Dipl.-Ing. Leo Kirchner

Loudspeaker measurement technology

I thank my team
for their help with the measurements:

Ralf Freymuth
Jürgen Knopp
Michael Borowski
Sandor Finn

Translation from German by Eric Brewer

1. THE FREQUENCY RESPONSE

The aim of HIFI technology is the truest reproduction of a recording or transmission possible. So that the evaluation of sound quality is not just based on subjective assessment, a Measurement System is used to make an objective assessment.

In the measurement technology, the properties of a transmission chain are illustrated. In the loudspeaker technology, the transmission chain is a loudspeaker and the sound pressure level is measured. From the sound pressure level, the frequency response is derived and Illustrated as amplitude and phase frequency curves. The frequency curve describes the frequency response completely. The function F shows the relationship of an input value to output value. The most simple input- output equation is when the input and output has equal properties and the function is a frequency independent constant. This is given by an amplifier for the hearing range. The frequency response is then the amplification V.

$$F = \frac{U_A}{U_E} = V$$

For loudspeaker, the input value is the voltage U and the output value is the sound pressure level **P.**

$$F = \frac{P}{U}$$

The loudspeaker is driven by an alternating current to produce sound. That's why the frequency response has to describe the alternating current. The alternating current U(t) consists of a sinus wave with the amplitude U and phase α**.**

$$U_{(t)} = \sqrt{2}U\cos(\omega t + \alpha)$$

Alternating current calculation demands complex calculation using the complex number Z, consisting of the real part a and imaginative part *b.*

$$Z = a + jb$$

The result is the value of [Z], by loudspeakers the frequency range,

$$|Z| = \sqrt{a^2 + b^2}$$

and the phase φ , by loudspeakers the acoustical phase

$$\varphi = \arctan\frac{b}{a}$$

If the frequency response for the stationary state or swung-in state and at the same time the state of movement, the non swung-in state is to be described, calculation is done with the Laplace-Transformation.
As an example for the frequency response according to Laplace- Transformation, the frequency response for loudspeakers is shown. The method comes from the controlling technology.

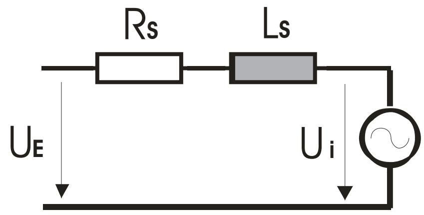


Fig. 1.1

$$\frac{P}{U} = F_{(p)} = \frac{v_1 T_1 p}{(T_1 p + 1)\left(\left(\frac{p}{\omega_0}\right)^2 + 2D_0 \frac{p}{\omega_0} + 1\right) + T_2 p} * \frac{v_S (T_S p)^2}{(T_S p + 1)^2}$$

$$T_1 = \frac{L_S}{R_S} \quad T_2 = \frac{(Bl)^2}{R_S C} \quad \omega_0 = \sqrt{\frac{C}{m}} \quad D_0 = \frac{b}{2\sqrt{Cm}} \quad v_1 = \frac{1}{Bl} \quad T_S = \frac{r}{c_0} \quad v_S = \rho c_0$$

R_S = Voice coil resistance, L_S = Voice coil inductivity

Bl = Force factor, m = dynamic moved Mass,

C = Compliance, b = Damping

c_0 = Speed of sound, ρ = Air density,

$r = \frac{a}{\sqrt{2}}$ a = Loudspeaker membrane radius

Fig. 1.2

Fig. 1.1 shows the substitute circuit of a loudspeaker.
The equation in Fig. 1.2 describes the frequency response of a loudspeaker without partial swinging of the membrane. With this formula, the behaviour of a loudspeaker over a large frequency range can be calculated.

The formula shows the behaviour of a single loudspeaker. By a loudspeaker combination with crossover, the calculation becomes very complex. For loudspeaker- boxes, measurement of the frequency response behaviour is the easier method.
In addition, the calculation of a crossover is not possible without measurements. First, when the CAAD program calculates with the frequency response of the loudspeaker is the result of any meaning. The frequency response is read into the program over the values for the amplitude, phase frequency ranges of the sound pressure level and also the amplitude, and phase frequency ranges of the impedance.

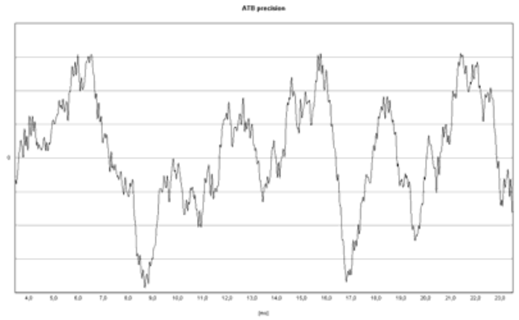
To measure the frequency response of a loudspeaker it is driven with an alternating signal.

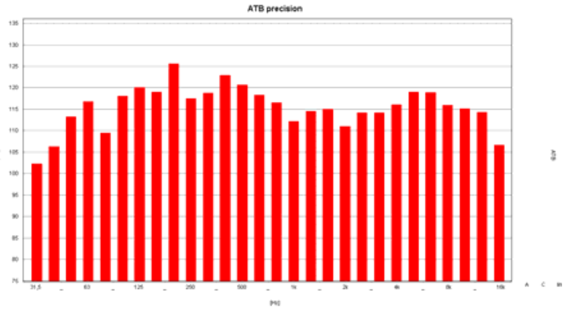
Measurement result = Frequency response x Signal

The equation shows that the signal is part of the measurement result. The signal has to be well known according to its properties, so that it can be calculated out of the result after measurement. This is achieved by correlation. A simpler method is to use a signal that has a constant frequency and phase range. These conditions are given using a sinus signal. As the amplitude of the sinus signal stays the same for the whole measurement range, the formula for the correlation is a constant. That why a correlation is not needed using a sinus signal for the measurement. For loudspeaker measurement there are other considerations when choosing a measurement signal. From the thought of it, a loudspeaker should be measured with the kind of signal it is meant to transmit. These signals are music, speech and noises. If the loudspeaker is driven with that kind of signal, the measurement results will relate to the hearing experience to be expected.

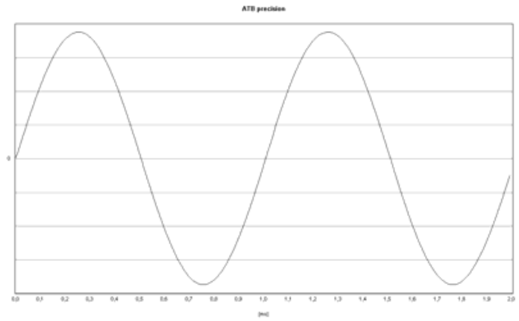
2. TNE MEASUREMENT SIGNALS

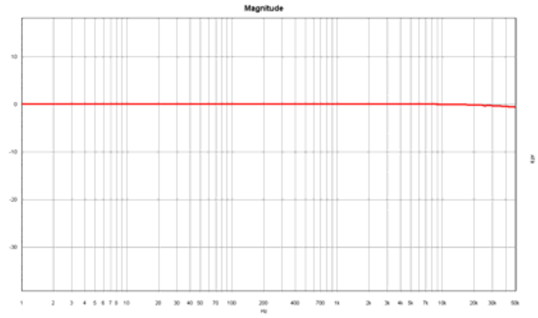
MUSIC



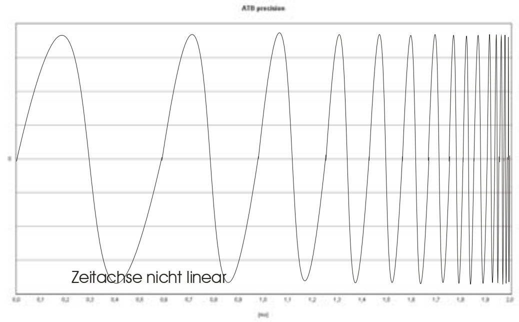


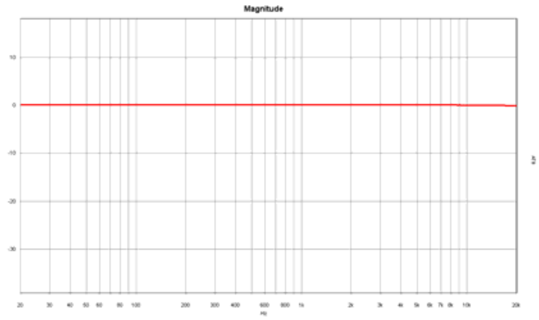
SINUS-STEP Plotter, LMS, ATB precision



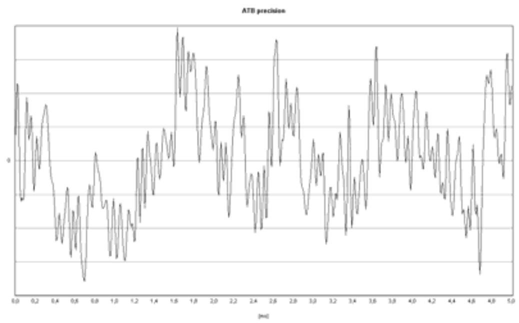


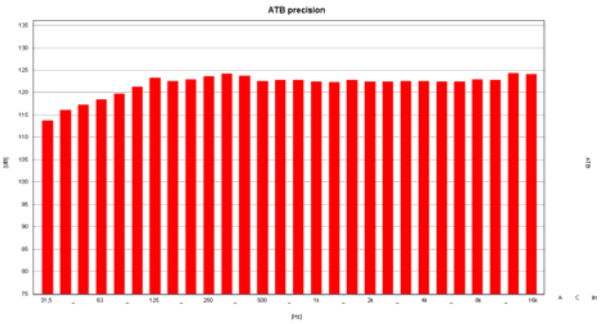
SINUS-SWEEP TEF, Room-Tools



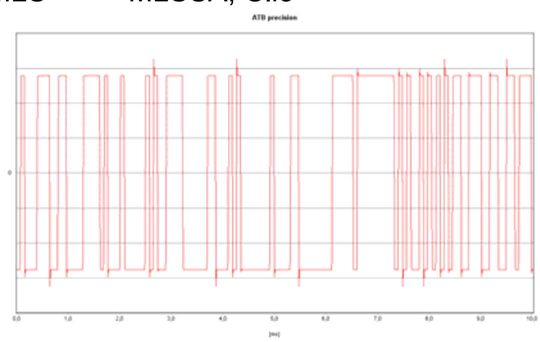


MULTISINUS ATB





MLS MLSSA, Clio



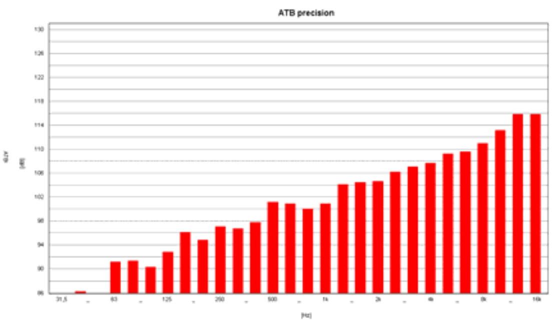


Fig. 2.1

The measurements in Fig. 2.1 show the signal in time and frequency range.
The time range is illustrated by an oscilloscope measurement.
The frequency range is illustrated in the right hand diagram. By the signals Music, Multisinus and MLS a 1/3 octave analyser shows the measurement of the frequency range and by the sinus signal a frequency response measurement the amplitude sweep.

3. THE MEASUREMENTS

The different measurement methods will be called after the measurement signal used.

3.1 SINUS-STEP

History:
The sinus signal is a proved method for measuring the frequency and phase ranges. By the sinus-step measurement, the frequency is raised step for step within a frequency range. Here the number of frequencies and with that the number of measurements and also the frequency range is predetermined. The measurement is very accurate. Earlier these measurements were carried out using a plotter. Neutrik produced the most well known one. New systems measure digitally using a PC or with an internal processor.

Usage:
For electrical measurement, sinus signals are the most accurate. The loudspeaker impedance should always be measured with a sinus signal, because this way the speaker reaches its swung-in condition. This is especially important by Thiele-Small measurements. For acoustical measurements, a non-reflective soundproof room is appropriate. Some measurement systems work with time-windows to eliminate room reflections.

Description:
During the measurement, the frequency of the sinus signal is raised gradually. As the plotter has a logarithmic frequency axis, the frequency steps are also determined logarithmic. For each frequency, the input signal is rectified and the value of the corresponding frequency displayed as amplitude.
By acoustical measurements, the curve shows dominant peaks due to the quality of the loudspeaker and the measurement environment. To be able to judge the characteristic of the measured loudspeaker better, the measurement curve is smoothed. Mechanical plotters smooth over the measurement speed or over the time constant of the rectifier the curve. Digital frequency plotters smooth the curve by averaging over a certain frequency range. Because of the logarithmic spacing of the frequency axis, the range is chosen as Octaves. Whole number parts of an octave are also common.

Phase Measurements:
Apart from the frequency range, digital measurement devices can also measure the phase with a sinus measurement. For electrical phase measurements is the most accurate method. By acoustical phase measurements, the running time, due to the distance between the microphone and loudspeaker, has to be preset. The distance is found using a square wave signal. The problem here is that usually the start of the signal is not right for the correct display of the phase. If the distance is chosen slightly behind the start of the signal, the phase is displayed most accurately. The distance is chosen correctly, when the phase shows the least phase jumps, or the phase shows a line over a large range.

3.2 SINUS-SWEEP

Knowledge: W1 THE ATB PRECISION WATERFALL MEASUREMENT

History:
The sinus-sweep measurement was developed from the sinus-step measurement to suppress resonances of the measurement environment. The extensive measurement setup by combination of several devices was first brought together in a single device, the TEF analyser, by the American Richard C.Heyser. The measurement method is also known as TDS, Time Delay Spectrometry. The TEF allows accurate measurements. An engineer though can only operate the program. The complicity leads to hot discussions about the correct setting of the time windows.

Usage:
For loudspeaker frequency range measurements the Sinus-sweep method is not so practical, because of the large number of single measurements.
The Sinus-sweep method is used to determine the acoustical properties of rooms. It gives an accurate picture of resonances, reflexions and the sound rebounding times of a room.

Description:
The Sinus-sweep measurement consists of several single measurements. The number of single measurements is the number of frequencies displayed. For each single measurement, a sinus signal is generated with constantly rising frequency, the so-called sweep. The signal contains all frequencies of a given frequency range. This signal is reproduced from a loudspeaker and recorded by a microphone. The electrical signal from the microphone passes through a band pass filter, set to a certain frequency. After filtering, the signal is displayed on an oscilloscope scribe. In the scribe, the signals are shown that contained the frequency setup in the filter. The form of display is described under W1, the ATB precision waterfall measurement. By the illustration the running time of the sound, due to the distance between microphone and loudspeaker, is subtracted at the start of the illustration. So the direct sound is shown as first. It displays the frequency range of the loudspeaker. The single measurements are shown in a 3D plot, the so-called waterfall diagram. The plot excises have following Units:
X-Axis = Frequency   Y-Axis = Amplitude   Z-Axis = Time
The waterfall diagram shows the real-time behaviour of the sound, as it is a combination of the time scribes of the oscilloscope measurements.
From the measurements all the acoustical parameters of the room are calculated.

3.3 MULTISINUS

KNOWLEDGE: W2 BASIC DEFINITIONS OF FFT

History:
Multisinus measurements were first possible because of the computer measurement technology. The foundation for the generation and evaluation is the FFT. The multisinus is a calculated noise signal, created by overlaying sinus signals.
The predecessor of the multisinus is the thermal noise signal. By thermal noise signals, the frequencies are spread stochocasticly. It is generated for instance from a semiconductor for instance a transistor. Looking at a long time the signal contains all frequencies at equal amplitude. By the FFT analysis the measurement shows a straight line, when a sufficient number of measurements are averaged. This signal is called white noise.

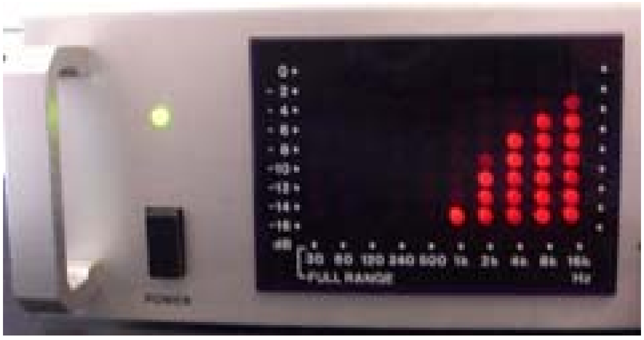


Fig. 3.1

Fig. 3.1 shows an analogue octave analyser. The measurement displays the frequency range of a white noise signal.

In the audio technology an octave or 1/3 octave analyser is used. When using a white noise signal, as a test signal, the above picture results. This is due to the logarithmic scaling of the x-axis in octaves. As each following octave has twice as many frequencies, we have following equation:

$$A_2 = \sqrt{A_1^{\,2} + A_1^{\,2}} = \sqrt{2A_1^{\,2}} = \sqrt{2} \cdot A_1$$

A_1 = Amplitude of the forgoing octave A_2 = Amplitude $\sqrt{2} =$ Factor for 3dB

As white noise is not well suited for such analyser measurements, pink noise was developed. Pink noise the FFT analysis shows a with –3dB falling frequency range for higher frequencies. An octave analyser measures a straight line.
Pink noise has a great advantage when measuring loudspeakers. Its spectrum is equivalent to that of music. So that the loudspeaker is work-loaded the same as when reproducing music. With white noise, the tweeter can easily be over powered falsifying the measurement result.
Thermal noise requires a great deal of averaged measurements to get reproducible results. To get reproducible results with just one measurement, the periodic noise, PN noise, was developed. The noise is calculated such, that during one period all the for the FFT analysis needed frequencies are contained with constant amplitude. This calculated white noise shows by FFT analysis an even frequency range, but is not suitable for loudspeaker measurement because of the high energy in the high frequency range. The for loudspeaker most suited, similar to music, pink noise shows by the FFT measurement a falling frequency range to high frequencies. Therefore, that it is not suited for the FFT measurement. As the PN noise is calculated, the physicist and mathematician Joachim Metzner has created a noise signal that stands up to the demands of signal with a music equivalent energy spread and a linear frequency range by the FFT analysis. This noise signal is known as the Metzner PN or MPN for short.
The multisinus also allows calculation of specialised measurement signals. The ATB Precision has three signals for testing Dolby Digital Surround systems. The signals are equivalents of the decoded audio tracks of DVD's.

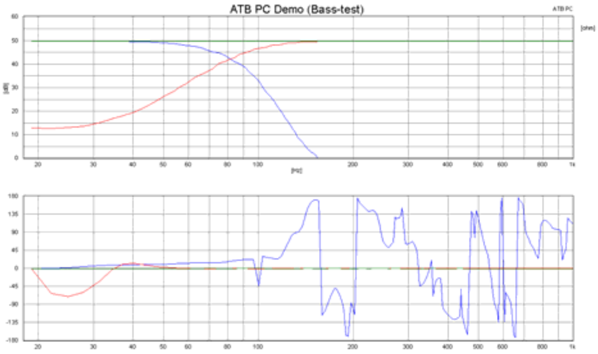


Fig. 3.2

Fig. 3.2 shows the three signals for testing Surround systems. They are Bass- Middle- High tone, Green, Middle- High tone, Red and the Subwoofer signal Blue. The phase jumps happen when there is no more signal amplitude there for phase evaluation. The phase of the signals is almost zero. So that the Bass and Middle-Hightone signals can be added to each other without breakdowns in the frequency range. The BMH signal is thus created.
The signals are calculated with the FFT filter. This can only be managed in real-time by fast computers because of the huge calculation extents.

