

audioTester V2.2

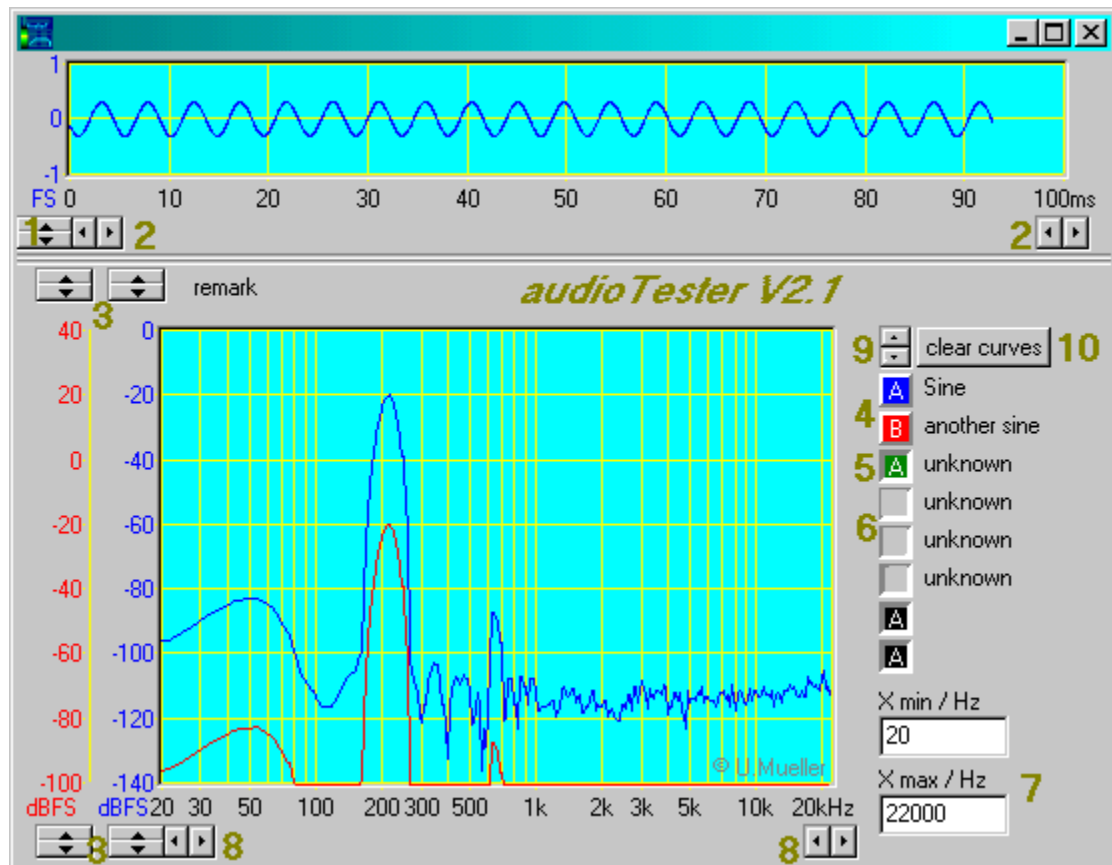
(c) Ulrich Müller

1 audioTester V2.2

2 Allgemeines

3 General

The 2D diagram



1. Adjust of the Y-axis (symmetrical) for the time diagram
2. Adjust of the X-axis values for the time diagram /ms
3. Adjust of the Y-axis values, you see the Y-axis groups A+B separately (see [curve dialog](#))
4. Curve buttons, you see two selected curves, that means, curves receiving measurement values and they are visible.

The characters in the buttons sets the group of the Y-axis scroller. If you change the scroller of, for example, group A, then change all of the curve, which are part of group A. Example [see below](#)

5. Is the button lowered, then is the curve visible and receives no measurement values, it is not selected.

With the [curve dialog](#) it is possible to select and show/hide the curves.

The [curve dialog](#) is reachable, if click on the curve name beside the button.

The curves (buttons) are movable, including its properties, with *Drag'n Drop* to another curve (button).

Example [see below](#)

6. Is the button greyed, so the curve is invisible and deselected
7. Adjust of the X-axis values via numerical input (only frequency diagram)
8. Adjust of the X-axis values via scroller /Hz

Moving and Addition of curves

Via drag'n drop it is possible to move a curve from one of the 8 places to one of the eight others.

Therefore you click on a curve button and drag it to an other button and drop it. All the properties are copy except the curve color.

If you hold the Ctrl Key while dropping the curve, they will be added. The addition is calculated with the phase, which is stored within the curve properties.

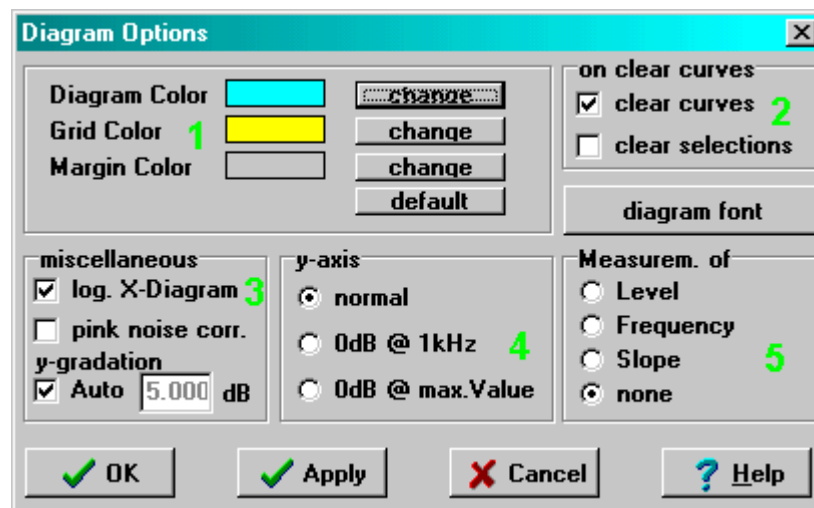
Example for selection of curves



In the figure above, you see the curves of the channels 1-4 visible, the channel named *unknown* is invisible. The channels 2+3 are selected, they are ready to show the actual values. Channel 1+4 belongs to the Y-axis group A and channel 2+3 to group B.

Diagram Dialog Box

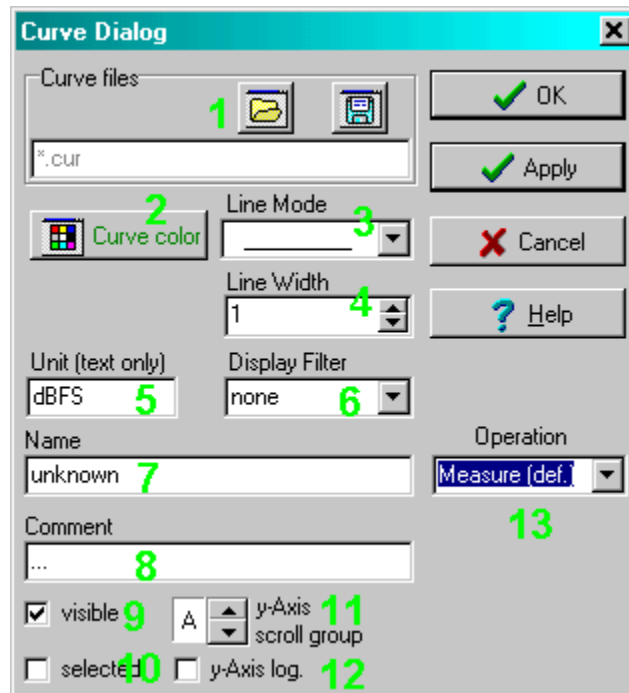
You enter this dialog box with a right mouse click into the diagram.



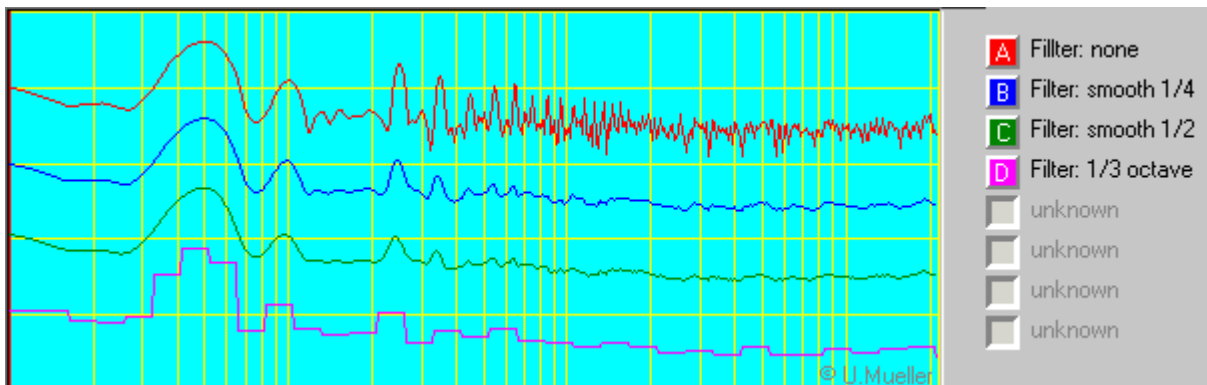
1. With a click on the buttons you start the color dialog for the diagram margin, the grid and the diagram surface.
2. If you click the *clear curves* button in the diagram, you can clear all curves and/or reset all selections, that means the first two curves are now selected and visible.
3. You switch between a linear and a log. X-axis. Apply a pink noise filter to linearize a curve measured with pink noise signal. Switch on/off auto gradation.
4. The scaling of the curves you can raise to 0dB. This is selectable for 0dB at 1kHz and 0dB at max. value.
The first curve of an Y-axis group sets the Y-Offset value. The values are set for each Y-axis group individual.
5. A click with the left mouse button you are able to measure levels, frequency and the slope of the curves.

Curve Dialog

You can set some properties of the corresponding curve.



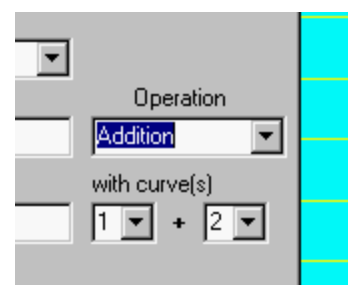
1. With the file open dialog you can load a curve. The *disk* symbol saves the actual curve.
2. Dialog for the curve color.
3. Selection of the *line mode*, solid, dashed, dotted etc.
4. Selection of the line width (1 - 20 pixel)
5. In the edit field *Unit* you set the unit text of the Y-axis.
It will be overwritten by the following measuring process and works for the document generation only.
6. In the dropbox *Display Filter* you select the curve filter. You can select between a none , a smooth 1/4 , a smooth 1/2 and a terce (1/3 octave) filter. The filter are only Display Filter. The filter have no influence to numerical measuring values.



7. The text field *Name* names the curve. The name is displayed in the diagram on the right side of the button.
8. In the text field *Comment* you can enter a comment.
9. A curve can be visible or invisible. An invisible curve is also deselected.
10. A curve can be selected or deselected. Selected means, the curve is ready to receive measurement values. At the same time it is possible for max. 2 curve to receive measurement values. If there are more than 2 curves selected, only the upper both of them are receiving values.
11. A curve belongs to one of 8 Y-axis groups (A-I). Each group has its own Y-axis, which is shown at the left margin of the diagram. The color of the scale is the color of the first curve of this group.
12. Set the y-axis to a logarithmic scale, to measure voltages or resistances in a wide range
13. Measure, default selection, curve take the measure values. Also possible curve show the peak value of another curve or the sum or difference of two other curves. It is calculate a complex operation, with the phase value. See picture below



curve x show the peak level of curve 1



curve x shows the sum of curve 1 and 2

Presets

With the menu point *Presets* you can load and store presets.

The presets have the extension *.atp (**a**udio**T**ester **P**resets)

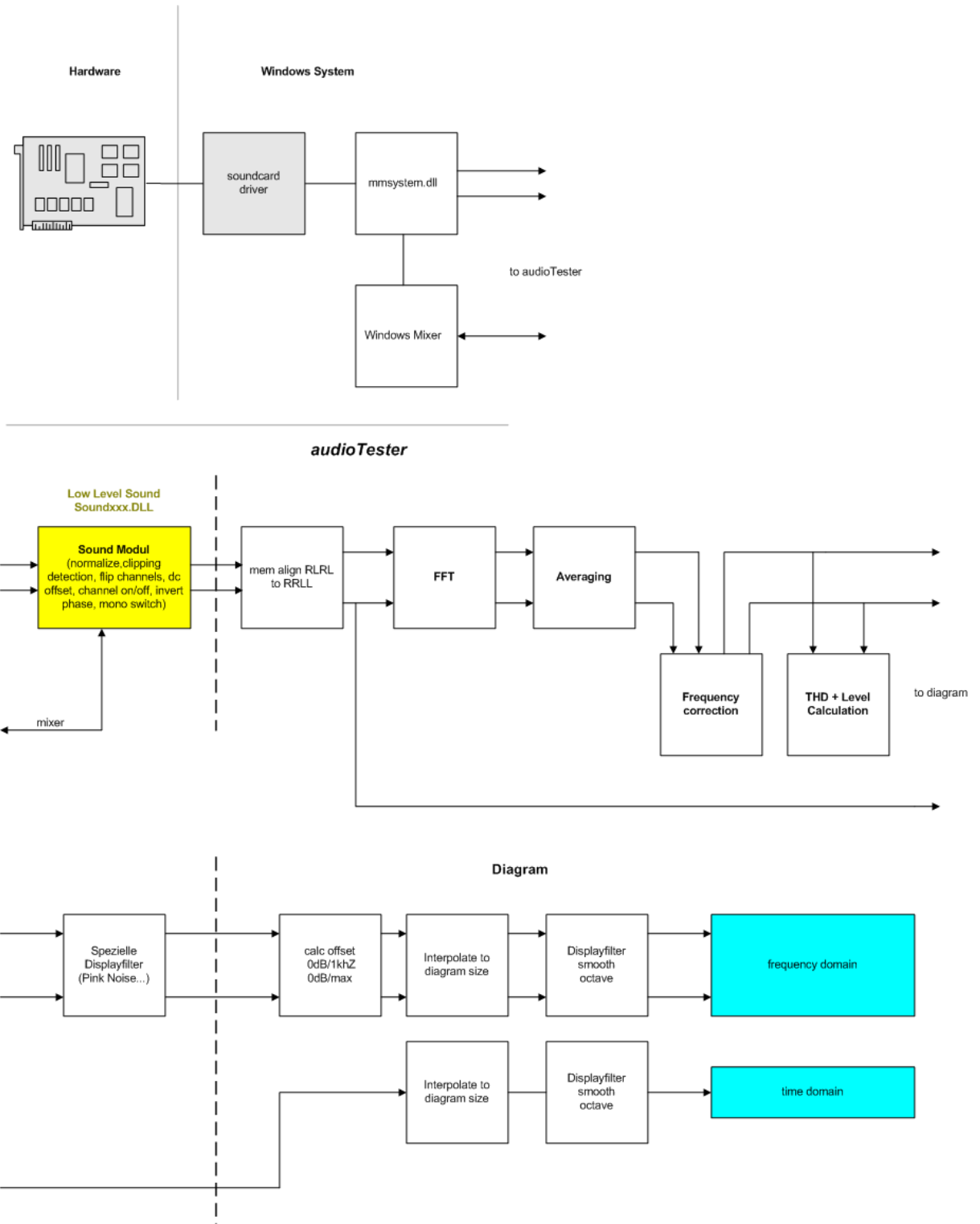
Save: Select *Presets/Save Preset* and store the actual settings with a file name of your choice.

Load: Select *Presets/Load Preset* and load a stored preset.

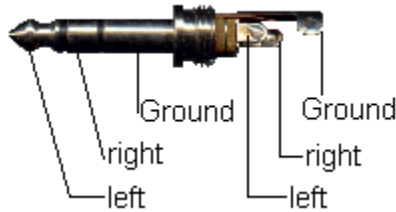
Predefined presets:

std2Dfft.atp :	Standard 2D spectrums analyse with 1kHz sine
IM60_7k.atp:	Measurement of intermodulation distortions with a interfere signal of 60Hz/-3dB and a main tone of 7kHz/-15db @ IEC 268 Part 3
MLS14RefFFT16k.atp:	Frequency measurement with a MSL 14 order and a 16k FFT
AsyncWob50s.atp:	Asynchronous sweep with a duration of 50 sec

Function diagram audiotester software

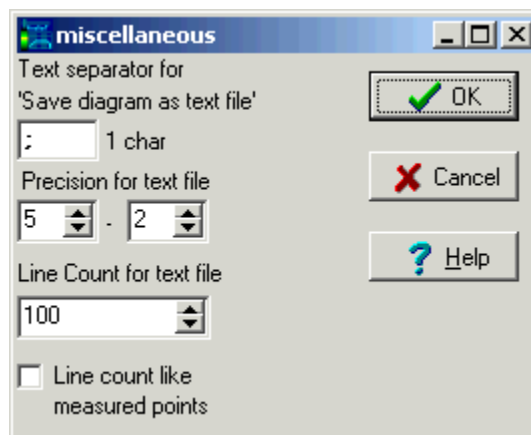


Phone jack



Miscellaneous Dialog

In this dialog are several settings, in the main thing *saving as text file* properties



If saving a diagram into a text file, the frequency and the levels are separated with a text separator, normally this is a comma. You can change it here.

The output format of the frequency and the level can be justified with the width and the precision declaration.

The first digit (in the example 5) is the width, its includes the sign, the decimal point and the precision. The second digit (in the example 2) is the precision, the digits after the decimal point.

The **line count** is the count of lines in the text file. Are there more or less measurement values as line count, so the frequency and the level are interpolated the line count values. Only visible values in the diagram are written. That means, if you decrease the diagram bounds, only the visible values are written. If at the lower and the upper diagram bounds are no measurement values (possible at sweep measurement) so the new boundaries are the valid upper and lower values.

If you select **Line count like measured points** the text file is filled with the count of the measured points (e.g. 2048 lines at a FFT with 4048 Points)

There is no interpolation necessary by the program. If at the lower and the upper diagram bounds are no measurement values (possible at sweep measurement) so the new boundaries are the valid upper and lower values.

Troubleshooting

Program breaks !

Please send me a screen shot of the error message. I will try to fix the error, please tell me also the version and build number of your audioTester. Most errors, except programming errors of me, are old sound card drivers. In this case please try to update the driver via internet.

Sound output, Sound input interrupted, hacked !

Please see that there is no Interrupt-Sharing between any PCI-Cards is.
Check this with the windows control panel.
Change this with the BIOS Setup.

Program shows not all details, and works in a strange manner !

Here goes something wrong, please delete the file aud20d.ini in c:\program files\audioTester\audioT

Calibration doesn't work !

Over is enabled, if the level reached the last bit, at 16Bit resolution that is 0x7FFF and 0xFFFF. Some sound driver doesn't reach this values and stop before. Please don't increase the level in this case, read out the value and calibrate it.

Problems with the registration !

Like any other software authors I must protect me for illegal using of my program too:

Therefore every customer get a key-file: The key-file has 256 Bytes and contains the name of the customer and some cryptic data, no program code or virus.

The key-file, it comes via Email normally and named *key.xuz*, you must copy into the directory:

x:\program files\audioTester\common . x means the drive where your program directory is, normally c:

In the headline is *SharewareVersion ! displayed* ?

1. If the audioTester is running while copying the key file, then close it and start it again
2. You use the wrong directory, please try it again.

Harddisk damaged, Key-File was gone ?

No Problem, send a mail - new key-file is coming.

Program shows 'cracked version' ?

Either the key-file is not made by me (then do not mail me),
but if You are a real customer, please send me a mail.

What is the license-key ?

A huge part of my program is selling with the help of the shareware-provider **www.shareit.de** . After You have buy a license you immediately get (since august 2003) a license-key in form of a string .

This string (i.e.

D359C5CDBDCF577264B05DC8FDD88424545AB6653C379BF661614757FAFD24BF069363B2AE1823D90)

copy (while audioTester is running) via *Copy and Paste* into a dialog box. This dialog box You will find in the menu help\Enter temp....

Your **audioTester** runs now 30 Days without any shareware breaks.

