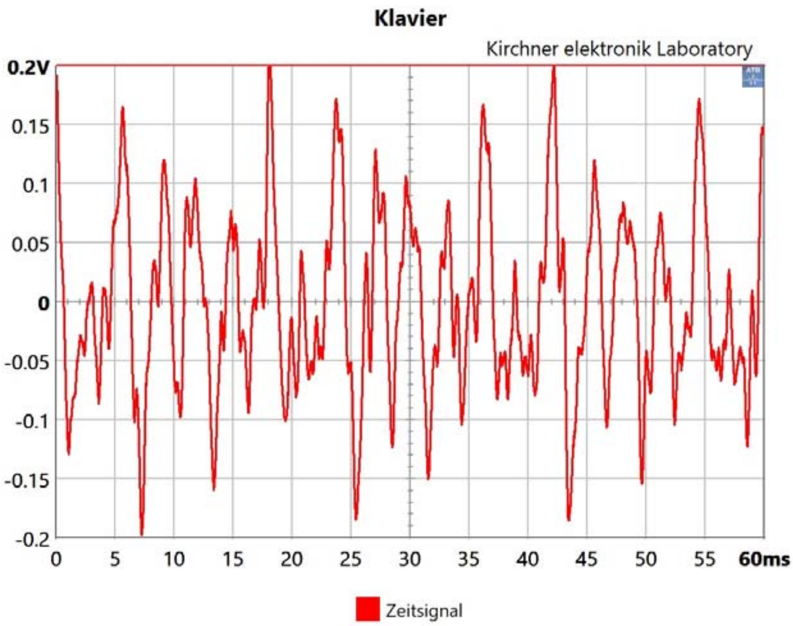
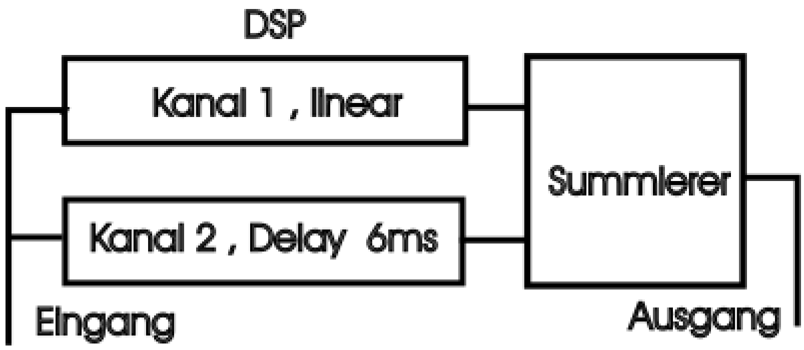
The development of the time-correct loudspeaker

An acoustic signal, music, speech and noises become audible through the vibrations of the air. The sinusoidal oscillation is described by the frequency, amplitude and phase. Oscillations are measured with an oscilloscope. The oscilloscope shows the oscillation over time. The frequency is determined by the time between two equal zero crossings, the period. F = 1/s, s = sec . The amplitude is the distance between the zero line and the maximum height of the oscillation. The phase is determined by the distance between the same zero crossings of two oscillations. One oscillation is the reference oscillation. The phase is specified in °. A period is defined as 360°. The distance is normalized according to the period of the measured oscillation.
A loudspeaker is intended to make the recording of music, speech and noises audible. These signals consist of several tones. The sound is an oscillation with a frequency. The oscilloscope is used to show the oscillations over time. As an example, the mixture of tones of a piano should be considered.