Usage:
The multisinus is used by computer controlled loudspeaker measurements. It has a short measurement time by an accuracy equivalent to the sinus measurement. This applies to amplitude as well as impedance measurement. The measurement is very well suited for quality controlling. With the adequate algorithmic the multisinus allows a from operator independent correct phase measurement.

Measurement:
With this measurement the loudspeaker is driven by an MPN signal. The sound is recorded by a microphone and transformed into an electrical signal. By the FFT analysis the signal must be preset with a time window that has the time window values needed for the FFT pickup rate. For accurate measurement the time window must be set synchronic to the measurement signal. Because of the running time of sound, the time relationship between measurement signal and measured signal becomes unprecise. One way of time relation is to measure the distance with the distance measurement function. That is quite extensive and is a new source for faults. That is why the ATB analyses the measured signal with a complex algorithmic and chooses the optimal area for the time window. This guarantees an accurate and reproducible measurement. Also when measuring noise for noise stress measurement the algorithmic has proved it's self for determination of the time window. Where as to date analysers need to average several measurements, the ATB achieves the same with one single measurement. This is a huge advantage when measuring changing sounds.

The Phase Measurement
The multisinus allows by acoustical measurements accurate reproducible results, independent of operator. The ATB precision can through complex calculation, such as inverse FFT,   compare the state of phase of measurement signals and recorded signals. The differences are then shown as phase of the loudspeaker. One could also say that the phase with value zero of the measured signal is found and the difference shown. Through extensive mathematics a microphone distance independent phase measurement, by loudspeakers is made possible.
The phase is described in capital W.5 PHASE.

3.4 MLS

History:
A known loudspeaker measurement uses the digitally generated maximum length sequence signal MLS.
The measurement was developed by Doug Rife, the founder of *DRA Laboratories.* The measurement system called MLSSA was made public in 1989. During time, the MLS measurement was adapted from several measurement programs.
The MLS is a periodic, binary and pseudo stochastic sequence. This digital signal has only two states, either -1 or +1. By this measurement, the Input signal is transformed with the output signal, over an autocorrelations function, the Hadamard Transformation, into a Dirac-Impulse. The calculated impulse contains all information about the frequency response. From this impulse the frequency range, phase range, impulse response and all acoustical parameters are calculated by room measurement.
This kind of measurement technology made possible at the beginnings of computer technology, the use of computers for loudspeaker measurement. The generation of the measurement signal needed no, in those days' expensive, A/D transformers. The generator consists of three digital IC's. The Hadamann transformation could be adapted to the first, in those days low performance, PC's. The measurement has been upgraded through more and more new software and evaluation methods. Nowadays there still are a large number of users.

Usage:
The MLS is used for acoustical measurements. By loudspeakers the frequency and phase range, the impulse and step response and the breakdown spectrum (waterfall diagram) are measured. By the room acoustic, all acoustical parameters are measured with this measurement. It is not appropriate for electrical measurements, because of the inaccuracy in the low frequency range.

Measurement:
By this measurement, the MLS signal is generated by digital generator. A loudspeaker reproduces the signal. It has recorded from a microphone and transformed by a computer. The Hadamar transformation calculates the impulse response from the measurement result. The frequency range is calculated from the impulse response per FFT. As the result is dependant on the A/D converter, the software right up to the operating system, some systems do a calibrating measurement before each measurement; others do a reference measurement on the second channel simultaneously. The measurement is strongly dependant on the number of FFT points and the time window chosen. As in the low frequency range by a 32000-point measurement the reproducibility of the measurement cannot be achieved, the measurement is split into two parts. The one part has a range from 20Hz until 100...300Hz and uses a MLS adapted for that range. The second measurement measures the range up to 20k...50kHz.
Contra to the multisinus measurement of the ATB, whose measurement parameters are setup in the program, so that the highest accuracy is achieved, the user plays an active role in the evaluation. That makes the results user dependant. That is why the systems that work with MLS cannot receive a certificate to the ISO 9001 industrial norm.
Before evaluation, a time window is set in the impulse response measurement window. For that the user must have knowledge of the basic rules of FFT, as described in W2 basic terms of FFT.

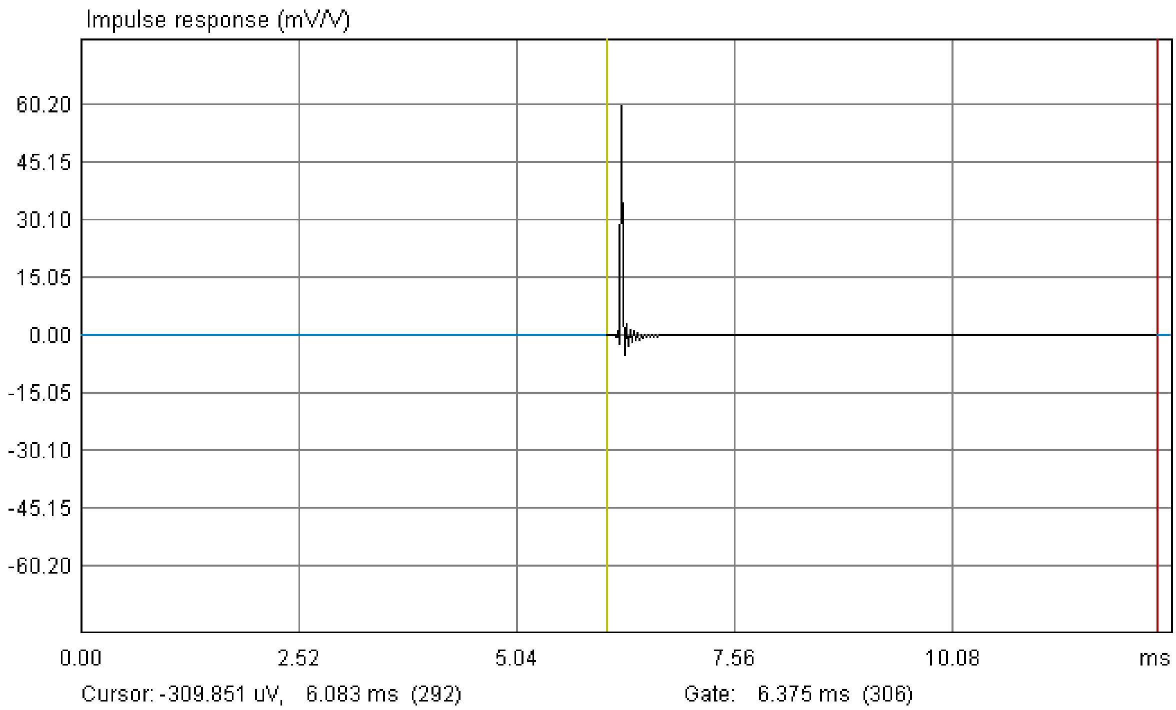


Fig. 3.3

Fig. 3.3 shows the impulse response of an MLS measurement. For the measurement, the input was short-circuited with the output, to test the electrical quality. The start of the time window was chosen as the start of the impulse. When measuring loudspeakers the end of the time window is set to the start of the first visible environment reflexions.

Measuring the breakdown spectrum
A further MLS measurement is the breakdown spectrum measurement also called waterfall measurement. The 3D graphic is build-up from the frequency responses calculated from the impulse response. The description will not go into the many different parameters that change the picture. It will try to describe the meaning of the measurement for the time behaviour. The 3D graphic shows on the X-axis the frequency, on the Y-axis the amplitude and on the Z-axis the time.

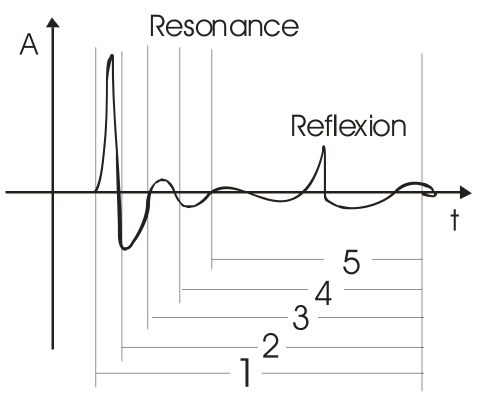


Fig. 3.4

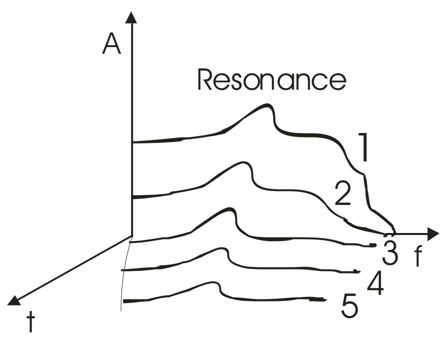


Fig. 3.5

Fig. 3.4 shows the impulse response of a loudspeaker. The numbers name the time windows for the FFT calculation. The longest window is called Nr. 1. It covers the whole range of swing-in behaviour, resonance, swing-out behaviour and reflexions. The behaviour in time cannot be shown by the frequency response in curve Nr.1, the 3D plot in Fig. 3.5. The behaviour in time is no more readable. In addition, the reflexion as described before is not recognisable any more. In the frequency analysis of 2,3,4 and 5 the resonance swing-out is shown. As the reflexion has a wide frequency spectrum it can also not be recognised on this curve. The MLS breakdown spectrum is interpreted from many developers as time behaviour. The developers are quite sure that a time behaviour that cannot be recognised also has no impact on developments. That this is not so will be shown in the following capitals.

4. ENVIRONMENT INDEPENDENT MEASUREMENT

Knowledge: W3 Multisinus measurements with the ATB Precision

When measuring the sound pressure level of loudspeakers, the environment and positioning has a great influence on measurement results. To avoid that influence loudspeakers are measured in soundproof rooms. For the development of single speakers, soundproof rooms are optimal. When developing speaker combinations, loudspeaker boxes, the tuning in soundproof rooms leads to results far from practical usage.
Tuning in soundproof rooms is not equivalent to real hearing experience, as the speaker usually stands on the floor and „sees“ a different environment to that of a soundproof room. The floor causes a gain of 3dB in the low frequency range. The ideal measurement seems to be only the free-field measurement. The loudspeaker stands raised on the middle of a large place with at least 20m far boundaries. The microphone distance is 1m. Interference noises are wind and environment noises.

4.1 TIMEWINDOW MEASUREMENT

A glass box is chosen for the example measurement. The box has hard walls to demonstrate the effect of inside reflexions best. Stone cabinet or boxes with tiles on the inside wall show similar measurement results. The measurements are carried out in a room with high enough ceilings comparable to free field measurements.

Fig. 4.1

Fig. 4.2

Fig. 4.1 shows the loudspeaker.
Fig. 4.2 shows the frequency response measured with the ATB.
By MLS, the measurement starts with the impulse response. In the time scribe for the impulse response, the window for the evaluation area is set.

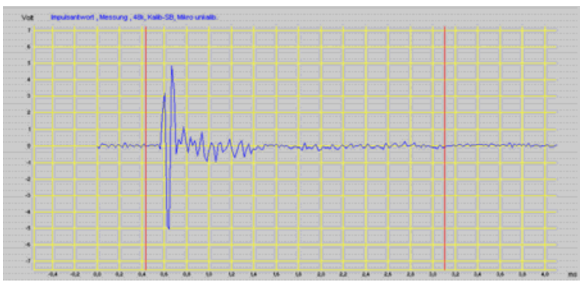


Fig. 4.3

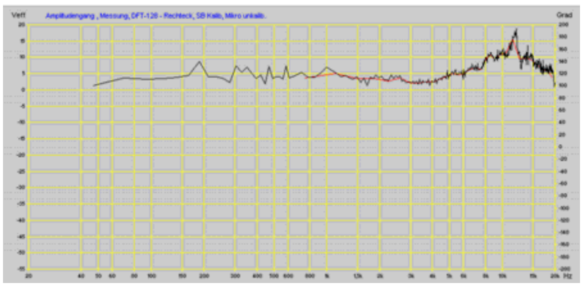


Fig. 4.4

Fig. 4.3 shows the impulse response of the loudspeaker as measured with an MLS system. The calculation of the frequency response is carried out with a large window, red curve, and a small window, black curve. The length of the large window is restricted by the already visible faults in the high frequency range. That is why no lower frequencies can be displayed. The curve shows a linear frequency range from 50Hz downwards. The comparison with the ATB measurement shows that that is wrong. That makes a second measurement for the low frequencies necessary. For the measurement a MLS with a high boundary of 300Hz is used. Measurement systems without the second measurement are of no use.
The small window shows the smoothing function. The resonances of the air in the cabinet are no more to be seen. This important information about the sound quality gets lost when measuring with the small time window.
By the following measurement, the influence of a room reflexion when measuring with time windows is shown.

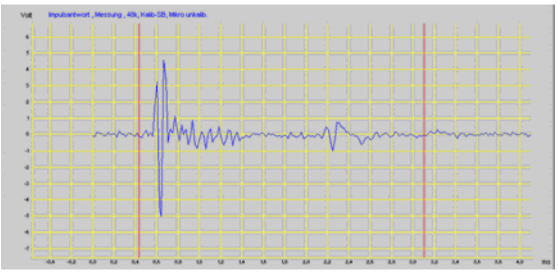


Fig. 4.5


Fig. 4.6

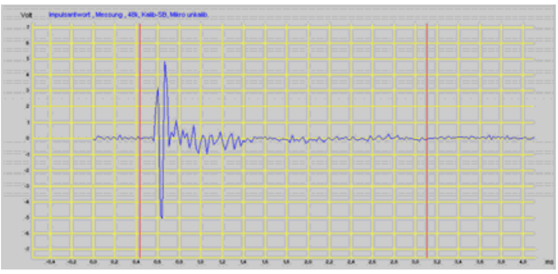


Fig. 4.7

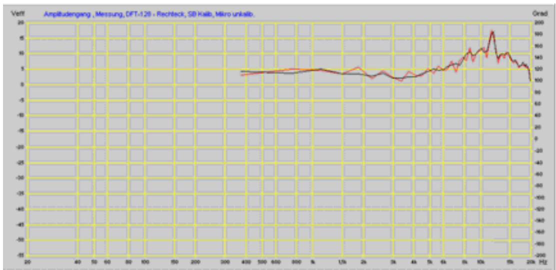


Fig. 4.8

The measurement shows the reflexion due to a glass plate behind the microphone, see Fig. 4.6. This is to simulate the strongest occurring reflexion in a room during measurement. This appears in the impulse response, Fig. 4.5, strongly visible. The impulse response in Fig. 4.7 shows no reflexion. In the frequency response, Fig. 4.8, both measurements with and without reflexion are compared. The reflexion that makes any measurement without time window impossible is hardly visible with time window. The measurement inaccuracy through the reduced density of the time window measurement even blurs the strong reflexions in the time window. From this, we combine that the effect of time windows is lesser the cut of room reflexions, but far more the strong smoothing of the measurement curve through reduced frequency density. However, the drawback of this smoothing is that it is linear and not logarithmic, contrarily to the frequency axis division which is in 1/3 octave or octaves. This way the low frequencies have a strong and the high frequencies hardly any smoothing. The statement that the measurement makes is thus misleading.
To attain reproducibility in the lower frequency an adaptive time window is used for measurements for frequencies lower than 300Hz. This kind of measurement equals more a

calculation. The measurement curves show nice and smooth frequencies responses, good for publication. The developer should though keep in mind that for the sound quality of the loudspeaker behaviour can hardly be recognised in such measurements.

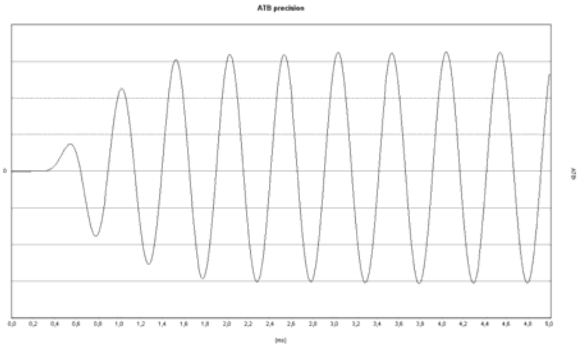


Fig. 4.9

Fig. 4.9 shows the swing-in behaviour of a typical woofer at 200Hz. The woofer needs 3 cycles to reach the amplitude shown later in the frequency response measurement.
The fault of time windows in the low frequency range is due to the swing-in behaviour of the woofer. The woofer needs up to 3 cycles to reach the amplitude to transmit the signal at the right amplitude. The time window needed for the swing-in time is so big that it takes up the whole FFT evaluation area. In the low frequency range the more correct results are achieved by near field measurements.

4.2 THE NEARFIELD MEASUREMENT

By the near field measurement, the physical property of sound getting weaker over the distance is made of use. A measurement microphone is placed 10cm away from the woofer. This way the direct sound is many times greater than reflected sound, so that only the direct sound is measured. The measurement is only carried out up to 300Hz and extended by another measurement for the frequency range above 300Hz. The second measurement is made at a distance of 1m to measure both mid and high tone speakers at the same time.

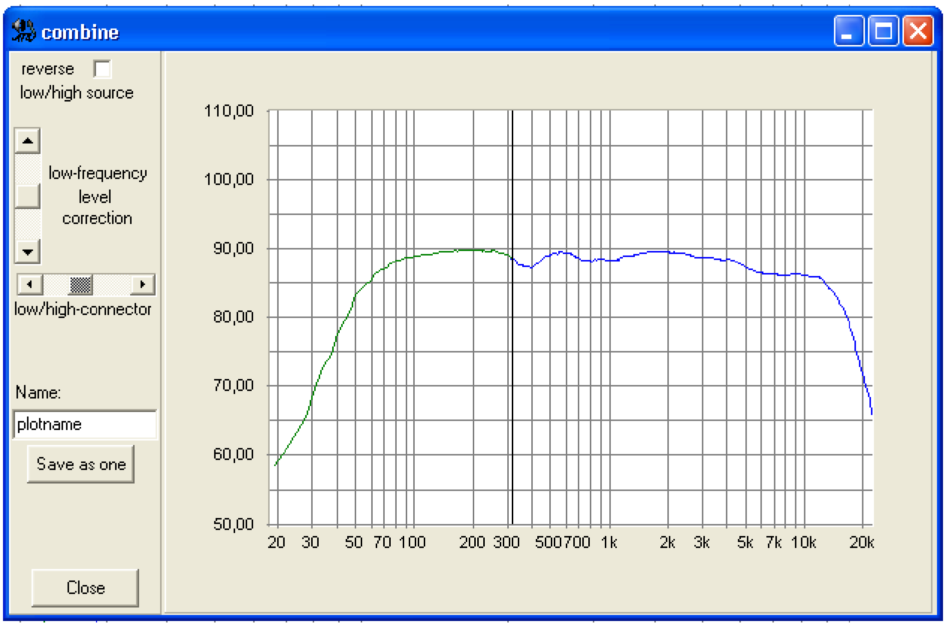


Fig. 4.10

Fig. 4.10 shows the menu Combine for the addition of low and high tone measurements. By the near field measurement, all low frequency sources must be measured. These sources include low- middle tone speakers when the crossover frequency is lower than 300Hz. For the measurement with ATB, the measurement constant measurement with averaging is chosen. During the measurement, the microphone is moved slowly from source to source. This way all sound sources are accounted. The path goes from the middle of the middle tone speaker to the middle of the woofer. By bass reflex, the relationship of the woofer membrane radius to opening size of the reflex tube is to be considered. According to the equation:

Pressure ~ Power x Area

A small opening will cause more pressure than the larger surface area of the woofer membrane. That is why the opening is only measured at the rim. How far from the rim the microphone must be cannot be easily calculated. Through the following description correct measurement results can be achieved.

In a small measurement room, a measurement is carried out with the boarder area microphone. The microphone is placed on the floor. As only the low frequency range is to be evaluated, the normal positioning is chosen. To suppress the first floor reflexion the boarder area measurement is used. This avoids the disturbing depletion above 100Hz. The first reflexion can also not be blended out by a time window. Because of the small time delay difference it lies within the time window.