If I get your order from www.shareit.de, I will send You the personal key-file (key.xuz)

Further proceeding with the key-file, see above.

The time between purchasing and getting of the key-file is normally 1 day

In my holidays it could be longer, therefore I have installed this procedure with the license-key.

4 Vollversion

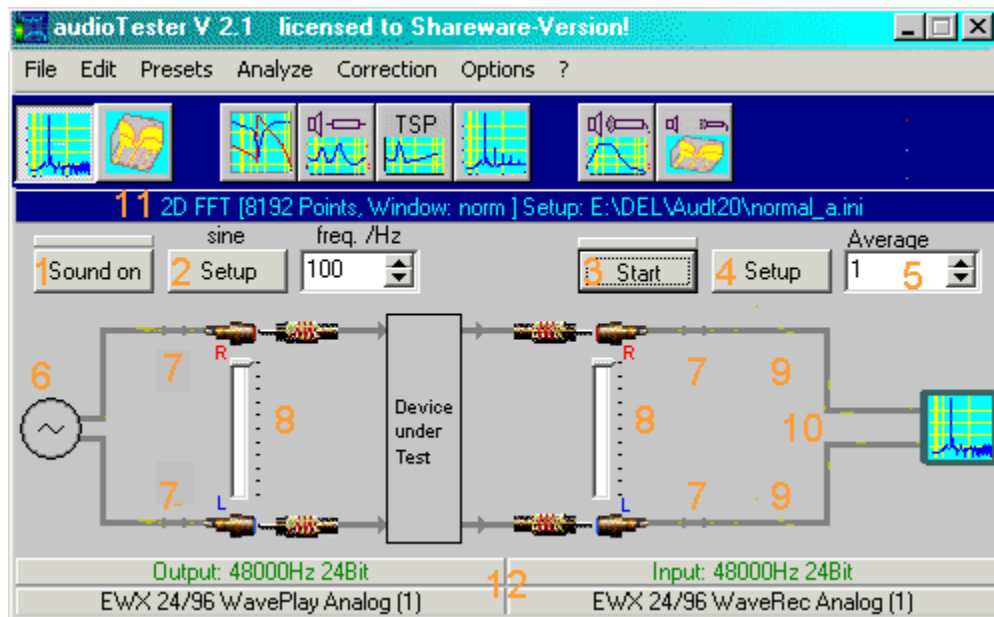
5 Full version

6 2D Spektrumanalyse

7 2D Spectrumanalyzer

Features:

- FFT points: 64 - 32768
- FFT-Windows: none, Blackman, Hamming, Rife-Vince ...
- Averaging
- wave generator 1Hz - 1/2 max. sample frequency (1Hz-96kHz @ SF 192kHz)
- wave forms: sine, square, triangle, white and pink noise ...
- switchable signal paths via schematic in main window

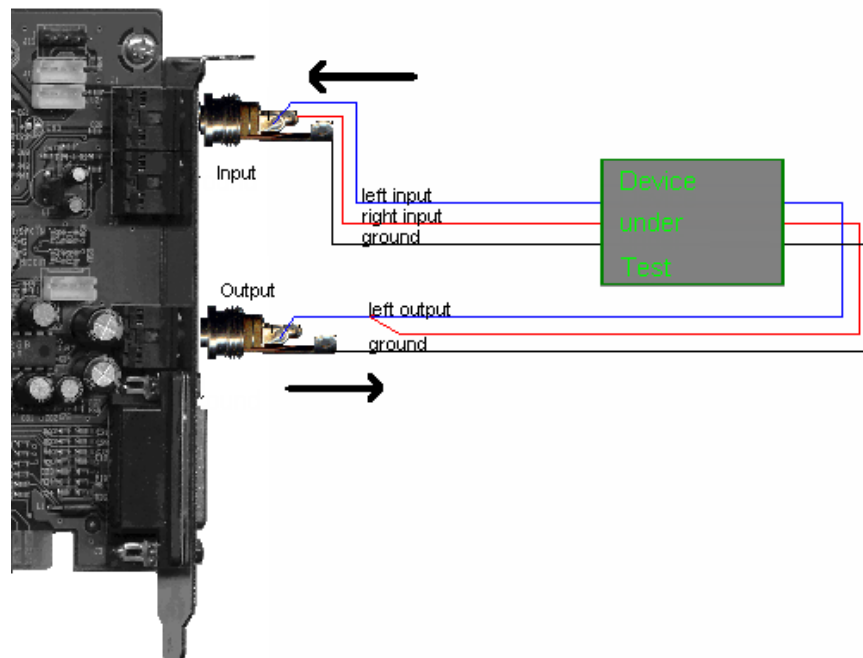


Description:

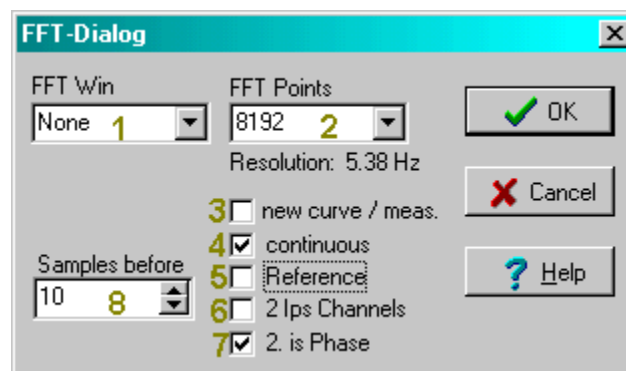
1. Start/Stop sound output
2. Setup sound output
3. Start/Stop spectrum analyzer
4. Setup spectrum analyzer
5. Count of the averages
6. Signal form, adjust by clicking the symbol (also by **setup**)
7. Channel on/off (also with **sound parameter**)
8. Adjustable In/Out level, only if mixer support = on
9. Switchable signal reversing (also with **sound parameter**)
10. Flip channels, mono via mouse click (also with **sound parameter**)
11. Display for the analyse parameter

-
12. Switchable sound parameter (SF, Data width ..) by double click (also via menu/options)

Connection to the test device

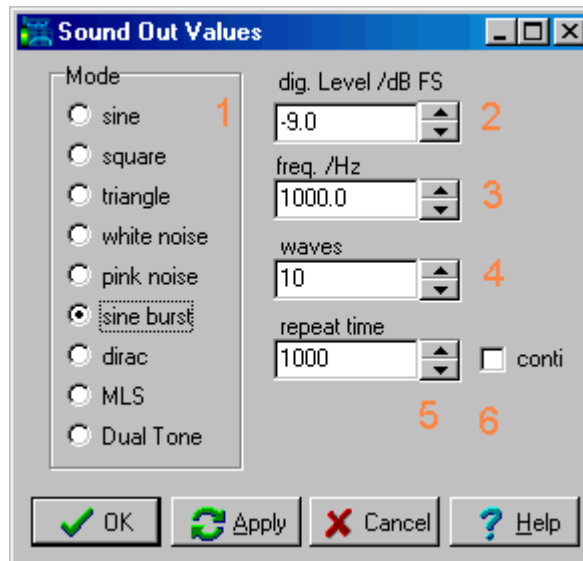


Setup FFT



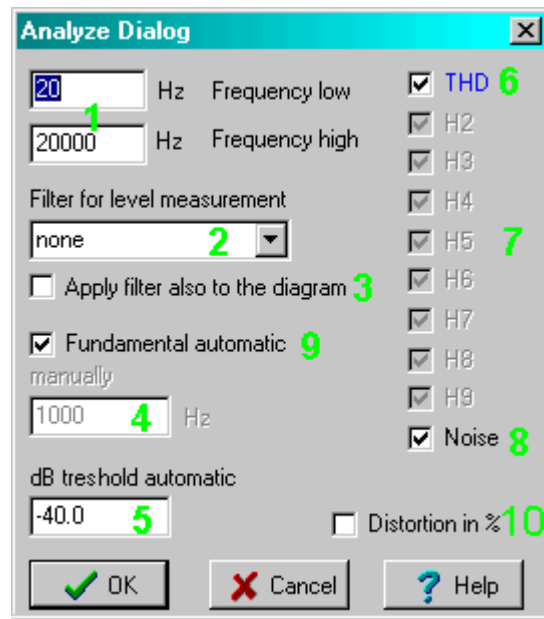
1. Selection of the FFT-Window (None, Hamming, Blackman, Rife.Vince ...)
2. Count of the FFT-Points (64 - 32768)
3. If new curve/measurement is selected, then there will be shown a new curve at each new measurement (Start-Button).
This is useable for compared measurement.
4. Continuous measurement
5. Reference Measurement
6. Two Channels selected, only for impulse measurement. No function here at the 2D spectrum analyzer .
7. The second curve is the FFT-Phase
8. Samples before, for impulse measurement with Dirac-Impulse. No function here at the 2D spectrum analyzer .

Setup Sound



1. Select the signal form
2. Digital level of the signal. Unit is dbFS (dB Full Scale)
3. Frequency in Hertz (except white/pink noise, Dirac, MLS)
4. Special parameter: at Burst -> Count of the full waves
at Dirac-Impulse -> Impulse width in count of samples
at MLS-Impulse -> Order (12 - 16)
5. *repeat time*, at the output of Sine-Burst, Dirac- and MLS Impulse you are able to set the repeat time. Only useful in case of (6) *conti*
6. *conti*, in case of the output of Sine-Bursts, Dirac- und MLS Impulses you are able to force a continuously output.

Analyse Dialog



1. Frequency low
That means the lower bound of the frequency while you calculate the noise component.

1. Frequency high
That means the upper bound of the frequency while you calculate the noise.

2. Filter
You apply these following filters, if you do the level measurement

NONE	no filter
A Weighting	noise voltage measurement, DIN 45412 (audible weighting)
C Message	transfer measurement, IEEE 743-84 (nearly flat)
CCITT - Filter	psychometric measurement IEEE Rec. 743-84
CCITT 0.41	
CCIR wtd	noise voltage measurement, CCIR Rec. 468-4 DIN 45405
CCIR ARM	NAB standard
RUMBLE wtd	record player sound voltage, DIN 45412
RUMBLE unw	record player sound voltage, DIN 45539
IEC Tuner IEC 315	tuner measurement, DIN/IEC 315
DEEM 50/15	CD-player, CCI Rec. 651
DEEMPH 50	noise voltage, DIN 45405 ARD
DEEMPH 75	noise voltage, DIN 45405 ARD
DEEMPH J.17	noise voltage, DIN 45405 ARD
CCITT J.17	
USER	you can define the filter and it loads itself

3. Applies the filter additional to the diagram

4. Fundamental Wave manual
If you don't select the Fundamental Wave automatically, you can edit the fundamental frequency here.

5. Threshold value for fundamental wave detection
Level value, for searching the Fundamental Wave automatically.

6. THD selects all harmonics (faster handle)
7. Selection of several harmonics H2 .. H9
Please notice the measurement bounds for the harmonics.
Valid H2 measurement only up to SF/4 (eg. 11kHz at SF 44.1kHz)
Valid H3 measurement only up to SF/6
Valid H4 measurement only up to SF/8
etc.
8. Additional measurement of noise
9. Fundamental Wave automatically
The fundamental wave is automatically determine, if you are doing the THD+N measurement.
10. The distortions are scaled in % Units, this is only valid for the scales in distortion sweep mode.

With the THD Analyses dialog you can determine the parameters of the THD+N and the level measurement. THD+N means Total Harmonic Distortion plus Noise.

The `Rife-Vinc 3` window should be selected at 4096 points, if you do level and THD+N measurements and desire the best accuracy.

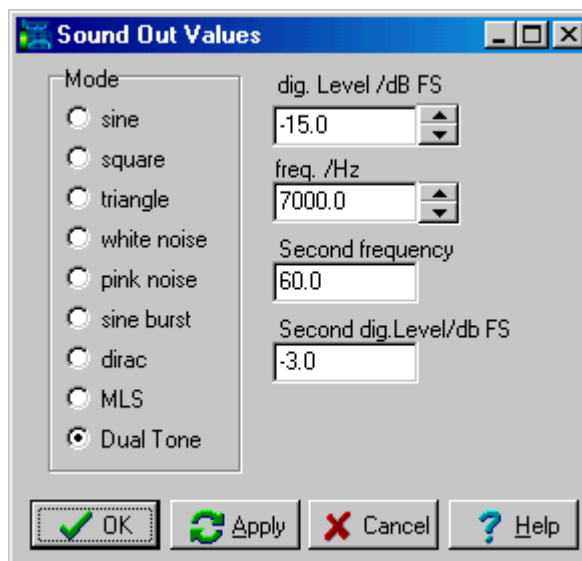
Measurement method THD+N

The fundamental wave is removed from the frequency spectrum, then have the effective voltage value over harmonics d2 thru dn and the noise between the frequencies 'low' and 'high' are summed. Then this value is divide by the total effective voltage value (that means with the fundamental wave and without any frequency spurs) now you get the THD+N.

Measurement method Intermodulations Distortions

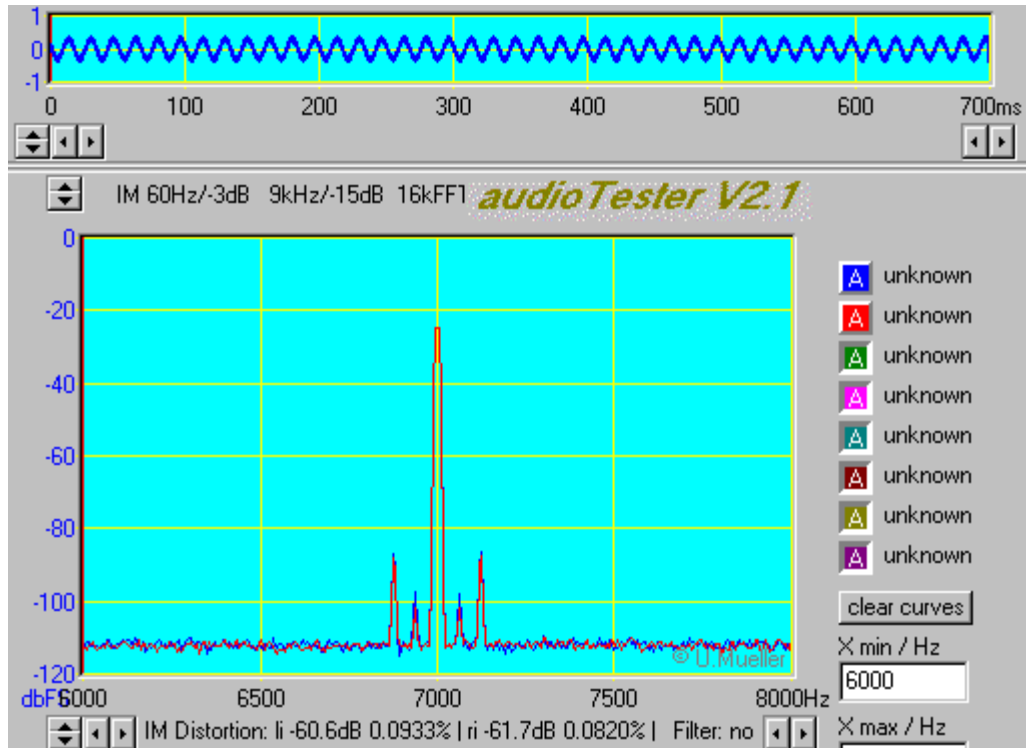
Please select in the sound dialog **Dual Sine** and enter at **freq/Hz** the **main frequency** (eg. 7kHz) and at **Second frequency** the interfere frequency (eg. 60Hz). The IEC 268 Part 3 says that the interfere frequency should be 12db louder than the main frequency.

Eg. dig Level = -15dB second dig. Level = -3dB



Please take care that the addition of the level (de logarithm) never reach the 0dB value (eg. -12dB

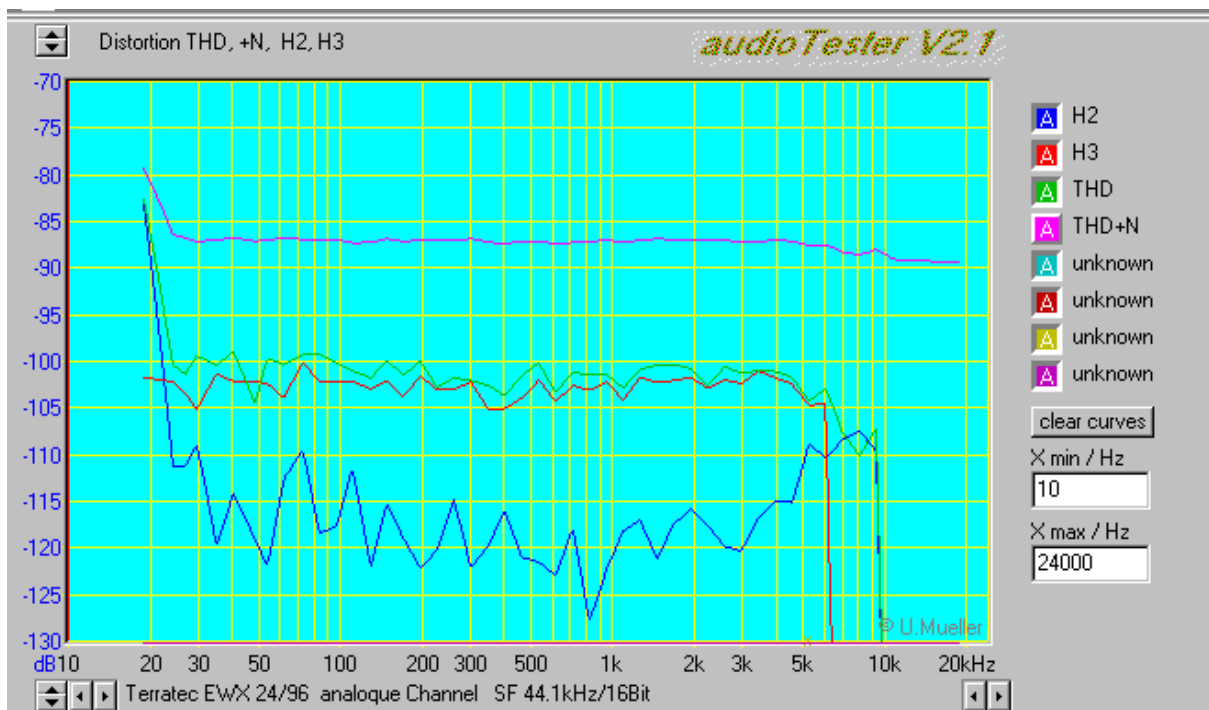
and 0dB don't work). The result is the quadruple sum of the intermodulations products (2. and 3. order at IEC 258 Part 3) and will be displayed in dB and %. Select here a 16384 Points FFT. At the diagram select a linear representation around the main signal. See example below.
For a predefined preset see [here](#)



Measurement method Level

The effective voltage level is produced by the frequencies between 'high' and 'low'. You can apply weighting filters to the measure values.

