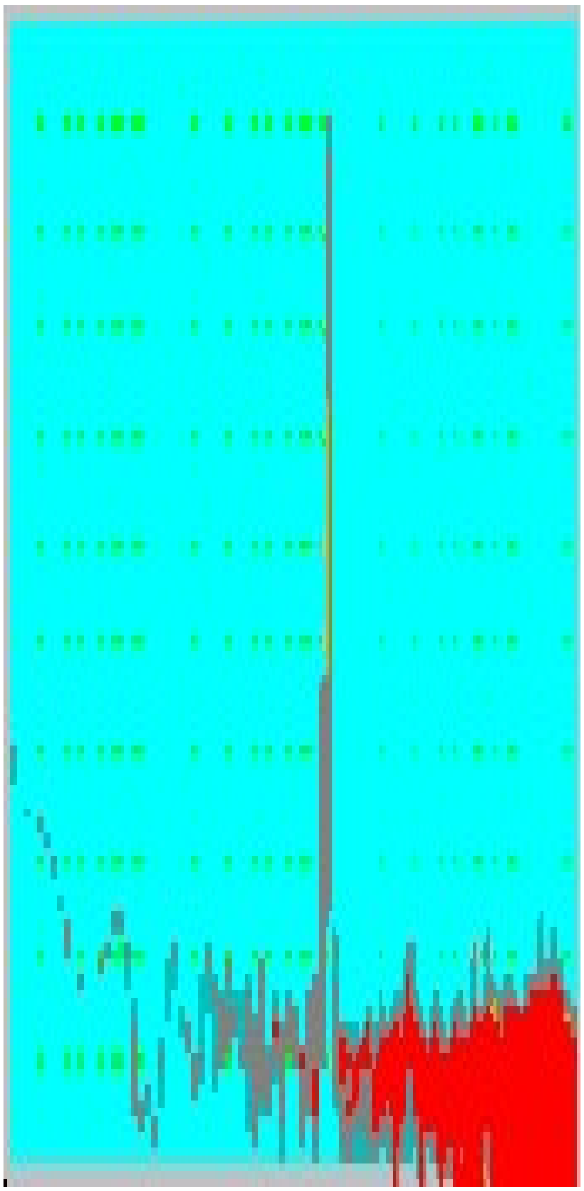


audioTester V3.0

(c) Ulrich Müller

1 audioTester V3.0

The ***audioTester*** is a low cost but high quality tool in audio measurement.

- 2D Spectrum analyzer, level and distortion measurement
- 3D Spectrum analyzer
- Sweep measurement, amplitude and phase
- Impedance measurement of speakers
- Determine of Thiele & Small Parameter
- Distortion vs. frequency
- Distortion vs. power and level
- 2D Impulse measurement
- 3D Impulse measurement - waterfall diagrams
- Oscilloscope
- Integrated function generator
- Time Domain Filter for input and output

What's new in Version 3.0:
- Rework Sound DLLs
- Audio properties access via status bar
- Sound output with audio files (*.wav, *.mp3)
- Time Domain Filter for input and output
- Extended functions for distortion vs. power and level
- New design
- New Sound-DLL **Direct Sound Version**
- Better support for **WIN 7** and **Win 10**
- Integration of the oscilloscope
- Improved Help with many measurement examples
- Wiring diagram and remarks for every measuring mode

System requirements
- Windows XP or Windows 7 / 10
- DirectX 3.0 or higher
- CPU 1.6Ghz or faster
- RAM 2 GByte or more
- Installed size 38 MByte on HD/SSD

Acknowledgements
- Windows® is a registered trademark of Microsoft Corporation in the United States and other countries.
- This product includes software developed by the OpenSSL Project for use in the
- OpenSSL Toolkit http://www.openssl.org
- SoundASIO DLL used the openasio.dll by www.martinfay.com
- SoundDirect.DLL used the bass.dll by www.un4seen.com

- All named firms and company names are registered trade-marks of the respective companies.

1.1 First Start

Read this first, it answers some questions for licensed users

In ***audioTester V3.0d*** the settings and the key-file are stored in
c:\Documents and Settings\user\Application Data\audT30d ('user' is your user name)
In Windows Vista and Windows 7:
C:\Users\user\AppData\Roaming\audT30d

The old Key-File of the Version V2.2 is no longer valid. Please purchase a new one see here.

1.2 Full version

1.3 Main window General Trouble Shooting

The main window

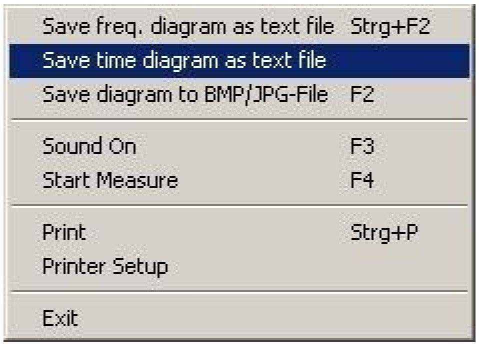
On the left side are the buttons which initiates the several measuring modes.
For each measuring mode, there are separate help page with wiring diagrams and hints (right mouse click on the button)

The button 'Sound on' starts the sound output from the sound card.
With the button 'Setup' you specify the waveform, the frequency and other parameters. Details [see here]()
In the input field you can enter the frequency directly.
These buttons are not available in all sweep modes.

The button 'Start' starts a measurement, the system is now sampling input from the selected sound card.
The button 'Setup' selects the different parameters for the specific measuring mode. Please refer to the help page of that measuring mode.
'Average' in some modes, enables an averaging of an number of measuring results.

Menu

File



The diagram can be stored as a text file (for export to excel sheets) or as picture (BMP and JPG). For special setups see here
All the selected curves in the frequency diagram are stored.

Instead of the windows buttons, you can start sound and measuring with F3 and F4.
Print prints the diagram.

Edit

You are able to copy the diagram into the clipboard.

Presets

With the menu item *Presets* you can load and store presets.
The preset files have the extension *.atp (**a**udio**T**ester **P**resets)
Save: Select *Presets/Save Preset* and store the your actual settings with a file name of your choice.
Load: Select *Presets/Load Preset a*nd load a stored preset.

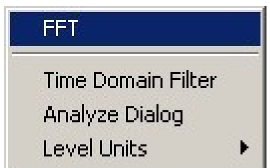
Predefined presets:

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

Measure

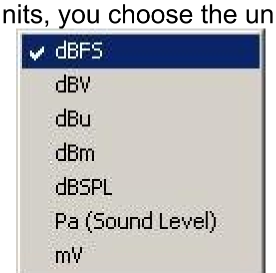
The entries activate the measuring modes exactly as the buttons on the left of the screen.

Analyse



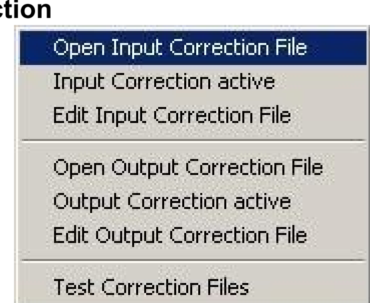
FFT opens the FFT-Dialog to adjust FFT-Parameter, see here
Time domain filter opens a dialog to modify and switch on/off filter for the time domain for input and output
Analyze Dialog opens a dialog to adjust the different analyzes parameters, see here

Level Units, you choose the units of measure of the results



For all units, except dBFS, the system must be calibrated for exact results.

Correction

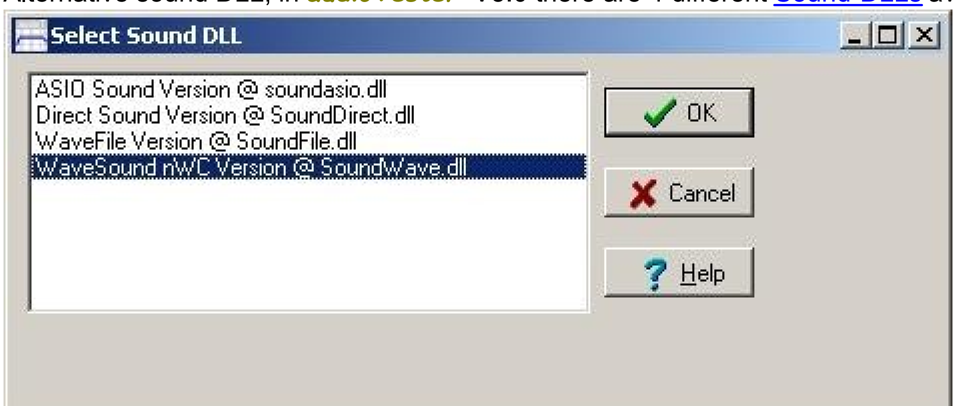


Here you are able to correct the frequency response of measuring microphones or bad sound cards, see here

Options

Miscellaneous parameter, see here

Alternative sound DLL, in ***audioTester*** V3.0 there are 4 different Sound-DLLs available.



In V3.0 ***audioTester*** is 'Direct Sound Version DLL' the default Sound DLL
This Sound DLL requires Microsoft DirectX Revision 3 or later for output.
This DLL works only with the Float-Format (24+8Bit).

Audio-Out-Device, select sound card for output.
Audio-Out-Parameter, Sample Freq., data width, latency ...

Audio-In-Device, select sound card for sound in
Audio-In-Parameter, Sample Freq., data width, latency ...