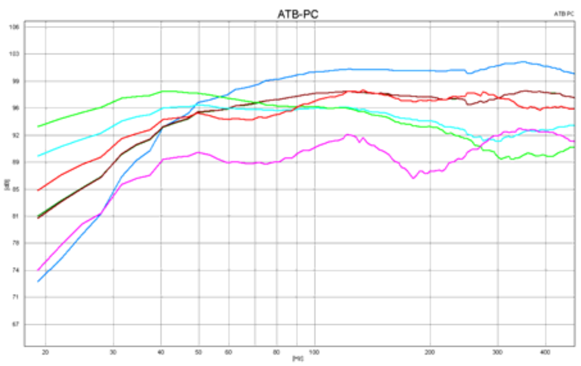


Fig. 4.11

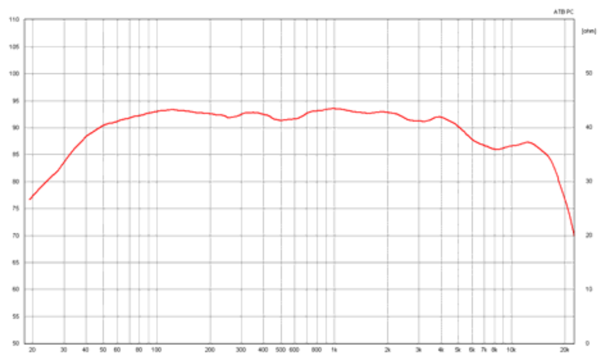


Fig. 4.12

Fig. 4.11 shows the measurement of the bass reflex opening at different distances. The follow up of distance to the rim of the basreflex opening is green, light blue brown and blue. The measurement was carried out with near field correction. The red curve is the boarder area measurement and violet the measurement at 1m. Both measurements were carried out without correction.

Fig. 4.12 shows the resulting frequency curve of the loudspeaker. It is made up of the brown curve, which equals the boarder area measurement and the violet curve. The curves were connected together in the Combine menu.

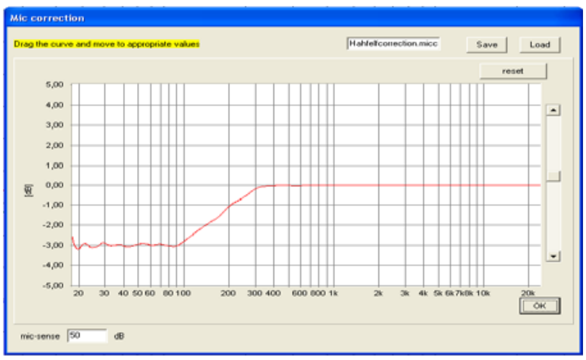


Fig. 4.13

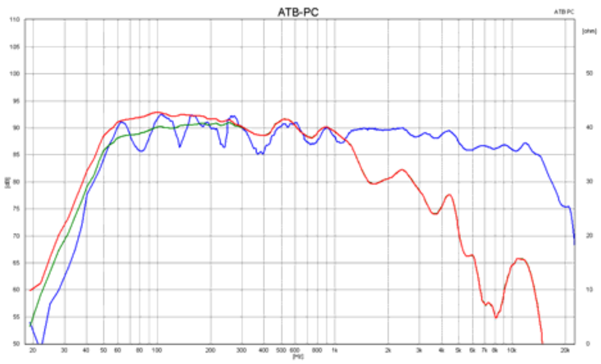


Fig. 4.14

A problem with the near field measurement is the measure of bundling, a gain of sound pressure in the near field area. The gain can be up to 3dB in the low frequency range. The ATB PC measurement system has a Mic-correction function. Here a correction curve for near field measurement is read in. The Fig. 4.13 shows the correction curve that is read into the Mic-correction. Once setup the Mic-correction curve is called up before near field Measurements. The frequency range measurement, Fig. 4.14, shows the near field measurement without correction, red and with correction, green. The green curve equals the blue curve of the measurement at 1m distance. With the described measurement process, the low frequency range can also be correctly measured in normal rooms.

4.3 THE BOARDER AREA MEASUREMENT

By the boarder area measurement the measurement microphone is operated as boarder area microphone. The microphone lies flat on the floor and has that way a half ball characteristic.

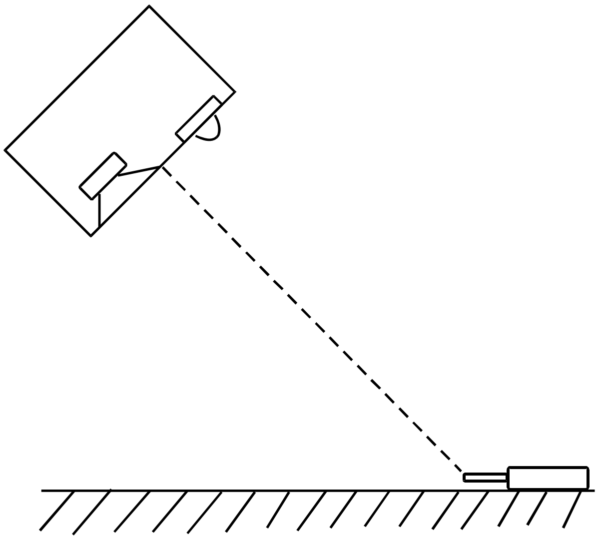


Fig. 4.15

The picture shows the positioning of loudspeaker and microphone.

The sound waves of the loudspeaker hit the floor at an angle and reflect into the room. This is especially in the mid- and high tones positively noticeable. Also in the low frequency range, where room resonances build up, only half the energy is picked up from the boarder area microphone, so that they do not influence the measurement so much.
As far as the room resonances go, the rule is the bigger the measurement room the better.

5. THE TIME BEHAVIOUR

Knowledge: W4 Phase
W5 Dynamic measurement

To completely describe the transmission path, the frequency and phase response has to be measured. The frequency response shows the transmisson, the amplitude size and the phase relationship in time of oscillations. In the loudspeaker technology, it is common to only observe the frequency sweep. The phase response is considered unimportant for the sound quality. This is in conjunction with the difficulties in measuring the acoustical phase. Earlier the measurement took hours and needed special measurement equipment. By computer-aided systems, the phase is calculated along with the frequency response measurement by the FFT. However, by the display of the impuls, for instance by the MLS, the distance is not found correctly and with that a curve created without real meaning. 20 years ago a test about the ability of hearing phase displacement was carried out. At this test, two sinus signals were played, and their phase varied. Here curves are created in the oscilloscope display through overlay of the signals. The test persons could not hear differences in the curve form because of overlay. From this it was followed, and to date still propagated, that the phase cannot be heard. For the test two frequencies in swung-in state were used for the judgement. This is not adequate for an absolute statement

5.1 THE CROSSOVER

.
As an example for time behaviour two loudspeakers with identical speakers and cabinet volume are built up.

Fig. 5.1

Fig. 5.2

Fig. 5.1 shows the loudspeaker „Analog.on Richtig“. The box was developed to demonstrate the Dynamic- Measurement program. After a very good test result in „Klang+Ton“ most of these loudspeakers are used in very many recording studios. A copy is also available in the USA at one of the largest recording studio suppliers.
The loudspeaker „Demo“ is built up with the speakers of the Richtig, Fig. 5.2. By the construction the cabinet volume of the Richtig were generally taken over. According to the state of loudspeaker technology, the speaker was built up with a flat soundboard. The crossover is designed according to the Linkwitz theory, with 24dB and Butterworth characteristic.

The Step Signal

The time measurements are always carried out with the oscilloscope measurement. A well-known test is measuring the step behaviour.

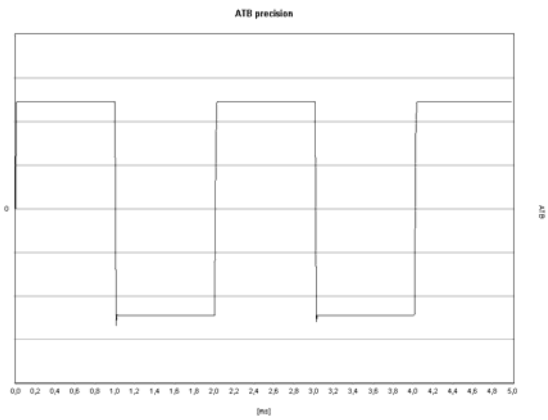


Fig. 5.3

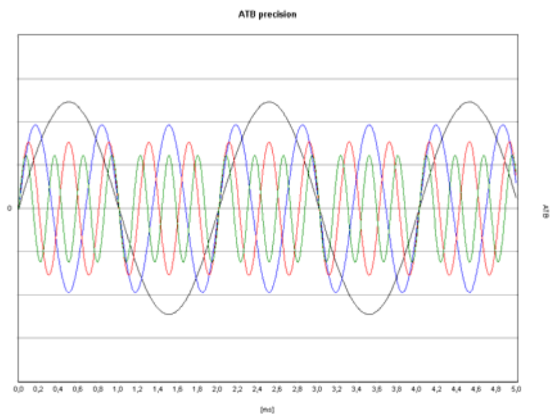


Fig. 5.4

Fig. 5.3 shows the time run of a step signal.
In Fig. 5.4, it has shown that the step signal is a combination of overlaid sinus waves. It is made up of the main wave = black and the odd number over waves 3 = blue, 5 = red, 7 = green and other not shown over waves. A step-like loudspeaker signal shows a linear phase with constant amplitude. That only goes for frequencies ≥ of the main wave. Also only the swung-in state is shown.

In the following pictures it will be demonstrated how step signal behaviour is made up of frequency response and phase.

Richtig

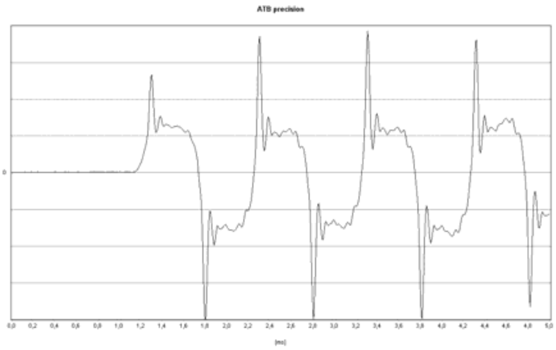


Fig. 5.5

Demo

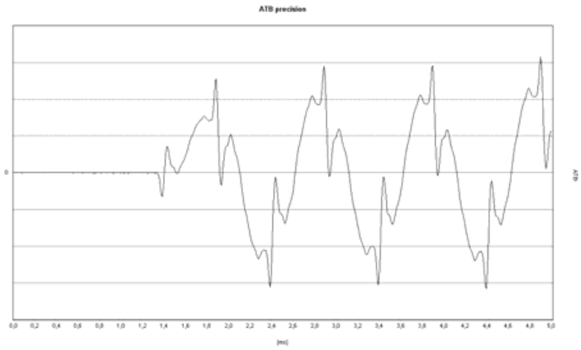


Fig. 5.6

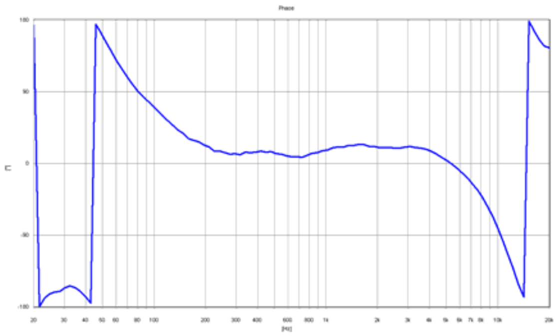


Fig. 5.7

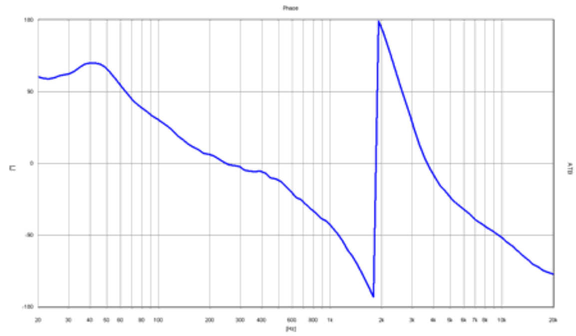


Fig. 5.8

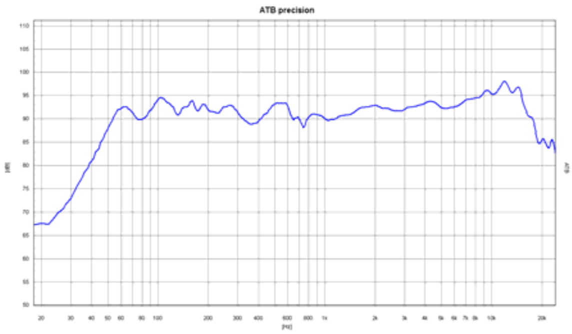


Fig. 5.9

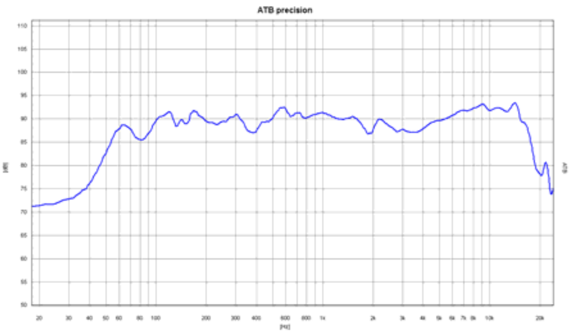


Fig. 5.10

The Richtig shows a good step signal behaviour, Fig. 5.5. The single waves of the step signal are transmitted across at the same time. That is shown on hand of the linear phase, Fig. 5.7. The peaks at the start of the step signal show the gain at 15kHz, Fig. 5.9.
Fig. 5.6 shows that the step signal behaviour of the Demo is bad. This also shows in the strong turn of phase, Fig. 5.8.

The Step response

Mathematically the step response is derived from the impulse response by integration.

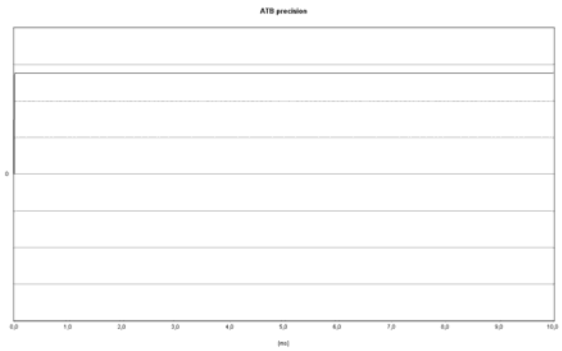


Fig. 5.11

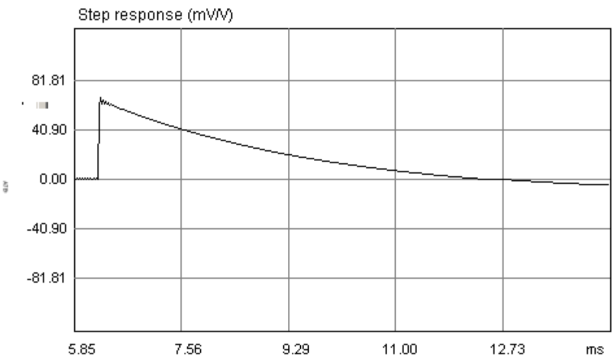


Fig. 5.12

Fig. 5.11 shows the measurement signal for the step response. The step response starts at zero time and has for time = zero constant and positive amplitude.
The step response has as in the impulse response all the information about the transmission path. One drawback of the step response is the very complex illustration. The behaviour of single frequencies is not to be made out. That is why the 3D display, Dynamic-Measurement program, was developed. The program is described at W5.
The step response is the most important illustration of transmission path time behaviour. For digital signal processing the step response is also recommended for testing. As to be read in following book:

The Scientist and Engineer’s Guide to Digital Sound Processing.

Fig. 5.12 shows the electrical measured step response of the MLS measurement. Contrarily to the oscilloscope measurement the calculated MLS step response builds a high pass. As the MLS, in accordance with FFT, has only few frequencies points in the low frequency range, mainly high frequencies are shown in the step response. As the display of low frequencies is a function of the MLS frequency points and the chosen time window, a measurement-based assessment is hardly possible. As in the electro-technology the step response is clearly mathematically defined, calling a MLS measurement step response is clearly false.

Step response measurements on example boxes.

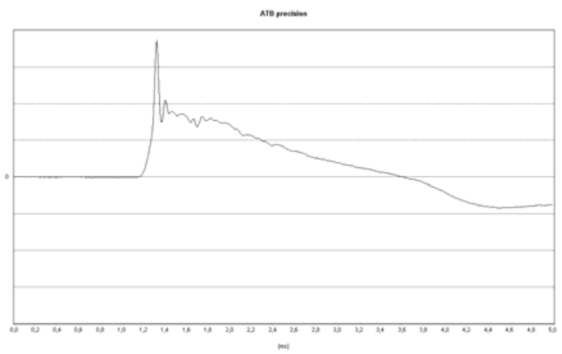


Fig. 5.13

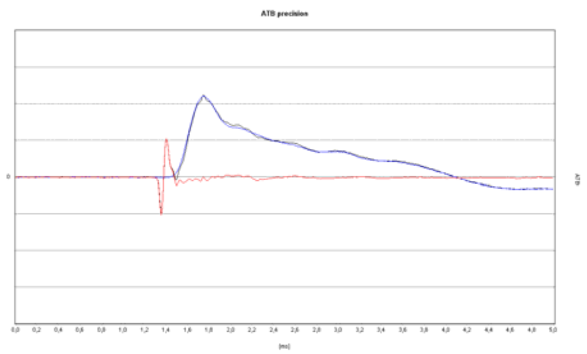


Fig. 5.14

Fig. 5.13 shows the ideal step response of the Richtig box. The signal rises suddenly to decrease evenly after a peak. The decrease is due to the high pass function of all loudspeakers. Only frequencies above the boarder frequency are transmitted because of the transmission resistance.

Fig. 5.14 of the Demo shows the detachment in time of the middle and high tone speakers. The high toner comes first, red. The middle toner follows blue. The black curve of the system step response cannot be seen because of the accuracy of the measurement as the curves overlay exactly.

Electrical time measurement with Sinus-Burst

The Sinus-Bust is very well suited to show the properties of a single frequency in a transmission path. So that the crossover behaviour at the crossover frequency can be aimed for assessment. A long burst is used as the swing-in behaviour is important in this case.
In the Demo loudspeaker, a crossover according to Linkwitz with 24dB/Octave is used. The crossover has four energy depots. The four energy depots are ideal to show the time behaviour of the crossover with an electrical measurement.

