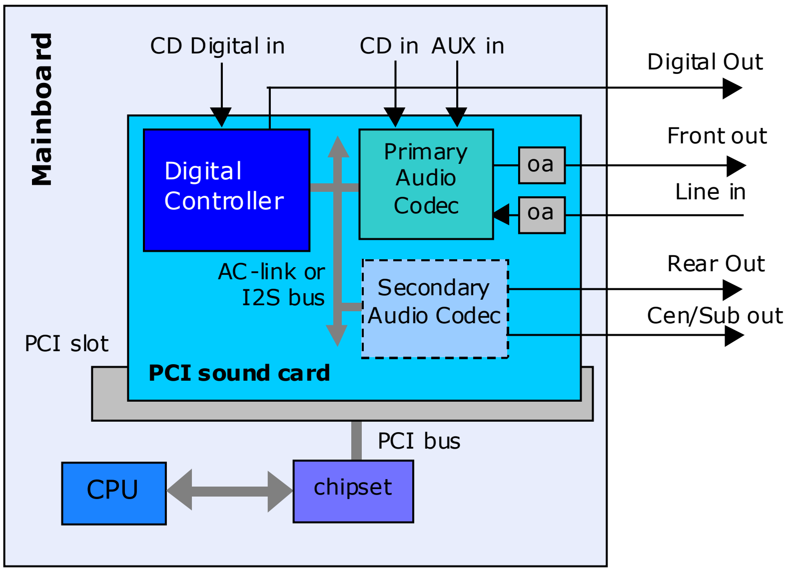
RightMark Audio Analyzer soundtest explanation

Maxim Liadov (maxim@ixbt.com)

Sir RMAA project manager

About sound quality

Look at the diagram. Sound goes a long way through different devices. If in one of them sound becomes worse the resultant sound will be also bad.



So, soundcard's quality generally depends on 4 things:

1. quality of drivers (data format transfers and conversions)
2. quality of Digital Controller (processing algorithms)
3. quality of Audio Codec (which has analog-to-digital and digital-to-analog converters)
4. quality of operational amplifiers and PCB layout

In 99% cases specs of sound cards show only ideal Audio Codec’s parameters (measured by codec manufacturers in certain laboratory conditions).

But in real life bottlenecks are possible in some places. That is way we need to test a real soundcard’s performance in the real conditions.

How we can test sound quality

It is known that the right sound contains minimum distortions and add-ons. There are some standard technical parameters of quality for estimating quality of any sound system. A special test signal passes through a test chain and then is compared with the original one. So, by comparison of those two signals we can find out the difference between them.

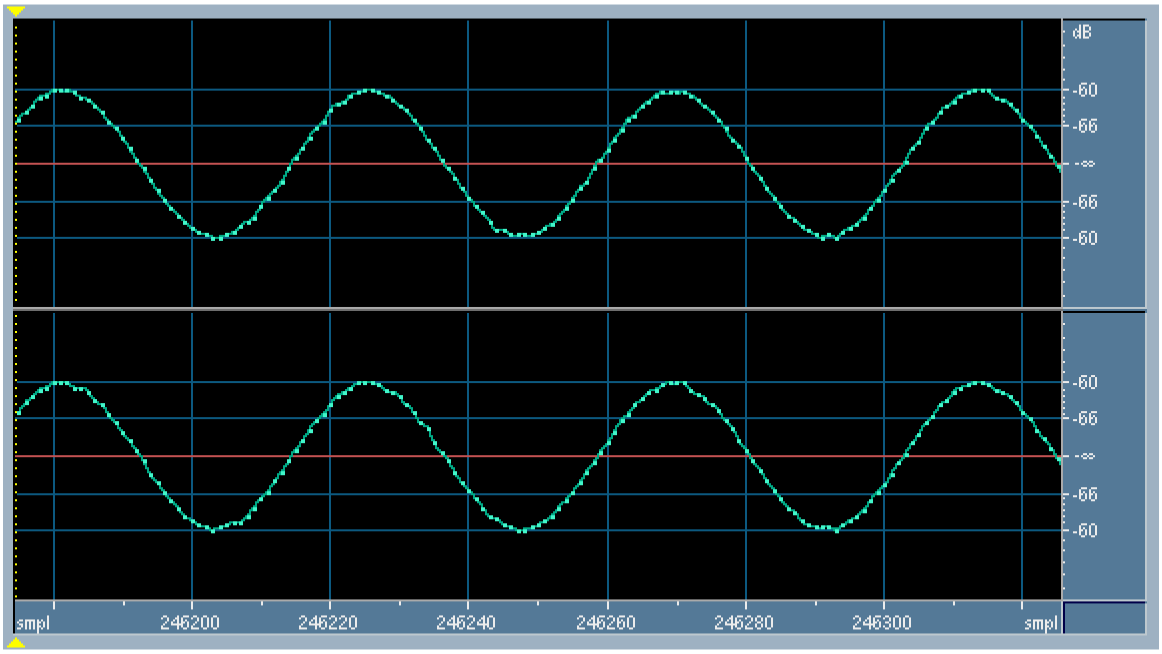
The simplest way is to connect an output of a soundcard to its input. Then we can playback the test signal and record it by the same soundcard.