Audio-In-DC-Offset, compensated DC-Offsets of cheap sound cards

Link AudioIn/Out-Dialogs, if selected there is only one dialog for the sound card selecting and one for the sound parameter available

Mixer Support, on/off of windows mixer support
Wave In Channel, if Mixer Support is on then select the input channel

Calibration, this opens the calibration dialog

Help

Contents, call this help file

Wiring diagrams, show the wiring diagrams and some hints

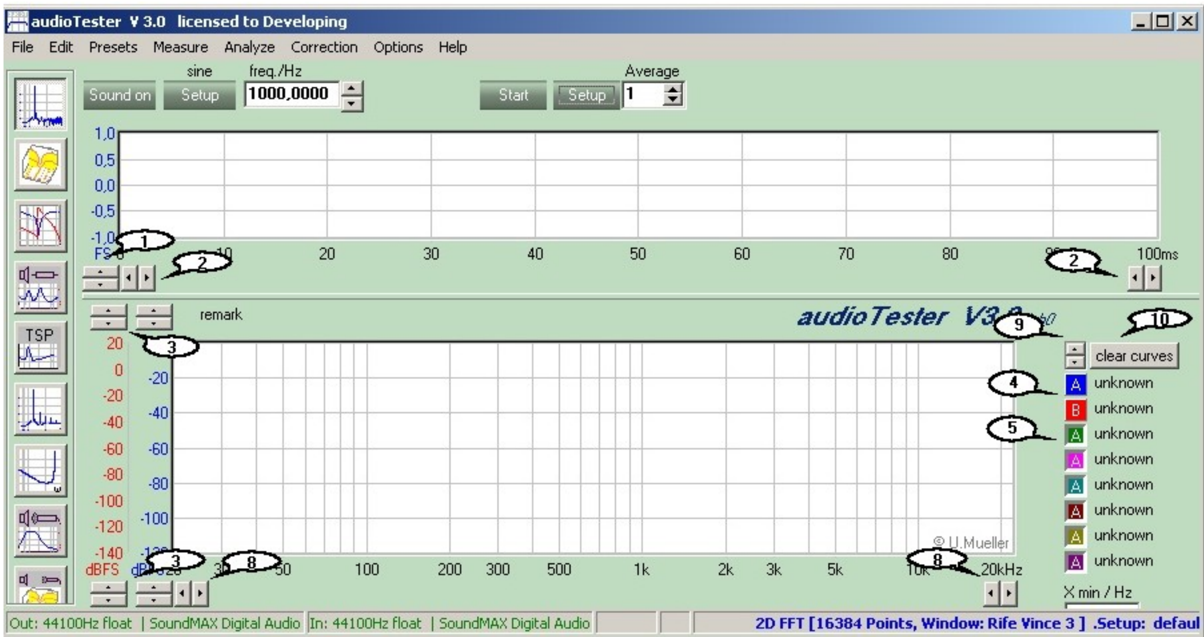
Purchasing audioTester, a link to the order page. Only visible in the shareware version.

Enter temporary license code, input dialog for the input of the immediately received key from shareIt .

Info-Dialog:link to www.audiotester.de , sending emails to the author and starting of the windows explorer in the home directory.

The diagram

The diagram is the main output medium of the ***audioTester***
All changes in the diagram can be made while running. All single curves can be set individually like colour, line width, display filter...
A right click into the diagram opens the diagram dialog (background colour, grid colour, x-axis lin/log ...)

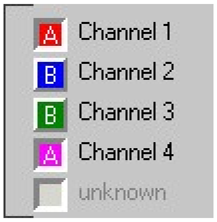


1. Adjust of the Y-axis (symmetrical) for the time diagram
2. Adjust of the X-axis values for the time diagram /ms
3. Adjust of the Y-axis values, you see the Y-axis groups A+B separately (see curve dialog)
4. Curve buttons, you see two selected curves, that means, the two curves are receiving measurement values and they are visible.
   The characters in the buttons shows the group of the Y-axis scroller. If you change the scroller to, for example, group A, then change all of the curve, which are part of group A. Example see below
5. If the button is lowered, then the curve is visible, but receives no measurement values. The curve 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. If the button is 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
9. Moving the selected curve up and down to switch to a new pair of curves.
10. Clearing all curves in the diagram see here

Moving and Addition of curves
Via drag'n drop it is possible to move a curve from one place to an other. Therefor you click on a curve button and drag it to an other button and drop it. All the properties are copied except the curve colour.
If you hold the Ctrl Key while dropping the curve, they will be added. The addition is calculated included

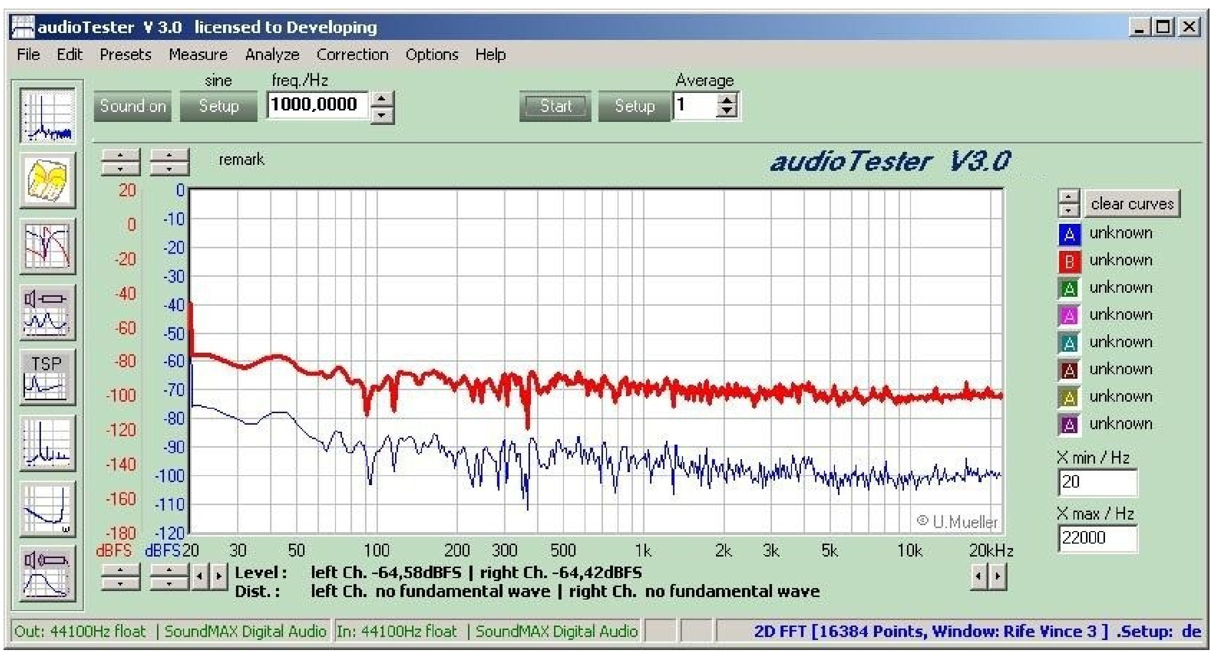
the phase, which is stored within the internal curve properties.

Example for selection of curves

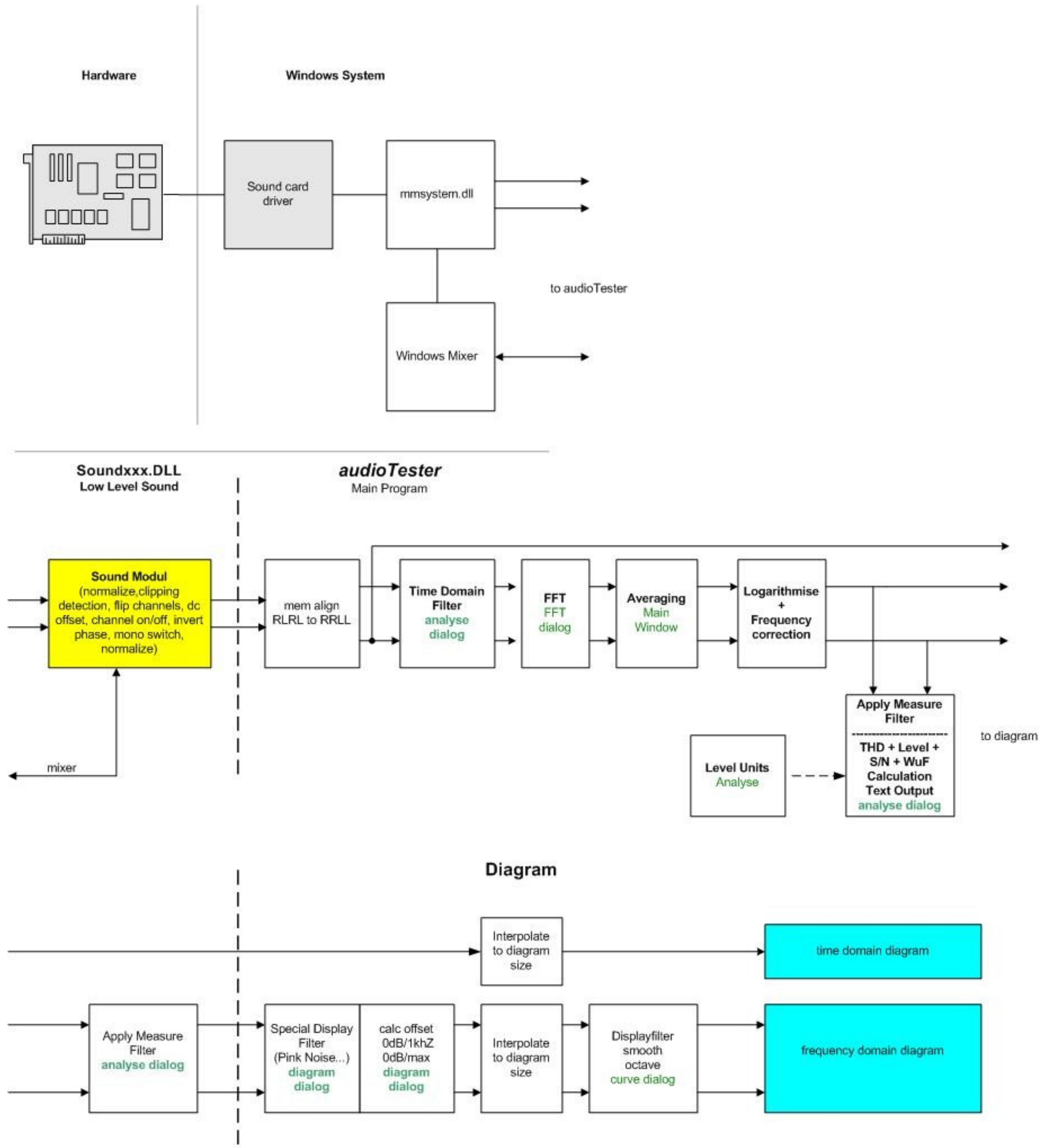


In the figure above, you see the curves of the channels 1-4 visible, the channel named *unknown* is invisible. The channels 2+3 are selected, they are ready to get the actual measurement values. Channel 1+4 belongs to the Y-axis group A and channel 2+3 to group B.