Example:



Distortion measurement of a sound card

With the menu Analyze/Level Units you select the displayed absolute Level Units:

dbFS

Level refers to digital Full scale

-135/127 at 8 Bits

-32768/32767 at 16 Bits

dbV *

Level refers to 1V 0dbV = 1V

dbu *

Level refers to 0.775V 0dbu = 775mV, this normally is 1mW across 600 ohms

dbm *

Level refers to 1mV at 600 Ohm = 775mV

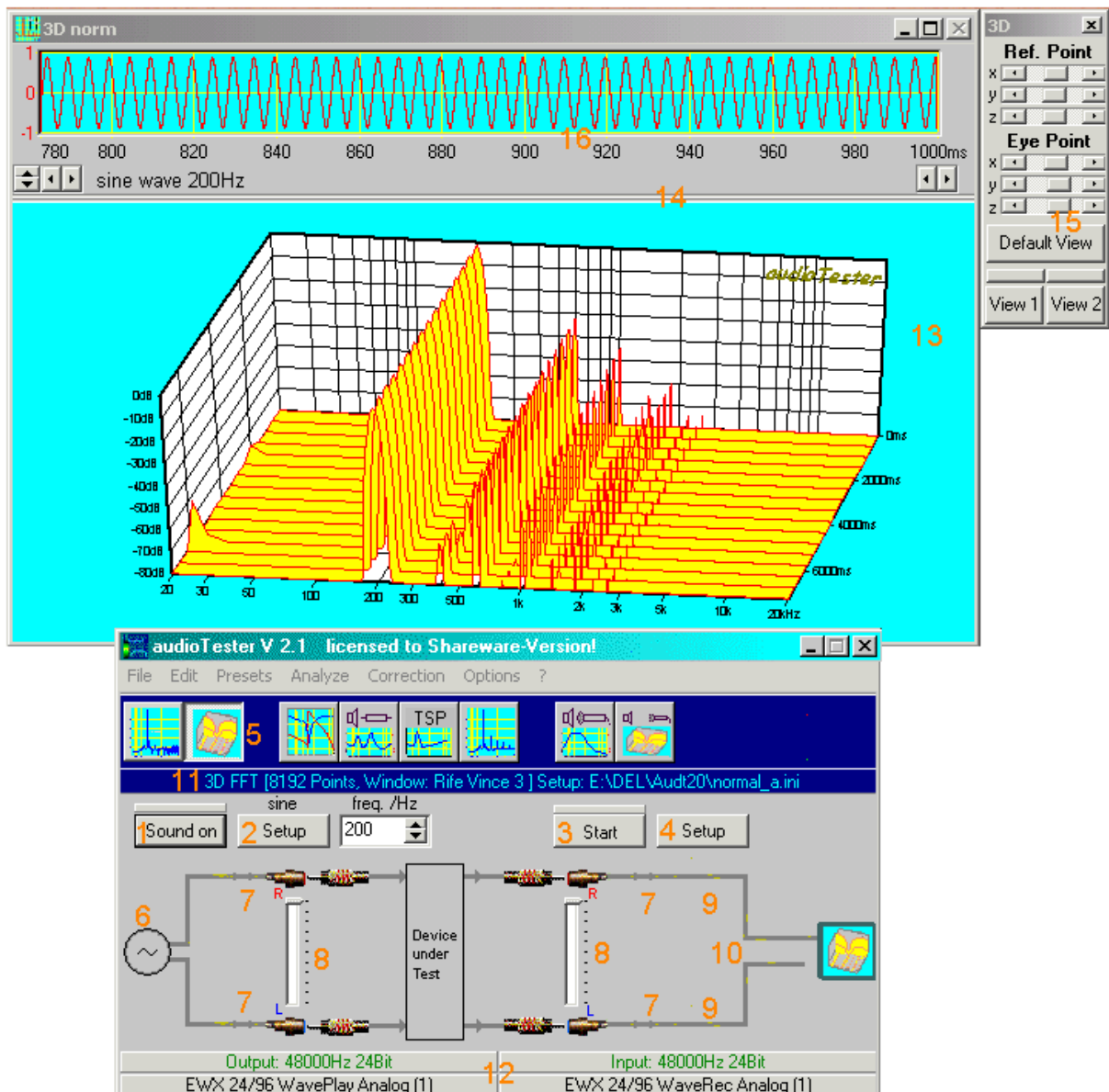
* System must be calibrated in order for these values to be accurate (see [Calibration](#))

8 3D Spektrumanalyse

9 3D Spectrumanalyzer

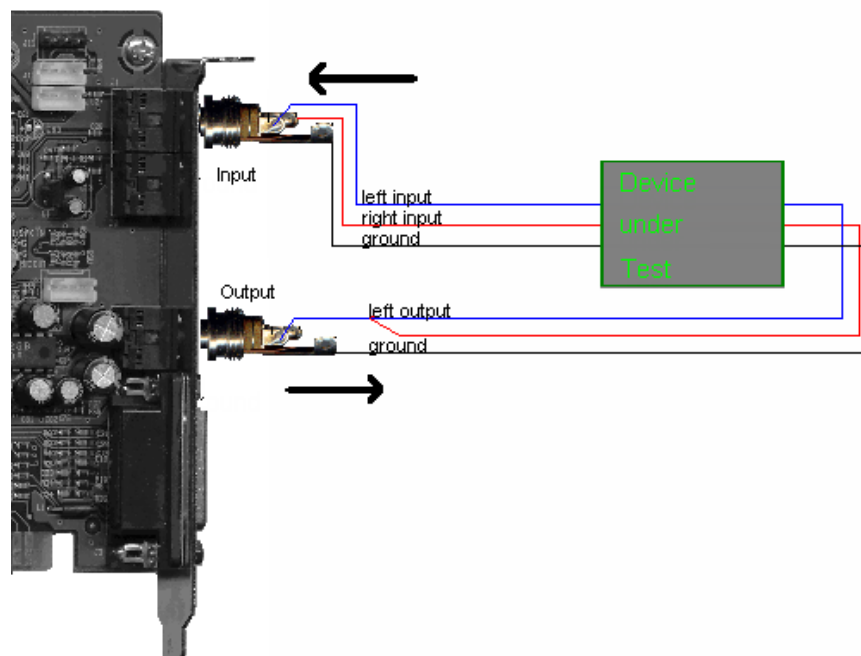
Features:

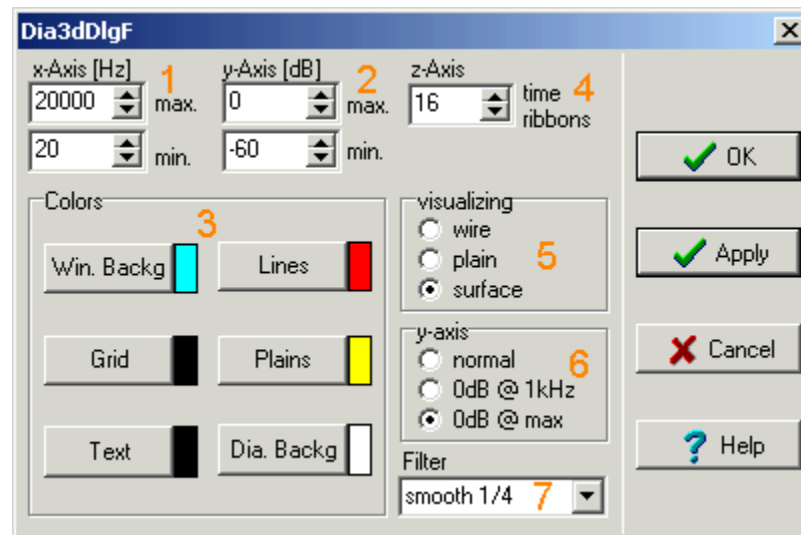
- FFT with 64 up to 32768 Points Bullet.Bmp FFT-Windows: none, Blackman, Hamming, Rife-Vince ...
- Up to 64 time ribbons
- Free rotation of the diagram
- Wave generator 1Hz to 1/2 max. sample frequency (1Hz-96kHz @ SF 192kHz)
- Wave forms: sine, square, triangle, white/pink noise ...
- Switchable signal paths via the schematic in main window



Options:

1. Start/Stop sound output
2. Setup sound output
3. Start/Stop spectrum analyzer
4. Setup spectrum analyse
5. Button for the 3D spectrum analyse
6. Wave form, adjustable via mouse click (also via Setup (2))
7. Switchable output channels (also via via sound parameter)
8. Input/Output level adjustable, only if mixer support is on
9. Switchable signal negation (also via sound parameter)
10. Channel flipping, Mono via mouse click (also via sound parameter)
11. Display for the analyse parameter
12. Switchable sound parameter (SF, Data width ..) via double click (also via Menu/Options)
13. 3D-Diagram, diagram-options with a right mouse click
14. Splitter to move the time and the 3D window
15. Selectors to rotate the diagram in the window
16. Diagram for the time domain

Connection to the test device**3D-Diagram Dialog**



1. Limits of the x-axis - frequency in Hz
2. Limits of the y-axis - level in dB
3. Selection of the diagram colors, window background, the lines, the grid, the plains, the text and the diagram background
4. Count of the time ribbons
5. Visualization of the time ribbons as grid model, plains or as surface
6. Pushing the y-axis to 0dB
7. Selection of the Display Filters

10 Wobbel-Generator

11 Sweepgenerator

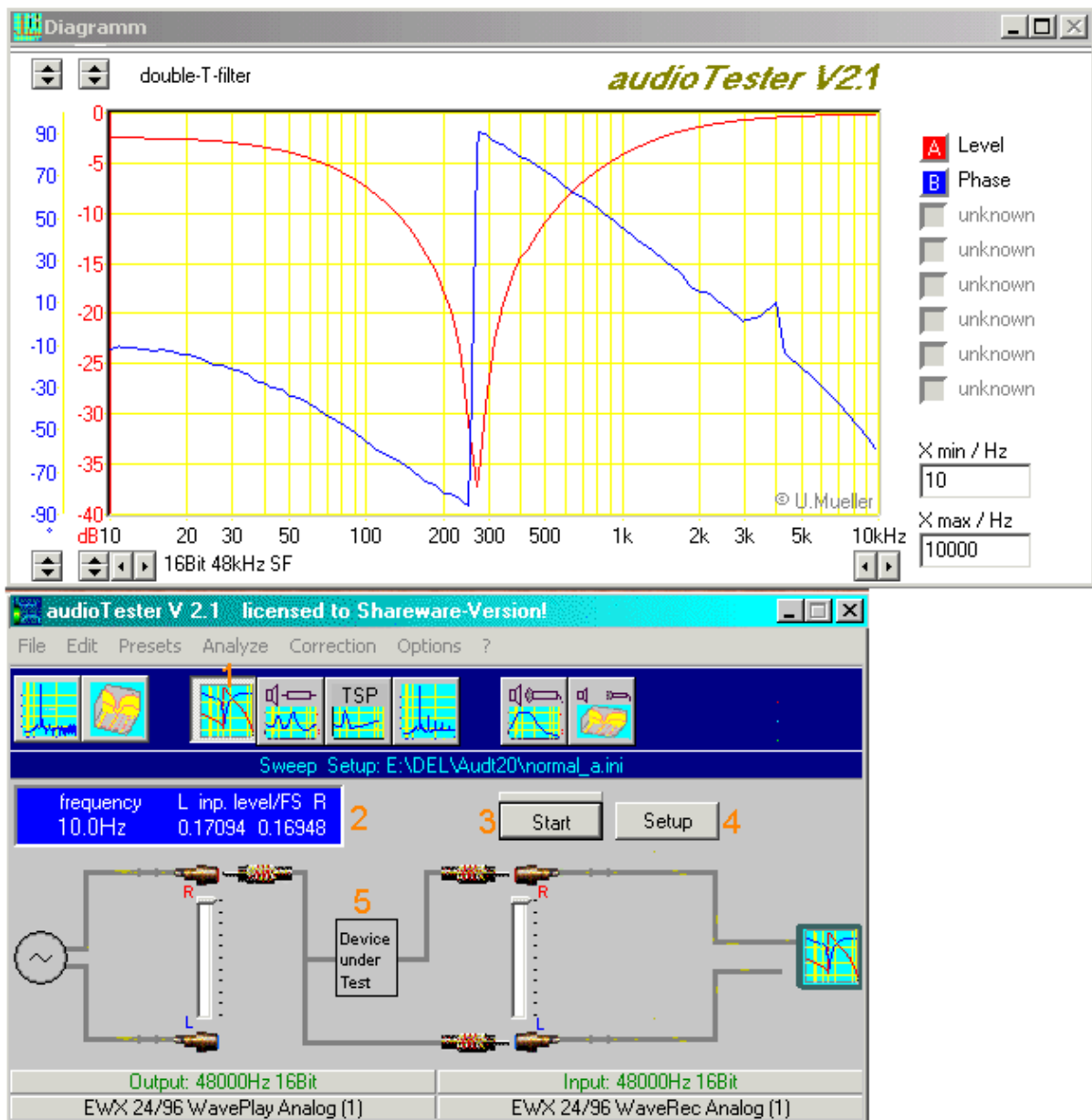
More themes in this topic:
Asynchrone sweep

With the Sweep-Generator you are able to measure frequency responses. Therefore a sliding sine-wave is applied to the measure object, and the read out level is shown in the diagram over the frequency. There is the possibility to use one channel as a reference, so you can eliminate non linear frequency responses of the sound card. With the reference measurement you are also able to measure phase shifting between in- and output.

We have two modes of measurements - synchronous and asynchronous.

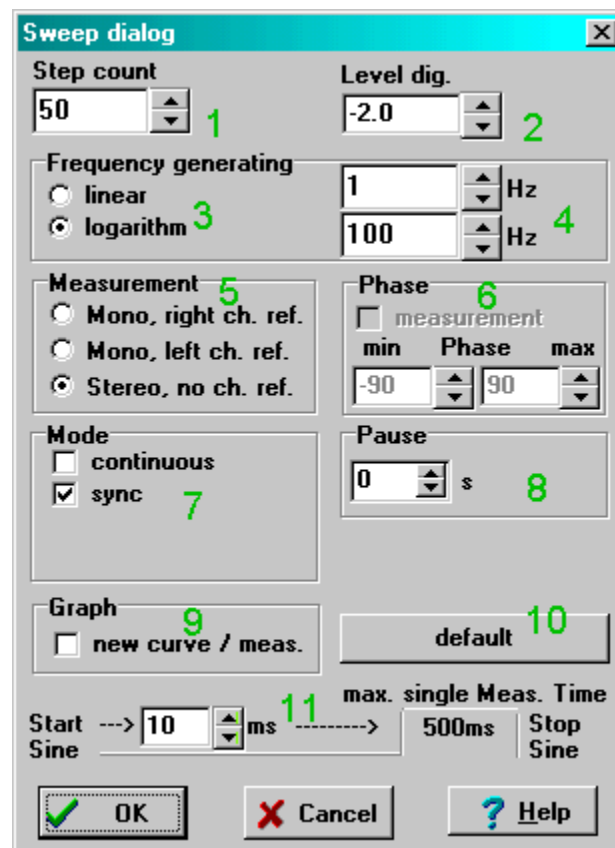
Synchronous measurement: the **audioTester** applied the sliding sine to the measurement object.

Asynchronous measurement: an extern device applies the signal. Example: frequency response of a CD-Player with a measurement-CD or a MP3 Player with a right file.



1. Button to select the Sweep-Generator.
2. Display the actual frequency and level. Adjust the level to get values from 0.1 to < 1.
Red levels indicates over driven signals
3. Start of measurement
4. Sweep Setup ([see here](#))
5. Schematic with switching areas

Sweep Setup



1. Count of the frequency steps
2. The digital level of the measurement signal in dBFS (Full scale)
3. Stepping of the measurement output signal: linear or logarithmic
4. Frequency range of the measurement output signal
5. Measurement modes:
Mono (+ phase) with right or left channel as reference
Compensates nonlinear frequencies of the sound card.
Stereo measurement (with any phase measurement)
6. Selection of the phase measurement, with input of the phase ranges (at the moment no function).
7. Selection of continuous measurement (see also point 8)
Selection synchronous mode: audioTester applies the measurement signal
Asynchronous mode: measurement signal is applied from an extern source. Max. measurement time see point 11.
8. Only cont. measurement - pause after each turn, e.g. to set parameters at the measurement object.
9. Only cont. measurement - new pair of curves after each measurement.
10. Button to recall the default parameter
11. Adjustment of the measure delay and display of the max. measure time at each measure step.
At the asynchronous measurement: Display of the whole measure time

Meas ---> 49 s
Time

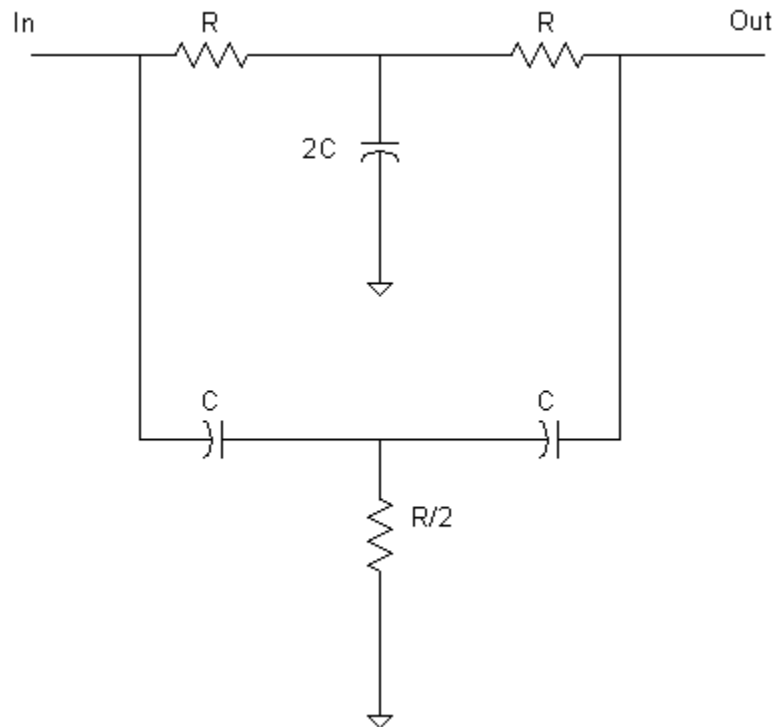
Asynchronous measurement: Please select the total duration time of your external sweep signal
Example: external sweep duration is 60sec. Select 60sec
Measurement with pilot tone: Only the duration of the sweep (without the duration of the pilot tone)

Tips, Tricks

Often there are indistinctness if cables are correct or are the channels flipped or are the setups ok and so on ...

Then there is a small circuit and the correct diagram therefore a good help.