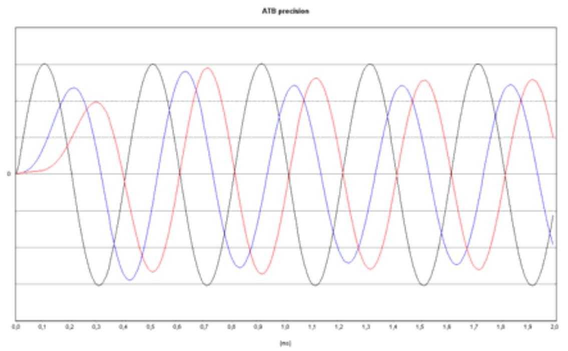


Fig. 5.15

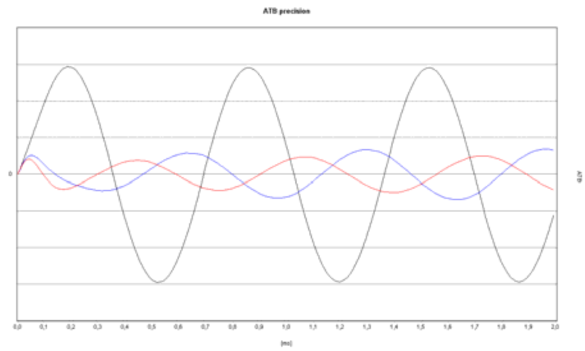
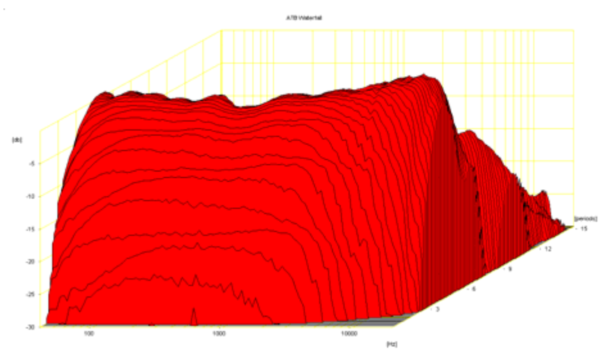


Fig. 5.16

In the oscilloscope display of the low frequency crossover, Fig. 5.15, the Sinus-bust of the input signal from amplifier is black, the signal between the coils blue and the signal at the output of the crossover red. The blue and red curves show the time delay of the signal because of the energy depots. The time delay can be corrected by the positioning of the speakers. The swing-in behaviour of the crossover influences the sound quality the most. The burst of the output signal, red, starts with a small amplitude, then over-swings with a high amplitude to then swing at the amplitude that later appears on the frequency response measurement. The low amplitude of the first half wave shows the bad impulse behaviour. A short signal is comparable with that half wave and would be transmitted with too low amplitude. Meaning that a part of the signal is lost over the frequency range.
Fig. 5.16 shows the high pass. The high pass has a forward running phase, as the output signal, red, shows. Through the compression of the signal at the start new frequencies are created, also called swing-in faults.
With the 6db crossover of the Richtig, delays or swing-in is hardly measurable.

The 3D Display

By Sinus-Bust measurements only one frequency is displayed. To illustrate a frequency range the 3D display is used.
The 3D display shows the oscilloscopes for single frequencies in a graph. The axis's have following meaning. Y = amplitude, x = frequency and z = time or cycle.
Main two measurements are used.
The waterfall display uses a Cosinus-Bust as generator signal and shows the behaviour of the swung-in state. The measurement is described in W1. The dynamic measurement mode shows the non swung-in state. The measurement is described in W5.
The waterfall diagrams for the example boxes are shown. Interesting here is the different time behaviours. That is why the swing-in behaviour is shown.



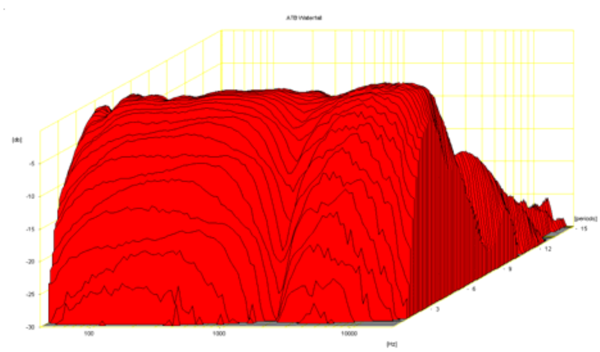
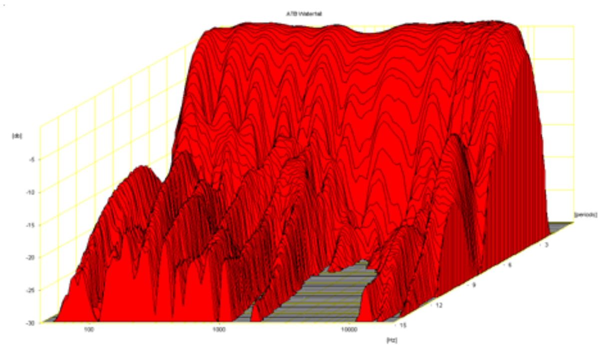


Fig. 5.17 Fig. 5.18

The loudspeaker Richtig, Fig. 5.17, shows the same time behaviour for nearly all frequencies. The lines of the same time run parallel. A light delay because of the tweeter getting slower can be seen from about 10kHz onwards.
The loudspeaker, Fig. 5.18, demo shows a strong time delay at the crossover frequency from middle to high tone. The middle tone speaker is not adapted to the tweeter. The breakdown means, as in the electrical Sinus-burst measurement, the suppression of short impulses. That is why the middle tones sound very clean, as parts of the voice or the overtones of musical instruments that fall in that area are not reproduced properly. The tweeter though is running ahead of time, making the high tones very distinct. They have no connection to the middle tones.

The swing-out behaviour



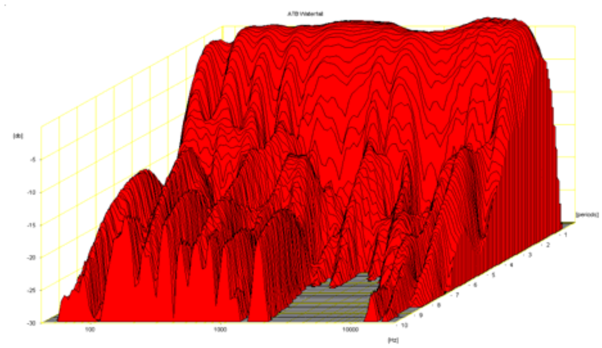


Fig. 5.19 Fig. 5.20

The Fig. 5.19, the swing-out behaviour of the Richtig also shows the hardly noticeable crossover from mid- to high tones. There is no bending of the sound on the cabinet edge to be seen. By case, curving above the low-mid toner the bending of sound is accounted for in the transmission behaviour of the speaker. The theory for this comes from horn loudspeakers. These can be developed greatly by use of the waterfall measurement.
Fig. 5.20 shows the swing-out behaviour of the loudspeaker Demo. An additional resonance is to be seen in the transmission range. Here the time play between the low pass and high pass is not correct.
In both pictures a forward running mountain is to be seen. As both boxes were measured from the same position, this is a reflexion of the flat soundboard. As the acoustic optimised soundboard does not show this resonance, those that propagate flat soundboards should think about it.

3D Step Response

In the 3D Step Response (Dynamic Measurement) measurement the behaviours of loudspeakers in the non swung-in state are shown. This measurement is very important to judge the reproduction of impulses. A half sinus is used as measurement signal. This signal compares to impulses in a music signal. This can be well seen in the oscilloscope diagram in chapter 2. The measurement Signal.

Swing-in behaviour

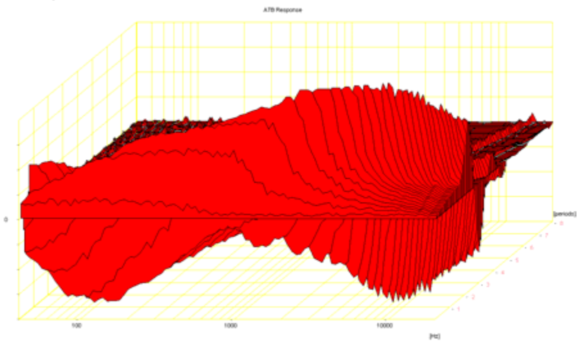


Fig. 5.21

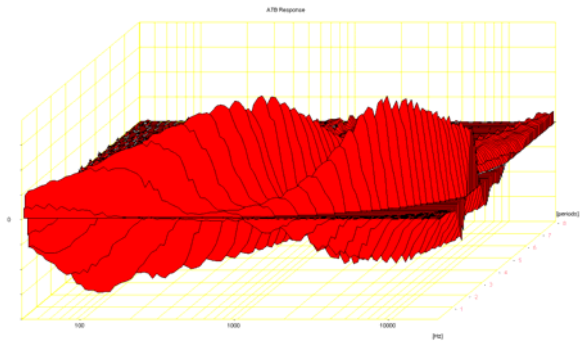


Fig. 5.22

The pictures show the swing-in behaviour of both loudspeakers.
By the Richtig in the left hand Fig. only one positive mounting comb is to be seen. This means that all impulses are transmitted at the same time. The light delay at high frequencies does not disturb the even reproduction. The swing-through that has to be seen in the negative mountain is due to the high pass function of the speaker and cannot be physically avoided.
The right hand picture shows the loudspeaker Demo and has two positive mountain combs running through the graph. One shows the middle toner and the other the high toner. The impulses in the transmission range are first transmitted from the tweeter and then from the mid range.

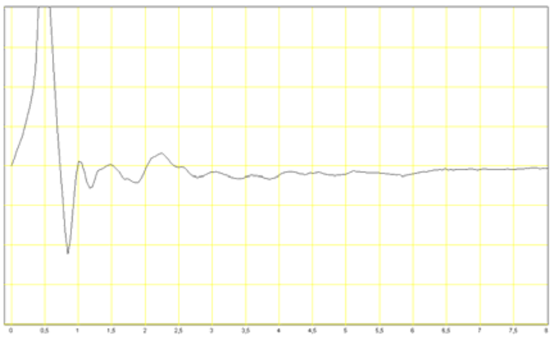


Fig. 5.23

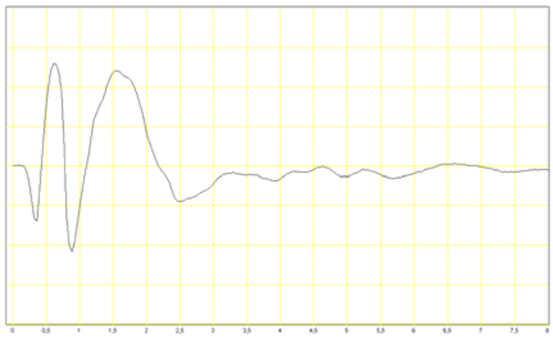


Fig. 5.24

The pictures of the analysis function of the dynamic Measurement show a cut from the 3D of the Demo. The frequency is 3700Hz. Fig. 5.23 of the Richtig shows just one impulse with a light through swing. Fig. 5.24 of the Demo shows a copped up impulse. The swing-in starts with negative amplitude, followed by a positive over-swing. After the next negative swing, the amplitude goes to the positive amplitude of the middle toner. As both impulses do not fit together in time, the picture shows an impulse with double width. This disturbs the signal additionally.

Swing-out behaviour

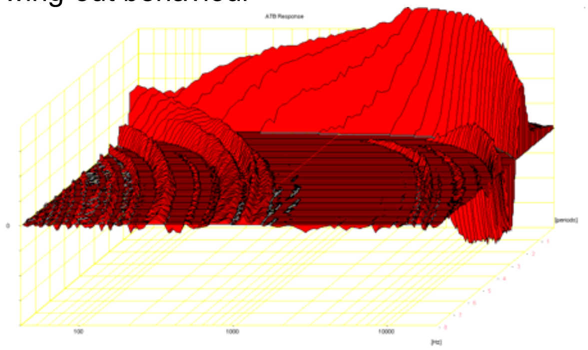


Fig. 5.25

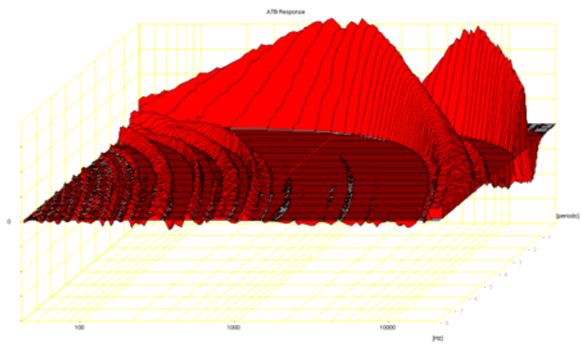


Fig. 5.26

In the pictures the swing-out behaviour of both loudspeakers are shown. Fig. 5.25 shows the Richtig. The mountain in the low frequency range shows the sound pressure of the bass reflex tube. By the Demo, Fig. 5.26, the copping of the impulses is even clearer as in the front side view. The long stretched mountain of the middle toner shows another quality of the crossover. For high frequencies it creates a resonance system that can be excited by impulses beyond the hearing range.

5.2 THE SOUNDWAVES IN LOUDSPEAKER CABINET

The soundwaves in loudspeaker cabinet also show time behaviour. The air creates a spring and mass system. The physicist Helmholz describes this. The Thiele-Small theory describes the effect of the closed in air of the loudspeaker cabinet. Here the air is considered as a homogenous swinging mass. However, this is not the case. The air movement is strongly effected by the form of the cabinet and the position of excitement (Loudspeaker position). The swinging air always has a back effect on the loudspeaker. In addition, the running time inside the cabinet must be considered. The loudspeaker and the air build a swinging system with resonances and correlating swing-in and swing-out behaviour. This behaviour is visible in the sound pressure frequency range of the loudspeaker. The loudspeaker transmits the reflexions that push on the speaker, especially by modern speakers with hard membranes. Driven by a sinus bust the sound pressure level over time is illustrated. To study the effect of sound in cabinet good 500 documented measurements were made. Here the results are just slightly gone into.

Two loudspeaker cabinets are setup with microphones.

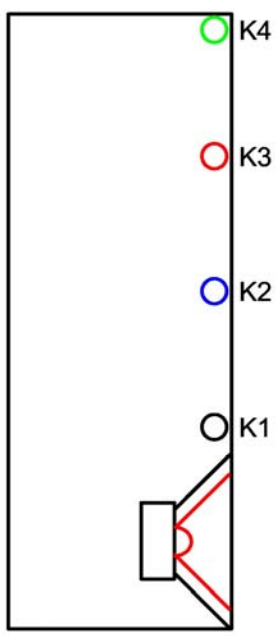


Fig. 5.27

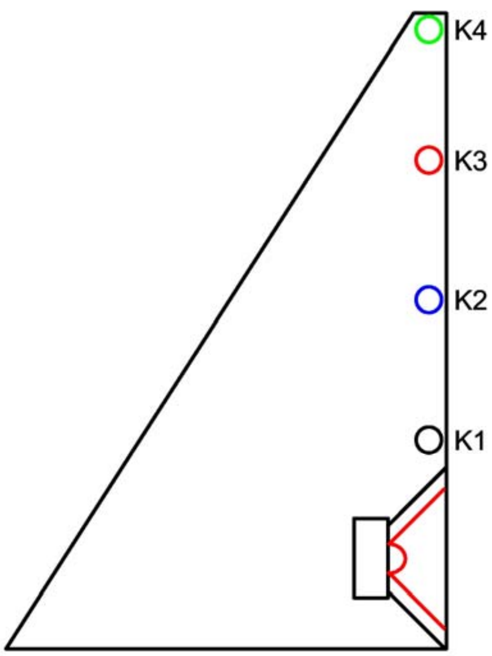


Fig. 5.28

The picture shows the cabinets with loudspeakers to measure the time behaviour of the sound waves. The cabinet Fig. 5.27 shows a conventional four-cornered loudspeaker casting. Fig. 5.28 shows the triangle cabinet of the TQWT box.

The sound movement in the cabinet

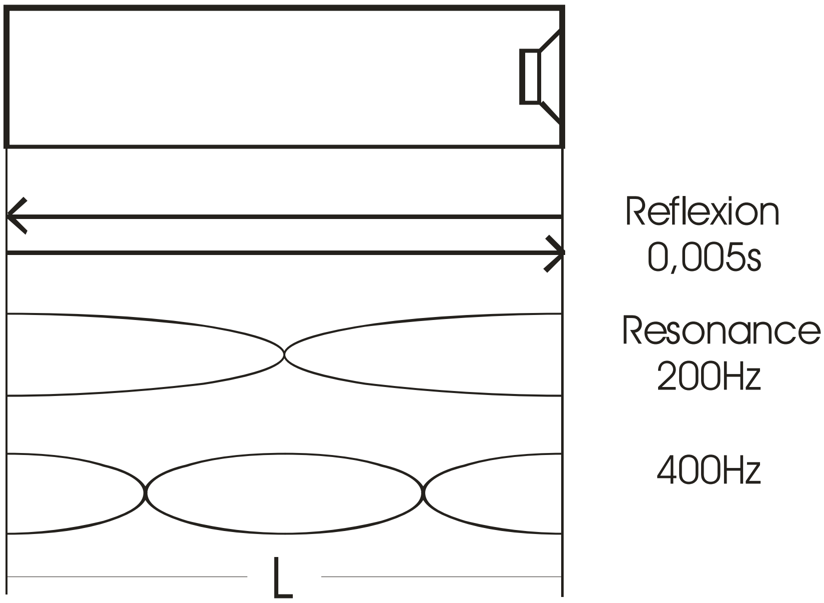


Fig. 5.29

Fig. 5.29 shows the spread of sound in a four-cornered cabinet.

The time of the reflexion is calculated with the equation:

$$t = \frac{2L}{c} = \frac{2 * 0{,}84m}{340\frac{m}{s}} = 0{,}005s$$ *L = Cabinet length*

The resonance is calculated with the equation:

$$f = \frac{340\frac{m}{s}}{2L} = \frac{340\frac{m}{s}}{2 \times 0{,}84m} = 200\frac{1}{s} = 200Hz$$ *L = Cabinet length*

By excitement, the generator frequency can be in a wide range around 200Hz

The following measurements show the sound level frequency ranges of the microphones when the speaker is driven with a sinus-burst.

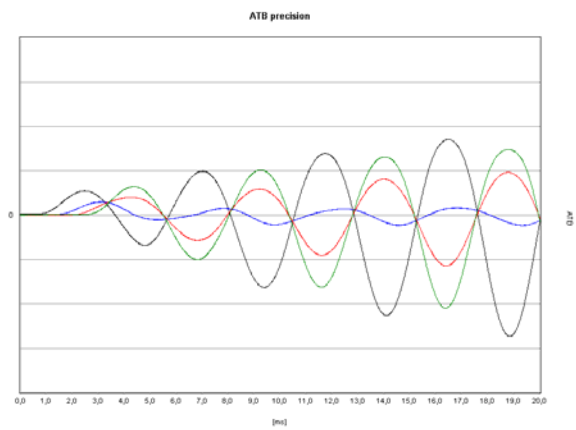


Fig. 5.30

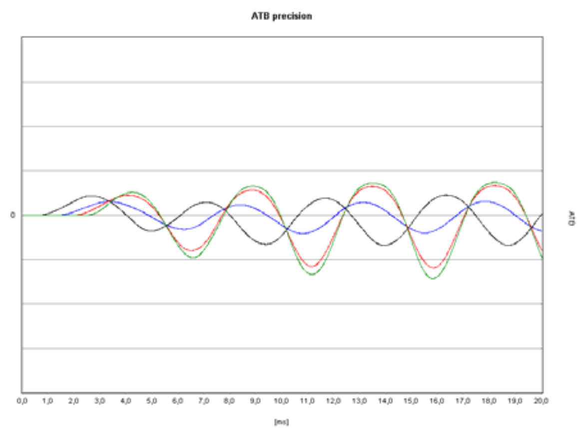


Fig. 5.31

The pictures show the oscilloscope diagrams of sound pressure levels measured in the cabinet. Fig. 5.30 is the four-cornered cabinet. At the start, the running time is well recognisable. The highest microphone shows the longest delay. Further, more the rising amplitudes are noticed. After even more than four cycles the system still has not swung-in. Through the even spacing of the microphones, the resonances are well seen.

Fig. 5.31 shows the triangle cabinet. Here the running time is also to be seen at the start. Against expectations, the sound wave has not disappeared before reaching the highest microphone. It even has the highest amplitude. Standing waves cannot build up in the cabinet. Resonances are not to be seen.

For further studies, a near field measurement per waterfall diagram is made.