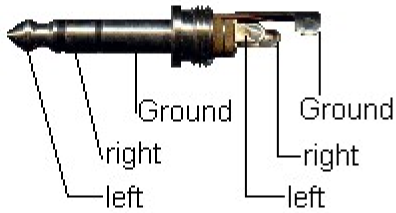
In the diagram below you see two curves with the same data values. The differences are the line width and the Y-axis scaling.
All this is set in the corresponding curve dialogs. A curve can be also the sum or the product of 2 curves, see curve dialog



Function diagram audiotester software



Phone jack



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*** (You will found it in the Info Dialog). 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 !
Try to increase the latency value in the Sound-Parameter dialog
Please see that there is no Interrupt-Sharing between any PCI-Cards.
Check this with the windows control panel.
Change this in the BIOS Setup.

Program shows not all details, and works in a strange manner !
Open the info dialog, got to the last entry there and click the displayed data path.
The windows explorer appears and then delete the file *audiotester30a.ini*
The data path should be *c:\documents and settings\user\applications data\audiotester30a* ('user' is your user name)

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, but rather read out the last value and calibrate it.

Problems with the registration !
Like any other software authors I must protect my software for illegal using :
Therefor every customer gets a key-file: The key-file has a size 512 Bytes and contains the name of the customer and same cryptic data, no program code or virus!
The key-file normally comes via Email normally and is named *key.bin*,
You must copy into the directory: *c:\documents and settings\user\applications data\audiotester30a* ('user' is your user name)
Or you drag'n drop the file onto the main window.

The headline shows *Shareware Version !* ?
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.

Hard disk damaged, Key-File was gone ?
No Problem, send a mail - a new key-file is coming.

Program shows 'cracked version' ?
Either the key-file is not made by me (then don't mail me),
but if you are a real customer, please send me a mail, I will send You a new key-file.

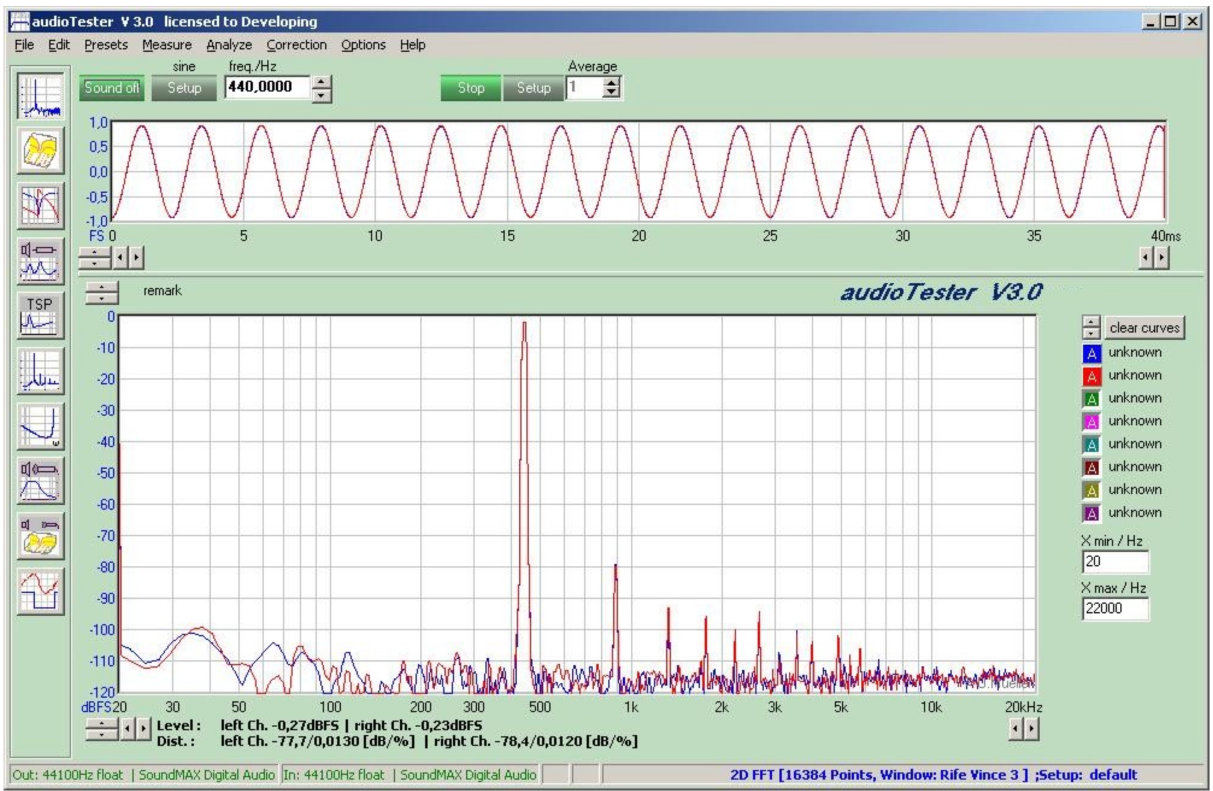
What is the license-key ?
A huge part of my program is selling with the help of the shareware-provider **www.shareit.de** . After You have buy a license you immediately get (since august 2003) a license-key in form of a string . This string (i.e.
D359C5CDBDCF577264B05DC8FDD88424545AB6653C379BF661614757FAFD24BF069363B2AE1823D90)
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.sharit.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.

2 Spectrum analyzer

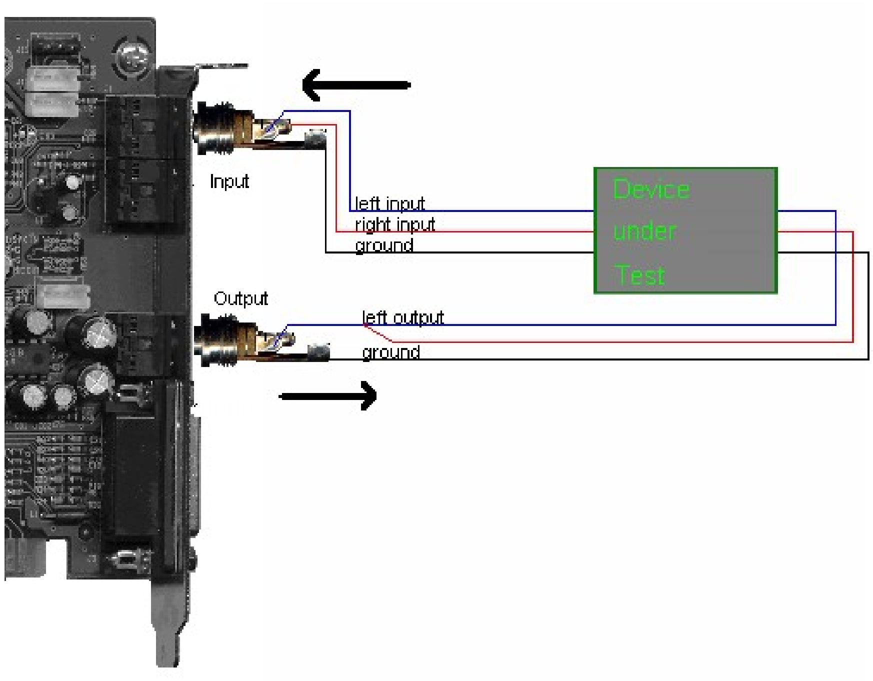
2.1 2D Spectrum analyzer

Features:
- FFT points: 64 - 1048576
- 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 ...

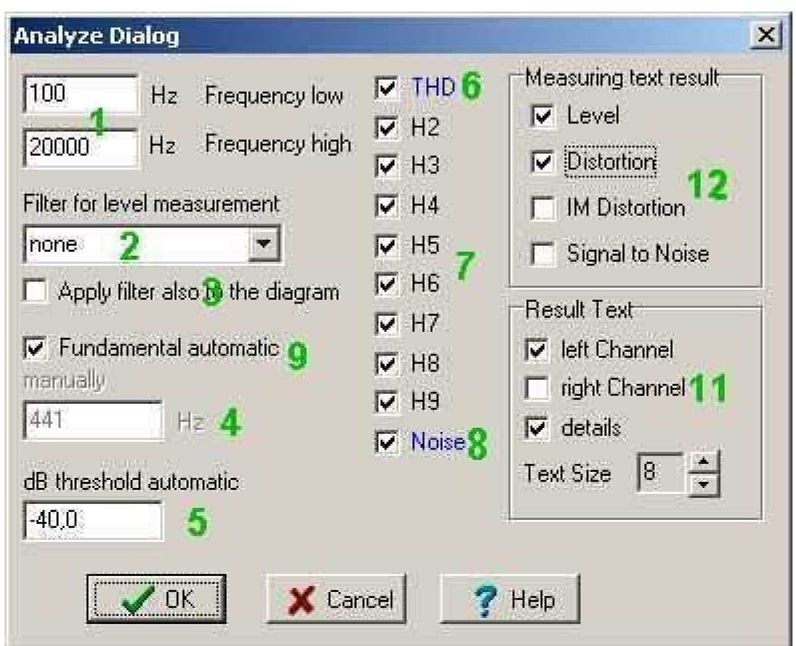


Description:
- Start/Stop sound output
- Setup sound output
- Start/Stop spectrum analyzer
- Setup spectrum analyzer
- Count of the averages
- Signal form, adjust by setup)
- Channel on/off with sound parameter
- Adjustable In/Out level, only if mixer support = on
- Display for the analysis parameter

Connection to the test device



Analyze Dialog



1. Frequency low
This is the lower bound of the frequency while calculating the noise component.

1. Frequency high
This is the upper bound of the frequency while calculating the noise.

2. Filter
You apply these following filters, for 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 additionally to the diagram

4. Fundamental Wave (for manual setting)
If you don't select for automatic Fundamental Wave searching, then you can set the fundamental frequency here.