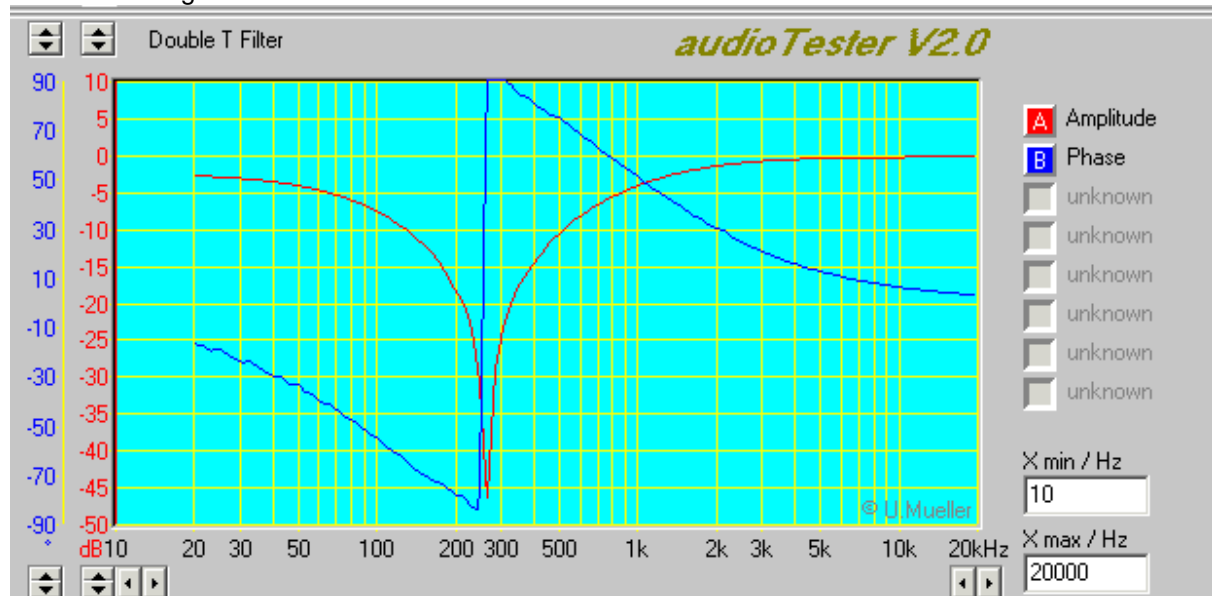
This Double-T-Filter



$C = 0.47\mu\text{F}$

$R = 1.5\text{k}\Omega$

makes this diagram:

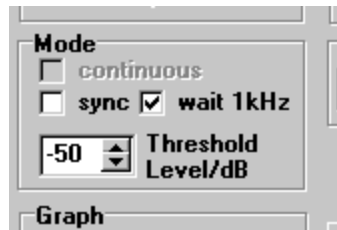


Asynchron Measurement

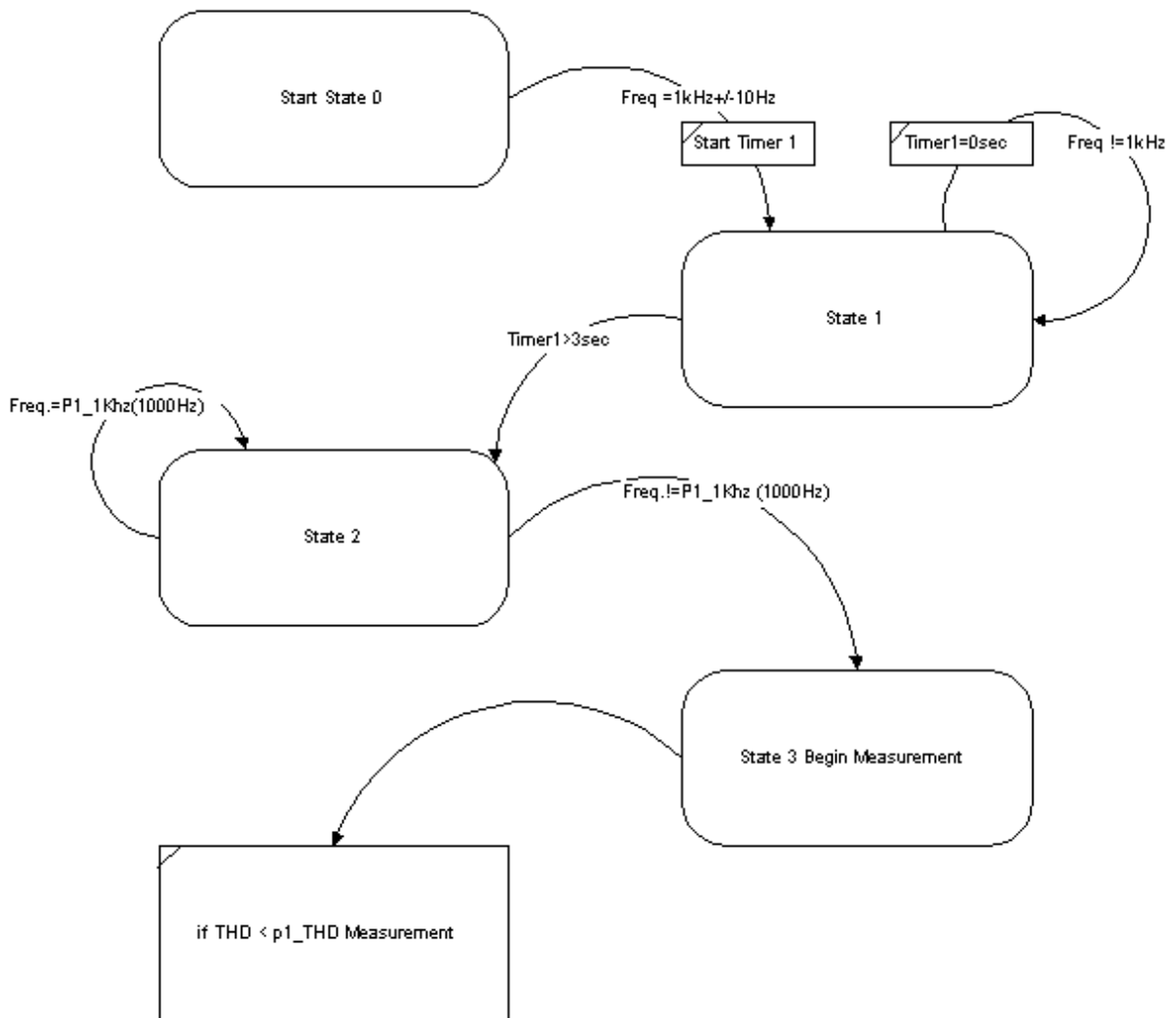
You must use the asynchrone measurement, when the signal comes from the device under test itself. For example a CD-Player with a measurement CD and a sweep from 20Hz til 20kHz.

You can choose between a sweep with or without a 1kHz pilot signal.

Measurement without pilot tone: During the selected time (default 50sec.) every frequency value, which have a distortion value under -6dB, will sorted stored and displayed immediately.



Measurement with pilot signal (see fig.): The obligate frequency measurement works with a threshold value of -50dB (default see fig.) It works with a state machine ->



At start (Start-Button) it will gone from state 0 to 1, if the signal is 1kHz.
 It will gone from state 1 to state 2, if the signal has more then 3sec. a frequency of 1kHz.
 It will gone from state 2 the the begin of the measurement if the frequency is not 1kHz.
 Please start the measurement with the Start-Button of the **audioTester** and start the the device under test (e.g. the CD-Player). A test signal you will find on every good Test-CD. As measuring time you must select the time of the sweep signal without the 1kHz pilot signal

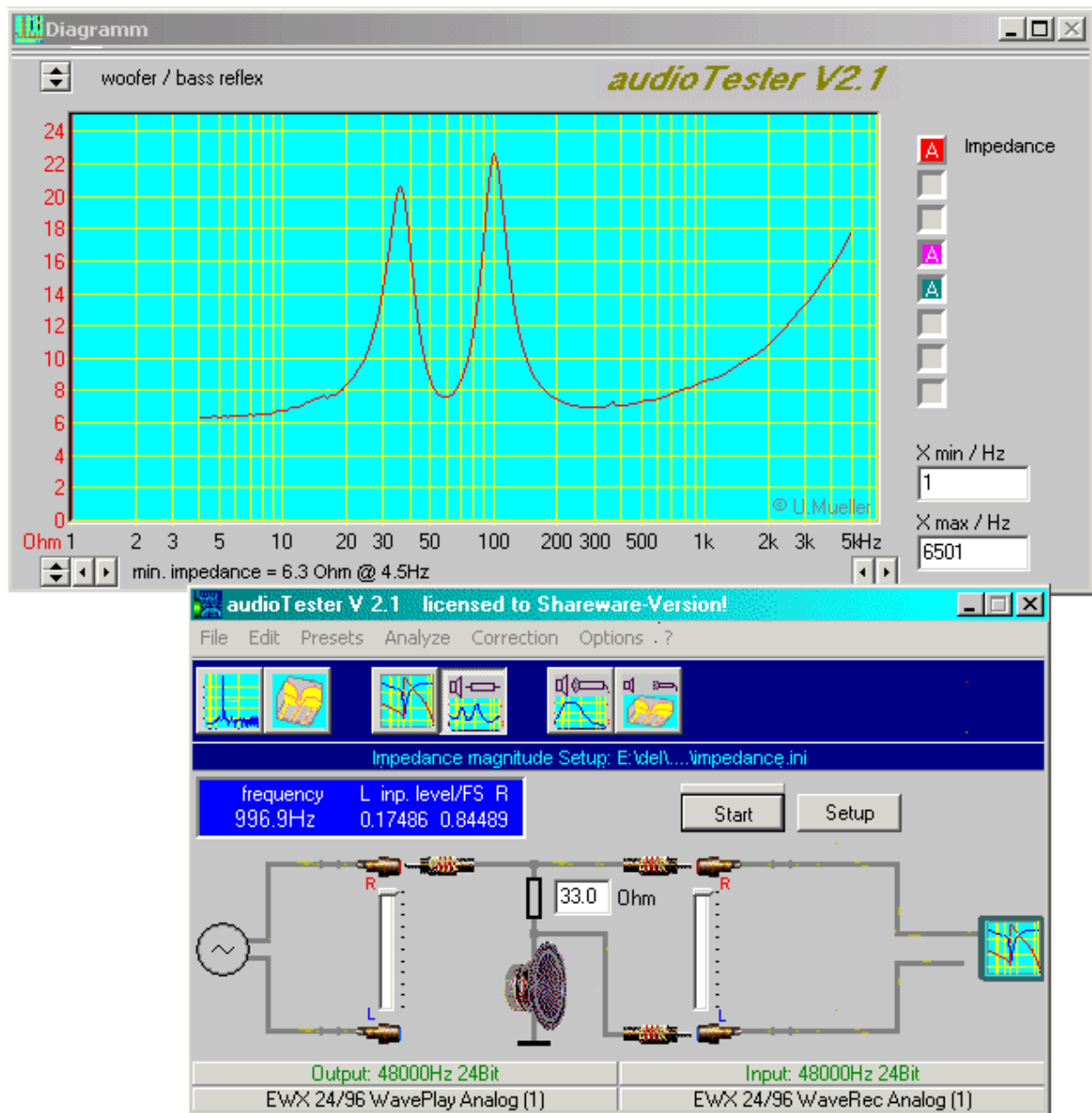
12 Impedanz-Messung

13 Impedance magnitude

more topics [TSP measurement](#)

Impedance magnitude

With this measuring mode you are able to determine impedances vs. frequencies (e.g.a speaker impedance). The use is similar to the use of the sweep generator. You must enter the reference resistor value (here 33 Ohm) in the schematic. The reference resistor should have a similarly value as the impedance. In the *setup dialog* are some settings disabled, otherwise its like the sweep dialog ([see here](#)).



Measuring Thiele-Small Parameters

Introduction

Thiele-Small parameters are an industry norm, they were developed in the 1970's. They are used to assist in the design of low frequency loudspeaker-enclosures systems, including both sealed and vented types. While most manufacturers will list the Thiele-Small parameters of drivers in their data sheets, older drivers may not have values available. It is therefore useful to know how to derive these parameters. The process of measuring the parameters is relatively simple and requires two step.



With this measuring mode of the **audioTester** you are able to determine the TSP .

The Thiele-Small-Parameter:

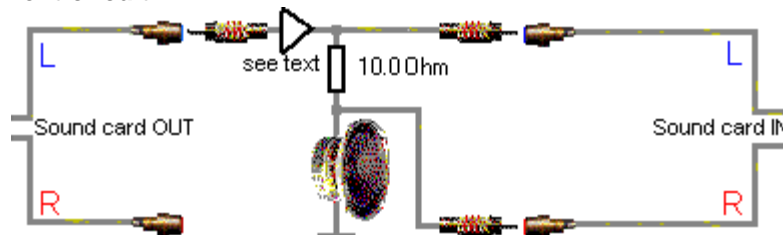
pretended:

Sd	is the area of the transducer diaphragm.	unit: cm ²
Mms	is the extra mass for the second measurement.	unit: g
Vb	is the test cabinet volume for the second measurement. (self build)	unit: Liter

determined by **audioTester**

fs	free air resonant frequency	unit: Hz
Zmax	impedance at resonant frequency	unit: Ohm
Rdc	DC-resistance	unit: Ohm;
Qms	mechanical Q of the speaker	
Qel	electrical Q of the speaker	
Qts	total Q of the speaker	
Mmd	mass of driver's cone	unit:g
Cms	compliance of driver's suspension	unit: mm/N
Vas	compliance volume of the speaker	unit: Liter

Measurement circuit:



Important Hints:

Dependent on the sound card it is possible to apply an amplifier at the sound card outlet.

Place the load speaker on a soft under-ground, so you prevent vibrations at div. resonances.

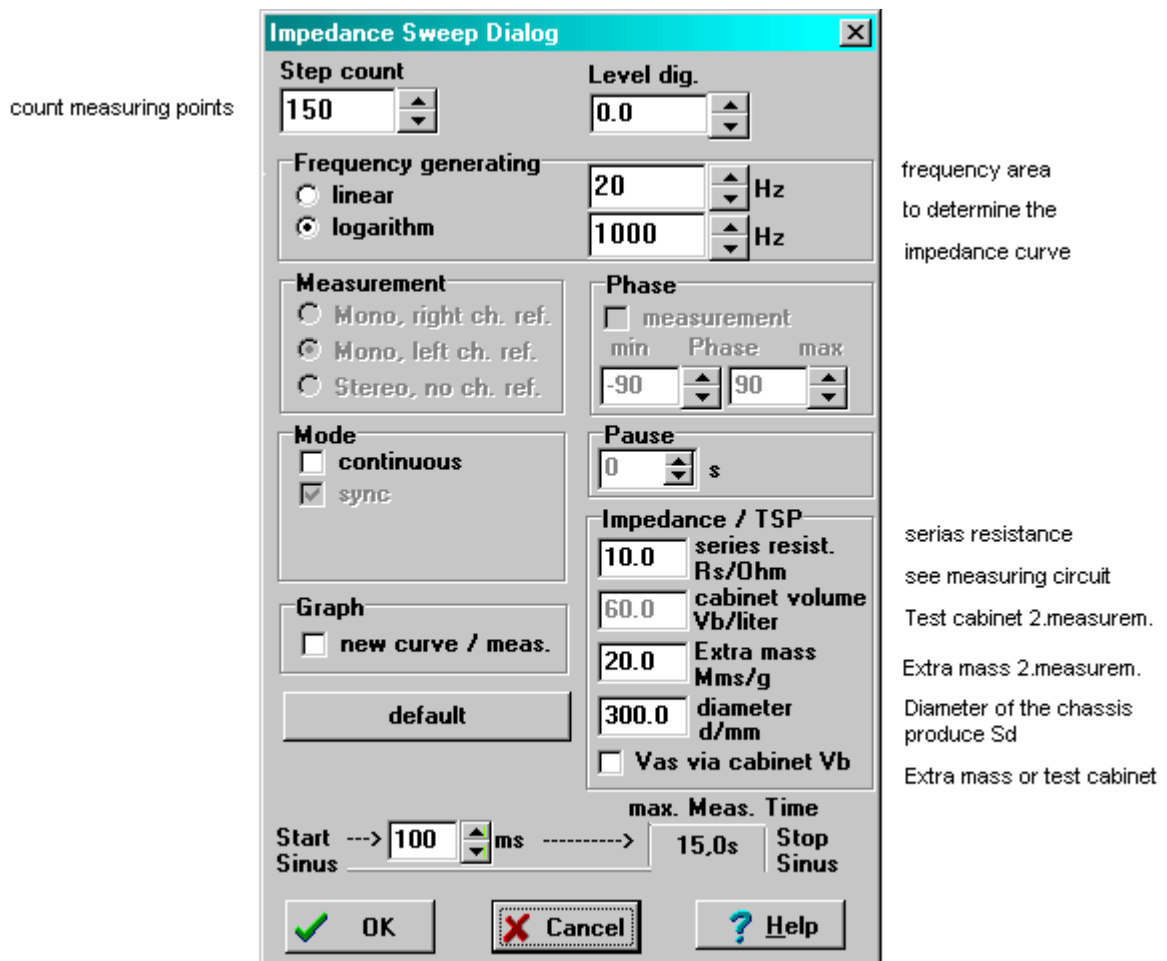
Use wires with a good cross section.

Please remove protector grid from the speaker ([see here](#)).

Do not use bad cables! ([see here](#)).

The extra mass ([modeling clay or so](#)) must be fastened secure at the chassis. Not correct fastened extra mass ([see here](#))

First setup the parameter of the TSP measurement



1.Measurement: Determine of f_s , Z_{max} , R_{dc} , Q_{ms} , Q_{el} , Q_{ts}

Apply speaker like described above and the start measurement.

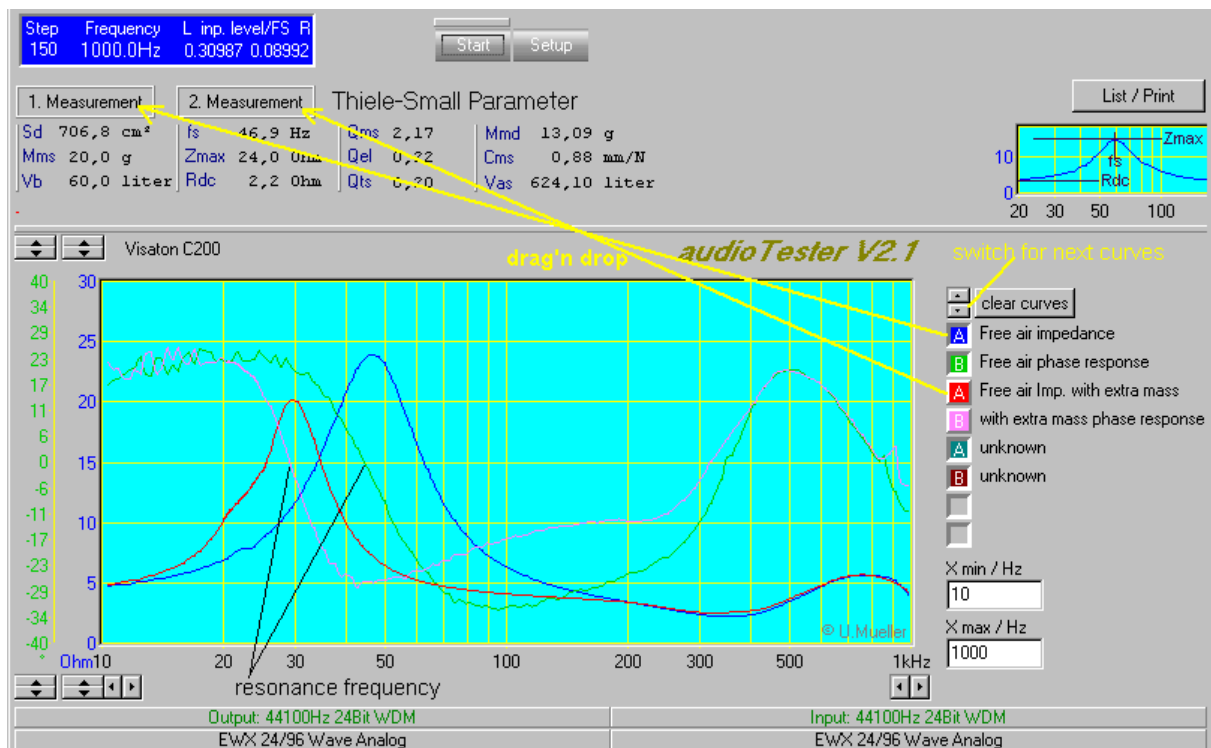
After the measurement drag the Free Air Impedance curve (in the example the blue one) to the panel '1.Measurement' (see picture below).

The values $f_s=48.4\text{Hz}$, $Z_{max}=19.2\text{Ohm}$ are immediately determined.

If an error occurred, a message appears below the measurement values:

Error: $f_2-f_1=0$	Impedance could not determined
Error: division by zero	Problems while determine Z_{max} and R_{dc}
Error: floating point error	General error
Error: f_s not found	Curve could not determined (no resonance)
Error: $R_{dc} = 0$	Resistor correct entered / plugged ?
Error: f_s not plausible	Curve could not determined (no resonance)
Error: f_s not plausible (f_1 , f_2)	Error while calculating of f_s (curve too short, more than one resonance ?)

Please repeat the measurement 3 or 4 times, until you are sure that the measurement is uninfluenced from any malfunctions and effects from outside. If the first measurement is good, then switch the curves for the second measurement (see below)



In the picture you see that the phase response at the resonance frequency goes through the zero line.

2.Measurement: Determine of Mmd, CMS und Vas

To determine of Mmd, CMS und Vas the speaker must build into a test cabinet or an extra mass (20g modeling clay) must fastened at the speaker.

For a chassis of 200-250mm (8"-10") use a 30 liter closed cabinet.

For a chassis of 250-300mm (10"-12") use a 60 liter closed cabinet.

In general the size of the test cabinet should be chosen, that the resonance frequency is 50% higher than the free air measurement resonance frequency.

Determine Vas over the extra mass measurement is much difficult as with an extra cabinet, but even easier.

The values are influenced from the temperature, moisture and so on. Make the decision for the measurement with a cabinet,

choose 'Vas via cabinet' in the *Setup dialog*.

After the measurement drag the free air impedance curve (in the picture the red one) to 2.Measurement panel and drop it here.

The calculated values are shown.

Hint:

At Measurement with extra mass the resonance freq. is lower than the free air resonance.

At Measurement with test cabinet the resonance freq. is higher than the free air resonance.

