: output power



In order to be able to load amplifiers in all typical variants, switchable resistor networks such as these are available for up to eight channels.

The strong negative feedback in most transistor amplifiers reduces distortion to an often extremely low level, only the maximum power limit is then of greater sonic relevance due to almost abruptly increasing distortion (clipping). Since the best music recordings contain level peaks up to 20 dB, and the distortion factor during clipping increases from 0.1 to almost 100 percent within a few watts, it is a bad idea to operate a transistor amplifier above its limit.

For a different view of nonlinear distortion, consider the distortion spectrum measured at a fixed level, with the first harmonic filtered out at 1020 Hz. It would also reveal the noise floor and mains hum, which is at integer multiples of 50 Hz, as well as interactions between signals (intermodulation).

Output power

We know that it causes drastic distortions to operate an amplifier outside its power limits. But where do these limits actually occur? That depends on the loudspeaker, the impedance of which can be well below

2 ohms on earlier Infinity or Apogee models, for example, but also an order of magnitude or more above. The former places high demands on the current delivery capability, the latter on the maximum output voltage of the amplifier. The speaker impedance response shows that both can come in tandem. The minimum impedance is just over 3 ohms, the maximum just under 25 ohms.

In addition, there are phase shifts, the maximum current does not flow simultaneously with the maximum voltage applied to the speaker. This is especially demanding on the amplifier, as the measurements on a load constructed by TestLab in the so-called stability diagram show. In this measurement, the so-called peak musical power required only for a short time is measured at 8, 6, 4, 3 and 2 ohms, whereby the phase rotation between voltage and current is varied from -60 degrees to +60 degrees in 20-degree increments. The cube then shows the maximum voltage the amplifier can output without clipping. From this, the output power is easily calculated.

The mains supply is stabilized down to the volt for measurements up to 3.6 kilowatts (230 volts/16 amps) to prevent measurement errors.

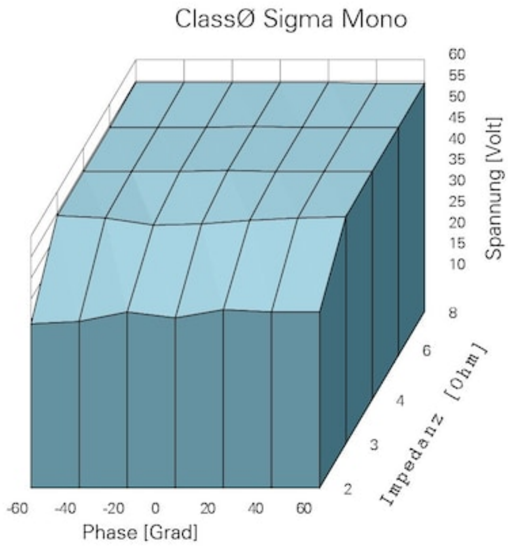
Amplifiers with high output impedance or limited current delivery capability show a more or less dropping cube towards the front and thus towards low load impedances. Consequently, such amplifiers should preferably be combined with loudspeakers of high impedance.

The combination of amplifiers whose cubes drop off to the right and left of the zero-degree phase line is also problematic. These amplifiers struggle to deliver their maximum power to loudspeakers with strong phase shifts in the impedance. This can be recognized by an unbalanced impedance response with large jumps within small frequency bands. As an indicator that can aide in the combination of amplifiers and loudspeakers, the test lab calculates the AUDIO-index (AUDIO-Kennzahl) from the 35 individual measurements of the amplifier output power. For loudspeakers, this is calculated from the efficiency, as the ability of the loudspeaker to convert input power into sound pressure, the impedance minimum and the maximum phase shifts of the impedance response. If the **AUDIO-index (AUDIO-Kennzahl)** of the amplifier is greater than that of the loudspeaker, the amplifier can drive the loudspeaker even at high levels.

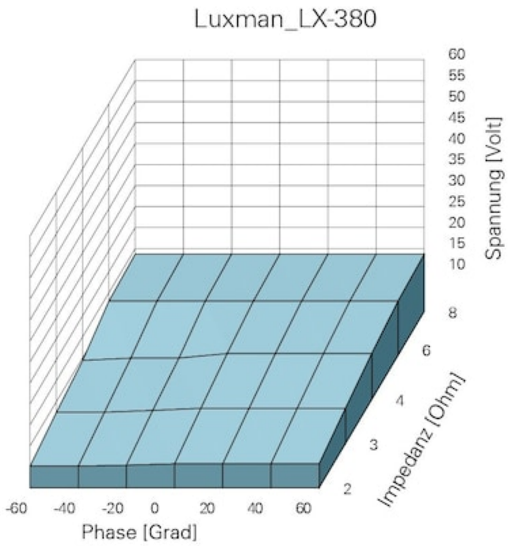
Conclusion

Whether one prefers the bandwidth-limited tuning of the frequency response or goes for broadband is just as much a matter of taste as the choice between a tube specific higher distortion with dominating second harmonic or minimal distortion. For magnet systems, however, the phono input should have a low input capacitance. More importantly, the amplifier should be able to handle the speaker. For highly fluctuating, low impedances, an AUDIO power cube with a flat surface is required; the power should match the efficiency so that the amplifier can really unfold its potential.

Sample stability diagrams



Stability diagram of a very powerful amplifier: At 8 ohms it delivers almost 55 volts, stable up to 3 ohms. Even at 2 ohms with 60 degrees phase rotation it still delivers 40 volts. Source: AUDIO



Stability diagram of a rather weak tube amplifier: No more than 15 volts are available. Source: AUDIO