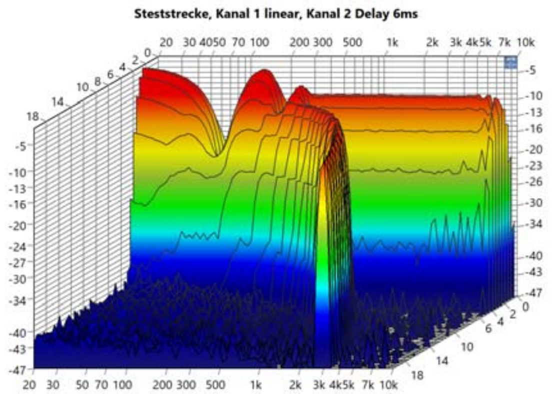


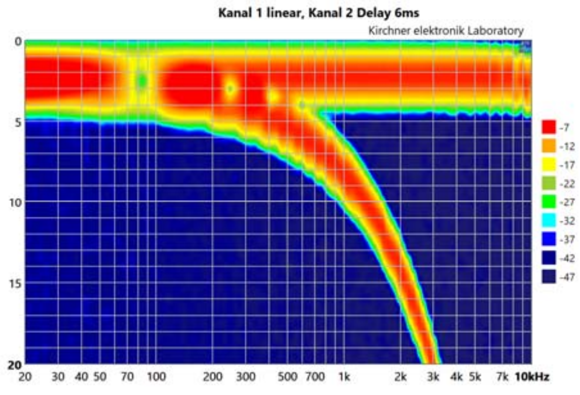
The fundamental tone is produced when a side is struck. However, the side not only vibrates at the frequency of the fundamental tone, a number of overtones are also produced. The vibration of the side also causes other sides to vibrate and produce tones. The piano is tuned to produce a harmonious sound. In the mixture of vibrations with different frequencies, all vibrations have a fixed temporal relationship. This relationship is described as phase. The sound of a piano is determined by the amplitudes of the individual vibrations and their temporal relationship, the phase. Rule of thumb for the superimposition of vibrations: At an angle of 180° the vibrations cancel each other out and at 0° and 360° the amplitude doubles. Just as the amplitude and the temporal relationship, the phase, determine the sound of the piano, this also applies to the loudspeaker. Both parameters must be correct during development.
Now let's take a brief look at the history of loudspeaker measurement technology.
Until 50 years ago, loudspeakers were developed using physically correct measurement technology. The measurements required expensive equipment and were not easy to operate. With the MLSSA measurement system, which worked with a PC, a fairly simple system came onto the market. Thanks to the MLS generator, which produced a digital signal, the circuit board could be manufactured cheaply. Thanks to the American company's good marketing and the claim that the digital signal corresponded exactly to the music, the device sold well. The fact that the MLS signal with very high energy in the high frequency range and low

energy in the low frequency range is exactly the opposite of the music was only recognized late on. The weak signal for the low frequency range causes significant measurement errors. Two measurements are therefore necessary, one for the low frequency range and one for the mid-high frequency range from 200Hz. The basis for the measurement of the MLSSA system is the impulse response. The evaluations show the frequency response and the decay spectrum, the waterfall.
New measurement programs avoid the error in the low frequency range and measure with the sine sweep. These are also unsuitable for loudspeaker development as they do not show any time response.
Most measurement programs for loudspeakers work with the impulse response. Their data is the basis for calculating various evaluations such as step response, frequency response, waterfall, reverberation time and other displays. This type of measurement is to be tested. Electrical measurements are carried out to ensure that the tests are not falsified by the acoustics.

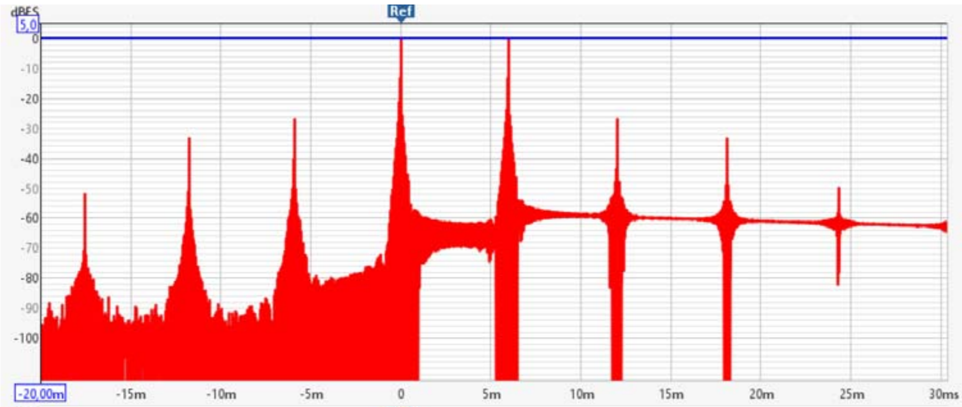


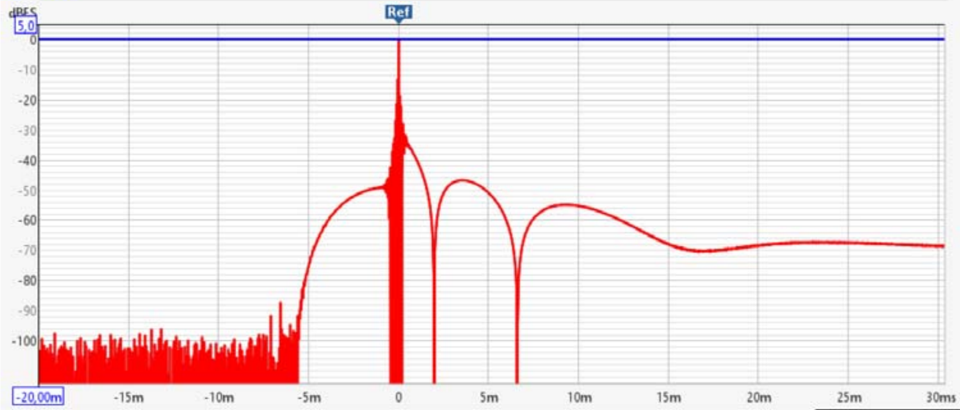
The structure of the test track consists of a DSP. The DSP is used to program the filters in the loudspeakers and the delays caused by the loudspeaker setup. As the drivers do not have the same acoustic plane, there are differences in the propagation time of the sound for the listener. The DSP uses 2 channels, corresponding to 2-way loudspeakers or a subwoofer system. The outputs are combined via an adding circuit. The output of the circuit has the signals of a microphone.
The ATB waterfall diagram shows the function of the test circuit without the filters.