5. Threshold value for fundamental wave detection
Threshold Level value, for searching for 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 measurements only up to SF/4 (eg. 11kHz at SF 44.1kHz)
Valid H3 measurements only up to SF/6
Valid H4 measurements only up to SF/8
etc.

8. Additional measurement of noise

9. Fundamental Wave automatically search
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.

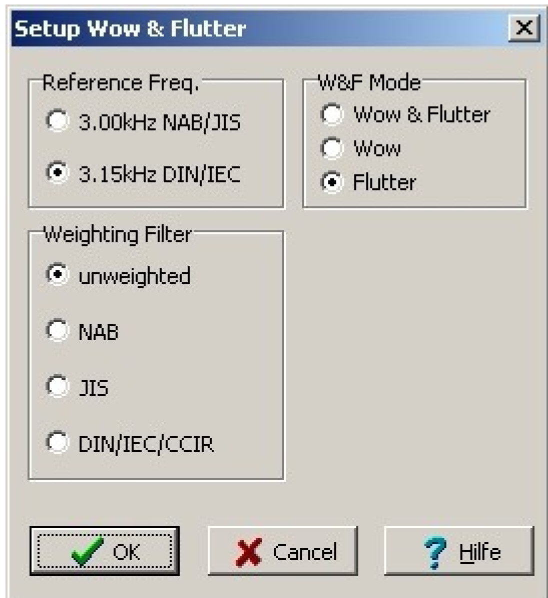
With the THD Analyses dialog you can determine the parameters of the THD+N and the level measurement. THD+N means Total Harmonic Distortion plus Noise.

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

11. and 12. You can choose the text outputs of measurement results: level, distortions, IM-Distortions, S/N and the Wow&Flutter measurements of tape recording machines.
It is possible to select more than one result text line below the diagram.

In the group **11** you can choose the font size and the result details (applied filter, fundamental frequency ...).
For Wow & Flutter Measurement there are Presets available with usefull entries for the diagram and the FFT Size (128k for optimal frequency resolution)
Here, Minimum requirements are CPUs faster than 1.6GHz

There is a special dialog for the Wow & Flutter Measurement.



In the group *Reference Freq.* you choose the fundamental frequency of the test tape (test cassette).
3kHz normally for measurement of the standards NAB and JIS
3.15kHz for DIN45507, IEC 386 and CCIR 409-2.

In the group *Weighting Filter* you can choose the filter for measurement

- unweighted - no filter
- NAB - measurement standard NAB Rec.
- JIS - measurement standard Japan Industry Standard
- DIN/IEC/CCIR - measurement standard DIN45507, IEC 386 und CCIR 409-2

Measuring method Level
Between the frequency low and high is calculated the rms value. It is possible to use a filter for the calculation.

In the menu point **Analyse/Level Units** you can select the level units:

dbFS
Level is related to Full scale (FS),
-128/+127 at 8Bit   -32768/+32767 at 16Bit   -1/+1 at float

dbV *
Level is related to 1V   0dbV = 1V

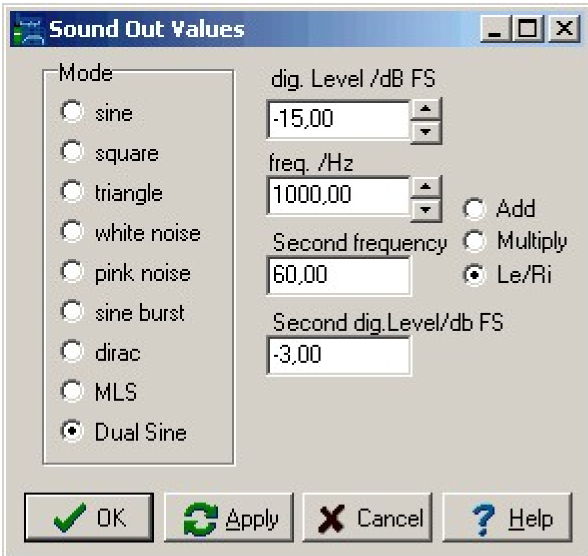
dbu *
Level is related to 0.775V   0dbu = 775mV

dbm *
Level is related to 1mW an 600Ω = 775mV
0dbm = 1mW/600Ω

*) The system must be calibrated before selecting one of the last 3 levels

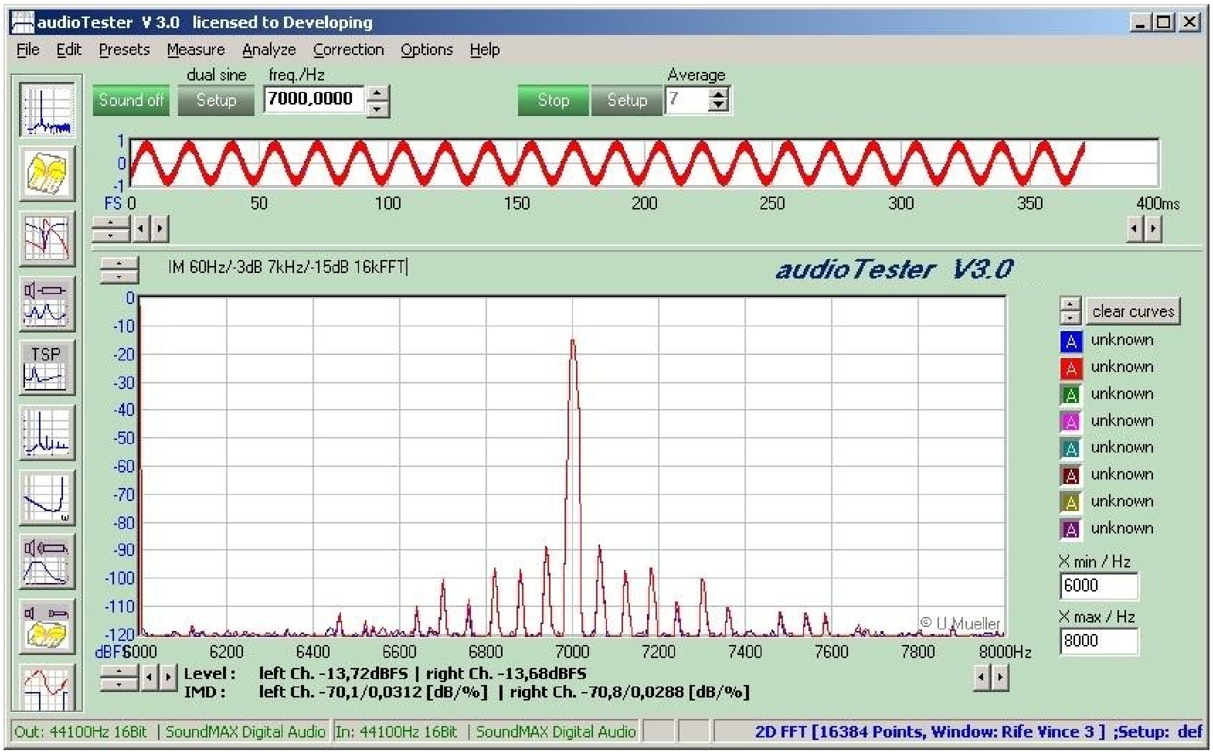
Measurement method THD+N
The fundamental wave is removed from the frequency spectrum, then the effective voltage value over harmonics d2 and the noise between the frequencies 'low' and 'high' are summed. Then this value is divided by the total effective voltage value (that means with the fundamental wave and without any frequency spurs) The result is the THD+N.

Measurement method Inter modulation Distortions
Please select in the sound dialog **Dual Sine** and enter at **freq/Hz** of the **main frequency** (eg. 7kHz) and at **Second frequency** for the interference frequency
(eg. 60Hz). The IEC 268 Part 3 says that the interference frequency should be 12db louder than the main frequency.
Eg. dig Level = -15dB   second dig. interference Level = -3dB



With the radio buttons add, multiply and Le/Ri you can change tone modulation
add: the sine waves will be added (see **Inter modulation Distortions**)
multiply: the sine waves will be multiplied
Le/Ri: the first sine wave is applied to the left channel and the secondary sine wave is applied to the right channel

Example measurement IM-Distortions:



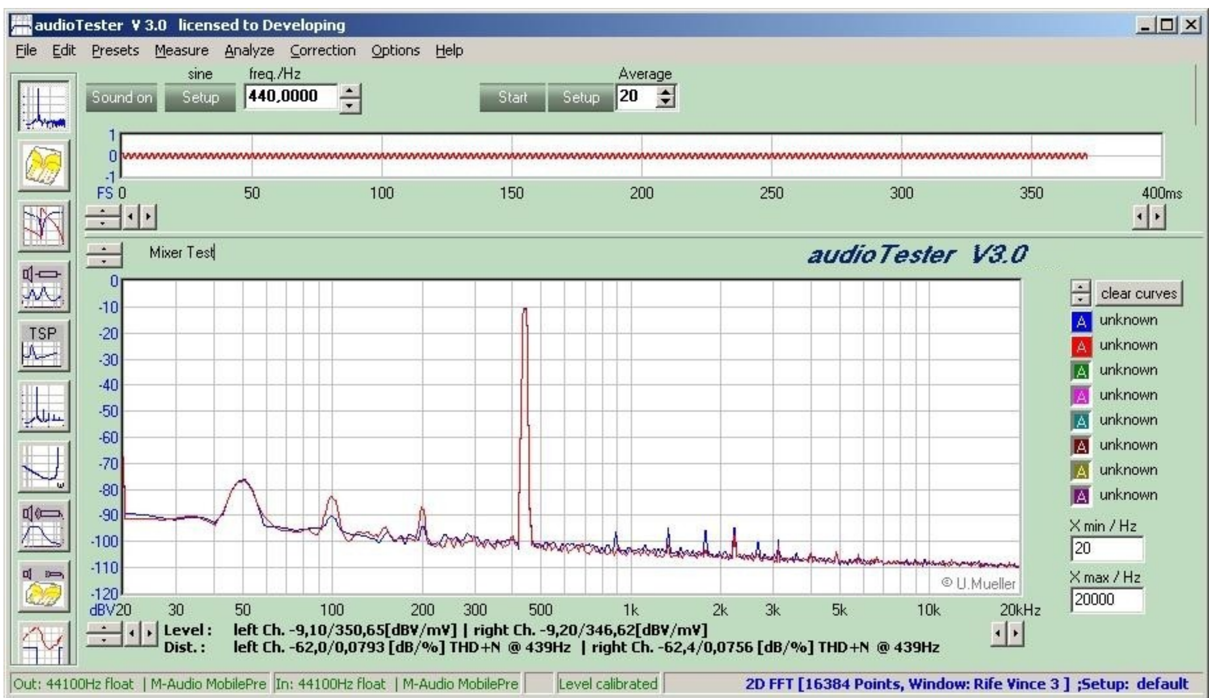
Examples:

