Audio amplifier evaluation parameters

Premise

An amplifier is an electronic instrument that is responsible for increasing the amplitude of an input signal.

Beyond what is believed in the world of audio, an amplifier is nothing more than an electronic device, and for this reason it must be judged according to scientific methods and parameters.

This sheet was created with the intention of explaining why they behave differently despite some parameters being the same. The solution is very simple: manufacturers don't provide enough specifications. Consequently, here we will list all those necessary to understand how to select them.

It should be specified that an amplifier is first of all an ideal voltage generator. Keep in mind that an amplifier for audio use is not a simple electronic instrument that increases the amplitude of the signal, it requires much more specifications than those normally indicated, since it does not have to deal with capacitive loads, inductive and reactive ones.

Parameters

The parameters are divided into two categories: acoustic and non-acoustic. The acoustic parameters are those necessary to understand how the amplifier behaves with musical signals, the non-acoustic ones instead how the amplifier behaves in its implementation: for example, by connecting a specific power supply to it, or its efficiency, or its stability.

Acoustic parameters

1) Fundamental distortion

The fundamental distortion is measured by inserting a fixed frequency signal at the input of the amplifier (usually 1kHz or 20kHz), and its output is observed when the amplifier is connected to a load (by standard, in power amplifiers it is 8 Ohm, in headphone amplifiers it is 32 Ohm).

This type of distortion is calculated with a ratio between the voltage of the fundamental frequency and the voltage of the harmonics produced: the higher the voltage of the harmonics produced with respect to the fundamental signal, the greater the distortion. It can be expressed as a percentage or in decibels.

From the electronic point of view of an amplifier that must drive particular loads such as loudspeakers, this parameter is of little importance and of little significance. On the contrary, in specific electronic devices or DACs, this parameter is essential to determine the accuracy of the object.

2) Intermodulation distortion

Intermodulation distortion is measured in the same way as fundamental distortion: it is the result of the ratio between the introduced signal and the harmonics produced. However, in its measurement, two or more signals are introduced simultaneously at the same amplitude.

Some signal generators and spectrum analyzers introduce over 10-15 signals simultaneously. The more signals are introduced simultaneously, the more precise the measurement of the intermodulation distortion.

The intermodulation distortion has a high impact on the performance of the amplifier, and is a value indicative of its accuracy.

3) Memory (thermal) distortion

Thermal or memory distortion has nothing to do with the previous ones, and is the most influential type on sound quality in an amplifier.

To explain how it works, it must be understood that an electronic component varies its performance as the temperature varies. For example, a resistor varies its value, or, a transistor varies its gain or internal resistance.

This type of distortion indicates the quality of an amplifier to maintain its performance as the temperature of the components varies. The smaller the variation in performance, the less thermal distortion.

A component varies its temperature as the passage of the signal varies, moreover, the thermal error is multiplied by its gain. In fact, it is very important to correct the thermal error where the gain of the stage or component is very high.

This is the most important type of distortion, because it shows us how an amplifier behaves with a signal that is not of constant amplitude, such as the musical one.

To give a more practical example, components of a differential could be perfectly selected, but without an adequate current mirror, since the signal and the currents between the two inputs will never be the same, then the thermal distortion will be high (while it could be however very low the fundamental one).

Normally, thermal distortion is simulated through temperature sweeps between components. Otherwise, one of the methods to measure it in reality would be to introduce two or more non-regular amplitude signals to the input and verify their output over time.

This parameter strongly determines the enjoyment of the outgoing musical signal. This is the strong point of tube amplifiers, as the gain of the stages is usually very low and the temperature at the passage of the signal is almost constant. However, if a solid state amplifier is cared for enough, the thermal distortion can certainly be much less than that of tube amps.

4) Slew rate

The slew rate represents the reaction speed of an electronic circuit, the higher it is, the higher the quality of the amplifier. Its lack represents a type of distortion that is hardly measured through a Fourier transform.

Slew is measured in volts per microsecond, to measure it, a square wave is introduced to the amplifier input.

Although the slew, the higher the better, too high a slew could mean instability for the amplifier if it does not have adequate structure to support it or its compensation is not adequate.

The lack of slew represents both inaccuracy at high frequencies and a lack of output power at high frequencies.