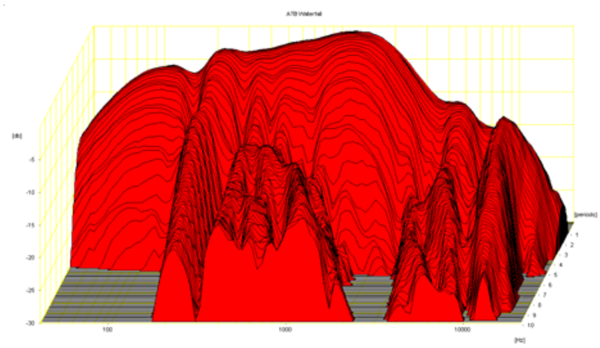


Fig. 5.32

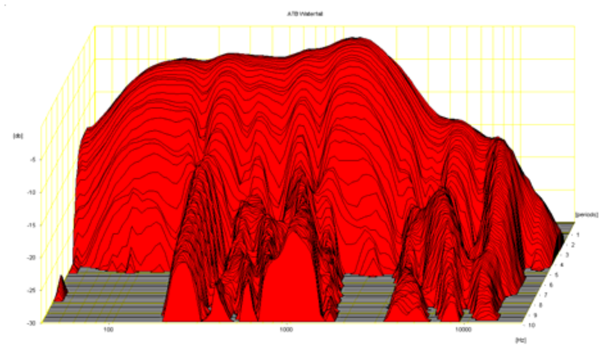


Fig. 5.33

In Fig. 5.32, the waterfall diagram of the four-cornered box is shown. The resonances at 200Hz and 400Hz are clearly visible. The break down of amplitude at 200Hz implicates a reflexion.
The waterfall diagram Fig. 5.33 of the triangle cabinet does not show those resonances. Still the reflexion at 200Hz is still there. It is recognisable by the big mountain and the flat decrease at 200Hz.
The resonances/reflexions at 1000Hz are suppressed by dampening material by loudspeakers.

The dynamic measurement program shows the reflexions quite clearly.

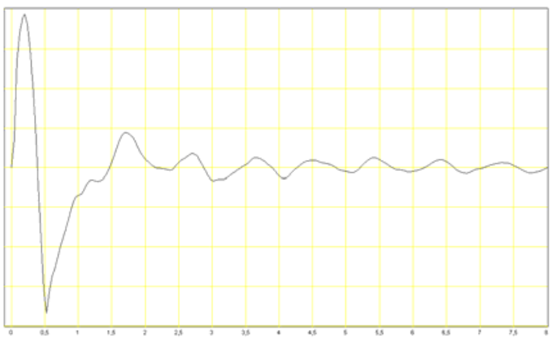


Fig. 5.34

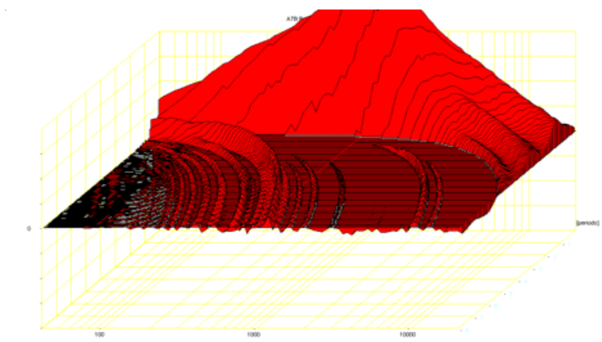


Fig. 5.35

The 3D step response Fig. 5.35 of the triangle cabinet shows the behaviour of a broadband loudspeaker. It consists mainly of one mountain. The left hand picture shows the analysis at 170Hz. It correlates also to a broadband speaker and shows a sinus half wave with a single swing through. The behaviour could be considered ideal if not for the forward running mountain at low frequencies. That is undoubtedly a reflexion. The analysis shows the constantly weaker going signal. Between two amplitudes, the running time of 0.005s is always to be measured.

The reflexion is clearly recognisable in a frequency range measurement together with a phase measurement.

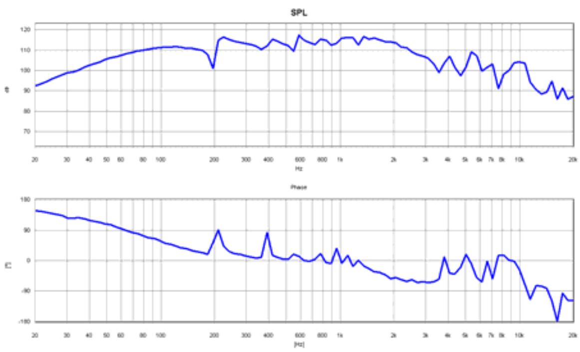


Fig. 5.36

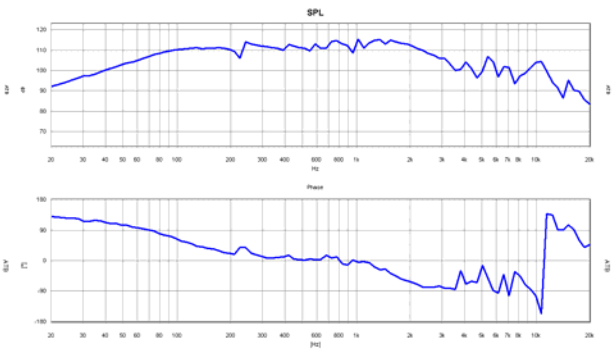


Fig. 5.37

Fig. 5.36 shows the resonances of the four-cornered cabinet. Fig. 5.37 of the triangle cabinet shows no cabinet resonances. The sling at 200Hz in the frequency range and phase indicates the reflexion.

The measurements show that in the back tapered cabinet the sound waves do not run dead. The resonances build up less. Remaining are the reflexions that are even amplified. This is especially damaging for correct sound reproduction, as tones are echoed over a broad frequency range.
The study does not generally speak against the use of triangle cabinet. The speakers were placed at the broad end of the cabinet, conform to the most loudspeaker constructions. At this position the reflexions have the greatest effect on the speaker. That applies also to the four-cornered cabinet. By positioning at three-quarter length, the reflexions do not hit the speaker membrane rear side quite so directly. This is the best position for the speaker. Another build-up to avoid reflexions in the sound quality is the folding of the cabinet. This corresponds to the construction of the Nugget from the „Photo Story „. Both cabinets' designs avoid resonances and reflexions.

5.3 DIGITAL CROSSOVERS

Digital signal processing is already standard in the Audio area. It allows an unlimited processing of audio signals. In digital applications, the analogue signal is converted into a digital signal by an A/D converter. After conversion the digital signal is processed by the, for fast processing developed processor, DSP. The calculations are conforming to that of equalisers, dynamic compressors, sound setters and crossovers. After processing the signal is converted with a D/A converter and can be amplified and reproduced from loudspeakers. The standard book about digital signal processing is:

The Scientist and Engineer’s Guide to Digital Signal Processing

As an example, a look into the frequency and time behaviour of a digital crossover is made. A DSP board is programmed as a high pass. The high pass is configured as an IIR and FIR filter. The properties of an IIR filter are those of an analogue filter. The FIR filters have a linear phase, which can only be reached by analogue filters with great effort.

The frequency range

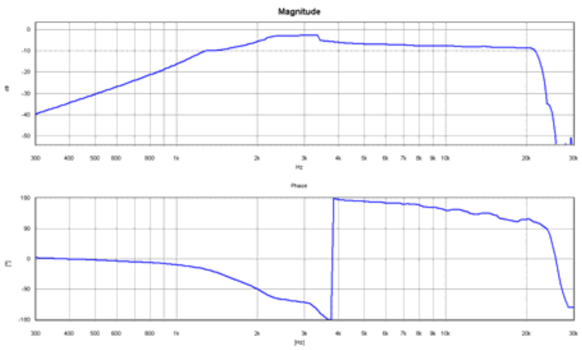


Fig. 5.38

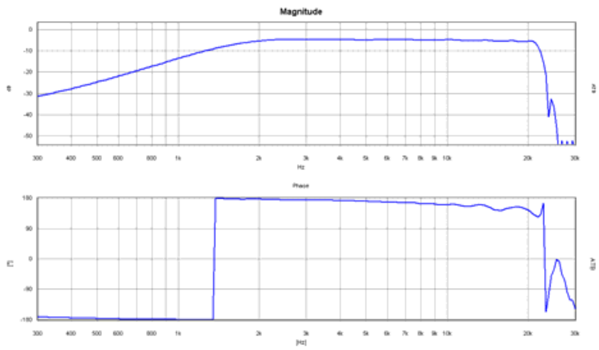


Fig. 3.39

Both pictures show the frequency and phase range of the filters. Both high passes have a crossover frequency of 1.5kHz. The IIR filter Fig. 5.38, shows the typical behaviour of an analogue filter with high grade, a peak of the frequency response at the crossover frequency. The decrease of 12dB/octave and 180^0 turn of phase comply with a filter 2nd Grade. The FIR filter, Fig. 5.39, shows nearly no turn of phase. The phase jump comes from the Illustration in the range between -180 and +180 degrees. As the display ends at -180^0 the curve continues at $+180^0$. The curve is around -180^0, as the controller inverts the signal. The area above 20kHz is also very intresting. The behaviour of the low pass filter of the D/A converter is shown here. The type of filter calculation also seems to have an effect on the low pass filter.

The Sinus-Burst

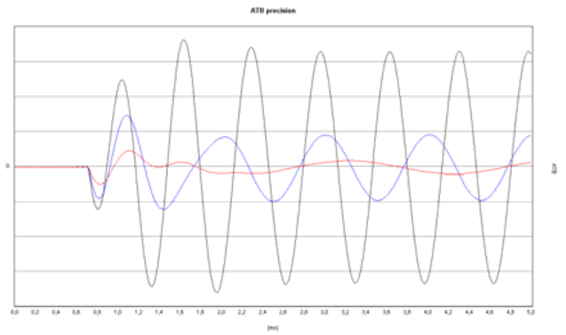


Fig. 5.40

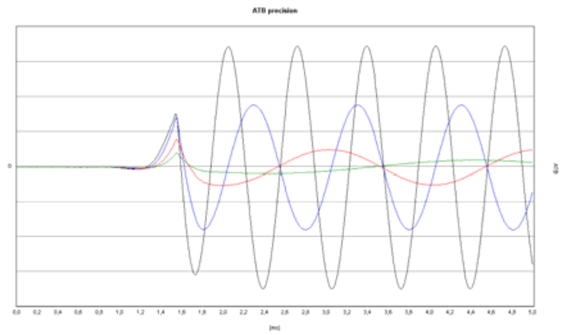


Fig. 5.41

The pictures show the Sinus-Burst with following frequencies:
Black = 1.5kHz, Blue = 1 kHz and Red = 500Hz.
Fig. 5.40 shows the IIR filter with analogue behaviour. All waves start at the same time with negative amplitude, because the circuit inverts.
Fig. 5.41 shows the FIR filter. The picture hardly shows the negative amplitude. The tip of the first wave shows behaviour non-typical of analogue circuits.

The step behaviour

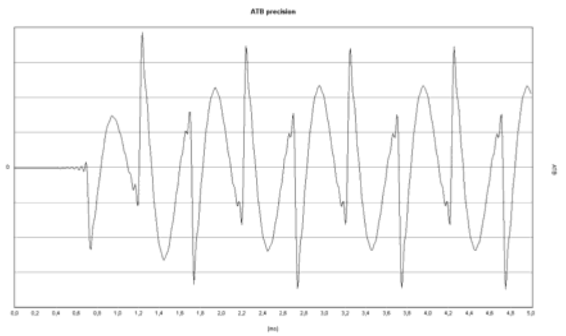


Fig. 5.41

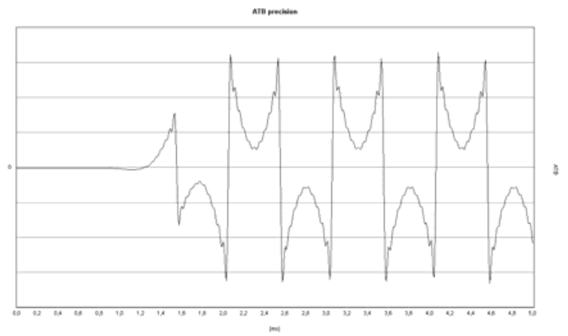


Fig. 5.43

Fig. 5.41 shows the bad step behaviour of the IIR filter due to turn of phase. In Fig. 5.43, the close to step behaviour of the FIR filter can be seen. At high frequencies, the behaviour gets closer to step behaviour. Also by the step signal the negative first wave is not to be seen.

The impulse response

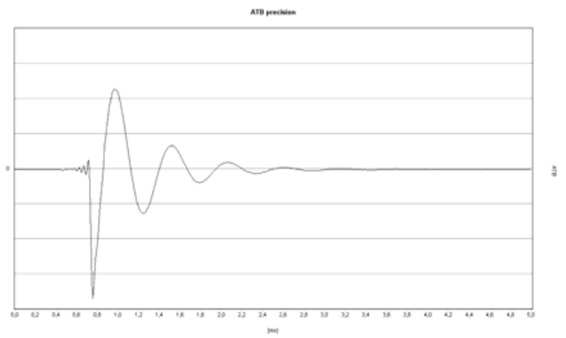


Fig. 5.44

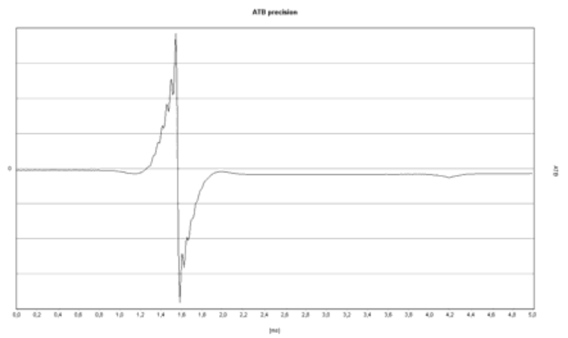


Fig. 5.45

In addition, the Fig. 5.44 of the IIR filter complies with analogue filter behaviour. The high grade is recognisable on the long swing-out.

Fig. 5.45 shows on the one hand the behaviour of an all pass, the jump from positive to negative amplitude, as well as high pass behaviour after the jump. The swing-in behaviour cannot be reached with analogue crossovers. The low frequencies are transmitted before the high frequencies. This means for music reproduction, the ground wave, of for example a guitar string, is to be heard before the actual stroke. As this is unusual for the human to hear, the sound quality can also not be felt as natural.

The Swing-in behaviour in waterfall diagram

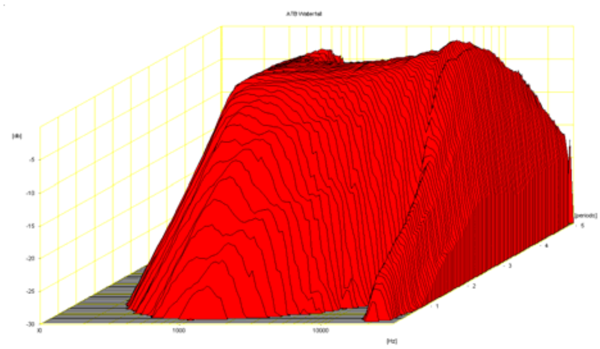


Fig. 5.46

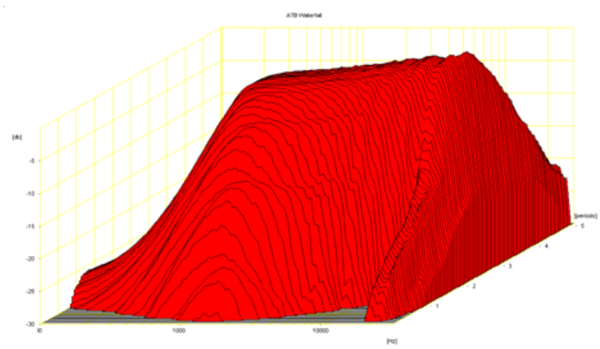


Fig. 5.47

The swing-in behaviour in both filters, Fig. 5.46 and Fig. 5.47, hardly differs. Surprisingly as this was not to be seen in the frequency response, the swing-in behaviour of the D/A converter is here to be seen. The forward running mountain is created by the calculation algorithmic of the filter.

The swing-out behaviour in waterfall diagram

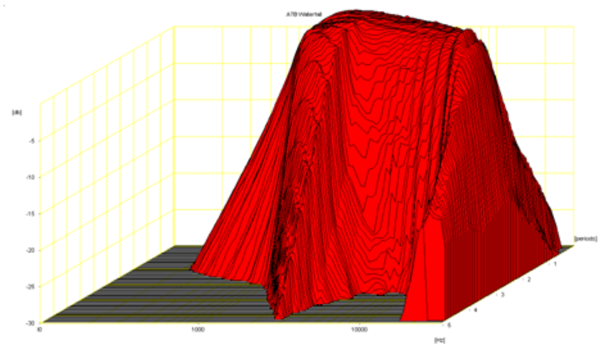


Fig. 5.48

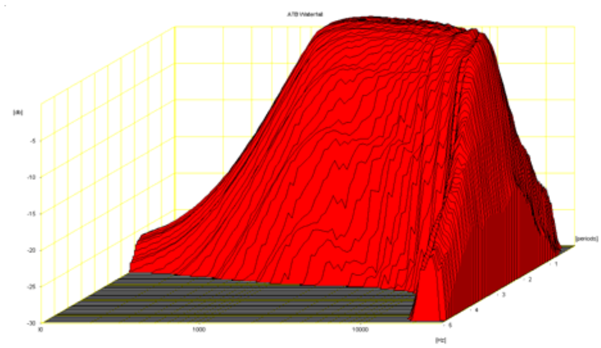


Fig. 5.49

In Fig. 5.48, the high grade of the IIR high pass filter can be seen in the long after waves at 2kHz. This shows that, especially by digital filters, the time behaviour cannot be seen in the frequency response. The developer should at least, if he has not used of the ATB precision waterfall measurement, make a step response measurement. In addition, in the swing-out of the low pass a mountain of the D/A converter can be seen. The forward and afterward running of the signals show the behaviour of a calculated filter.
This behaviour cannot be recognised in the frequency response. As the forward and afterward running signals lie in a non-hearable range, this behaviour is acceptable. However, that these filters can be heard is shown by hearing comparisons and the Dynamic Measurement program.

3D Steppresponse

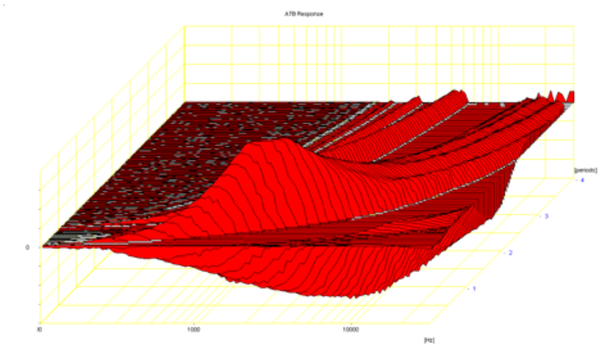


Fig. 5.50

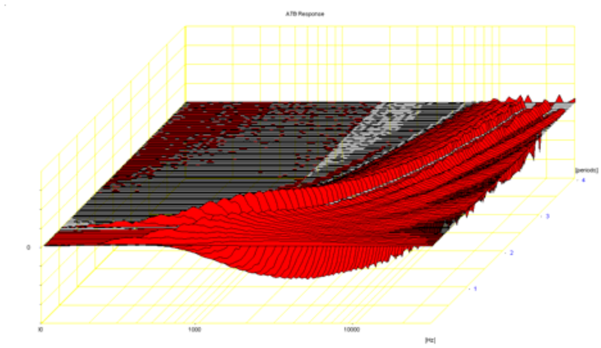


Fig. 5.51

The two pictures Fig. 5.50 and 5.51 of swing-in behaviour from the Dynamic- Measurement, plot show completely different behaviour. As an overlay of low and high pass pictures are too unclear the oscilloscope diagram for one frequency analysis is shown.