The advantages of this method are simplicity and repeatability (by everyone). The disadvantages are impossibility to reach absolute results coinciding with the specification. But a good soundcard always has good test results. On the other hand, there are bugly devices with awful sound and excellent specifications produced by stupid marketing departments. A more advanced technique involves reference professional soundcards or even special test equipment for $20000.

As to software, there is a well-known freeware open-source test suite named **RigthMark Audio Analyzer,** or shorlty **RMAA** (the official supporting site is at http://audio.rightmark.org). This world audiotest standard is currently used by Anand Tech, Tom’s Hardware, Clubic.com, Digit-Life, GZeasy, etc.

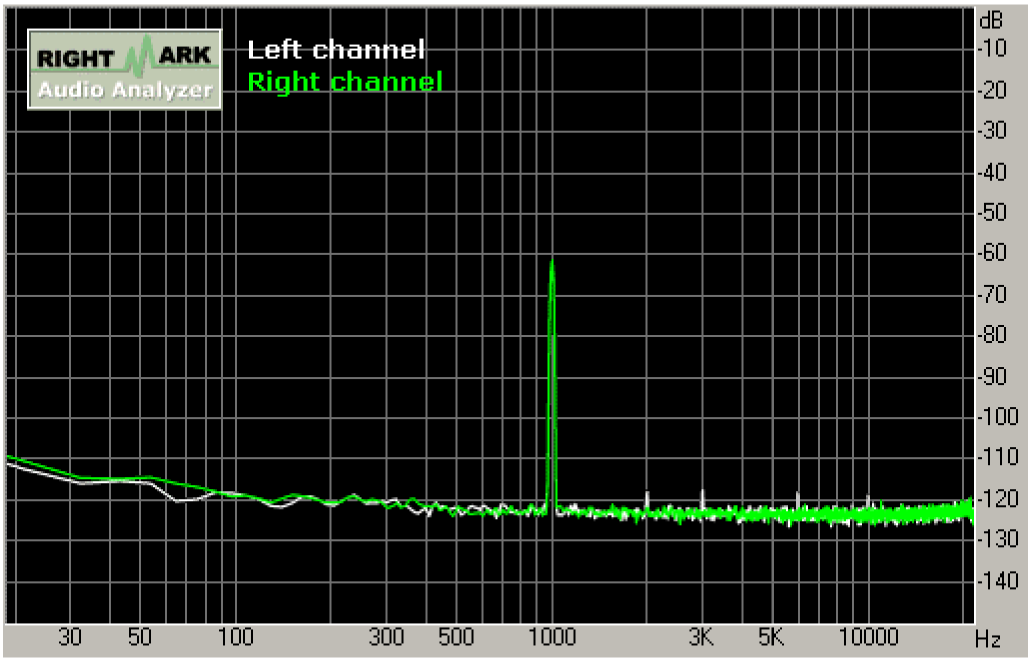
Sound spectral analysis in detail

Let's take a look at a traditional plot of signal in a time domain. The recorded test signal doesn't seem ideal, but it's hard to say what's exactly wrong with it.



Traditional signal in time domain:
sound level in dB vs. time in discrete samples

Now look at the same signal in a spectrogram (in a frequency domain). Everything is clear and looks okey. There is just some low-frequency even noise and lack of any big parasitic harmonics.