Distortion measurement with an old mixer:
Test device: ADC Disco Mixer SX-90
Sound card: M-Audio mobile Pre
Before the sound card input there was applied a voltage divider 10k / 680ΩΩΩΩΩΩΩΩΩ,
to avoid an overdriven sound card input

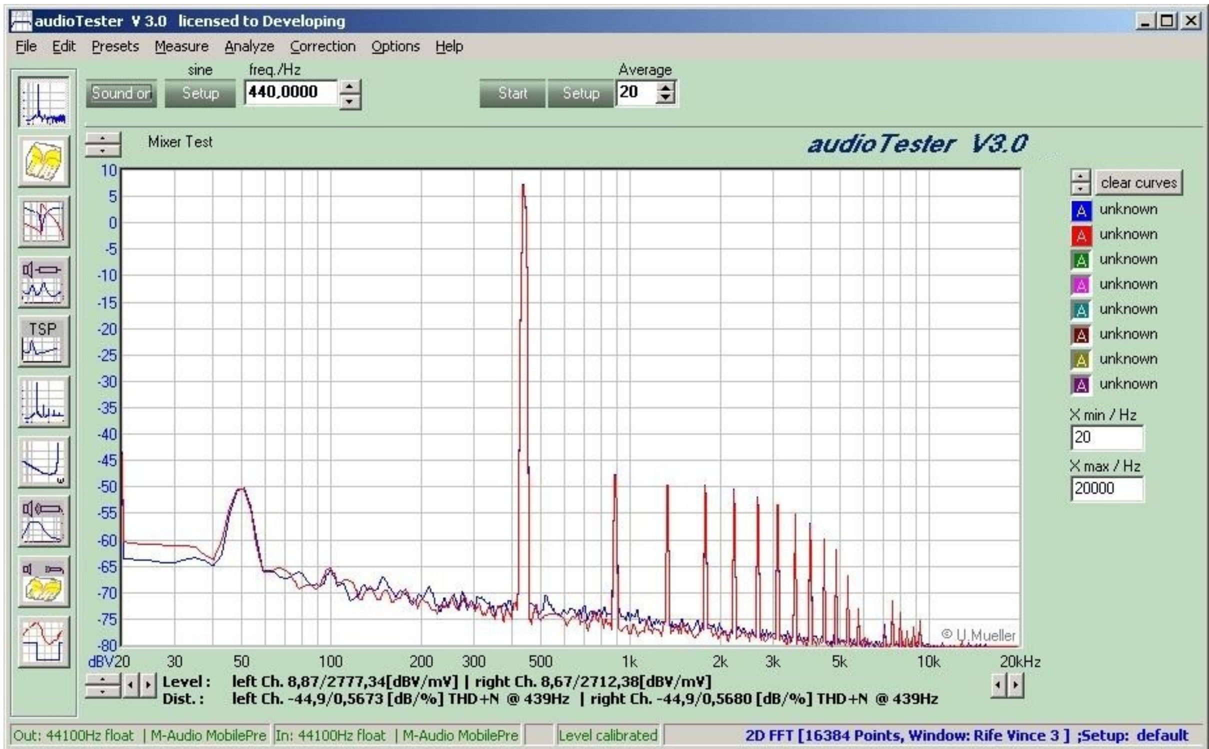
Input Channel 3 Line In: 400mV Fader Ch3 full (see photo below)
Measurement 1:Output Line Out: Master Fader scale 1   Output 350mV
Measurement 2:Output Line Out: Master Fader scale 4   Output 2800mV

Remarks: If level unit is dBV, the system must be calibrated for exact values
During the second measurement the distortion level rises up to extreme values, the sound card input are overdriven without a corresponding view in the
audioTester. Later we saw, that before the potentiometer of the sound card there was inside a preamplifier directly connected to the XLR plugs. This preamplifier was
overdriven. Then the voltage driver, described above, was used (10k/680Ω).

Conclusion: The bar graph-display of the mixer show correct value in dB.
In practice fade only up to 0dB. Above 0dB of the mixer display the distortion are bad.



Distortion and noise is acceptable

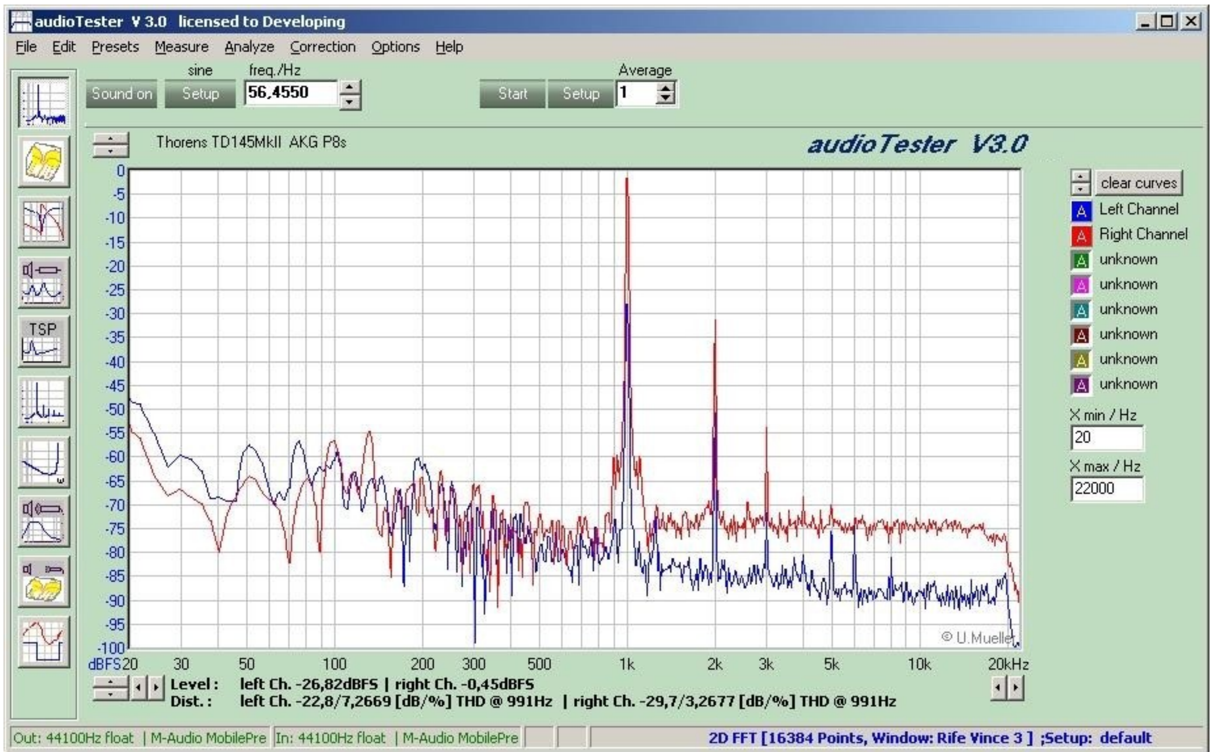


Increased master level about 18dB -> distortion much higher

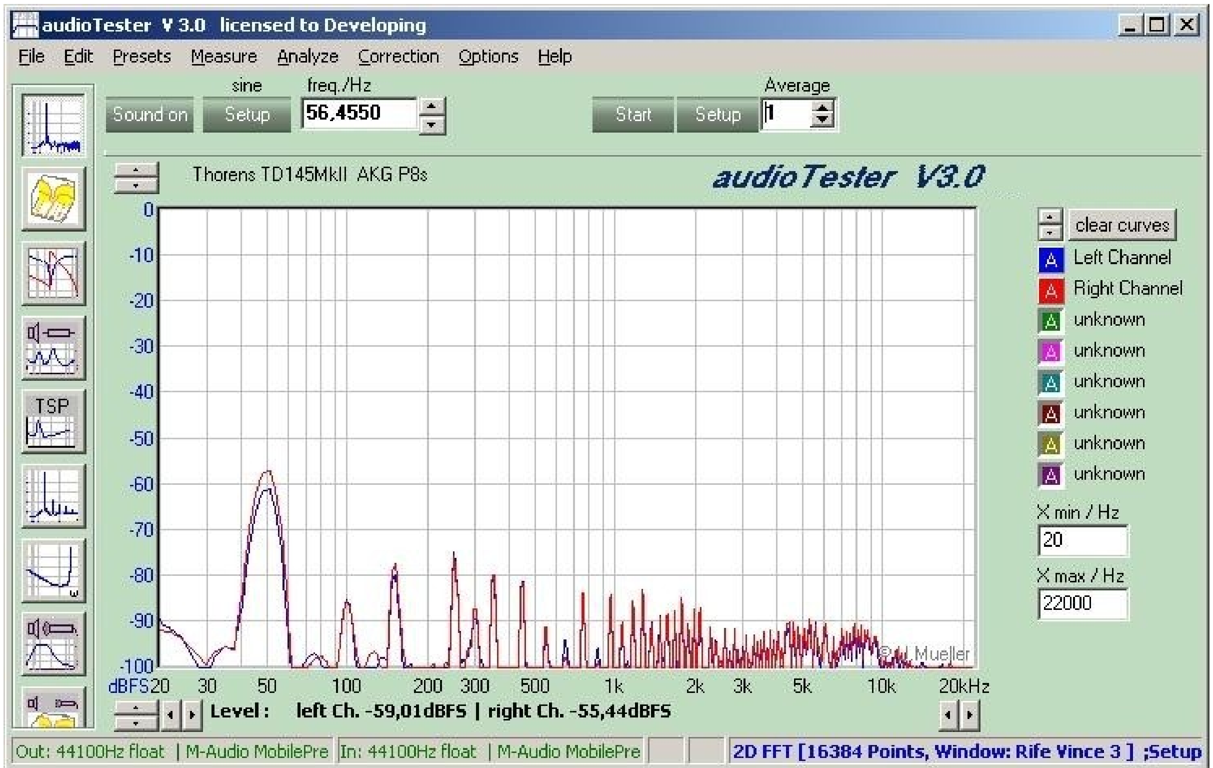
Test device:


DISCO MIXER
model SX-90
EQUALIZER
CH 1
CH 2
ECHO
CH 3
CH 4
TALK
LEFT
MASTER
RIGHT
FLUKE

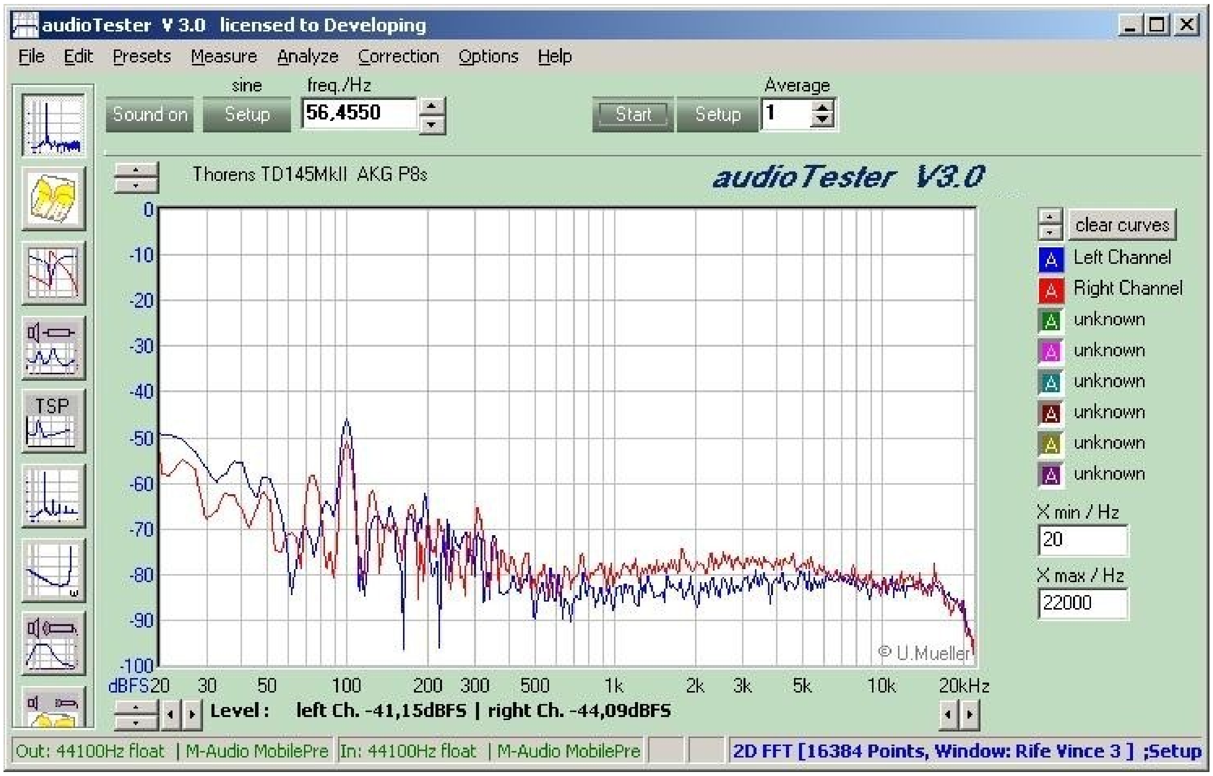
Measurement with a turntable
Test device: Thorens TD145 MkII, pick-up system: AKG P8s, all in a good condition
sound card: M-Audio mobile Pre Line In
dhfi Messschallplatte Band 2



Right channel 0dB 1kHz, left channel silence
Crosstalk R->L -26,4dB (see above in diagram)
Turns around 1% to slow (991 to 1000Hz)
Distortion 3,3% @ 1kHz (Band 2 is in the middle of the record)



Pick-up system over the running turntable



Playing a silent track

Turntable at work

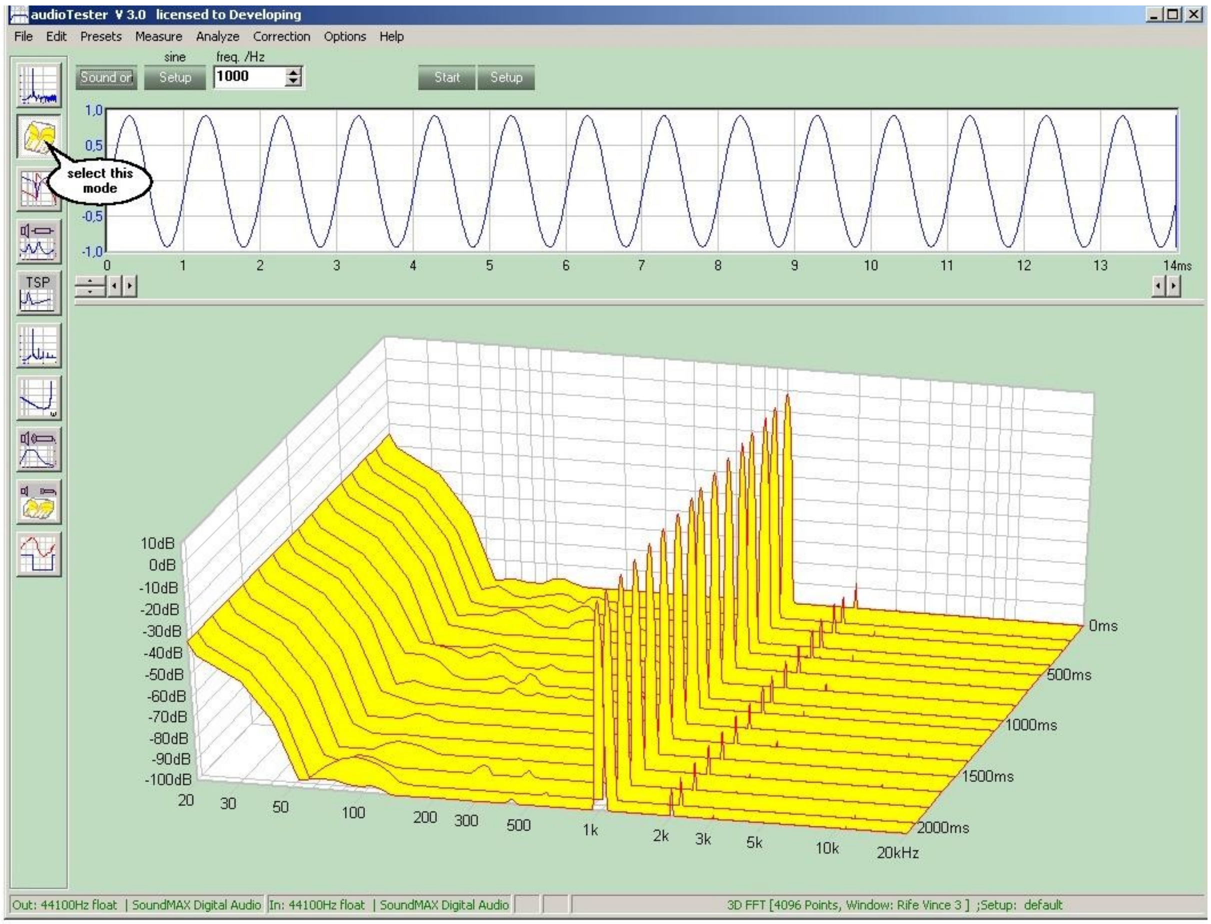
Measurement of a sub woofer

To the input of an active sub woofer is applied a pink noise signal.
The signal was received with a measurement microphone at 1meter distance.
In the audioTester diagram dialog is selected a pink noise correction see here
The measurement with this sub woofer is made also with the measuring modes: Sweep and 2D-Impulse measurement
Measurement was not made in an anechoic room, you see the resonance at 35Hz.

Measurement with a PC-Speaker:
To the input of the active sub woofers was applied a pink noise signal.
The signal was received with a measurement microphone in 30cm.
In the audioTester diagram dialog is selected a pink noise correction see here
The measurement with the PC-Speaker is made also with the measuring modes: Sweep and 2D-Impulse measurement

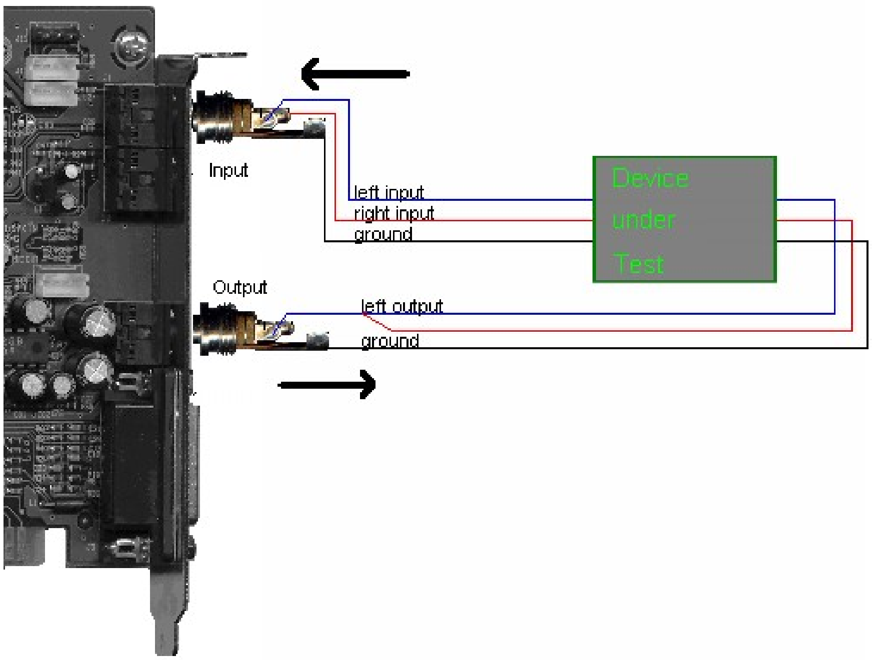
2.2 3D Spectrum analyzer

Features:
- FFT with 64 up to 32768 AnalysisPoints
- FFT-Windows: none, Blackman, Hamming, Rife-Vince ...
- Up to 64 time ribbon waveforms
- Free rotation of the 3D diagram
- Wave generator from 1Hz to half the max. sample frequency (1Hz-96kHz @ SF 192kHz)
- Wave forms: sine, square, triangle, white/pink noise ...