The reason why the acoustic distortion of the slew is not detectable by a simple Fourier analysis is because sinusoids are used in the analysis. In a musical signal there are signals much faster than sinusoids, for this reason one can measure it with a square wave. The acoustic impact of this parameter is remarkable on enjoyability and precision.

5) Input impedance

The input impedance represents the load that the signal source sees. The higher the impedance, the less current is required from the source to drive the amplifier. Ideally it should be infinite, however the higher the latter, the higher the input noise and the more sensitive the amplifier is to external disturbances.

It should be high enough not to strain the source and low enough to ensure very low input noise. The ideal value should be between 10k Ohm and 20k Ohm for modern sources (equipped with output opamps).

6) Output impedance (damping factor)

Output impedance is one of the most important acoustic factors along with slew rate and memory distortion. Ideally, that of an amplifier, being an ideal voltage generator, should be zero.

The output impedance represents not only how much the signal is attenuated as the load impedance decreases, but also the control the amplifier has over the speaker excursion.

This parameter is in fact inversely proportional to the damping factor. The lower the output impedance, the higher the damping factor. The output impedance varies with the load impedance, the lower the load impedance, the higher the output impedance. Furthermore, it is not constant on all frequencies, it normally rises as the frequency rises.

To understand how the damping factor works, and how an amplifier has control over the speaker excursion, one has to see the speaker as a reversible machine.

Normally, a speaker generates a certain amount of sound pressure based on the current flowing through it. However, it is a reversible machine: that is, by generating pressure on it, it produces tension at its ends.

Explaining it with a simple electro-acoustic transducer, that is a loudspeaker composed of a coil, a magnet and a cone, it behaves in the same way as a magnet that is passed inside a solenoid.

The magnetic field of the magnet moves the electrons inside the inductor, generating voltage across it. By applying pressure to the cone of a loudspeaker, it moves the coil inside the magnet, and thus generates a voltage across the winding.

Assuming to send a pulse from the amplifier to the loudspeaker, the coil will begin to move according to the direction of the signal current. Once the impulse is over, however, the speaker will not stop instantly, rather instead, it will continue to spring on its suspension until the friction stops the moving crew.

The springing shouldn't exist, because it's not part of the original signal. The correction then takes place by the amplifier: the voltage produced by the speaker ends up in the feedback circuit of the amplifier, which compares it with the original signal, correcting it by sending an opposite signal to the speaker, stopping it immediately.

The damping factor is therefore a fundamental component of the acoustic quality of an amplifier, especially important at low frequencies, where the suspension of the speaker is more relevant. Obviously, the higher the damping factor the better, and it is a determining factor in the accuracy of the low frequency amplifier.

7) DC offset

The DC offset represents the direct current voltage output from the amplifier. Ideally, it should be zero.

Normally, a good power amplifier does not have an output offset greater than 10mV, a headphone amplifier normally less than 1mV. If the output offset is too high, it could damage the speakers, as well as damage the output signal and thus the sound quality.

8) Bandwidth

The band represents the frequency response of the amplifier. In it there are two fundamental parameters to consider: linearity and extension.

It is measured at the output, with a constant-amplitude sweep of frequencies on the input.

Linearity is represented by the variation of the output amplitude with respect to the input amplitude. Example: introducing an amplitude 1 signal at two different frequencies, the output signal of the two could be 1.1 and 0.9 respectively. An amplifier with better band linearity has a variation of 1.05 and 0.95 respectively, for example (i.e. a smaller tolerance). An amplifier with perfect band linearity, on the other hand, will always have output of 1.

The extension, on the other hand, represents the amount of frequencies at the same amplitude that the amplifier is able to reproduce. It is normally calculated at -3dB with respect to the central test frequency. For example, an amplifier could have a 20Hz band up to 100kHz, which means, I have a linearity of the band from 20Hz to 100kHz excluding extremes.

The wider the band, the better: it means less attenuation of the useful frequencies of the sound and is normally related to less distortion in the audible frequencies. For example, an amplifier with 100kHz of bandwidth normally has much less distortion than an amplifier with 50kHz of bandwidth over 20kHz. However, too broad a bandwidth, if not properly compensated and with inappropriate components, could result in an unstable amplifier.