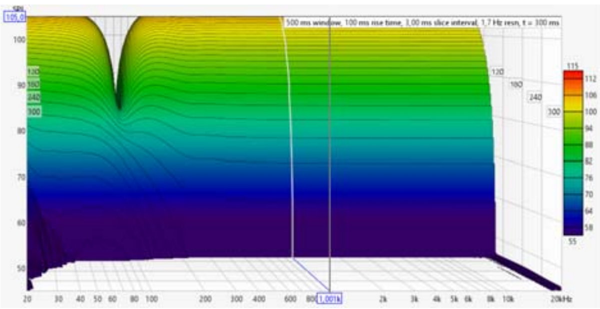
The waterfall diagram shows the linear channel 1 at the top. The delayed signal from channel 2 runs to the front. The spectrogram also shows the superposition of the signals from the two channels.
The test track is measured with a modern measurement program that calculates the measurements from the impulse response.

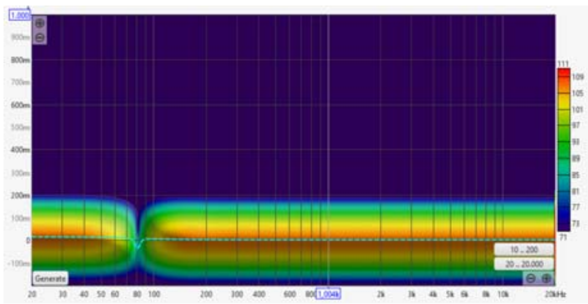




The first image of the impulse response shows the test section described above. The two channels can be seen through the high lines at a distance of 6 ms. In the second measurement, filters were programmed into the DSP.

Channel 1 is programmed with a 24dB Linkwitz-Riley low-pass filter, 80Hz. Channel 2 is programmed with a 24dB Linkwitz-Riley filter, 80Hz. Channel 1 retains the delay of 6ms. The delay of channel 1 can no longer be seen in the measurement. This was also tested with new programs, with the same result. This shows that the pulse measurement has no significance with regard to time, even because one channel is no longer measured. The following evaluations show that the measurement also makes no statement about the time.



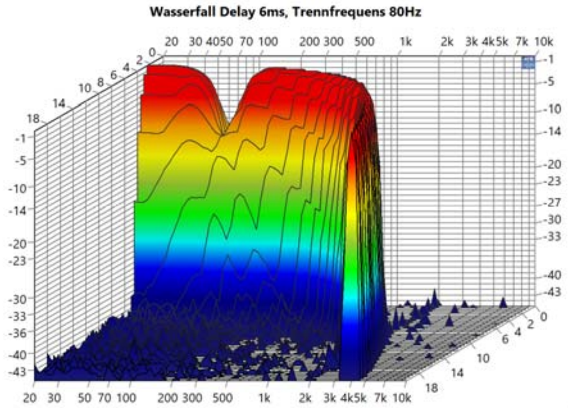


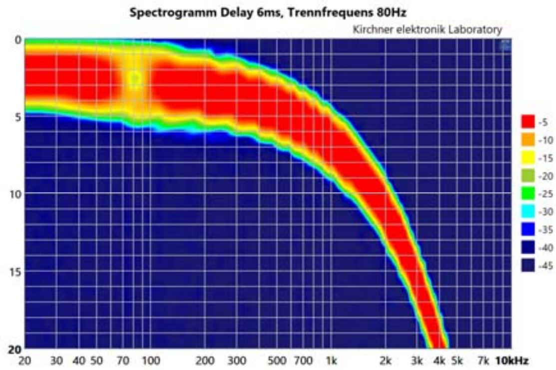
The waterfall and the spectrogram have no time information. The delay of channel 2 is not shown.

The waterfall shows the frequency response with the first line at time > 0. Other measurements can only show resonances. The temporal behavior, the phase, is not shown.

The phase shows the phase rotation of the filters, the different travel times of the sound from the individual drivers to the listener and the transient behavior of filters and loudspeakers. The frequency adjustment only covers part of the characteristics of the loudspeaker that determine the sound. The temporal relationship between the individual oscillations, which also determines the sound, is not shown.

The ATB measuring program does not use the impulse response. The graduate engineer in measurement technology, Leo Kirchner, developed the ATB measuring system together with the programmer Elmar Meyer-Carlstädt, a gifted mathematician. Even after 50 years, it still sets the standard for many companies. The program can be used to measure time correctly and to develop loudspeakers that can reproduce the true sound of music.
The ATB measurements show the correct behavior of the test track.





This information is used to develop a loudspeaker that meets the requirements for real sound reproduction. Since the temporal behavior can be measured.

Leo Kirchner