Spectrogram in frequency domain:
the same sound level, dB vs. frequency, Hz

So, there is a special mathematical method named Fast Fourier Transform (or FFT) for analysis of recorded signals. This method allows us to get an amplitude-versus-frequency characteristic for any sound signal.

Here is the formula of the direct Fourier transform:

$g(\omega) = \frac{1}{2\pi}\int_{-\infty}^{+\infty} f(t)\cdot e^{-i\omega t}dt$, where *f(t)* is a continuous function in time domain

The computer has a discrete signal representation. So, the following formula is used for the discrete Fourier transform:

$G(\omega) = \sum_{i=1}^{N} F(i)\cdot e^{\frac{2\pi}{N}(i-1)(\omega-1)}$, where *F(i)* is discrete values of the continuous function $f(i)$

in point i, N - number of points, w - circular frequency. With this formula and some other additional methods we can draw an amplitude-vs-frequency spectrogram in a logarithmic scale for both axes.

What is dB (decibel) and why logarithm is used in spectrograms?

An apparent sound volume is estimated via its level:

$$L = 10\,log\frac{I}{I_0}\text{, [dB]} \qquad \text{where } I = 0{,}5\frac{dp^2}{\rho a} \text{ is acoustic intensity.}$$

According to the psychophysical law of Weber-Fechner, for a human this value is in direct proportion to subjective perception of volume changes. But in most cases a volume level is measured by the sound pressure:

$$L = 10\,log\frac{p^2}{p_0^{\,2}} = 20\,log\frac{p}{p_0}$$

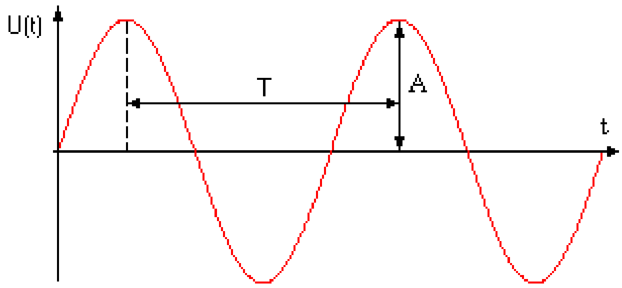
That is why a signal's amplitude is plotted in a logarithmic scale. The same concerns frequency. The human sense of a pitch of tone is also logarithmical by nature.

So, a test spectrogram is not a kind of science fiction, but the real illustration of our aural perception.

What parameters are needed for testing

Flatness of amplitude-versus-frequency characteristic

Sound is invisible waves which spread out in the air mostly because of oscillations. Any simple oscillation (sine wave) has its amplitude (A) and its frequency (1/T). The higher the number of oscillations, the higher the sound. The number of oscillations per second is called frequency and measured in *Hertz* [Hz].