Error messages:

Error: Calc Mmd

Error: Calc Cms

Error: Curves with and without mass are identical

Error: Curves with and without extra cabinet are identical

Error: please make first measurement

Floating error at calculating Mmd

Floating error at calculating Cms

you drop the wrong curve

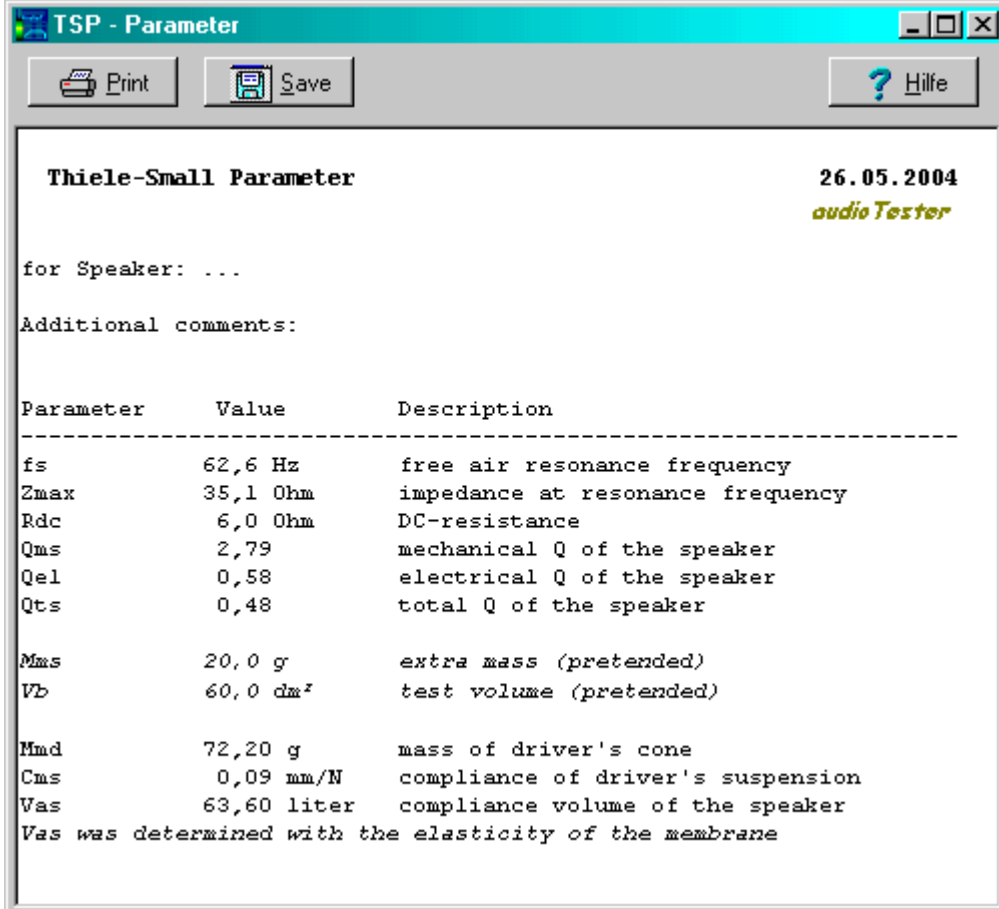
you drop the wrong curve

do it

The summary is, that the TSP measurement reacted very sensitive on parameter changing in

measurement wiring and environment. That a look on same cables and same speaker positions while comparing different speakers.

With the button *List/Print* the determined values are listed for further use.



The screenshot shows a software window titled "TSP - Parameter" with a menu bar containing "Print", "Save", and "Hilfe". The main content area displays the following text:

Thiele-Small Parameter 26.05.2004
audio Tester

for Speaker: ...

Additional comments:

Parameter	Value	Description
<i>fs</i>	62,6 Hz	free air resonance frequency
<i>Zmax</i>	35,1 Ohm	impedance at resonance frequency
<i>Rdc</i>	6,0 Ohm	DC-resistance
<i>Qms</i>	2,79	mechanical Q of the speaker
<i>Qel</i>	0,58	electrical Q of the speaker
<i>Qts</i>	0,48	total Q of the speaker
<i>Mms</i>	20,0 g	extra mass (pretended)
<i>Vb</i>	60,0 dm ³	test volume (pretended)
<i>Mmd</i>	72,20 g	mass of driver's cone
<i>Cms</i>	0,09 mm/N	compliance of driver's suspension
<i>Vas</i>	63,60 liter	compliance volume of the speaker

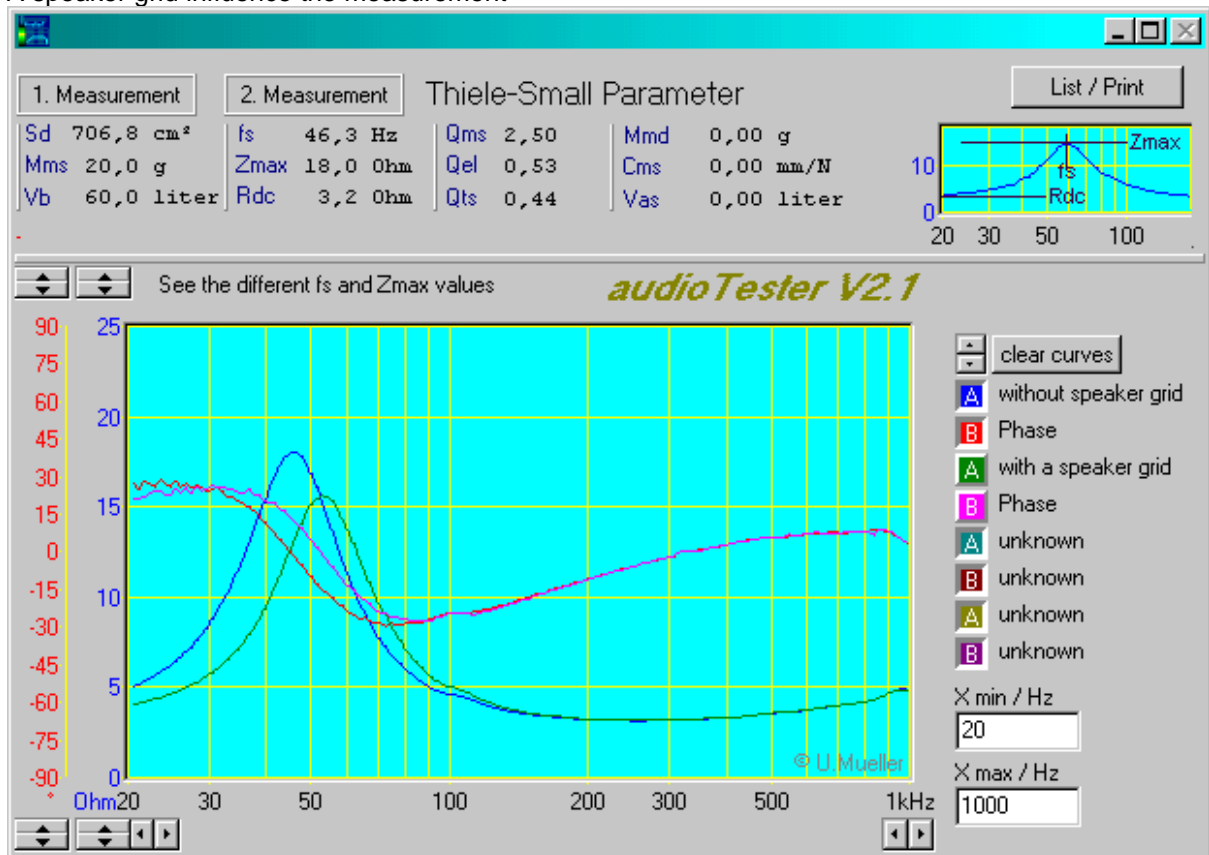
Vas was determined with the elasticity of the membrane

The speakers without and with an extra mass

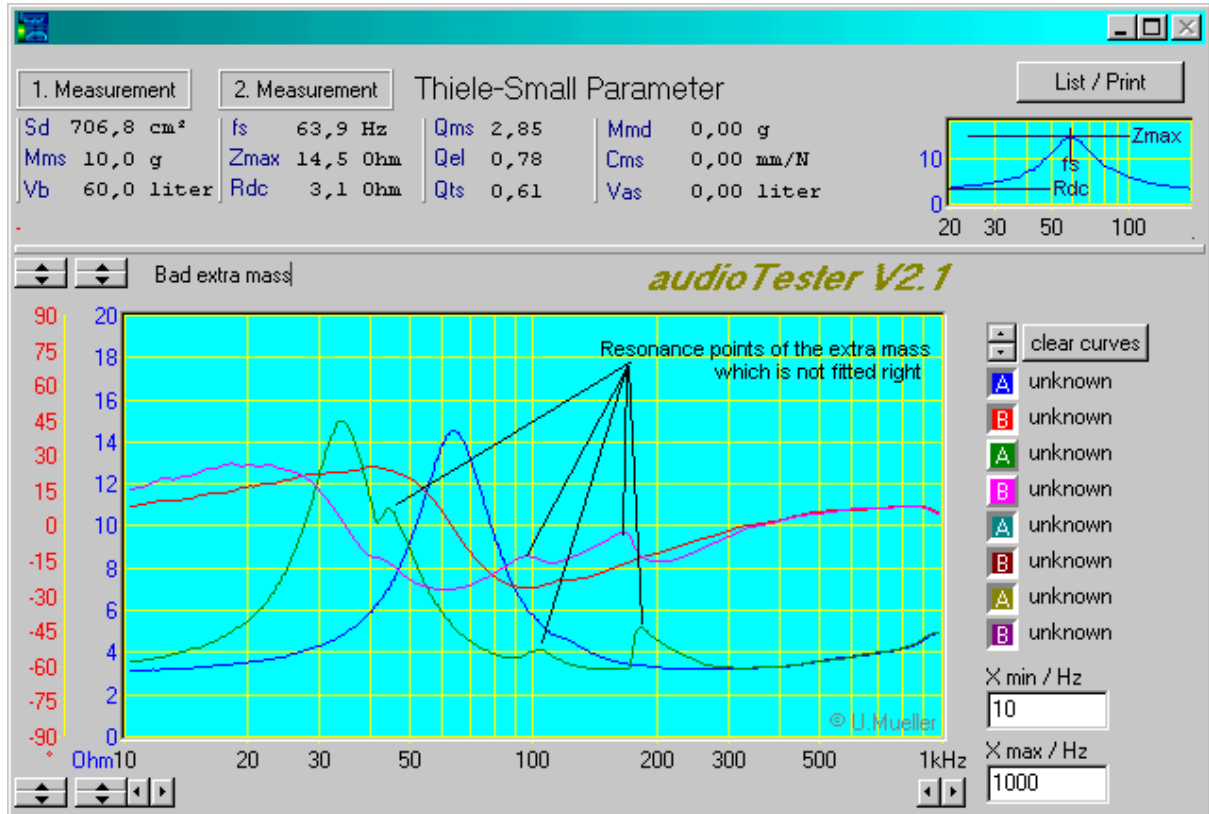


Examples for measurement errors:

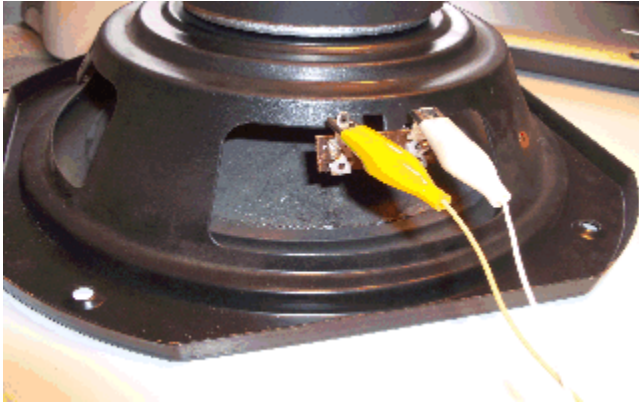
A speaker grid influence the measurement



Not correct fastened extra mass



This cables are bad

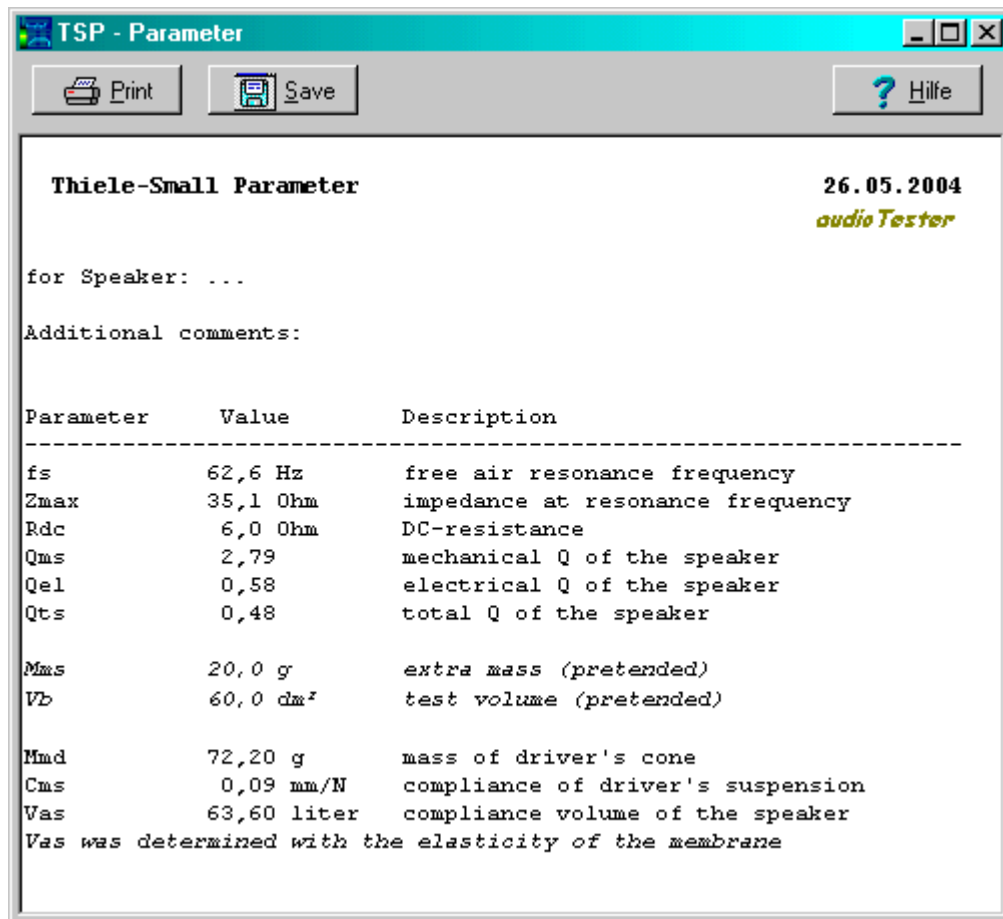


Better



Attention: Membrane to the bottom side - only for this photos, **Membrane always to the upside while measuring!**

Thiele-Small print out

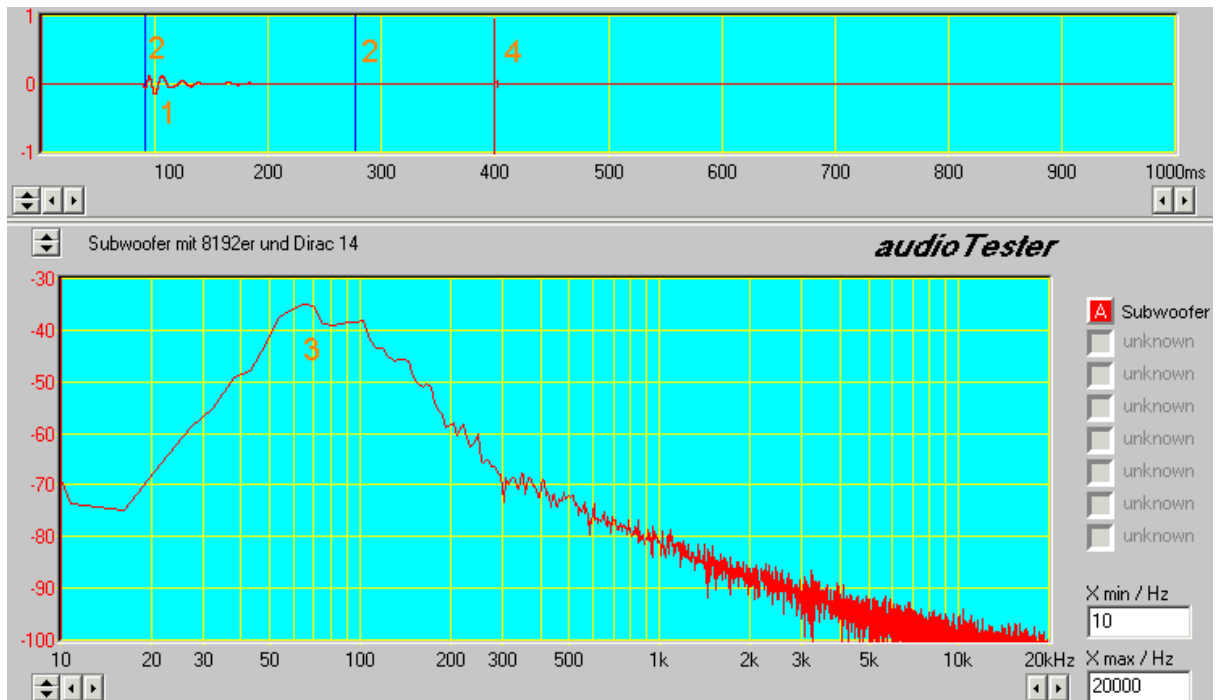


14 2D Impulsmessung

15 2D impulse measurement

Also on this page measurement of delays and measurement with limiters
 With this measure mode you are able to measure impulse responses of loudspeakers and filter systems. As a stimulus you can select a dirac impulse, a MLS-Impulse (Maximum Length Sequence) or a sine wave burst. To select, click the setup button on the left, it opens the Sound-Setup.([see here](#))

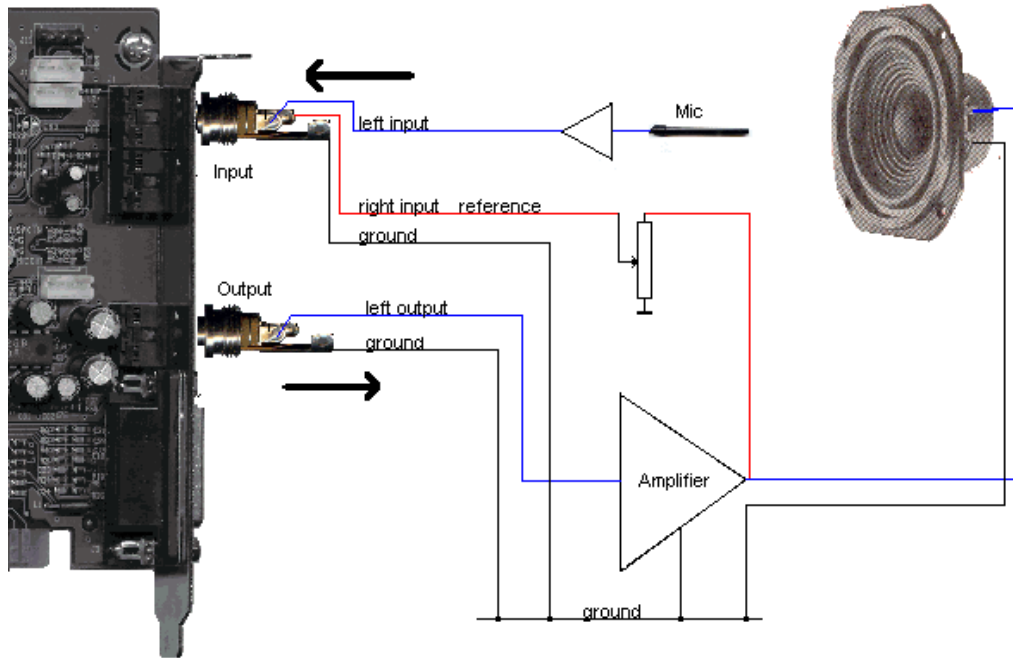
In the example (see below), you see the measurement of a subwoofer via microphone in a distance of 1 m. Stimulus is a dirac impulse with a width of 14 samples. At a sample frequency of 44.1kHz and a width of 14 samples the measurement is ok till 1,5kHz, good enough for a subwoofer. With the averaging of more then one impulses you can increase the Signal to Noise Ratio. Each doubling of the count of measurements gives you an improvement of 3dB (2 Meas. =3dB 4 Meas. =6dB 8 Meas. =9dB)



Measure example:

1. Impulse response in time domain
2. Measurement window at a FFT with 8192 points (FFT Window: none)
3. Impulse response in frequency domain
4. Manuel marked echo point ([see also here](#)) (normally not so far away ;-). If you marked an echo point, so you must trigger a new calculation with the *Manuel* button. The FFT is execute till the echo point, with the renunciation of some FFT lines. ([see fig.](#))

Measurement schematic



Important hints:

Input (right) is only necessary if you use the reference measurement. Effect of the reference measurement [see here](#)

Please pay attention to the maximum input voltage of the used sound card.
Perhaps use a attenuator before the reference input.