Options:
- Start/Stop sound output
- Setup sound output
- Start/Stop spectrum analyzer
- Setup spectrum analysis
- Button for the 3D spectrum analysis
- Input/Output level adjustable, only if mixer support is on
- 3D-Diagram, diagram-options with a right mouse click
- Splitter to move the time and the 3D window
- Selectors to rotate the diagram in the window
- Diagram for the time domain waveform

Connection to the test device

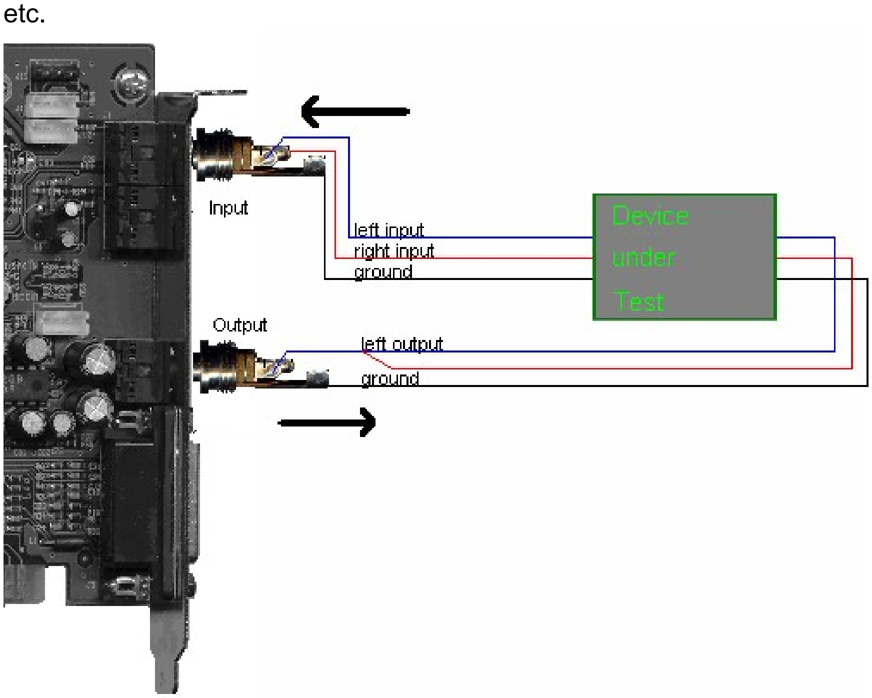


2.3 2D FFT Wiring Diagram

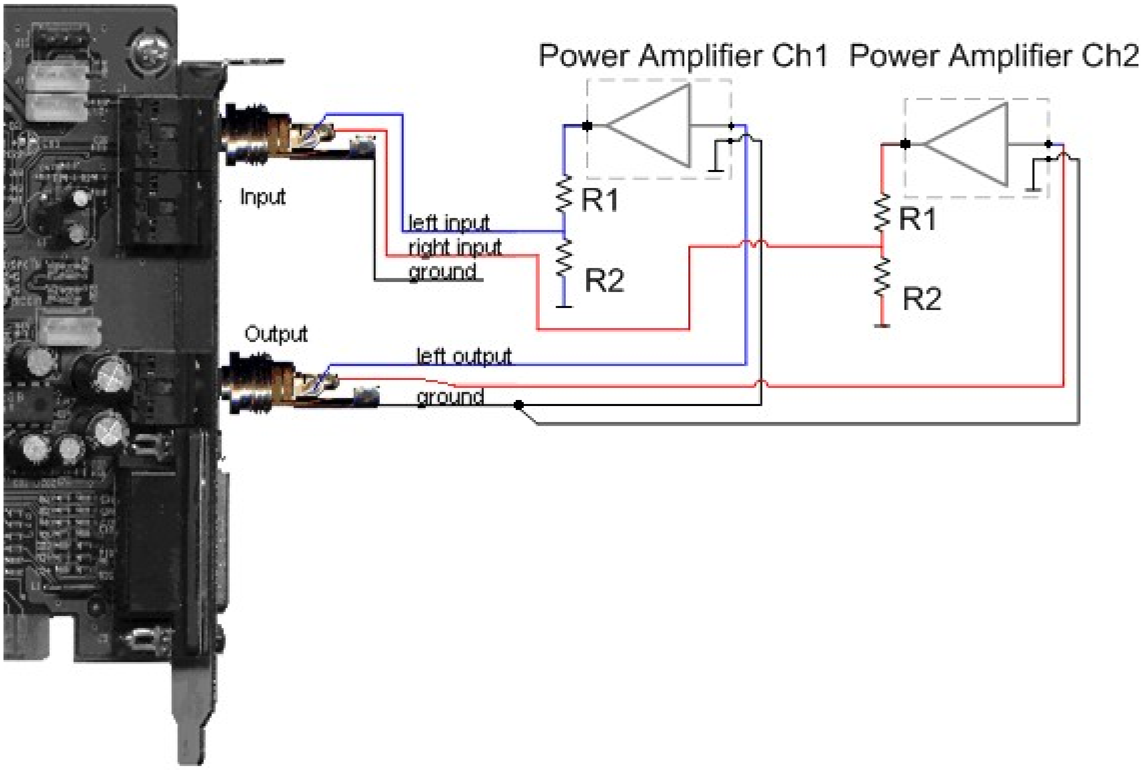
Help

Typical wiring:

Devices under test with low signal levels like frequency crossovers, mixing devices or pre amplifiers etc.

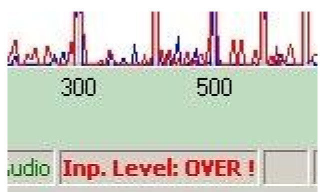


Secure wiring for amplifiers and devices with high signal levels:



Remarks:
The 2D FFT measuring mode normally is used to measure levels and distortions. Please think about the estimated signal levels **before** you begin to measure.
See in the above pictures the wiring of the sound card and the devices under test.

The first problem may be that the test device is over driven! You **won't** see it, as in the picture below! You may only noticed it as high distortions in the spectrum **without** that the ***audioTester*** shows it on the Status line as seen below.



Immediatly you should take precaution to protect the input sensitivity of your test device, by decreasinge the ***audioTester***s output level. You can do this digitally with the [Sound out/Set up Dialog]. But a better solution is with the a voltage/volume potentiometer. The digitally decrease of the level is not the best way, because it decreases the number of bits of the output signal and that increases distortions of the outgoing signal.

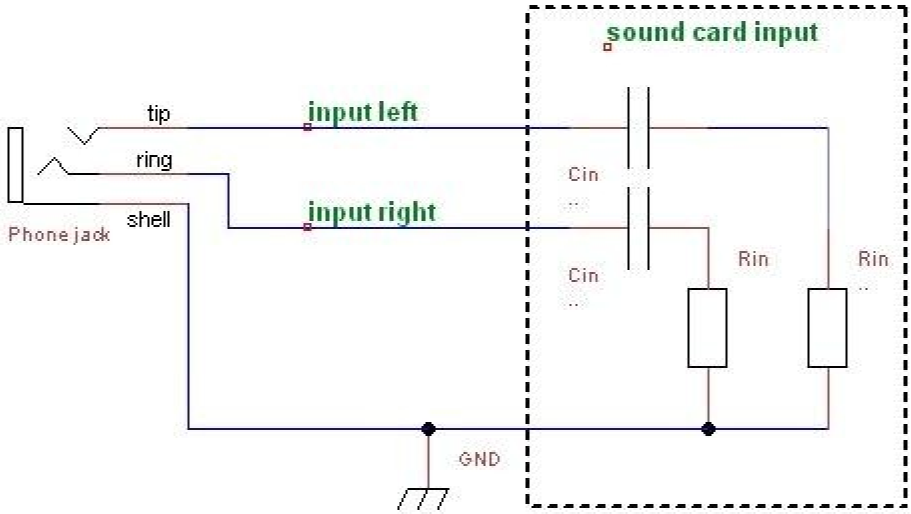
Very often it happens that the input of the sound card is overdriven. In fact, this is shown by the ***audioTester*** (see above), BUT it can DESTROY your sound card! So you must know what you do, especially for measurement of amplifiers. If there is, for example, no signal applied at the amplifier and then a start-up glitch of the amplifiers destroys the sound card. The best solution is to use a potentiometer at the sound card input.

How should you choose the Resistors R1 and R2?
Example:
Assuming input sensitivity of the sound card is 1V.
Max. power of the amplifier is (P) 100W at (R) 8Ω.
At the amp output is seen the voltage U=sqrt(P x R) = 28.3V .
This max. voltage must be divided down to 1V.
The amp voltage must divide by (R1+R2)/R2
Therefor you chose R1 with 3.3k and R2 with 120Ω --> divide by 28.5.
The input resistance of the sound card is much higher than the impedance of the voltage divider, so that there a no problems. The load of the 3.4kΩ (3.3+0.12) of the voltage divider is no problem for a power amplifier. Now you must calibrate the sound card with the new potentiometer.