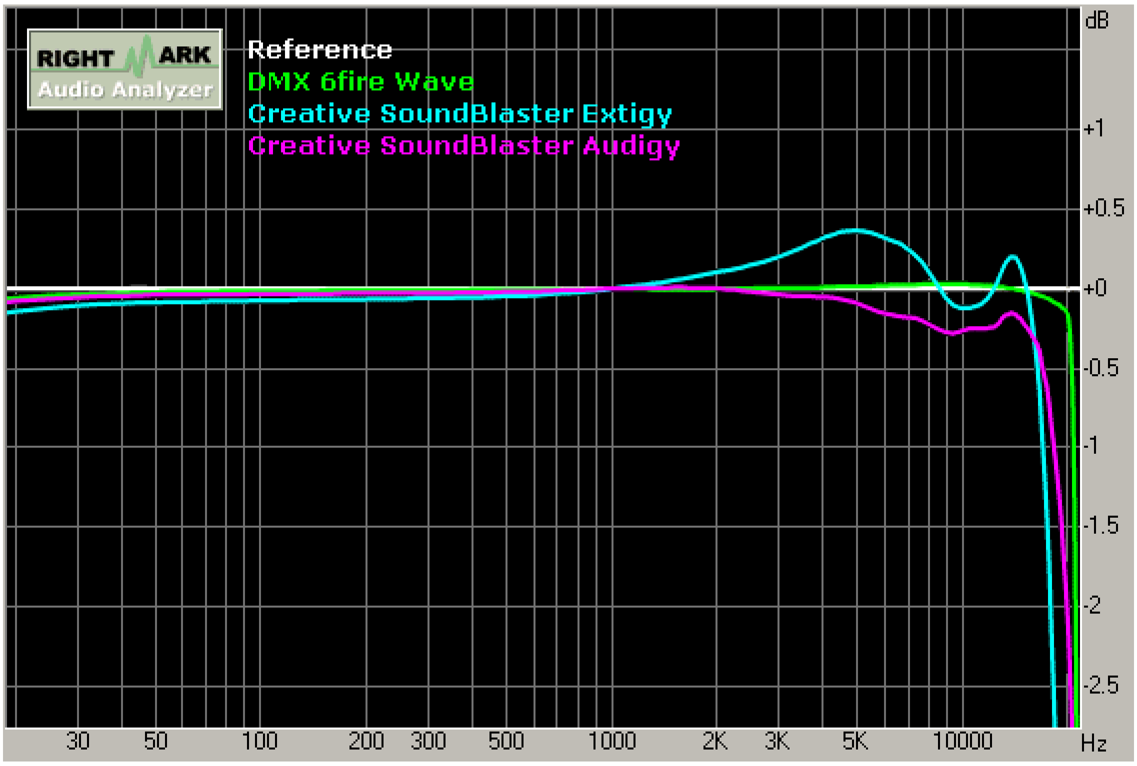


Sound heard by human is, in general, a set of waves in a finite audible range. It is important that the sound in any part of the audible range has the same amplitude as an original one.

So, amplitude-frequency characteristic is variation in a signal level as the frequency varies. The plot is drawn by measuring the output amplitude when a harmonic signal (usually it is a swept sine) with a constant amplitude is applied. Ideally, it should be a straight line. There is no an absolute correspondence to the reference signal that matters, but a deviation from the most even part. It allows judging how accurately a signal amplitude is delivered at different frequencies. If necessary, it's corrected with a multiband equalizer. Unfortunately, equalizers usually bring in some unpleasant distortions into sound.

The full audible range is, in general, 20 Hz – 20 kHz, but as the human hearing is more sensitive in the middle range, for a usual user it is more interesting how a soundcard plays in the 40 Hz–15 kHz range. But musicians and uncompromising users need the full-frequency range, i.e. 20 Hz – 20 kHz.

An excellent amplitude-versus-frequency characteristic it is not a big problem for modern transistor circuits , but in some cases (or some modes) soundcard characteristics can be imperfect which bring in inflatness.



Spectrogram of frequency response (FR) obtained in 16 bit 44.1 kHz mode

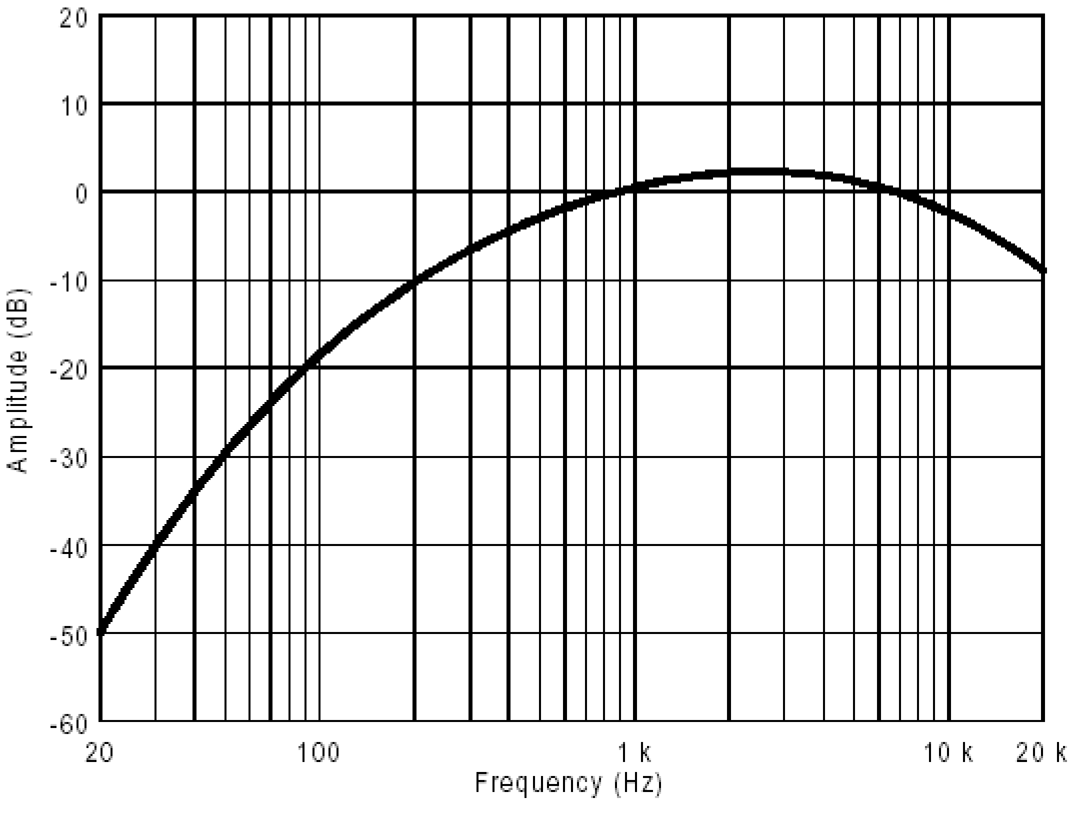
The human sound sensitivity is, in fact, limited to 1 dB. So, inflatness in the range of +/-0.5 dB is good enough for all purposes, except professional usage where full accuracy and precision are required.