Please increase output level slowly.

Do not use a bridged power amplifier.

The left and the right output of the sound card is equally.

Time Diagram Options

With a mouse click in the time diagram a popup-menu will be opened. In the menu you be able to select some settings of the time domain diagram.

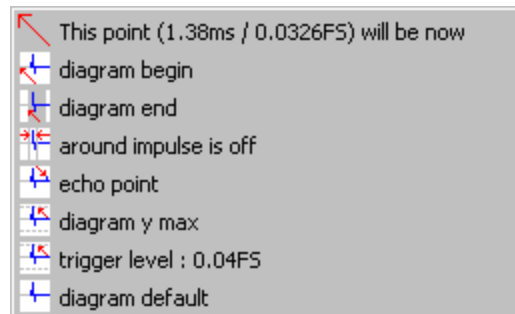


diagram begin

diagram end

around ips

echo point

New begin of the time diagram at the clicked position (here at 1.38 ms)

New end of the time diagram at the clicked position (here at 1.38ms)

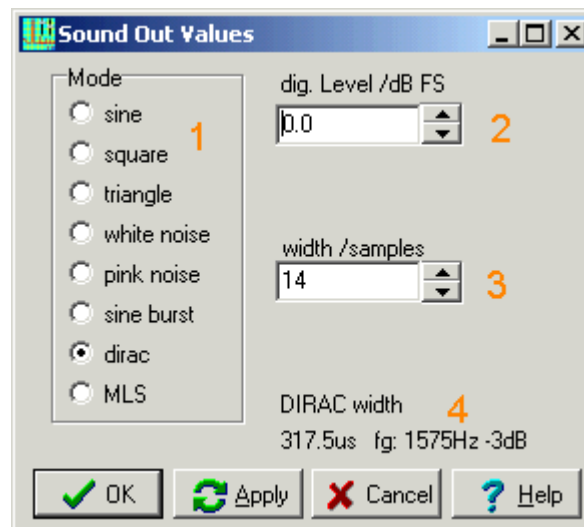
The displayed time range is set to new values, which makes it possible to see the environment around the impulse.

Setting of an echo point while impulse measurement, only till this point the

diagram y max	FFT works (shown by a red, vertical line) New Y-scale at the clicked position, here Ymax = 0.0326 FS and Ymin=-0.0326 FS
trigger level	Threshold for an automatic impulse recognition -> automatic setting of the FFT begin
diagram default	reset of all parameters, clears the echo point

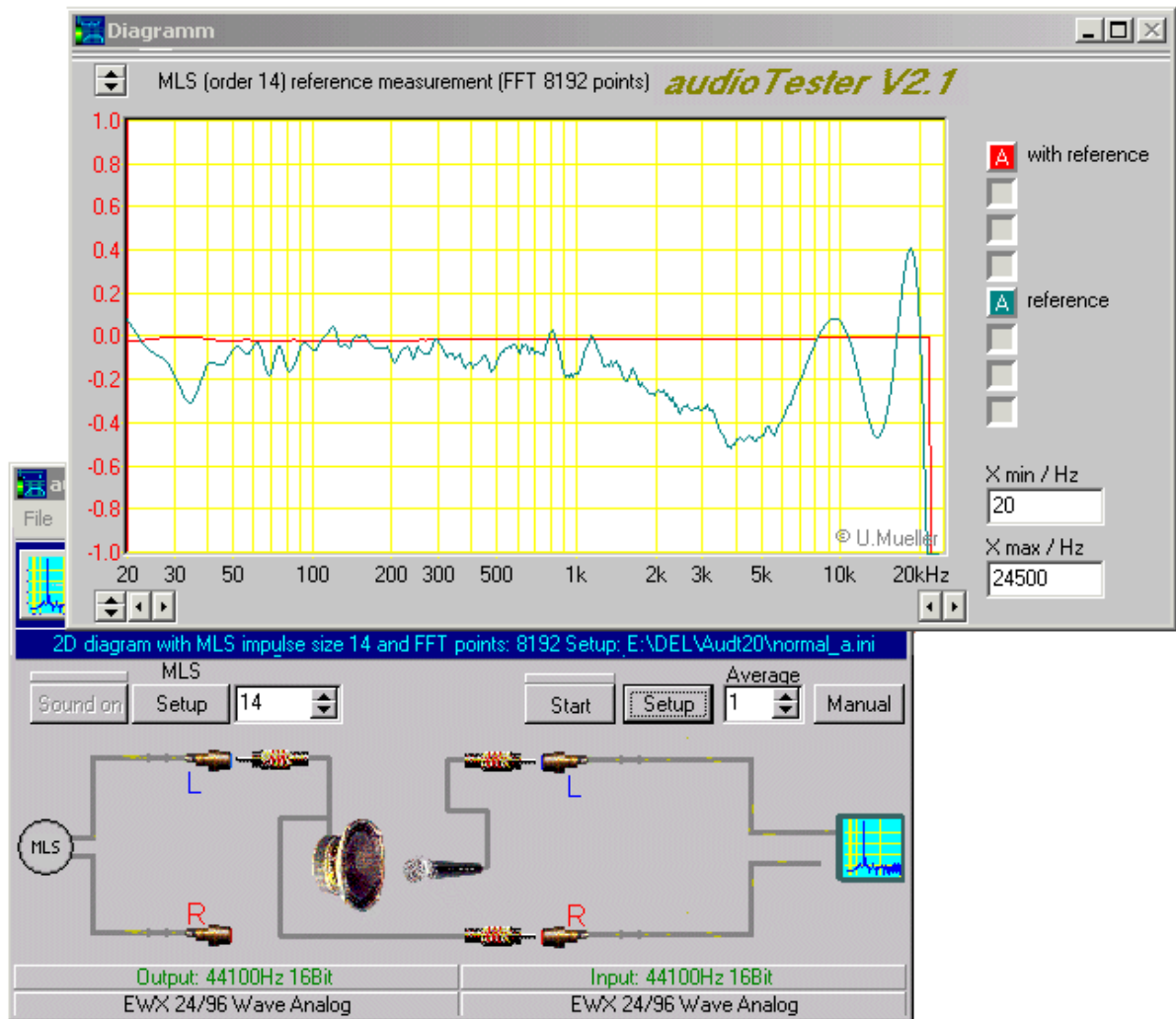
If you click the first row (This point ...), you will leave the menu without any changes!

Sound setup



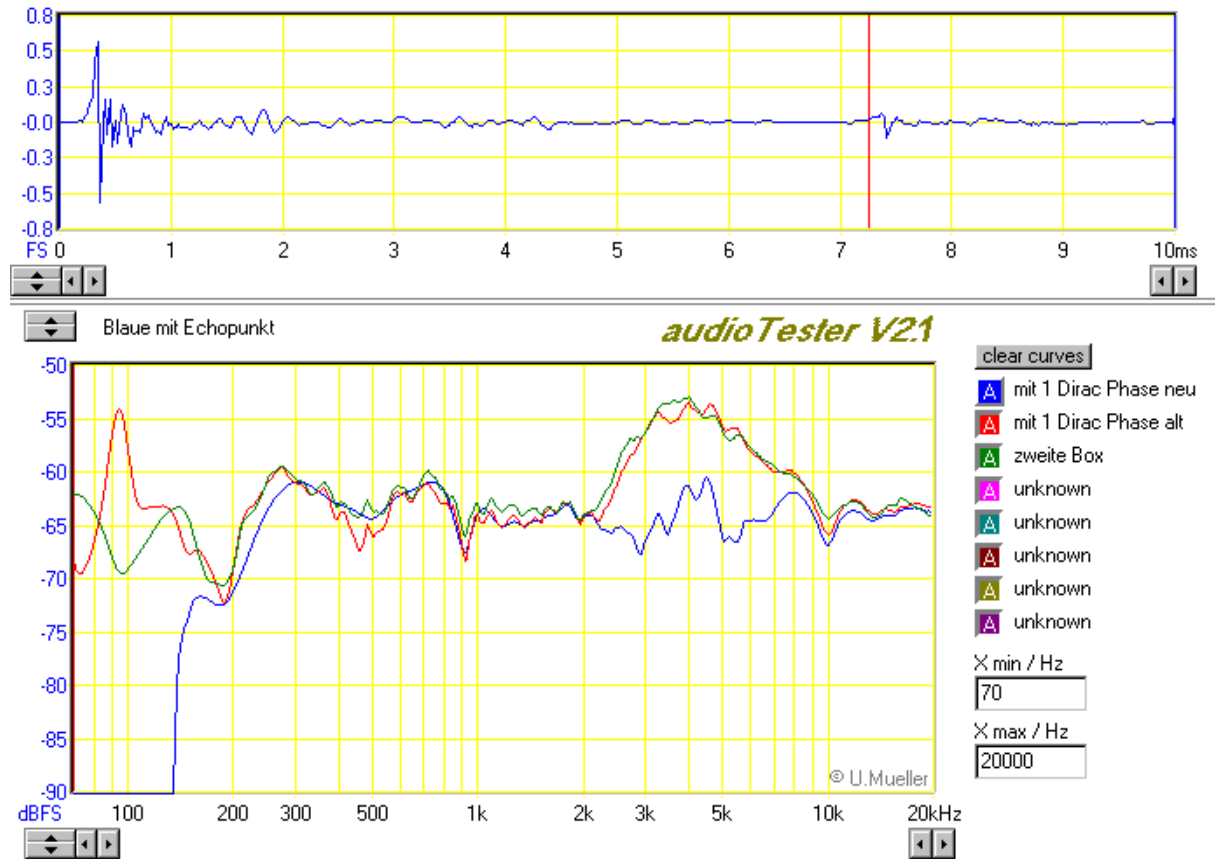
1. Selection of the wave form
2. Adjust of the digital signal level. Dirac and MLS have a level of -9dBFS. In the case of Dirac and MLS the the level will be set to -9dBFS .
3. Selection depends from the selected wave form, here you adjust the width of the dirac impulse
4. Display of some calculated values, it depends of the selected wave form, here the max. reasonable frequency range up to 1.5kHz
 For the Dirac impulse is $fg = SF[Hz] / (Width[count of samples] * 2)$
 e.g. SF = 48kHz, Dirac width = 1sample -> $48kHz / (1*2) \rightarrow fg = 24kHz$
 For the MLS impulse you see the duration of the impulse:
 MLS Duration = $2^{MLSOrder} / SF$
 e.g. MLS Order = 15 SF = 44.1kHz -> $2^{15} / 44.1 = 743ms$

Reference measurement of piece of 'wire'



red graph: originate frequency response of the 'wire'
 green graph: signal reference input (R)

Echo point example

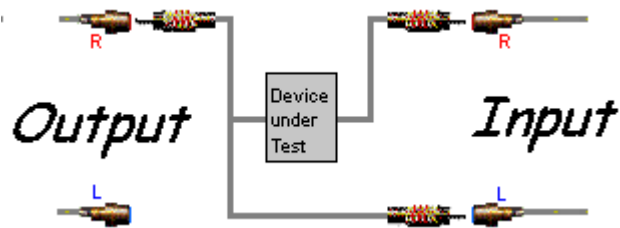


The blue line is calculated with the echo point (see red line in time domain diagram) and a click on the *manuel* button.

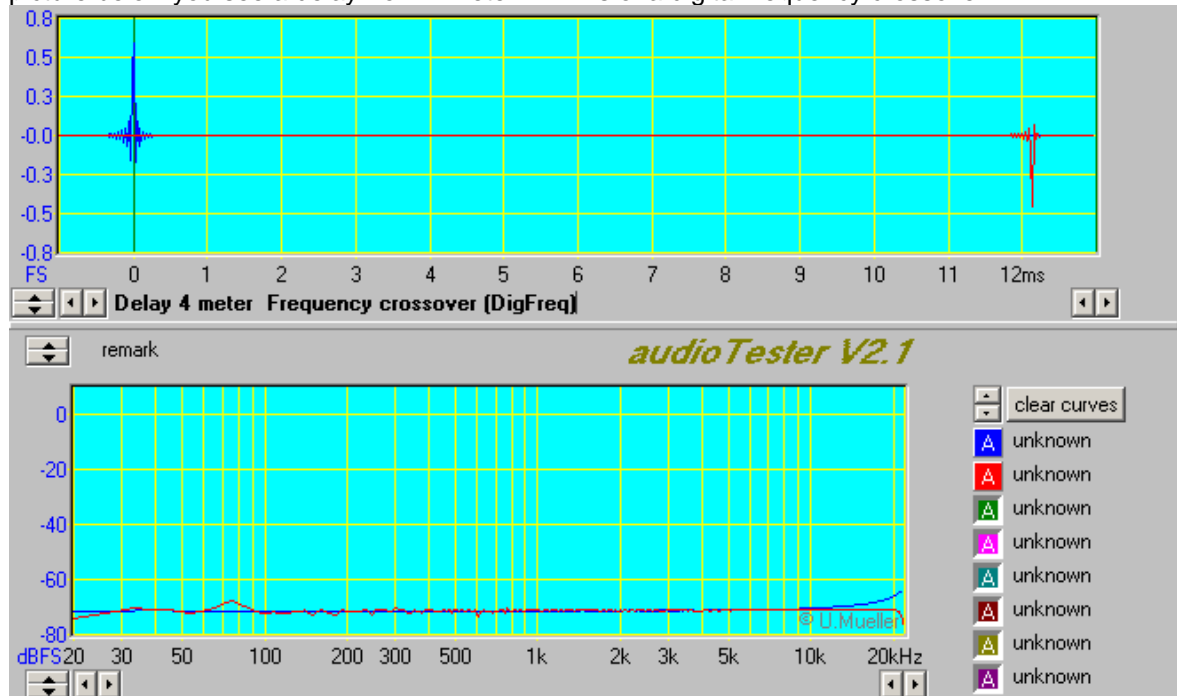
You see the missing low frequencies, depend on the decrease of the FFT-lines. The red and green line are calculated without an echo point, but with the same real echo. You see the strange frequency responses below 150 Hz.

Measurement of delays

With the delay measurement you are able to determine delays in audio devices. See the wiring diagram below to connect the audio device, it looks like the wiring of a reference measurement. But you don't select in the setup dialog the the reference measurement !

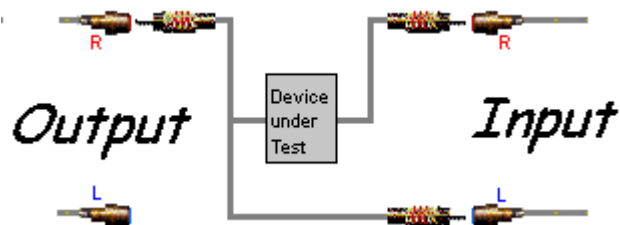


We use the time domain diagram, which is now extend to 2 curves. The audio device is stimulated with a dirac impulse, we see then the undelayed and the delayed impulse in the diagram. In the picture below you see a delay from 4 meter = 12ms of a digital frequency crossover.



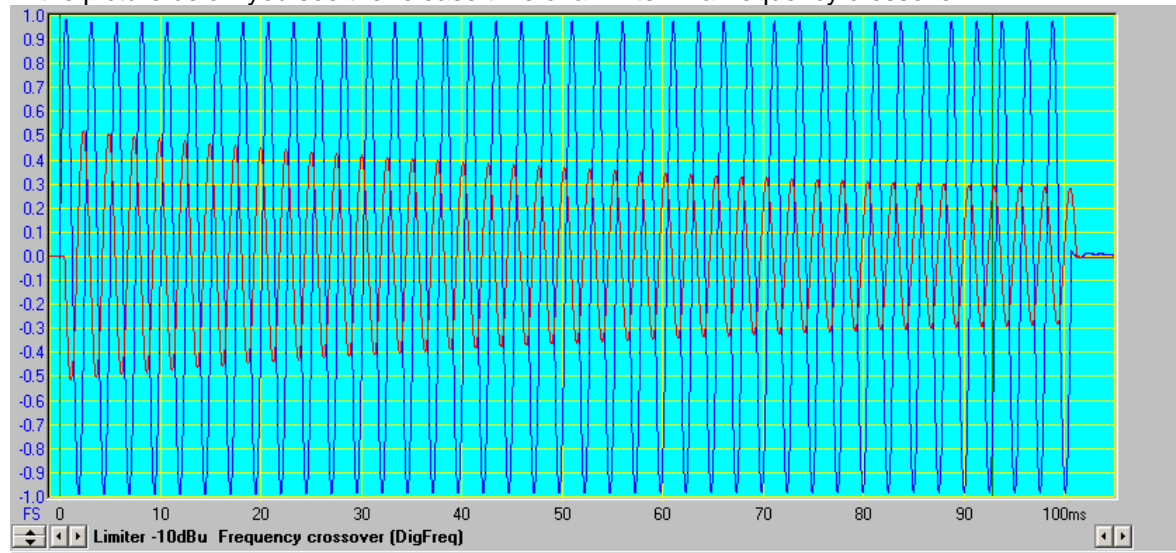
Limiter measurement

With the limiter measurement you are able to determine attack and release time of limiter of audio devices. See the wiring diagram below to connect the audio device, it looks like the wiring of a reference measurement. But you don't select in the setup dialog the the reference measurement !



We use the time domain diagram, which is now extend to 2 curves. The audio device is stimulated with a burst impulse (!), we see then the attack and release progression in the time domain diagram.

In the picture below you see the release time of a limiter in a frequency crossover.