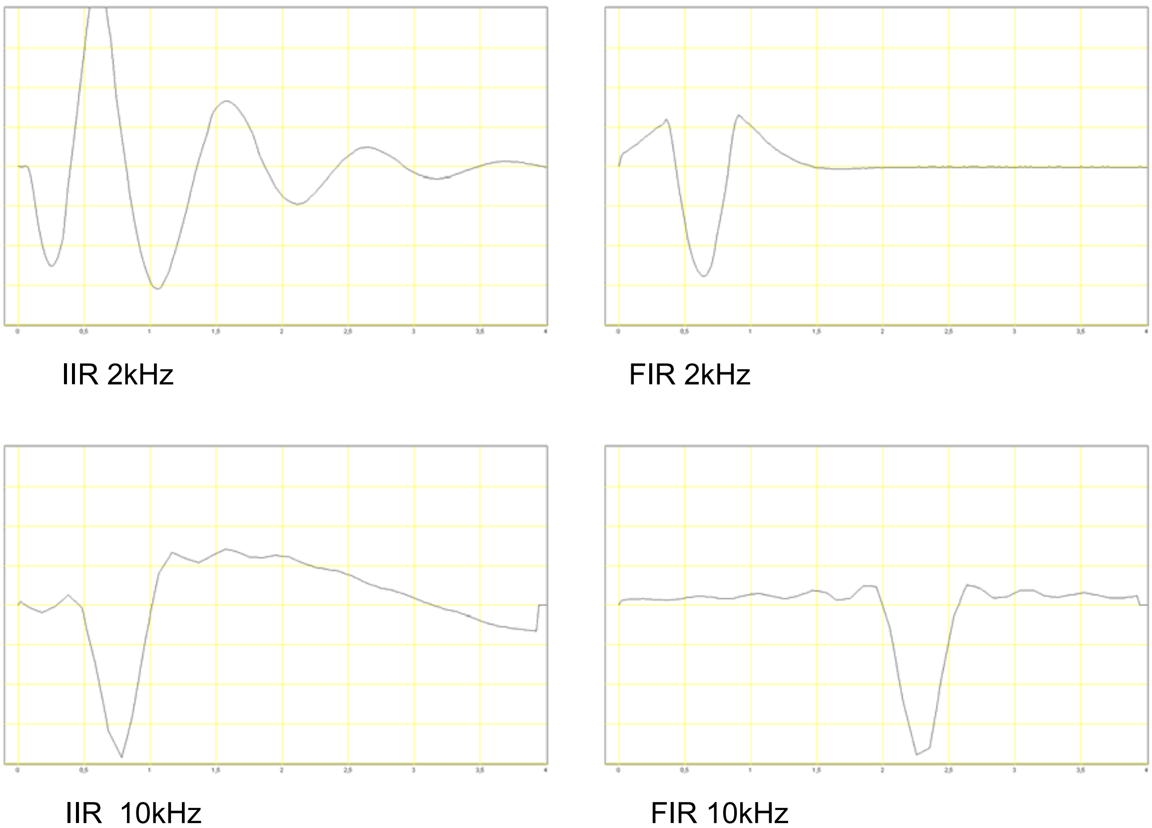
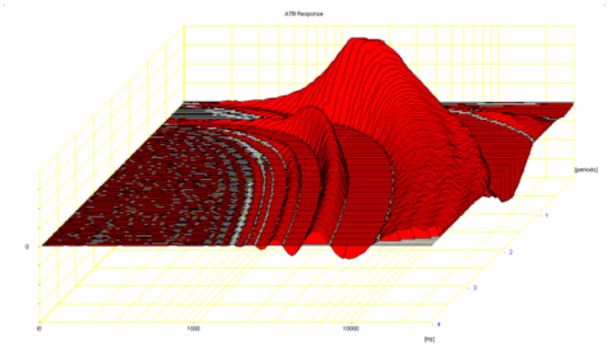
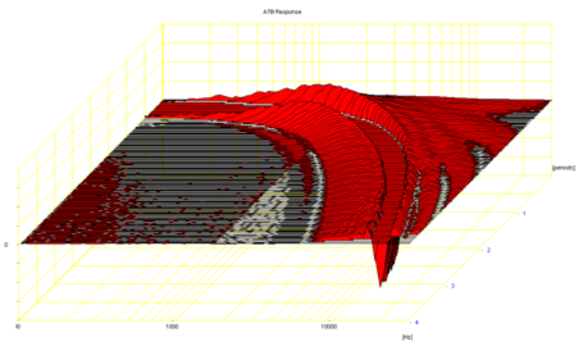


Fig. 5.52

Fig. 5.52 shows the oscilloscope measurement with a sinus half-wave measurement signal. The IIR filter complies with an analogue filter in this measurement as well. The signal starts with negative amplitude and shows with a positive mountain comb the over-swing at the crossover frequency. This is the normal behaviour of a high pass. At 10kHz, the low pass of the D/A converter can be seen. The FIR shows the forward running positive signal. Already at 10kHz both filters can see the low pass due to high frequency alternation. The filter that lies above the hearing range still influences the hearing range. This is noticed by experience with CD players. By some players, the filter characteristic is switchable. Each filter setting has its own sound character. Audio DVD players move the filter, due to the Upsample function, in frequency range to 50kHz. So that its influence is far away from the hearing range. That is why the audio DVD player cannot reach the sound quality points of a CD player by reproduction.



IIR Filter



FIR Filter

The measurements will show that the frequency range is not adequate to judge digital crossovers. First, the oscilloscope measurement shows the behaviour decisive for the sound quality.

W1. THE ATB PRECISION WATERFALL MEASUREMENT

The waterfall diagram is also measured with the ATB PC precision. The ATB and the TEF Analyser show the same results. The measurement procedure of both systems is different. Though by the ATB the Cosinus-Burst is used for measurement. The Cosinus-Burst is won from a band pass filtered sinus sweep. This complies with the TEF measurement. By the ATB precision the band pass filter is not needed, as the filter is already contained in the measurement signal.

THE JOURNEY FROM OSCILLOSCOPE TO BREAKDOWN SPECTRUM

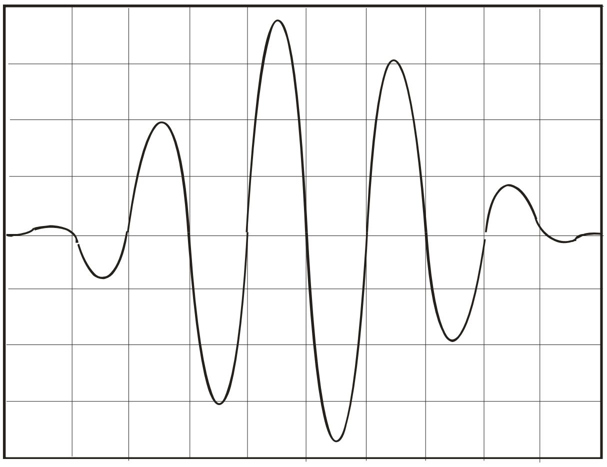


Fig. W1.1

Fig. W1.1 shows the time run of the Cosinus-Burst.

The measurement signal is rectified after measuring the oscillogram with accurate time relation of the generator to measurement signal.

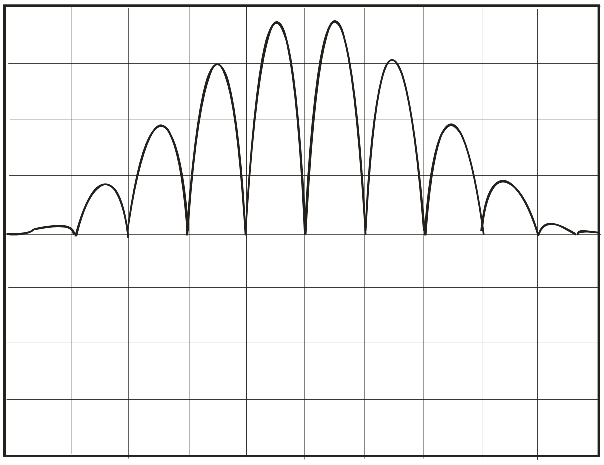


Fig. W1.2

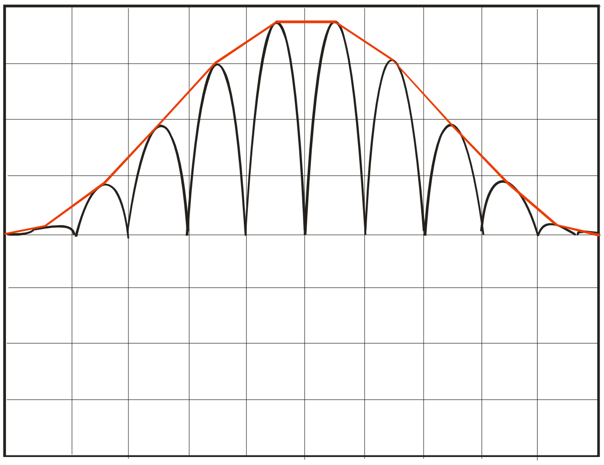


Fig. W1.3

Fig. W1.2 shows the function of the rectifier. Fig. W1.3 shows how the rectified signal is converted to a hull curve with a digital filter. For the display, the amplitude values are converted to a logarithmic value.

The single Hull curves are displayed in the breakdown spectrum. Each Hull curve shows the time behaviour of one frequency.

To display low frequencies with long periods and high frequencies with short periods at the same time, the time axis is normalised and scaled in periods. By normalising the time axis, it is possible to display the whole audio frequency range in one diagram. The time *t* for each frequency *f* is calculated with.

$$t = \frac{1}{f} \times Periode$$

In the waterfall diagram, lines connect equal time points (Period).

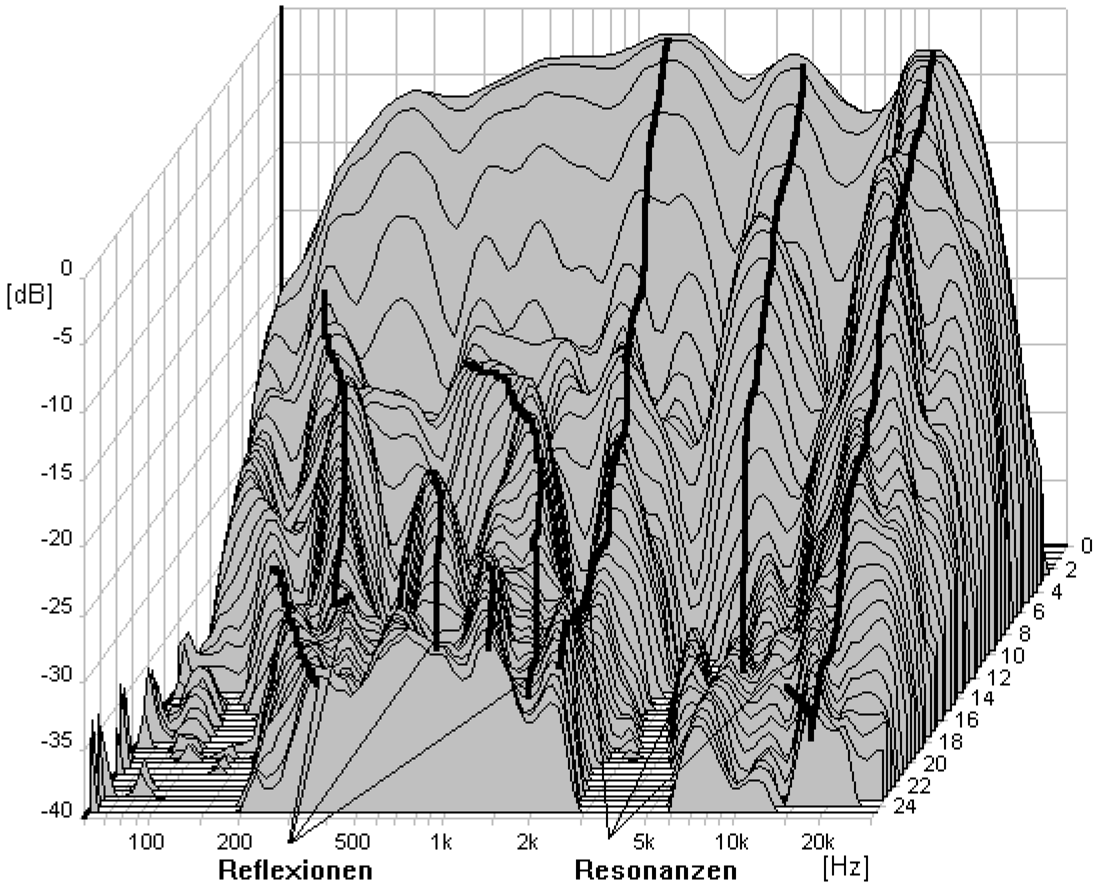


Fig. W1.4

Fig. W1.4 shows the waterfall display. Through scaling the time axis in periods, it is possible to differ between reflexions and resonances. The resonances cause a mountain in direction time (Period) axis. A reflexion is displayed as a (bent) mountain crest that runs to the right. Reflexions have a constant delay in relation to the direct sound. This delay time is built-up in the waterfall diagram with a time axis in a mountain comb parallel to the frequency axis. With the period axis the mountain comb does not run parallel, as the display is frequency independent. By low frequencies, the constant time is displayed by short and by high frequencies by a longer path. So a reflexion is recognised by a from back left to forward right running mountain comb that is bent because of the logarithmic frequency spacing.

Through a change of displaying the time (Period) axis the transmission behaviour, that is covered up by the swing-out mountain, becomes visible. The meaning of this measurement will be shown in the following examination.

W2. BASIC TERMS OF FFT

Time range for FFT calculation

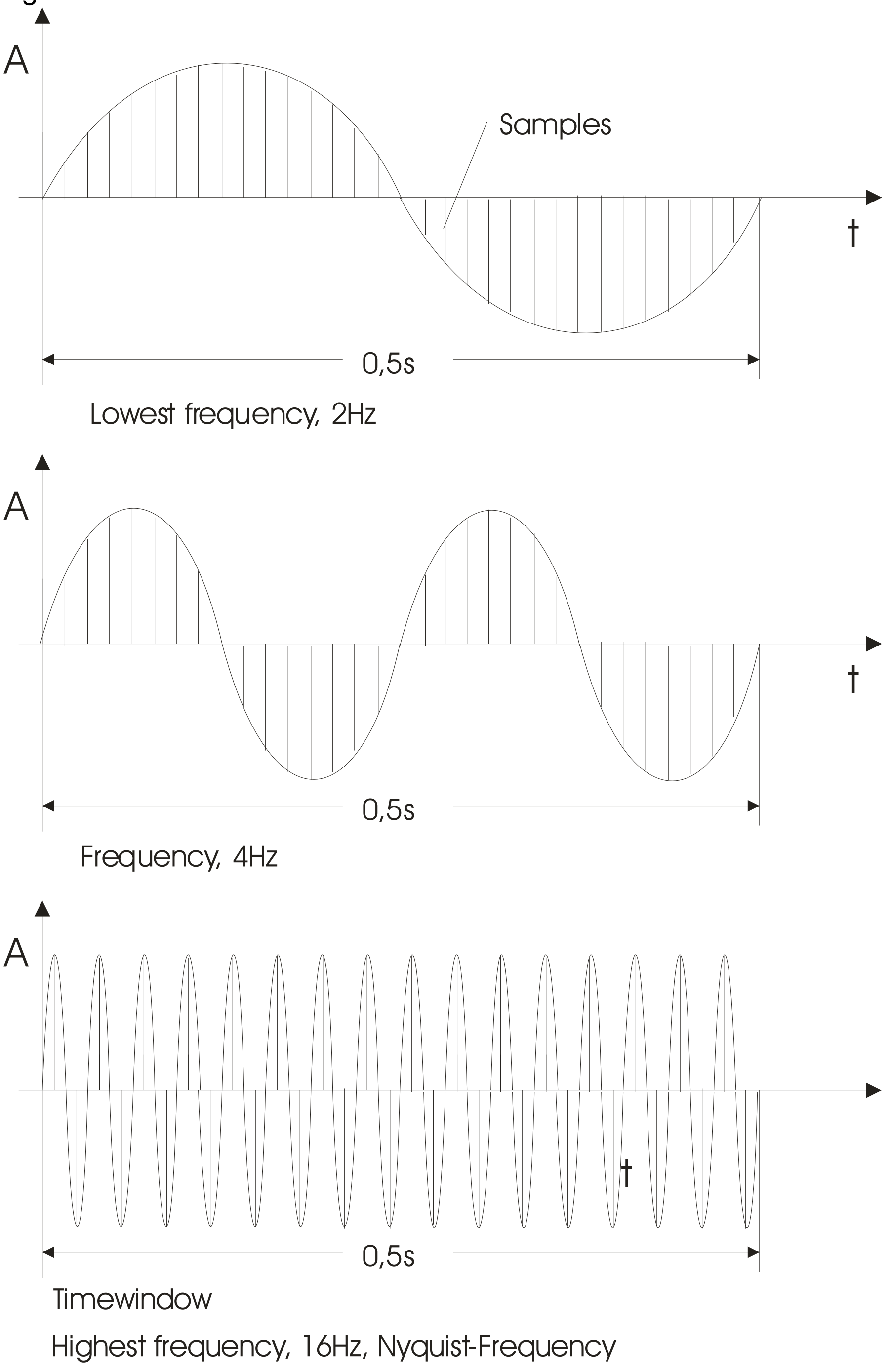


Fig. W2.1

Fig. W2.1 shows the time range of sinus waves with the pickup values used by the FFT.

Frequency range of the FFT

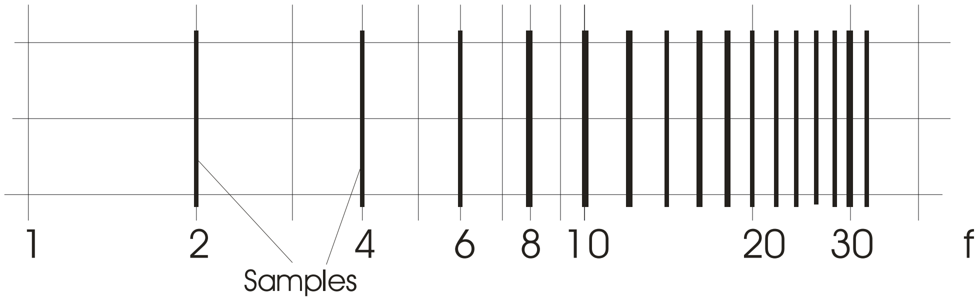


Fig. W2.2

Fig. W2.2 shows the 16 frequencies of an FFT with N = 32

DEFINITIONS:

FFT:
Algorithmic for fast calculation of the Fourier analysis. For this, the length of input data field must have potency of 2 (2, 4, 8………16k, 32k, 64k).

FFT size or block length N:
Number of pickup or data points that build the input field of the Fourier transformation. As mentioned above N is a potency of 2.

FFT-Order:
Size of the potency of 2, that leads to the FFT block length. An FFT 4'th Order results in $N = 2^4 = 16$.

Windowing:
Multiplication of the time data with a certain equation, to let the values of the considered time range run to zero. Through that, the unreal side humps in the spectrum are reduced, although at cost of frequency density.

Pickup frequency or pickup rate:
Rate, with which the time signal is digitalised. Soundcards convert with 48kHz. Making 48000 pickup points per second.

Nyquist-Frequency:
Highest frequency component that is calculated by the FFT. That is half of the pickup frequency.

The pickup values for evaluation:
By the FFT the in the time area converted data values are transformed into the frequency area. For a 32 point FFT 32 data values build the FFT input signal. It is quite common, that not the whole FFT data block is filled with measurement values. The data fields not filled with measurement values are set to zero.

The FFT result:
By the FFT, the pickup values of the time area are transformed into the frequency area.
Simply said: The FFT calculates the frequencies contained in a signal.
The example shows a 32 point FFT, which is picked up with 64Hz. That results in 16 frequency points with 2Hz spacing.

The time density:
The number of pickup values, which build the input of the FFT, determines the time density of the FFT.