The higher the band linearity, the better: they mean fewer variations from the original signal amplitude.

9) Phase

There are two phases in an amplifier: the first is strictly related to the band, the second is related to the voltage compared to the current.

The first phase, being linked to the band, varies as it varies. The greater the band extension, the smaller the phase change in the audible frequencies. This phase effectively represents a delay of certain frequencies in the resulting output signal. A high phase could cause several frequencies to overlap together, thus resulting in a substantial loss of signal quality.

The second determines the phase between the voltage and the output current, a phase shift represents a decrease in the power factor, or a lowering of the output power. The smaller the phase difference between voltage and current, the greater the output power factor of an amplifier, the less attenuation at the specific frequency.

10) SNR

SNR is the signal to noise ratio. It is understood as the electrical output noise of an amplifier as the signal passes. It should not be confused with the Noise floor, which instead represents the output signal of an amplifier when no signal is present at its input.

The lower the voltage noise produced, the greater the amplifier dynamics. The total dynamic perception, however, is the sum of different components, as well as of the noise: to it must be added the intermodulation distortion, the thermal one, maximum gain and slew.

Non-acoustic parameters

1) PSRR

The PSRR is also referred to as the Power Supply Rejection Ratio. It is a factor that determines the rejection of disturbances deriving from the power supply, as well as the rejection of noise, also that of ripple.

The greater the rejection, the less the variation of the amplifier performances will be with the variation of the power supply performances.

It is measured by placing the amplifier inputs to ground, and inserting a variable frequency voltage on one of the power supply branches. The greater the attenuation of the signal on the amplifier output, the greater the PSRR. This normally decreases as the frequency rises. Also, an amplifier with a lot of feedback will be less prone to power disturbances, as these will be corrected by the output from the feedback itself.

2) CMRR

The CMRR is the Common Mode Rejection Ratio, it is typical of amplifiers with differential inputs, it measures the tendency of the device to reject the input signals common to both inputs.

It therefore also identifies the amplifier's ability to reject external disturbances, as well as voltage variations and various fluctuations between the differential inputs.

3) Rated output power

The rated output power represents an AES (Audio Engineering Society) standard. It is a complex calculation based on:

- The power supply voltage
- The maximum output voltage

- The efficiency of the amplifier
- The maximum heat that can be dissipated by power devices and their SOA (Safe Operating Area) curves
- From the package surface of power devices
- From the size of the heatsink and the efficiency of the heat transmission
- From the impedance of the load
- Other variables

The AES standard states that the rated power, delivered or absorbed by an audio device, is the power that the device can deliver or absorb for at least two consecutive hours, without going into protection, overheating or breaking.

4) Rated output current

The rated output current is calculated on:
- From the output current curves of the power device, multiplied by the number of existing devices.
- From the supply voltage and the maximum output swing.
- From the temperature.
- From the delivery time and the output frequency.

5) Maximum output voltage

The maximum output voltage is the result of the maximum input voltage multiplied by the internal gain. The maximum output is calculated on the supply voltage considering the drop-voltage of the amplification stages.

6) Load-invariancy

It is the tendency of an amplifier not to vary its performance when the connected load varies, whether this is purely resistive, reactive or capacitive-reactive. Furthermore, it identifies the quality of not varying the performance as its impedance varies.

Ideally, an amplifier is load-invariant. However, specific classes vary greatly in performance, as well as in distortion, even in the bandwidth.

7) Compensation-invariability

Compensation allows an amplifier to remain stable in the high frequencies. If there were no compensation, the internal devices and stages would be prone to oscillations and free from any restrictions in the reproduction of high frequencies.

Each active device present in the electronic circuits has a maximum band of reproducible frequencies before their break-down. The lack of adequate compensation could lead these devices to deliver as much voltage, current or power as possible (according to their SOA), resulting in a melting or breakage of the devices themselves, especially if power devices.

An adequate compensation design requires the use of adequately positioned poles and specific components: such as the use of NP0 (C0G) capacitors, which do not vary their capacity as the voltage and temperature vary, and which also do not add further distortion.

8) Thermal stability

Thermal stability indicates the ability of an amplifier to remain thermally stable, and its propensity not to overheat with consequent thermal drift.