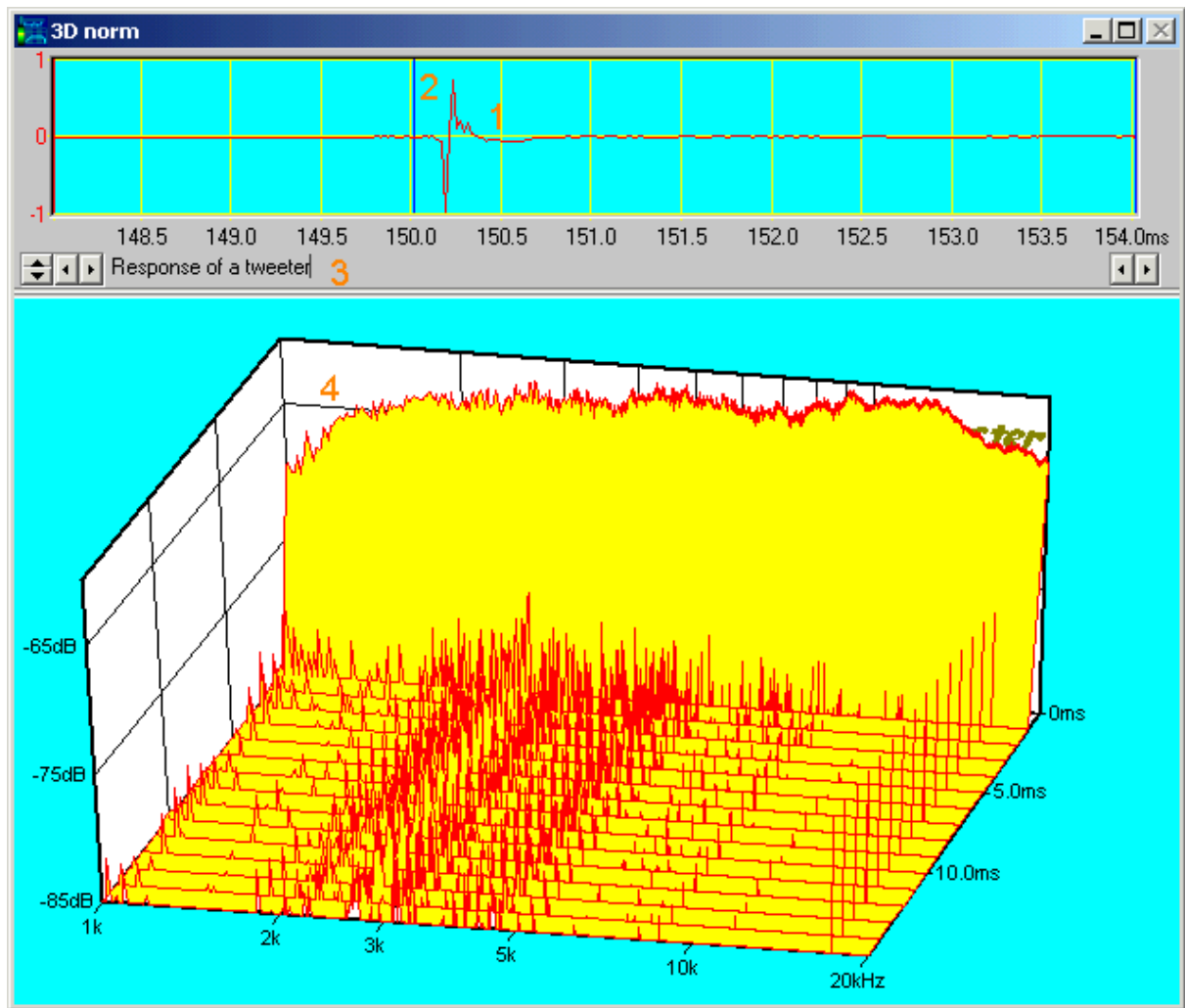


16 3D Zerfallsspektrum

17 3D waterfall plot

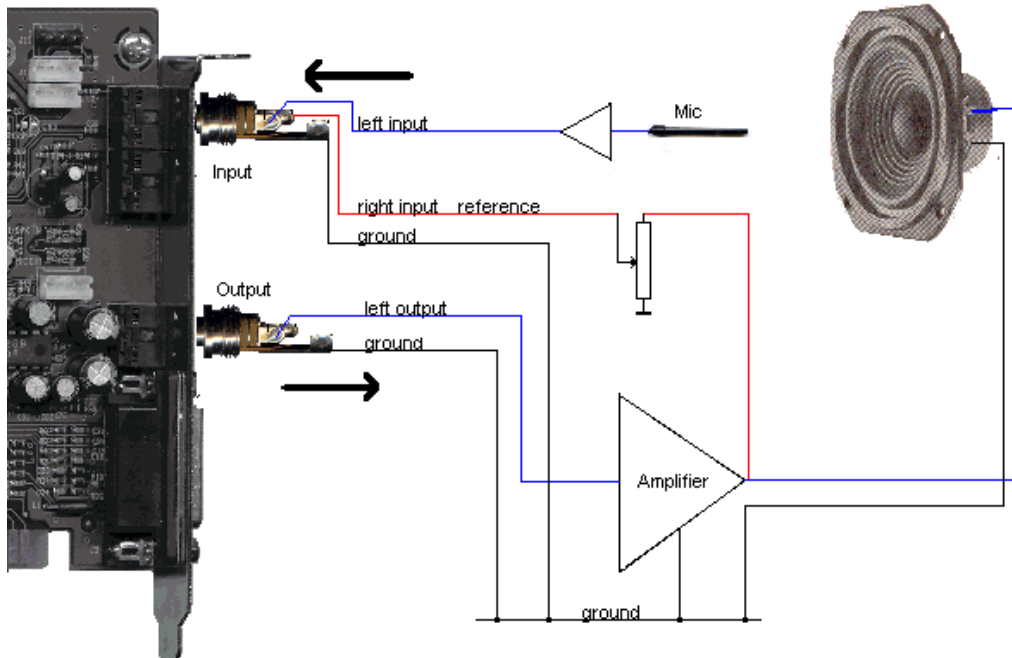
With this measuring mode you are able to measure impulse responses of loudspeakers and filter systems and its disintegration over the time. As a stimulus you can select between a dirac impulse or a MLS-Impulse (Maximum Length Sequence). To select, please click the setup button on the left, it's opens the Sound-Setup.([see here](#))

Example: Impulse response of a tweeter



1. Impulse response in the time domain. parameter for the time diagram see [here](#)
2. FFT-Window, further parameters see [here](#)
3. Space for your comments and remarks.
4. Impulse response in the frequency domain, parameter for the waterfall plot see [here](#)

Measurement schematic



Important hints:

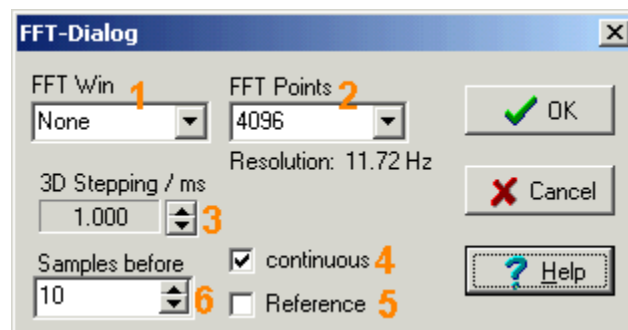
Reference input (right) only necessary if you use reference measurement. Please pay attention to the maximum input voltage of the used sound card. Ever use a attenuator before the reference input.

Please increase output level slowly.

Do not use a bridged power amplifier.

The left and the right output of the sound card is equally.

3D FFT Dialog



1. *FFT-Windows*, here better *none*.
2. Count of *FFT-Points*, from 64 till 32768. See below for the steps in Hz, which depends on the sample frequency and the *FFT-Points*
3. Steps of the frequency ribbons. The step size depends on the sample frequency
4. Choice between single or *continuous* measurement
5. Measurement with reference channel. Schematic - see above
6. Count of samples before impulse, this is the begin of the FFT

18 Soundeinstellungen

19 Sound adjustments

Menu Options



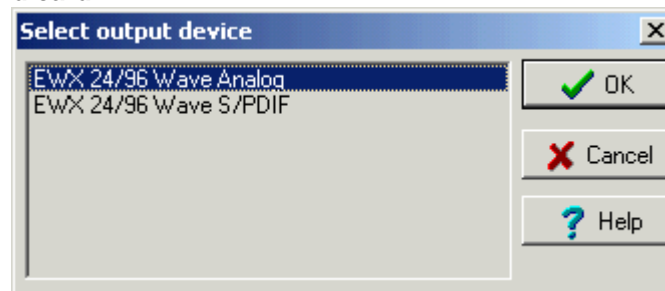
In the menu point *Options* you are able to reach the dialogs to adjust the sound card parameter, to switch on/off the mixer support and get the calibrations dialog.

Miscellaneous	Dialog for various setup values e.g. saving diagrams as text file
Alternative sound DLL (please email me)	This point appears if you have an optional sound DLL like ASIO
Audio-Out-Device	selection of the sound card, see here
Audio-Out-Parameter	Adjustment of the sound parameter (sample freq. ...) see here
Audio-In-Device	selection of the sound card, see here
Audio-In-Parameter	Adjustment of the sound parameter (sample freq. ...) see here
Audio-In-Offset	remove DC-Offsets on cheap sound cards see here
Wave In Channel	if Mixer Support = On, so switch the channel here.
Mixer Support On/Off	On/Off Mixer Support see here
Calibration	Calibration of the sound card input, see here

The selection of the sound card and the sound properties are also reachable with a double click on the panels at the bottom of the main window

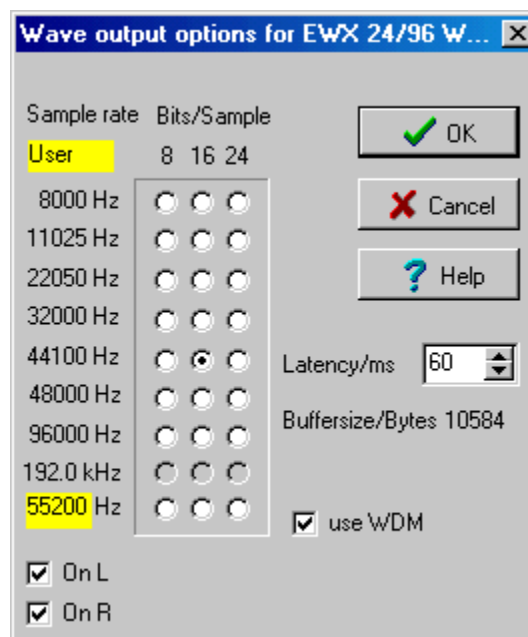


Selection of the sound card



You can select a different sound card for sound in and sound out.

Changing of output sound parameter



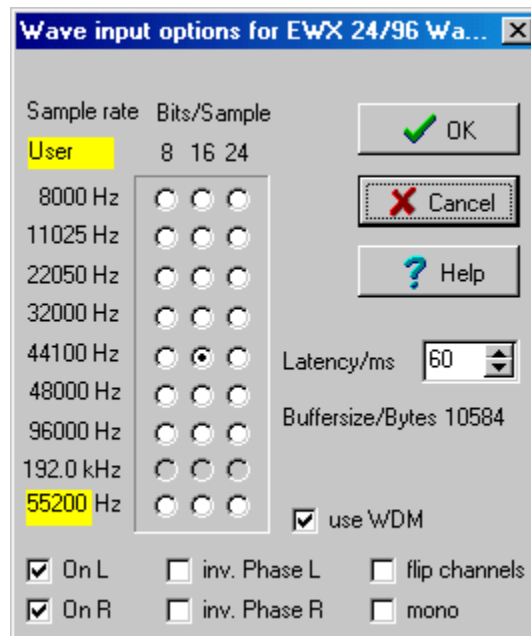
In the **matrix** you can select the possible **samplerate/datawidth** pairs

The **latency time** must be less then possible, but without any buzzing in sound.

A good value is 60ms, this value is preselected. The value depends on the computer performance, the sound card and the quality of the card driver.

With '**use WDM**' you select the WDM-sound card driver (WDM = Windows-Driver-Model). This may be necessary to select higher data width as 16 Bit. WDM-Driver are available since Windows ME.

Changing of input sound parameter



In the **matrix** you can select the possible **samplerate/datawidth** pairs

The **latency time** must be less then possible, but without any interruptions in the time domain diagram.

A good value is 60ms, this value is preselected. The value depends on the computer performance, the sound card and the quality of the card driver.

With '**use WDM**' you select the WDM-sound card driver (WDM = Windows-Driver-Model). This may be necessary to select higher data width as 16 Bit. WDM-Driver are available since Windows ME.

For data sampling there some more parameter available.

Both stereo channel are switchable separately **OnL / OnR**

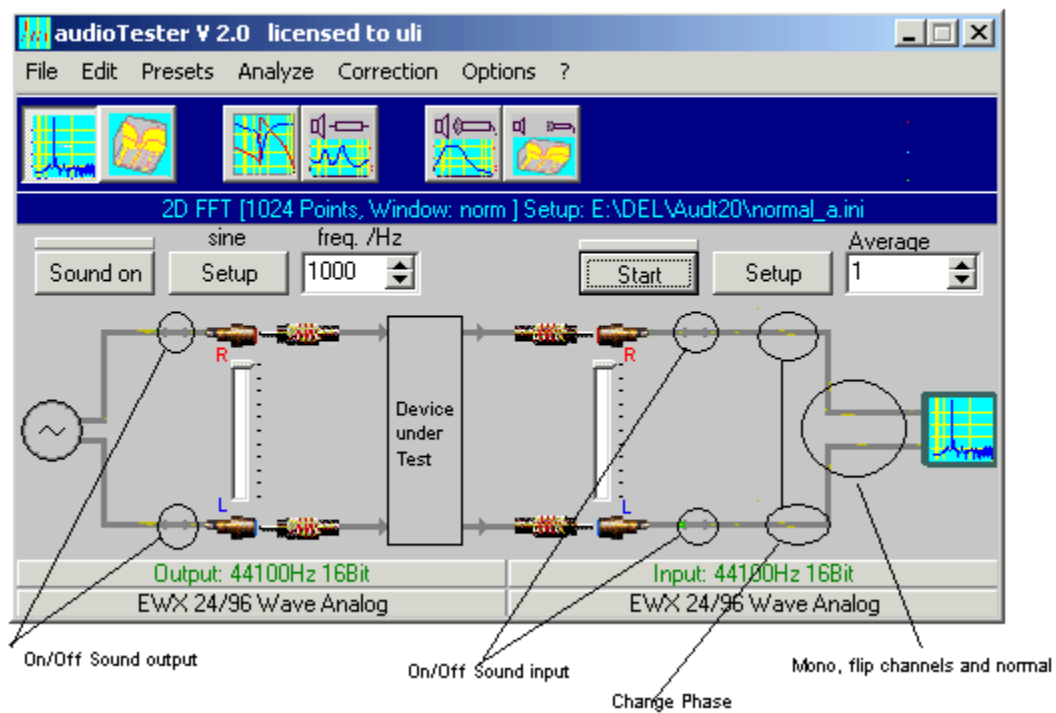
Both stereo channel you can inverted **inv Phase R/L**

The channel can flipped **flip channels**

With the **mono** selection both channels are added

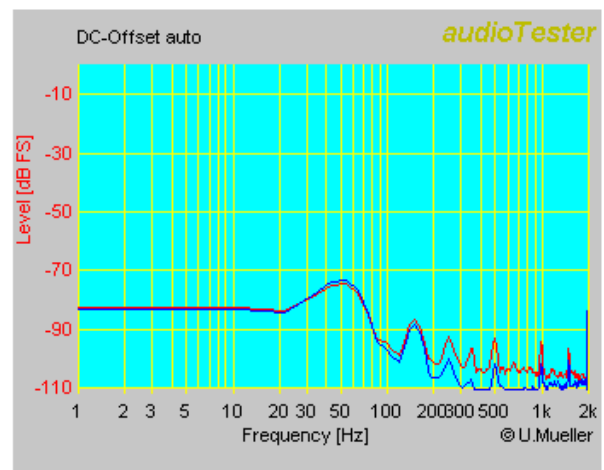
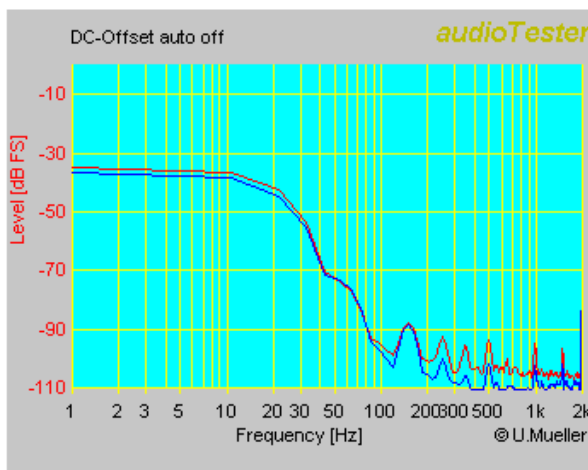
The selection above, you can also switch with a mouse click on the symbols in the main window.

Switch Areas



DC-Offset

Some sound cards have a DC-Offset in its AD-Converter. You can see that in the left picture. You are able to compensate this with the scroller or automatically, see the right picture..



Mixer Support

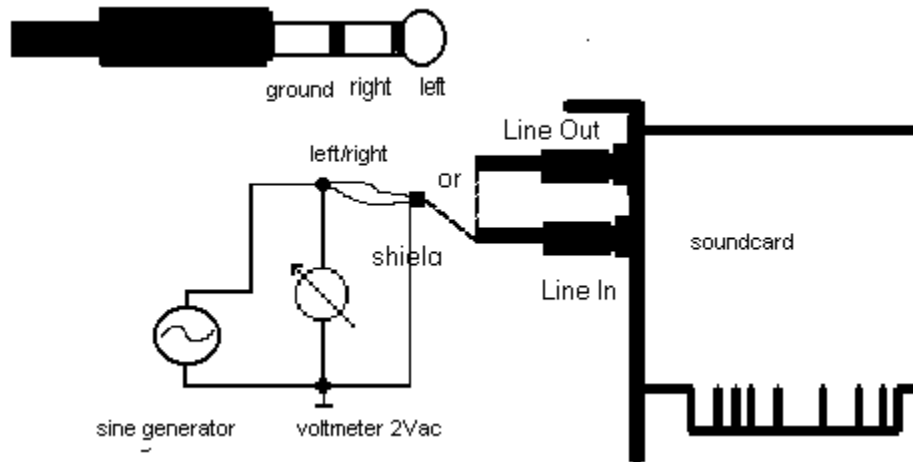
Mixer Support On -> the controller on the main window are linked to the Windows Mixer. The controlled input channel is selectable at the menu point *Wave In Channel*. The controlled output channel is always the *Volume Control*, at which only the channel *Wave* is selected.
 Mixer Support Off -> the controller on the main window are disabled, the adjustment must be made by hand.
 High quality sound card have often own mixers, select 'Mixer Support Off'

Calibration

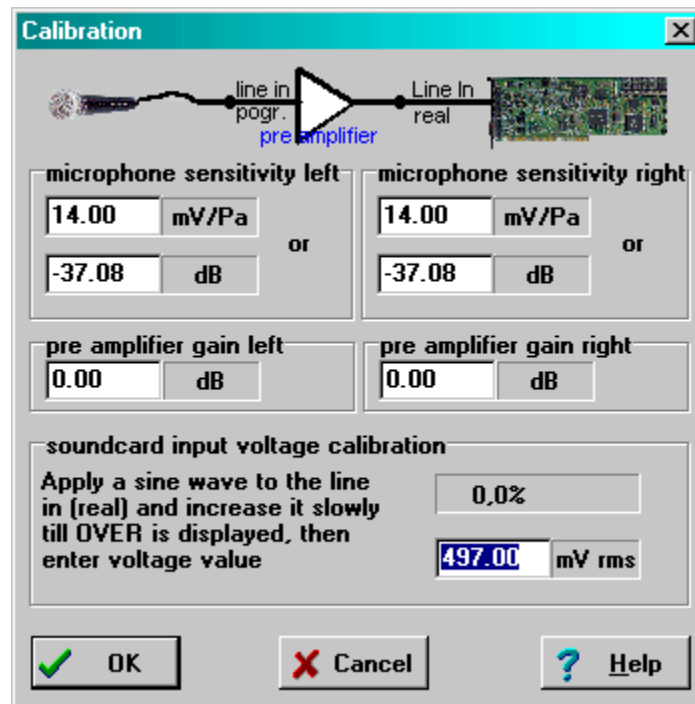
Before working with the **audioTester**, the Line - Input should be calibrated. To calibrate the Line - Input you need a multi meter with an AC Range of 2 Vac and a AC signal source, which has a sine waveform, for example the sine wave generator of the **audioTester**.

Please use a low frequency (like 50-60Hz) on the sine wave generator, because this is where the multi meter is most accurate.

The signal has to be wired to both the right and left channel of the Line - Input. After that adjust the pot for a voltage level of about 100mV . Then push the Line-Input Fader (Windows-Mixer) to the maximum . After calibration this is the **ONLY POINT** that is actually calibrated. This is indicated by **Level calibrated**.



In the menu options you have to choose the item *calibration*.



A moment later a red percent value (1) is represented, but don't concern yourself with this number, it is used internally by the program as a calibration constant.

Next increase the voltage with the pot until you see the text **OVER**. OVER means, that the A to D Converter of the sound card is overdriven (Beyond the maximum signal it can handle). The digits after OVER shows the number of overdriven samples. Please don't increase the level to much, better look for a reason why the values don't increase.

When the display changes between the percent values and OVER, make note the value on your voltmeter. Next write down the value you noticed in the input field (in mV). For example, a Soundblaster AWE 32 has a value of 260mV. (Soundblaster Live! around 490mV)

Over is enabled, if the level reached the last bit, at 16Bit resolution that if 0x7FFF and 0xFFFF. Some sound driver doesn't reach this values and stop before.

Please don't increase the level in this case, better is if you see that the percent values doesn't change.

Clicking the OK-Button will store the calibration value (also in the .INI file). The calibration value will be restored every time you start audioTester.