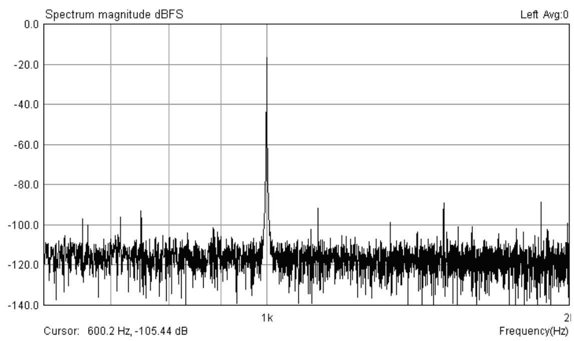
Time density = number of pickup values / pickup frequency

In the 32-point FFT example the time density is: $\frac{32}{64\frac{1}{s}} = 0,5s$

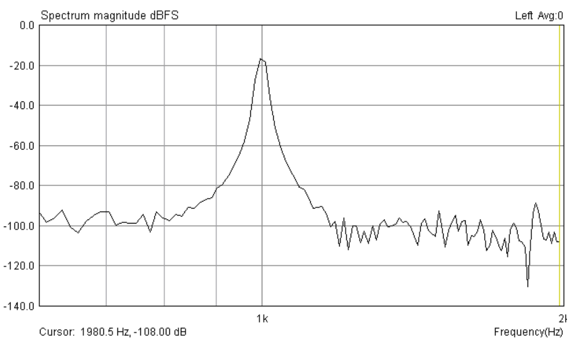
The frequency density:
The frequency density of the FFT is calculated as follows:

Frequency density = Pickup frequency / Number of pickup values

The frequency density is equal to the distance between the single frequency points. This applies only, when firstly all pickup values are evaluated and secondly a step window was used.
By the in the loudspeaker technology commonly used measurement in a time window, the real density is:
1/ time density
When for instance a 1024 point FFT for a time run of 3ms is carried out, then the first 144 points get pickup values (by a 48kHz pickup frequency rate of the soundcard) and the 880 points left over are set to zero. The time density is 3ms (window), the spacing of the frequency points 46.88Hz, the frequency density is though only 333.33Hz (= 1/time density = 1/0.003). The result of this FFT will be 512 frequency points from 0 until 24kHz with a spacing of 46.88Hz. Each frequency point has though a density of 333.33Hz. The point at 1000Hz has then the energy of 8833.33Hz to 111166.66Hz (1000+/- frequency density).



FFT mit n = 131072
Fig. W2.3



FFT mit n = 4098
Fig. W2.4

The Fig. W2.3 and W2.4 show the frequency density of the FFT in relationship to the number of pickup values.

The meaning of the windowing for the frequency density.
Through the windowing, unreal side humps are suppressed. Furthermore a „smearing" of the frequency points is avoided (Energy wanders from the frequency point, to which it belongs, to the neighbouring. This is caused by the supposed periodicy of the Fourier transformation). A windowing is carried out by multiplication of the time data with differently formed window functions.
As the multiplication in the time area complies with a folding in the frequency range, the result consists of the folding of the frequency contents of the input data with the frequency

contents of the window. The result is a broadening of the frequency points or as to say a reduction of the frequency density. The frequency density of the windowed data equals the non-windowed frequency density (1/ time density) multiplied with the bandwidth of the window.
The four-cornered window is in the practice not a window at all. It has a bandwidth of one and has there for no influence on the frequency density. The other windows have a bandwidth >1. The Hanning window for instance 1.5. The real frequency density of Hanning windowed data is there for 1/time density* 1.5.
The reduced frequency density has to be accounted for by loudspeaker measurement. The nice and smooth frequency curves of windowed measurements are due to the reduced frequency density.

W3. THE MULTISINUS MEASUREMENTS OF THE ATB PRECISION

By the acoustical frequency response measurement the multisinus also allows the measurement with time windows. Here is the ATB precision especially accurate. By the ATB precision FFT measurement the frequency range from 18Hz to 20kHz is split into four logarithmic spaced parts. A single measurement is made for each part. Here the time windows are set automatically such that the full frequency density is preserved.

The development of the multisinus KM-C signal:
The KM-C signal is calculated with the inverse DFT. If the phase is set to zero by the inverse DFT, a DIRAC impulse is calculated. The DIRAC is an impulse with an infinite amplitude and time going on zero. This impulse is only theoretical and cannot be used as a measurement signal. To calculate a signal adequate for measurement with the inverse DFT, the single frequencies are given random phase values. The variation of the phase position is not quite carried out at random. Through appropriate algorithmic it is achieved, that the relationship of the maximum amplitude to RMS, the Crest factor, is as small as possible. The amplitudes of the time signals are almost nearly the same size.

The properties of the KM-C signal:
The KM-C signal is an analogue signal for FFT measurement. It contains 512 frequencies with equal amplitude. Because of a maximum frequency range, it has low pass characteristic, so that the boarder frequency of the anti-aliasing filter of an A/D converter can be out of the transmission range. Through the even spread of amplitudes, the measurement path can be optimally regulated.

The FFT Measurement with the KM-C Signal:

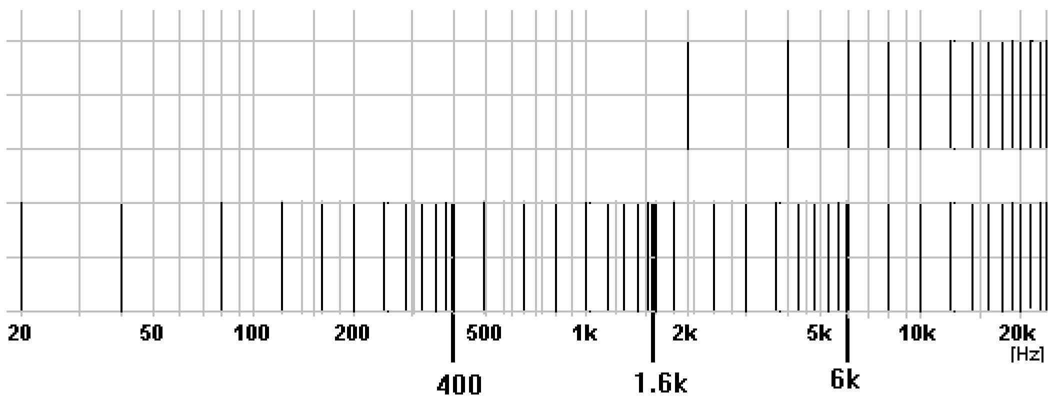


Fig. W3.1

Fig. W3.1 shows the comparison of the normal FFT measurement to the 4-fold FFT measurement. The illustrated FFT has 12 points or support places. Through the 4 fold measurement a nearly even frequency spacing is achieved, so that the measurement accuracy (number of support places) of low and high frequency is almost the same.
The frequency range from 18Hz to 24kHz is divided into four areas and for each area, a measurement with 512 points is carried out. The data is summarized and illustrated in a measurement scribe. The measurement has in the low frequency range a density of 512 x 64 = 32768 FFT points. The same density can be achieved by a FFT. By all FFT measurements, a time window must be set. The start of the time window is determined by the running time of the sound. With the DISTANCE function the distance to the loudspeaker and with that the running time, is measured. The measurement program determines the end of the time window. Through this operator, faults are avoided and correct measurement results guaranteed. From 305Hz onwards the time windows get so short, that by the

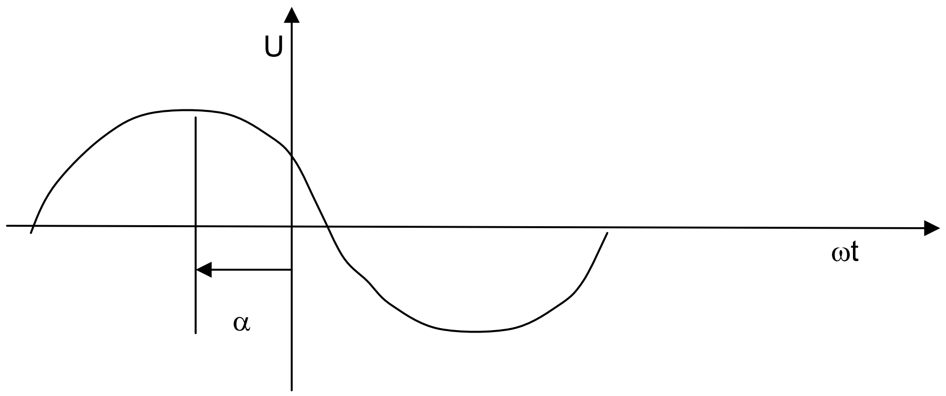
measurement of loudspeakers no room reflexions of measure influence the evaluation, so that the measurement result is room independent.
Below 300Hz, a near field measurement suppresses the influence of the measurement room.

The near field measurement is described in capital 4.2.

W4. THE PHASE

W4.1 Basic Theory

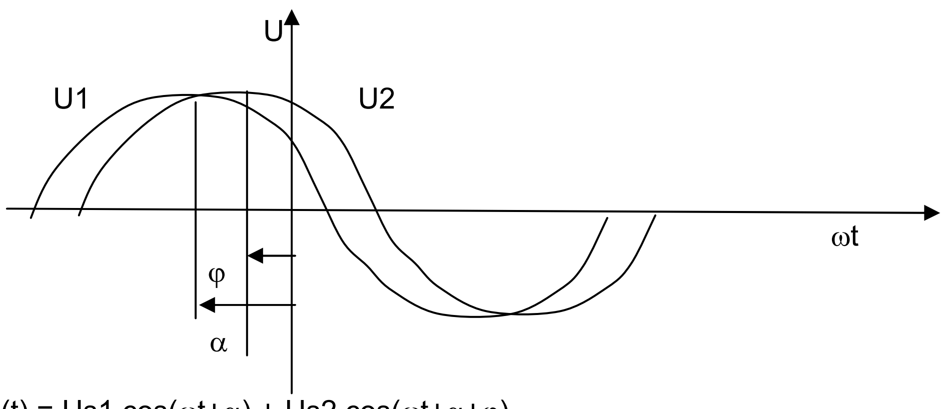
The phase is a point in time when describing sinus formed processes like mechanical oscillations, alternating current, radio waves and sound waves.



$U(t) = Us \cos(\omega t+\alpha)$
t Time, U(t) Momentary voltage value, Us Crest value (max. Amplitude),
f Frequency, T = 1/f Cycle time, $\omega = 2\pi f = 2\pi/T$ Circle frequency,
α Zero phase angle

Fig. W4.1

Fig. W4.1 describes the sinus wave.
The phase angle α already appears in the ground equation. There an orientation point wilfully exchanges it to zero time. By two or more waves, as in acoustical signals, the phase angle is important for the measurement description.



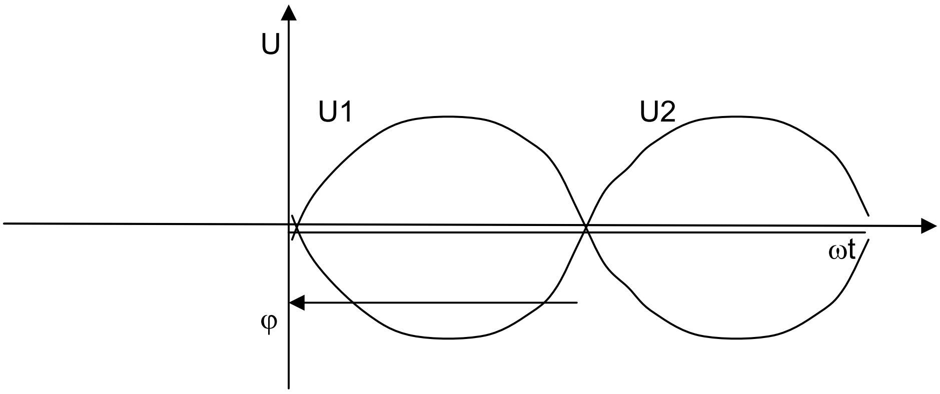
$U(t) = Us1 \cos(\omega t+\alpha) + Us2 \cos(\omega t+\alpha+\varphi)$

Fig. W4.2

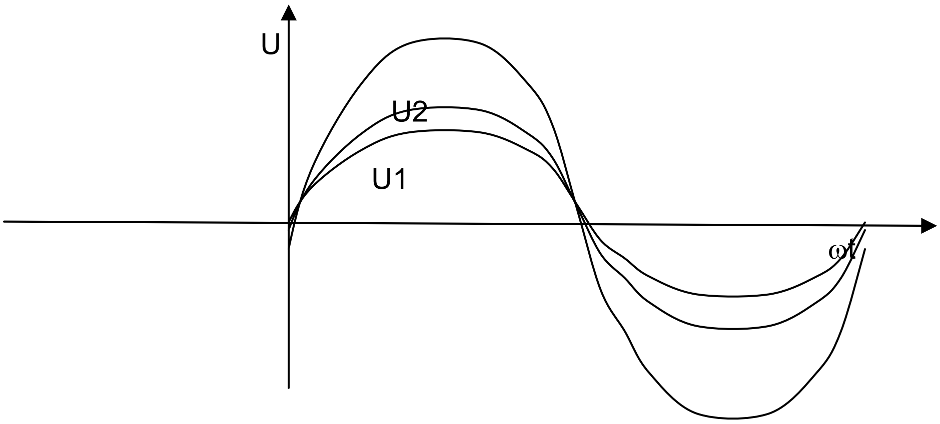
Fig. W4.2 shows that two waves are related to each other over the phase angle φ.
The phase angle φ determines how the signals overlay.

Examples for how the phase φ affects the wave.

$\varphi = 180^\circ$ => Deletion



φ = 0°, 360° => maximal Amplification



φ = 90°, 270° => semi Amplification or semi deletion

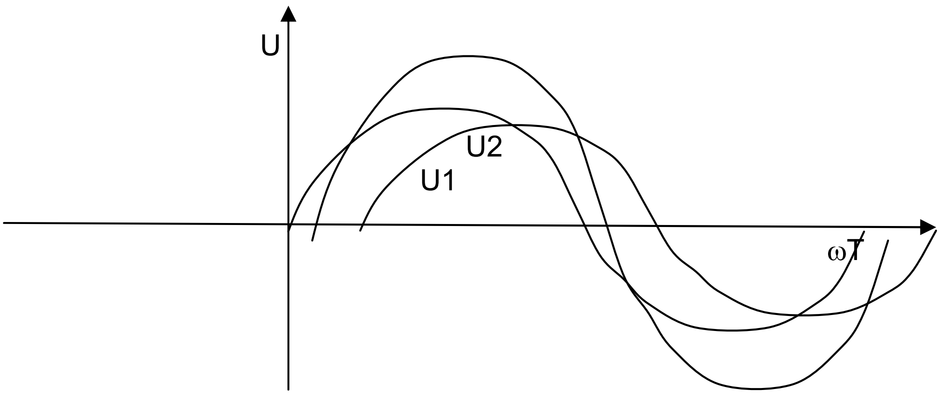


Fig. W4.3

The illustration in Fig. W4.3 shows the interference of waves.

W4.2 THE PHASE BY LOUDSPEAKERS

By loudspeakers the ground rules of physics also apply. To the correct description of transmitted sound belong the amplitude and the phase. The common theory, that the phase can be recognised in the frequency sweep, is false. Simple electrical circuits only give this. By loudspeakers the complex frequency response, that also has running time elements, inhibits the recognition of the phase in the amplitude frequency range sweep. A box with an absolutely flat frequency sweep can still have extreme and as such hearable phase jumps. The common practise for recognition of phase relationships by loudspeakers is following: To recognise the phase between loudspeakers, for instance a middle toner and tweeter, the crossover is made such that maximal deletion at the crossover frequency is achieved. When after one of the speakers has been repoled the frequency sweep is even, then the phase position is correct. However, here the phase lays 180°, 360° or 540° apart and the swing-in behaviour is bad.

Outside the crossover frequency, this method does not lead to success wich the following example will demonstrate.

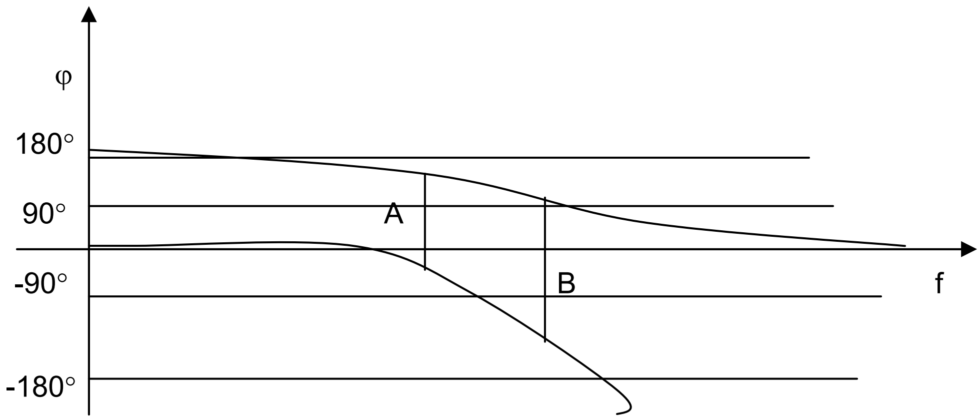


Fig. W4.4

Fig. W4.4 shows the phase behaviour of low and high pass.

A = In compliance with the above described method found phase of 180^0.
B = Phase angle of 270^0. At this angle, the sound components of middle high toner partly delete each other. A repoling of the speakers shows the same frequency sweep, so that the fault by normal crossover development is not found. As by a phase angle of 90° or 270° the single speakers are too loud a nervy sound quality is the result. In this case, the normal frequency response measurement has no saying.

All investigations show that an even phase is imperative for a natural sound reproduction quality.

W4.3 THE ILLUSTRATION OF THE PHASE

The graphic display of the phase in a plot takes place in an angle range of -180^0 to 180^0. This range is not adequate for the turn of phase. A larger range for the angle worsens the illustration, because of missing density of the Y-axis.

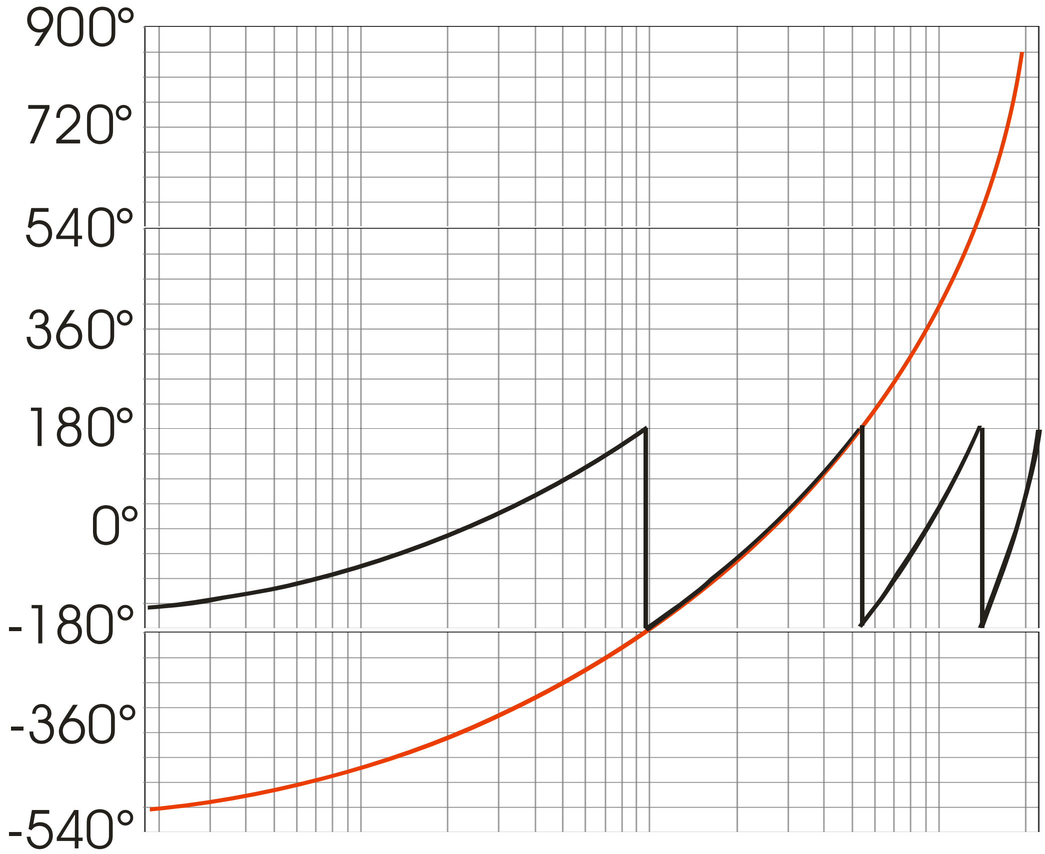


Fig. W4.5

Fig. W5.4 shows the real phase (red curve) and the displayed phase (black curve). The jumps in the illustrated curve are not phase jumps, but come from the display. This causes especially by acoustical phase measurements faulty interpretations, when the phase angle lies in range of 180^0 to -180^0. Then the curve jumps between both values. The blue curve shows a reverse poled loudspeaker.