Signal-to-Noise Ratio (SNR)

SNR is noise power without a signal but calibrated to the full-scale signal level (FS). However, human hearing is unequally sensitive to sound of different frequencies. For example, the maximum of our sensitivity to quiet sounds lies around 3 kHz. Sounds of these frequencies are perceived as louder ones. That is why we need to modify the technique of spectral measurements to make them closer to our aural perception. Such modifications are known as A-weighting. They are widely used in audio measurements (for example, for noise level or dynamic range). As a result, we get inaudible frequencies attenuated, and the most audible ones contribute more to the final results.

The SNR test estimates a noise level in silence in the test chain. A diagram of the noise spectrum helps to locate frequencies of interference caused by AC circuits, CRT displays and other electronic devices.

SNR is usually measured in dB FS A, which means sound level relative to the full-scale signal and is weighted with a special aural perception A-curve. Professional usage requires SNR more than –96 dB A, while just for music listening –90 dB is enough.



A-weighting curve

Dynamic Range (DR)

Without a signal some smart circuits can cut off any output signal to make the SNR excellent. To prevent this trick we can apply a test signal of a low amplitude. Such test was named Dynamic Range test. It works with a standard level of –60 dB FS. The test also lets us know how many distortions can take place in case of low-level signals.

In other words, the DR test estimates a noise level with a weak signal applied, and linearity of a sound device operating at low signal levels (which is very important for high-quality sound recording and playback).

For digital devices SNR is not so good parameter as the DR. Contrary to analog circuits, the main problem of digital ones is not noise, but distortions.

Total Harmonic Distortion (THD)

THD defines a level of unwanted harmonics generated in a sound device. Usually high-quality devices have a low THD value (lower than 0.002%), but there are exceptions. Many tube devices have a quite high THD level, which makes their sound "warm". But transistor devices must have a low THD value because their (odd) harmonics don’t make sound pleasant.

A THD test estimates amount of harmonic distortions that occur when a signal with a large (close to maximal) amplitude passes through a test chain. Musical ability and transparency of sound depend on amount of distortions and their spectrum. What is the most important on a spectrogram is the ratio of even and odd harmonics and availability of harmonics at unwanted frequencies.

Bottlenecks for THD of soundcards are creatred by operational amplifiers. That is why cards with the same DACs (and the same great specs) often have very different THD.

Intermodulation Distortions (IMD)

This test estimates amount of intermodultion distortions that occur when a complex signal passes through a test chain. The test signal consists of 2 harmonics of different frequencies. After passing through the test- chain, the resulting signal contains different harmonics and, possibly, a large number of harmonics resulting from D/A and A/D conversion oversampling filters and sample rate conversion inaccuracy.

IMD is the most difficult test for multimedia cards with an old version of AC'97 codec which has a fixed sample rate at 48 kHz. Not very good IMD test results are not crucial for game soundcards but very important for high-quality and professional ones.

Stereo channels crosstalk test

This test estimates leakage of a signal from one channel to another for various frequencies. If a sound device has poor crosstalk results, you can't get a good stereo image from your sound recording.