Hint:

You have to put in the rms value, which is shown by the voltmeter. Do not measure the voltage with a oscilloscope which will be peak to peak. It is also important that you do not measure a rectangular voltage with a voltmeter because unless the meter is true rms the value will be wrong

You are able to input the microphone sensitivity (2) in the unit mV/Pa or dB, you take the value form your microphone data sheet.

At (3) you must input the gain of the microphone pre amplifier. If you have no amplifier applied the value (3) is 0 dB

Both declaration are needed, if you will measure absolute sound pressure - dBSPL (SPL = sound pressure level)

0 dBSPL corresponds to $2 \times 10^{-5} \text{ N/m}^2 = 20 \mu\text{Pa}$ (Pascal) - 20Pa corresponds to 120dBSPL that very

loud.

Internally calculates the audioTester as follows:

$\text{PreAmpGain (dB)} + 20 \times \log(\text{INP (mV)} / \text{MicSens (mV/Pa)} / 2 \times 10^{-5} \text{ (Pa)})$

Correction Files

With the help of the correction files you can correct errors in the frequency response of the sound card (and any external devices). But you have to distinguish between the Input and the Output correction.

There is also only one input or output correction file available at the same time.

The correction file affects the frequency domain response.

It is active after the FFT Analysis in the spectrum analyzer.

It is only active in sine outputs for the output correction file.

The output correction is switched out, if there is a rectangle, a white/pink noise or User Wave Data, however the input correction can remain active.

Important:

If you switch the level higher while the Output correction is active, you have to decrease the digital level controllers to the maximal adapted level. This is important for the digital level controller of the sweep-generator in the Setup-dialog. It is also important for both of the digital level controllers of the Wave-generator.

The correction files are normal text files, where the values are entered in pairs. The data should have the extension *.cor. One line of the correction data consists of a frequency-value (Hz) that is separated by the **text separator** and followed by a level-value (dB).

You can change the **text separator** in the menu point /options/miscellaneous dialog (text separator).

Comment-lines are NOT allowed.

For example:

100 -4

200 -3

500 -1

1000 0

2000 -1

5000 -2

20000 -3

The correction values are applied like the values you find on your i.e. microphone measuring data sheet. The adjusted data values are a mirror image for the errors your device has. For example if your mic is -3 dB at 10000Hz the correction value would be 10000 -3. The actual data you view will be raised by the correct amount thereby producing a flat response. You can use up to 200 correction values pairs. You can load , enable/disable, edit and test correction data with the help of the menu Correction.

Begin with the lowest and end with the highest frequency you will measure.

How to load a correction data:

You must distinguish between the loading of an Input or an Output correction data file. With the corresponding menu you have the option to load a correction file. The extension *.cor is already automatically added. After loading of a correction file it is immediately active, this is shown by the menu Input/Output correction active. It is also possible to switch a loaded correction file on and off with the help of this menu.

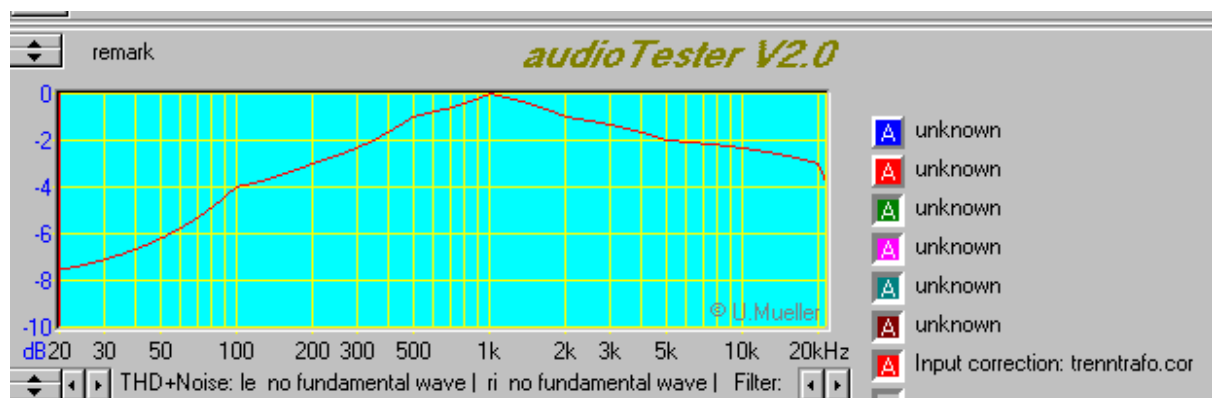
With the help of the menu Edit Input / Edit Output correction you can work with a correction file. If there is already a correction file loaded, it is edited. If the correction file is not loaded, the file dialog opens automatically. The edited correction file is then made active. If you are done with your work, you can test the correction file on the diagram, if you click the point Save & Test in the editor. The data is saved first then the correction factors are shown. Any data on the screen are overwritten.

How to test a correction data:

If you have loaded and activated the In/Output data, you can see the correction lines in the diagram, with the help of the menu Test correction file you can see these correction lines that are produced with the help of the corrections and Spline-Interpolation. The input correction file is presented in the color of the left channel; the Output correction file is presented in the color of the right channel. If you do a second editing or loading you have the option to do changes in a very easy way.

See the Diagram to the following input correction file.

```
10 -8
100 -4
200 -3
500 -1
1000 0
2000 -1
5000 -2
20000 -3
22000 -4
```



Caution!


It is important to know that the output level can only be corrected by up to 10 dB. So the lowest value in the Output correction data has to be -10dB. Areas of the curve (different frequencies) with the same level (for example 0dB) have to be assigned values because of the Cubical-Spline-Interpolation will create errors in the correction curve if only a few points are present. The exact degree of error will be shown when you test the correction.

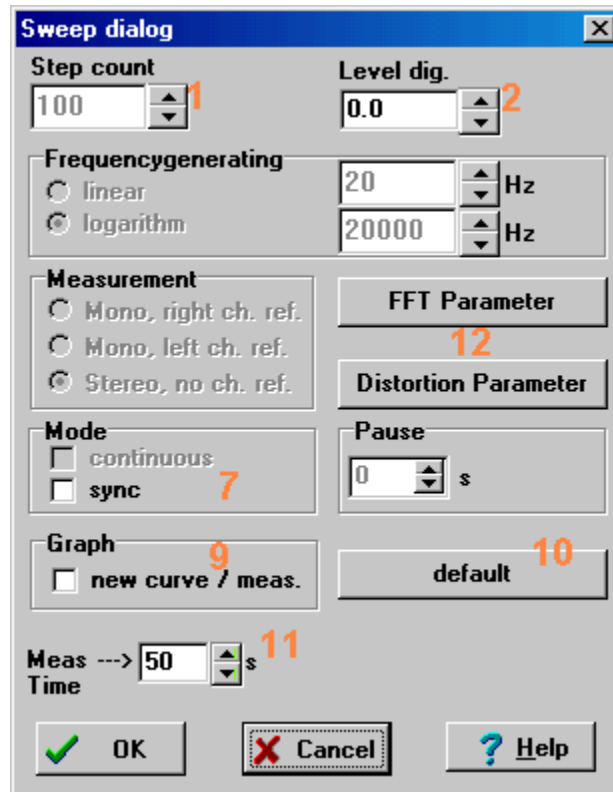
20 Klirrfaktor Messung

21 Distortion Measurement

Distortion Measurement in frequency domain

With the Version 2.0.c and later you can also measure the distortions over the

frequency. Therefore we have a new tool button  *distortion measurement*. The manner is like the normal sweep measurement, but You only can use a stereo measurement (see below). In the sweep dialog are the buttons (12) added.




With the button *FFT Parameter* (12) you call the known *FFT-Dialog*. The FFT-Points should be greater than 4096. The FFT-Windows should be Rife-Vince 3.

With the button *Distortion Parameter* (12) you call the extended *Analyze-Dialog*.

You do not use the asynchron measurement (foreign sweep, see Sweep Dialog p.t. 7 above), it doesn't work here..

Distortion measurement vs. Power

With the version 2.2 , you able to measure distortions vs. power.

Therefore we have new tool button  *Power Distortion Measurement*.

The procedure is a little bit other than the distortion measurement vs. frequency.

Please notice all the hints in the following text, at this measurement you could be damage your sound card, the computer and the measurement object (the amplifier)! Please read the text below and notice the hints for the test construction.

First Hint:

The sine level for the measurement is generated digitally, so if you use only a 16Bit sound card, it is possible that you measure self produced distortions instead of the distortions of the measurement object.

For example: you must produce a output level from -40,5 to -0,5dB for a measurement from 1mW to 10W. You use only 9-10Bit for sine generating at low levels and that makes the signal not even better. Therefore, use a 24Bit sound card for best results.

The measurement:

First, output of a test tone, with an adjustable level (default -30dBFS) and an adjustable frequency (default 1kHz).

Measurement of the voltage behind the amplifier.

With the load resistor we calculate the test power.

With this result, we are able to calculate the test tone level range.

You see the procedure, if you open the debug window.

For example :

Selected is a measurement from 1mW till 10W, test tone level is -30dBFS

Debug output:

1. 16:12:53 Expected voltage:0,0894427V - 8,9442719V @ 0,001W - 10,000W / 8,00Ohm
2. 16:12:53 Expected Input Level: -40,5dBFS to -0,5dBFS
3. 16:12:55 Determined power(L,R): 0,0253300W 0,0249249W @ -30,0dBFS Test Out level
4. Chosen Out Level for the selected power range: -44,0dBFS / -4,0dBFS

1. line: With the formula $P=U^2 / R$ we calculate the expected voltage at the input.

Here it is possible that you receive an error message:

The input voltage transcends the calibrations voltage of the sound card. The sound card is overloaded and will be damaged !

The measurement breaks immediately.

This is clear, if you see that the max. input voltage of a sound card is around 1V .

With 1V the measured power is $P=1^2/8 = 0.125W$.

So use a resistor divider and put it before the sound card input and recalibrate the audioTester.

You never use the windows mixer in this case to reduce the input level !

2. line: Here is shown the expected input level in dBFS with the results of (1) and the stored calibration level

3. line: The test tone is evaluated. With the -30dBfs test tone the power level is 0,025 W and so ...

4. line: the digital test level must be -44dBFS to -4dBFS for the selected power range.

Here it is possible that an error message appears:

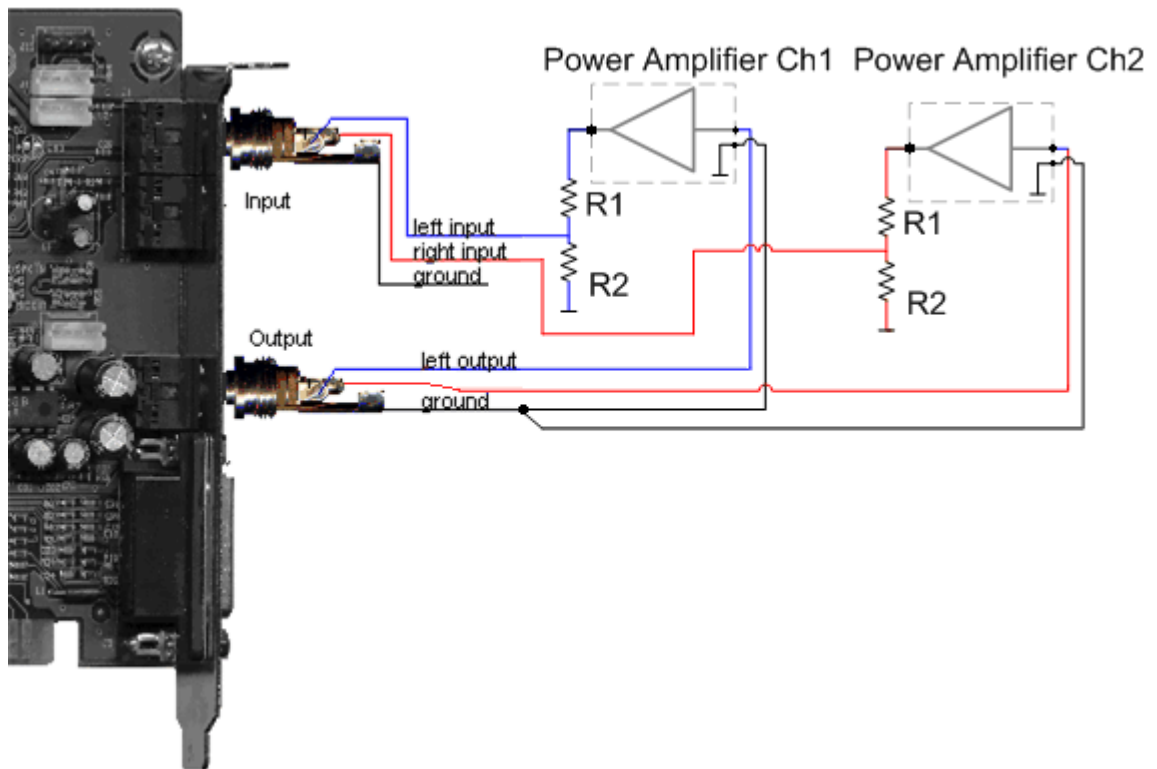
The digital test level never can be greater than 0dBFS.

The measurement breaks immediately.

What can we do? The test tone level can be increased with the windows mixer, with the mixer delivered with the sound card or via increase of the gain of the measure object.

Now the measurement begins.

Measurement schematic:

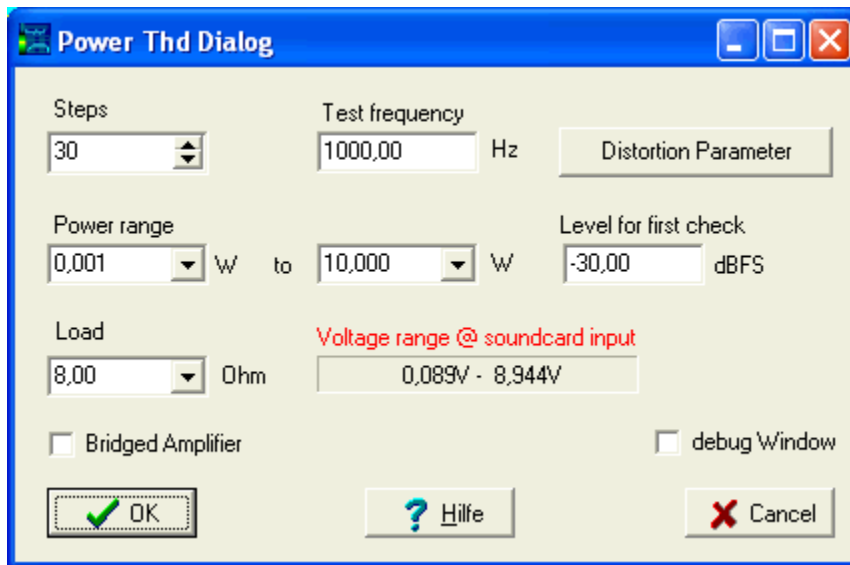


With this measurement schematic you can measure without any risk even a bridged amplifier. Avoid in any cases to connect the ground plug of the sound card input with any amplifier outlet! If your measurement object is a bridged amplifier, please activate the corresponding check box in the Power distortion dialog.

The resistor divider depends on the expected voltage at the sound card input:
 Example: The input sensitivity is 1V the max. power of the amplifier is (P) 100W at (R) 8Ohm.
 Max. voltage from the amp. is $U = \sqrt{P \times R} = 28.3V$ The voltage must be divided down to 1V.
 Therefore we choose R1 with 3.3kOhm and R2=120Ohm and get a ratio from 28.5
 The input resistor of the resistor divider is $3.42k\Omega \parallel 50k\Omega$ (of the sound card) = 3.2kOhm
 this is no problem to measure at an amplifier output.

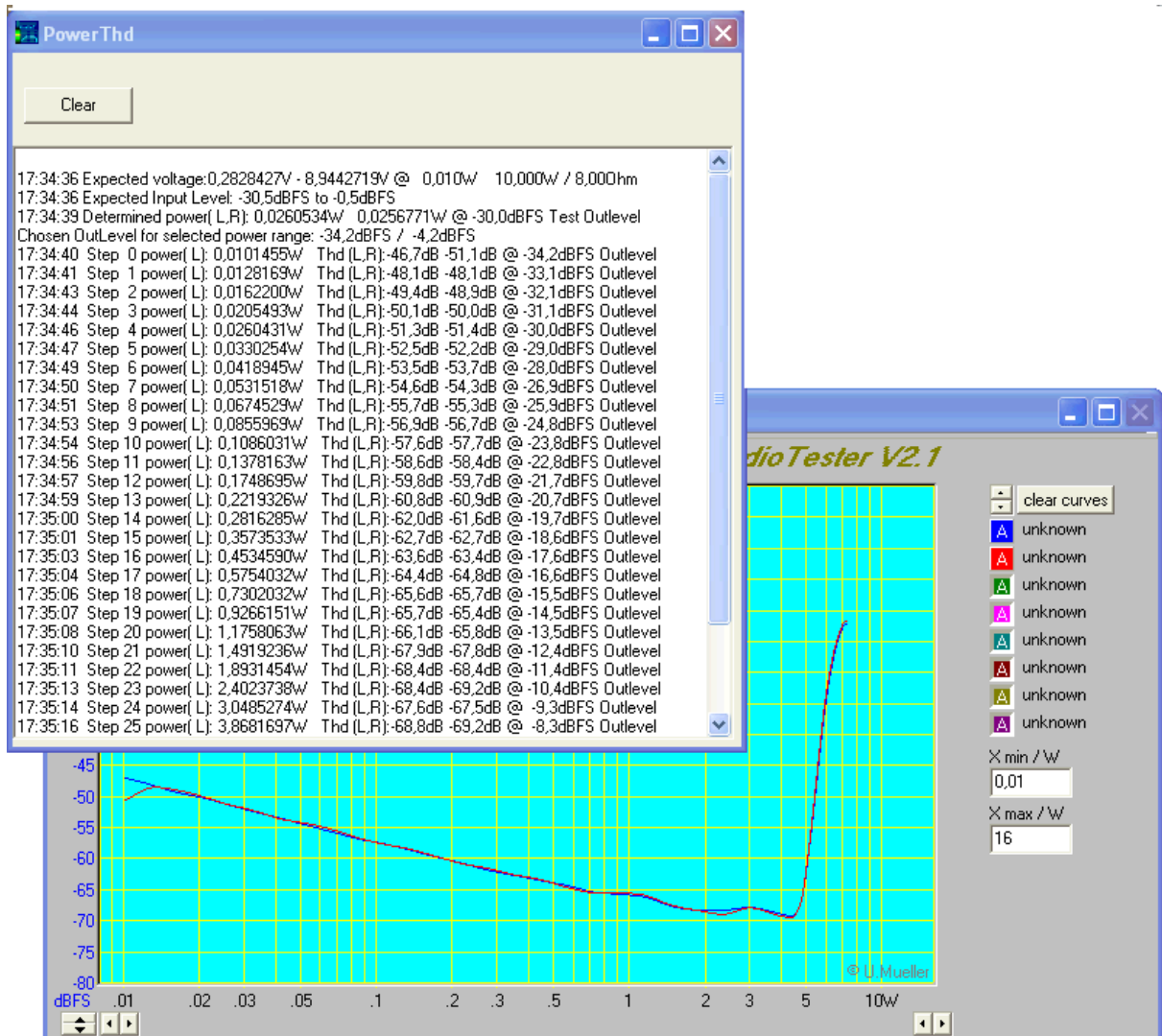
Now the sound card with the resistor divider must be recalibrated!

Power Thd Dialog



Steps:	Count of the measurement steps
Test frequency:	Frequency of the measurement tone
Distortion Parameter:	Dialog for the distortion measure parameter
Power Range:	Power range to determine the distortion
Level for first check: to measure	Digitally test tone level for a first measurement to determine all the parameter
Load:	The load resistor, to calculate the power. Never use a load speaker while measurement, the may be damaged.
Voltage range	Calculated range of voltage at the sound card input while measurement.
Bridged Amplifier only half of the voltage,	If selected, the power will be corrected (Px4). Is the measurement schematic like above, we measure at the bridge amp. and so only a quarter of the power. This will be corrected.
Debug Window an debug help, if there are	A debug window appears. You can see all terms while measurement. It also is problems while measurement.

Example: measurement with debug window



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