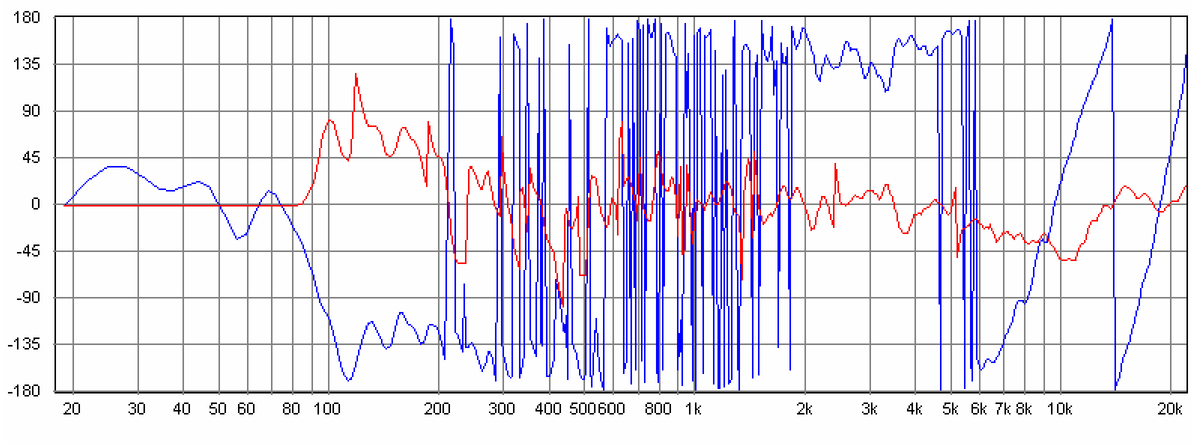


Fig. W4.6

W4.4 THE ELECTRICAL PHASE MEASUREMENT

By the electrical phase measurement, the phase of for instance loudspeaker impedance is measured. This is necessary for the malfunction free operation of the loudspeaker. Large phase angles show a capacitive or inductive behaviour of the loudspeaker impedance. The result is an overstrain of the amplifier. As well as that, the amplifier can become instable and cause oscillation.
After the system correction is available, the loudspeaker is connected to the poles of the test box. With the IMPEDANCE switching button the measurement is switched from frequency range measurement to impedance measurement and the phase measurement activated. By the phase measurement „electric“ is chosen.

By the phase measurement, the quality of the soundcard and CD/DVD player becomes known. For frequencies > 5kHz the measurement can be faulty. This shows on the too negative angle falling curve.
When measuring the loudspeaker impedance there should be no noises present. The noise could falsify the measurement, as the loudspeaker works as a microphone during the impedance measurement.

W4.5 ACCUSTICAL PHASE MEASUREMENT

By the acoustical phase measurement, the running time of the sound from loudspeaker to microphone has to be considered. The ATB PC pro automatically compensates the running time by calculation of the zero phase.
By the measurement systems that use MLS, the running time is found automatically or setup by the operator. This is done by setting up the time window. The start of the window determines the running time of the sound.

The start of the time window determines the running time of the sound.

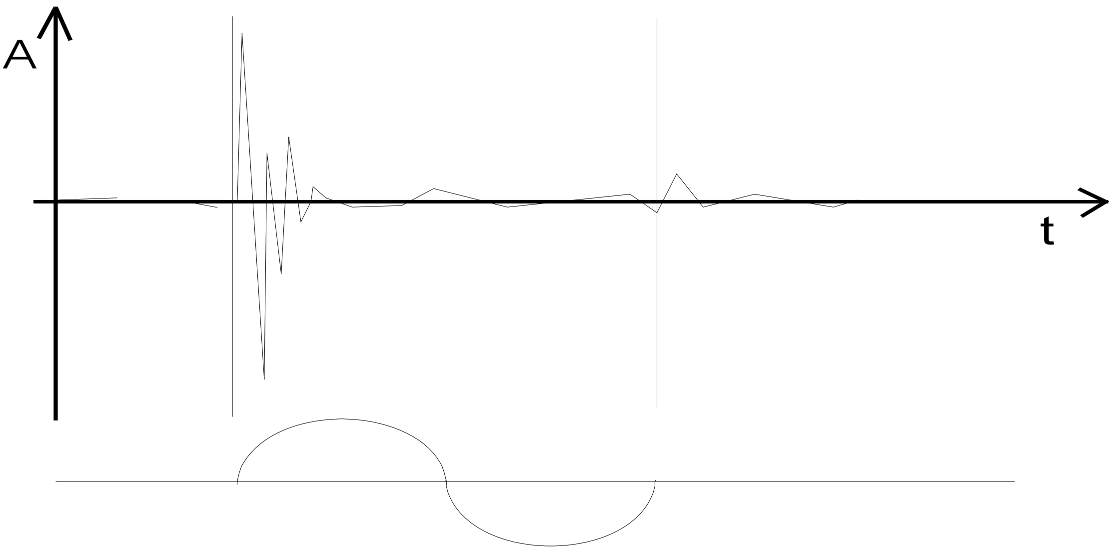
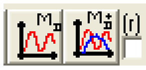


Fig. W4.7

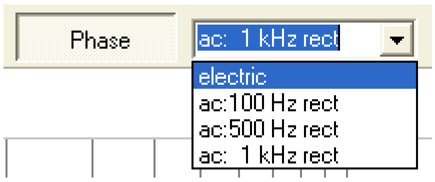
Fig. W4.7 shows the measurement for setting the time window. The impulse was calculated from the MLS signal. The time from the start of the time axis to the start of the impulse is the running time of the sound. The first part of the impulse is built-up from the highest frequency of the measured sound. The highest frequencies are often reproduced with delay, for instance from a single loudspeaker, that has a low pass function. By loudspeaker boxes with multiple loudspeakers, the tweeter has a forward running character; its signal comes too fast. The exception is the correct time loudspeakers. By all other loudspeakers by setting the time window either the delayed or forward running highest frequencies are used as relation point for the zero phase. That is why the display of acoustical phase is false.

Carrying out the Measurement

The acoustic phase measurement of a loudspeaker can only be carried out in a quiet room. It is not so sensitive to external voltage as the frequency response measurement. Strong reflexions in the measurement room also falsify the measurement results. In small rooms the distance should not be >0.5m.
The measurement should be carried out with a permanent measurement signal.



As the automatically phase calculation is highly complicated, false displays happen.
Next to the phase button, there are setup options to support the phase measurement. One of three ranges is chosen during permanent measurement:



The correct range is chosen when the phase curves stops changing. By the measurement false display can occur. This is recognised immediately by the copped up curves. If the curve is even at the high amplitude area, then a correct measurement can assumed. By the separate measurement of loudspeakers, only the run of phase can be compared. Phase differences of 180^0 cannot always be correctly found by the program. By measuring the loudspeakers at the same time the location is again correct.
As the distance measurement is not necessary with the ATB PC, all measurement technicians come to the same results.

W5. DYNAMIC MEASUREMENT

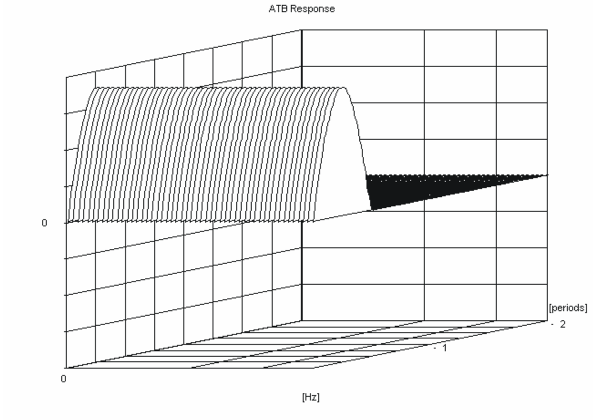
Theory of the 3D step response measurement

The sound quality of the loudspeaker made visible.

Loudspeaker developers, testers and consumers have more and more difficulties judging the sound quality of loudspeakers. The high value loudspeakers have an even frequency response, but are completely different in their sound qualities. This shows that the frequency response is totally inadequate for the judgement of sound quality. The frequency response is measured with non-changing (Static) signals such as sinus waves or calculated noise signals. Against that, a music signal consists of changing (Dynamic) signals. The study of music signals shows that the sound quality of a musical instrument is determined by impulses. Especially the first impulse, the stroke of a guitar string, the impact of a piano cord, the hit on a drum and the blow of an organ pipe or brass instrument is important for the sound. As the loudspeaker transmits these signals, they have to be tested with dynamic signals, step signals. The result is the Step-Response. In the Step-Response, all acoustic parameter information of the loudspeaker is contained. This consists of the frequency response, phase response and swing-in behaviour. There for the display is so complex, that the sound quality judgement based on the measurement is not possible. That is why the 3D Step-Response measurement was developed. By the measurement, the Step-Response is analysed and displayed in a 3D graph. The 3D measurement shows the impulse reproduction for each single frequency with use of an additional frequency axis. The measurement makes it for the first time possible to analyse the sound quality through measurement technology and is a non-dispersible help for developers, testers and consumers.

The measurement signal of the 3D Step-Response measurement.

The measurement signal of the 3D Step-response measurement is a sinus half wave. When observing music signals on the oscilloscope, the for sound quality most important first impulse, appears a signal complimentary to that of a sinus half wave. The impulse is according to the tone height of the played tone, narrower or wider. Therefore, the measurement is carried out with sinus half waves of different frequencies. This measurement is at the same time the analysis of the jump response.
The interpretation of the 3D Step-response is simple, by comparing the measured signal with the measurement signal. All differences to the measurement signal are faults made by the loudspeaker. By electrical appliances such as amplifiers, the measurement shows hardly any difference between input and output signals.

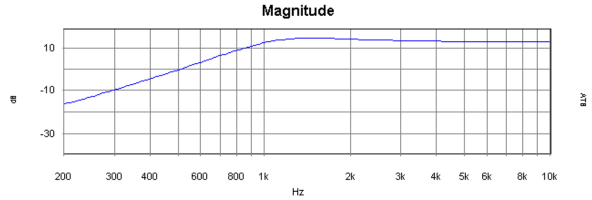


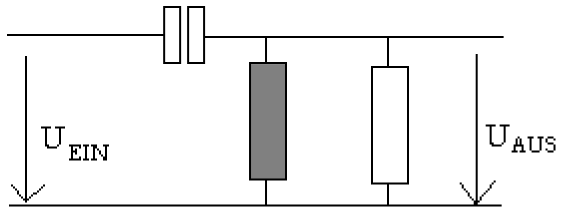
The graphic shows the measurement signal for the 3D Step-Response measurement. The y-axis is the amplitude as an absolute display with +/- so over and under pressure of the loudspeaker signal. The x-axis is the frequency. The z-axis is the time axis normalised in periods. So can low and high frequencies with their different period lengths be displayed at the same time. Apart from that, the acoustical phase is visible.
Fig. W5.1

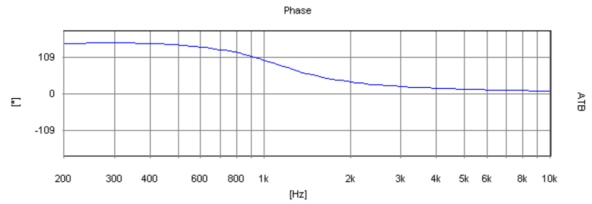
Examples for electrical 3D Step-Response measurement

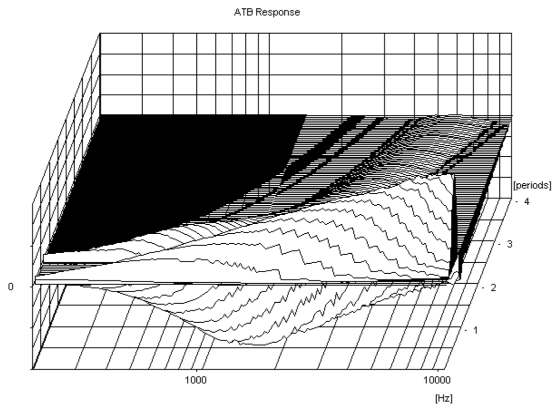
Three measurements are to be made on ground circuits.

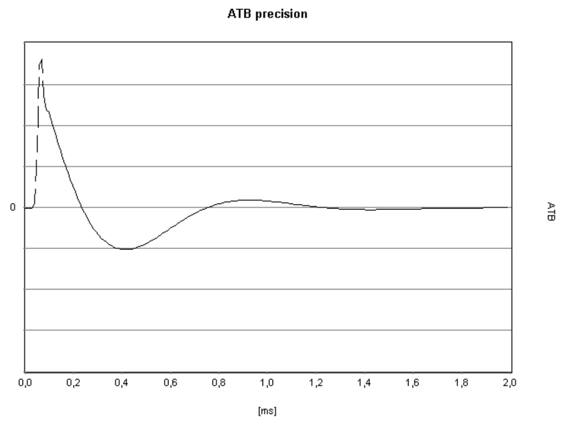
12 dB / Oktave Highpass



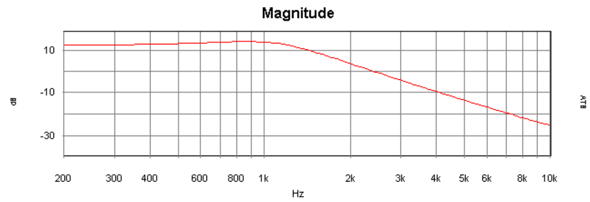


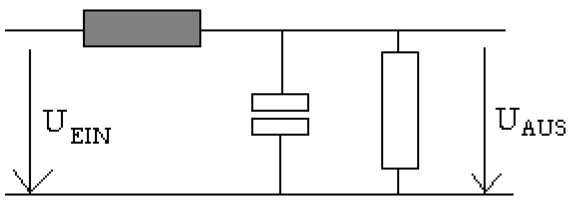


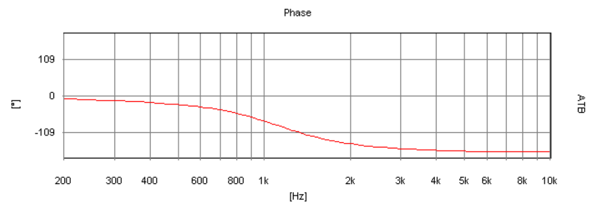


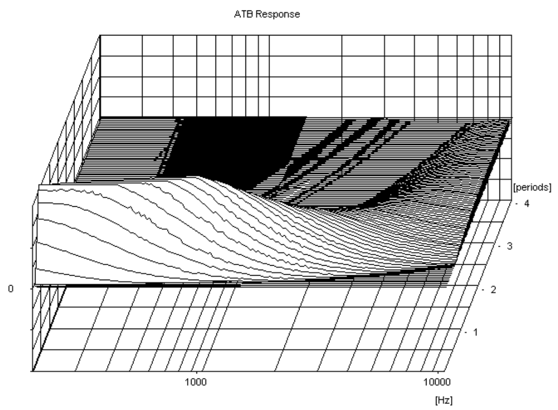


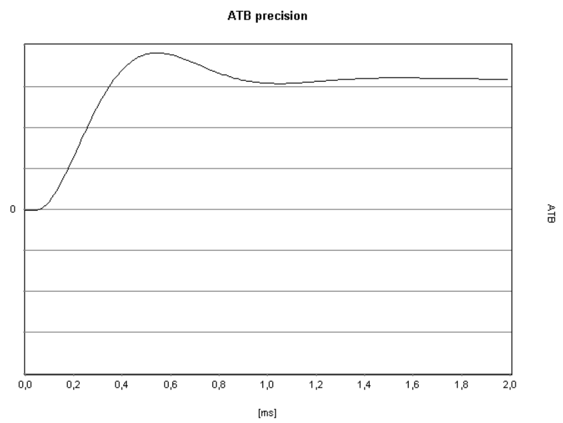
12 dB / Oktave Lowpass



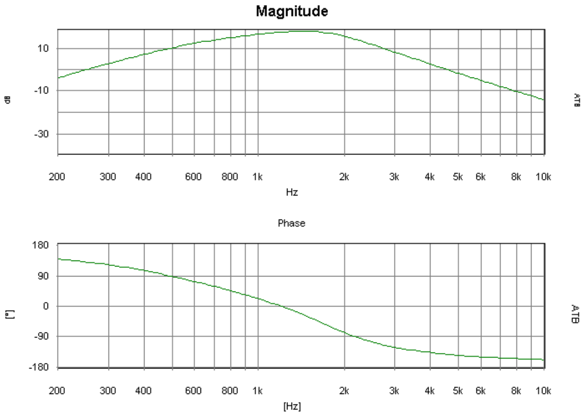


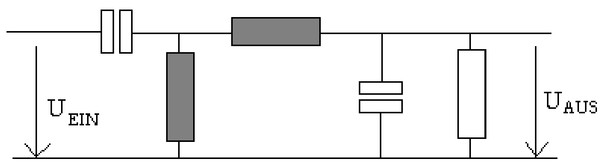


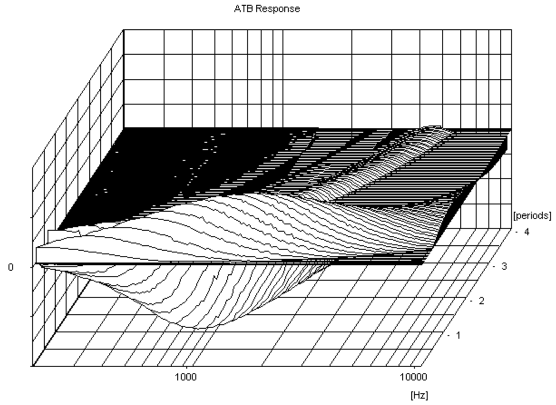


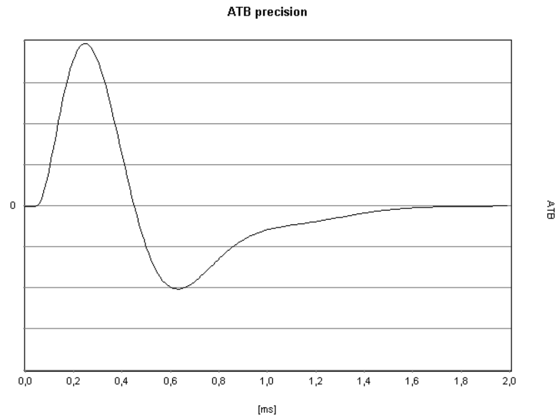


12dB / Oktave Bandpass









Allpass