Note that the thermal drift can be both positive and negative. In the case of certain devices, such as lateral MOSFETs, the thermal drift is negative. In contrast, BJT transistors are positive thermal drift devices.

A stable amplifier keeps the temperature constant, and a well-designed one does not change its performance as the temperature changes.

It is a factor directly related to memory distortion, the smaller the temperature variation at the passage of the signal, the lower the memory distortion.

9) High-frequencies stability

High frequency stability is linked to three factors: bandwidth and respective phase, slew rate and compensation. A high-speed, wide-band electronic device is more prone to oscillations than a slower, small-band one. However, this is not always said, in fact, often an amplifier with even a reduced band and a low slew, with incorrect compensation, could be far more unstable than one with a much wider band and higher slew.

10) Capacitive-load stability

A truly stable amplifier, it is with any load. The most difficult load is certainly the capacitive one. The test is performed by normally placing a low capacitance capacitor (1nF) on the output, with no other added.

The test aims to obtain the output response of a square wave, any ringing and oscillations.

This allows us to understand how the electronic device behaves in the absence of a load, or with very distant loads (where the capacitance of the conductor that connects it is very high).

11) Bandwidth and phase stability

Bands and phase, like the other performances, can vary according to the temperature, the internal variables of the device or, as often happens, to the variation of the load.

Often, as happens in switching amplifiers (class D, T, S, etc), the band varies a lot as the impedance of the load varies. This is because the output low pass filter is calculated on the generic impedance of a specific load.

It is a variable linked to the load-invariancy of the amplifier. Band stability is also linked to high frequency stability. With non-standard loads some amplifiers may have oscillations due to variation and the creation of phase peaks and rotations in the band.

12) Open loop gain

Open loop gain is typical of amplifiers with differential inputs. It is the gain obtained by placing one of the two inputs towards ground. It determines two things: the amount of correction that is achieved at a specific frequency, and the maximum closed-loop gain available to the amplifier.

Furthermore, it is possible to determine the general high-frequency stability of an amplifier from it, by observing the phase with respect to the gain of the frequency response.

13) Closed loop gain

In amplifiers without differential inputs, this is simply referred to as "gain". Determines the multiplication of voltage that the input signal receives with respect to the output signal.

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If the amplifier gain is 26dB, for example, the voltage multiplication will be 20 times the input signal.

In differential amplifiers, the gain is decided by a voltage divider placed between the amplifier output and the second differential input. The closed loop gain is necessarily always less than the open loop gain.

The higher the gain, the smaller the bandwidth, the lower the gain, the more likely it is that the amplifier will become unstable and the band will have a high frequency peak.

14) Efficiency

Efficiency is the ratio between the absorbed power and the delivered one. Given by the following variables:

- Supply voltage versus output voltage
- Operation class
- Quiescent current
- Power supply voltage of the driver stage in comparison to the power one

15) Operating class

The operation class of an amplifier first determines the topology, then how it handles the signal.

The amplification modes are divided into linear and switching amplification. In the first, the signal is amplified linearly. In the second, the signal is added to a high frequency triangular signal through a comparator, as a result a variable duty cycle square wave is obtained, this square wave is amplified and finally filtered to obtain a musical signal again. It should be specified that in both cases the signal management is purely analog.

Contrary to popular belief, the class does not determine the efficiency of the amplifier, rather the efficiency is a consequence of the same. The class determines, in addition to the amplification mode, also the conduction angle of the signal.

For example, class A amplifies, in the case of a sinusoid, a 360° signal, that is 100% of the portion of the signal.

Class B amplifies the exact half of the signal, 50%, in the case of a sinusoid it would be equivalent to an angle of 180°.

Class AB, on the other hand, amplifies a portion greater than half but less than the complete signal, therefore between 50% and 100%: $180° < X < 360°$.

Class D, for example, amplifies 100% of the signal like class A. Unlike class A, however, class D is switching, while class A is linear.

To specify, that the best amplification class does not exist, and that each class has its strengths and weaknesses.

The operation class can be applied: to the single device (transistor, mosfet, etc), to the single stage, or to the entire amplifier. For example, a "pure class A" amplifier means that all its devices and stages work with that specific class (or so it should be, it is possible that producers say this even without this being true).