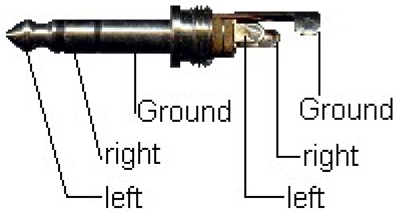
CAUTION:If you measure bridged amplifiers you **should not** connect the ground input connector of the sound card to any amp output, the amp output would be short-circuited . Here you must use only one amp output, and the ground of the sound card should be applied to the amp case. In this case you measure only the half of the voltage and a quarter of the power. This is also an important issue in Power THD Measurement.

Additional hints:
In the schematic below you see an input of a typical sound card. The input electrolytic-cap prevent measurement of d.c. voltages, or measurement of very low frequencies.
The input resistor of the sound card, here labeled Rin, is normally around 50kΩ, this should considered

if you must use a potentiometer.



Pins of a stereo phone jack

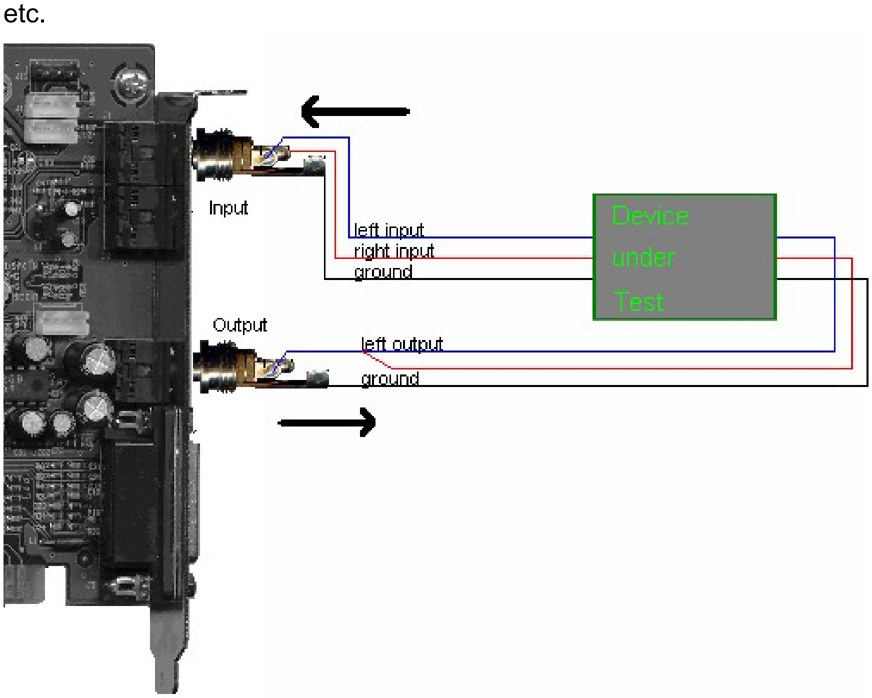


2.4 3D FFT Wiring Diagram

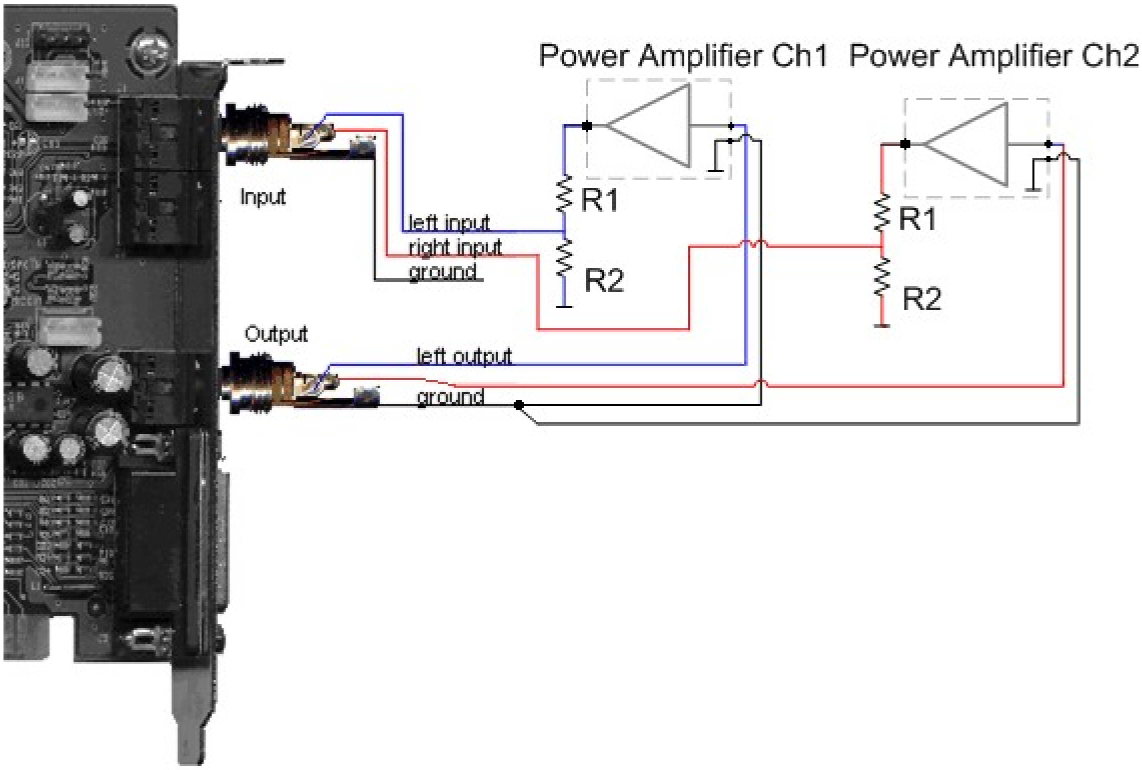
Help

Typical wiring:

Devices under test with low signal levels like frequency crossovers, mixing devices or pre amplifiers etc.



Secure wiring for amplifiers and devices with high signal levels:



see also Hints at 2D FFT

3 Sweep measurement

3.1 Sweepgenerator

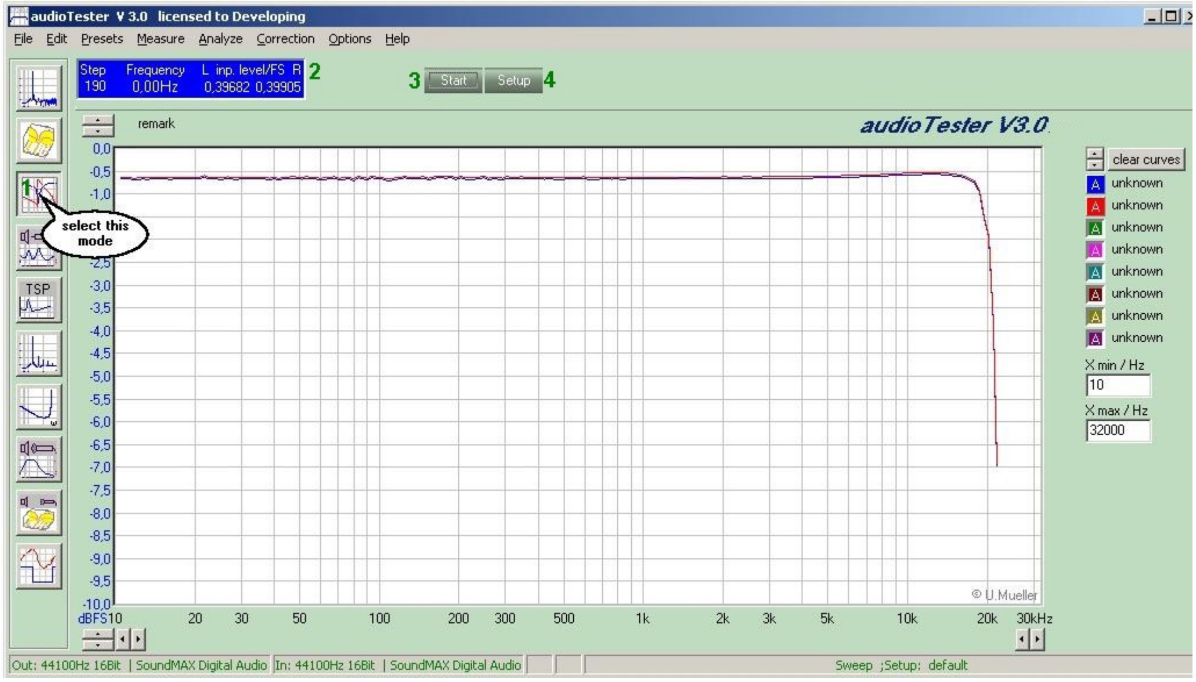
More themes in this topic:
Asynchronous sweep
Measurement example: iPhone

With the Sweep-Generator you are able to measure frequency responses. Therefor a sliding sine-wave (of progessively increasing frequency) is applied to the device under test, and the read out level is shown in the diagram over the frequency range.It is possible 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 shift between input and output.

The tone generation begins at 0.1Hz up to the sample frequency divided by 2. (SF/2)
The lowest Freq. of 0.1Hz is not reachable for normal sound cards, but for special rebuilt cards, this is an interesting feature.
The measuring time for one step at such low frequencies is a max. of 100sec.
Please use low sample rates for measuring of low frequencies.
For the measuring of any one tone, the system tries to use 25 full wavelengths. This is limited by the max. measuring time of 100sec.
The min. measuring time is 100ms, so for example, at a measuring freq. of 10kHz the system uses 1000 full wavelength to determine the level.

It is not possible to store the sweep into a wave file by using the soundFile.DLL.

We have two modes of measurements - synchronous and asynchronous.
Synchronous measurement: The ***audioTester*** applies the sliding sine wave to the measurement object.
Asynchronous measurement: An external device applies the signal. Example: frequency response of a CD-Player with a measurement-CD, or an MP3 Player with the right file.

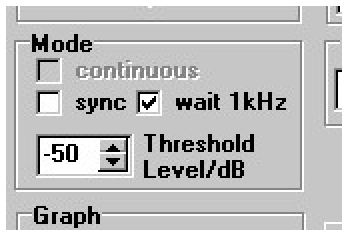


1. Button to select the Sweep-Generator.
2. Display the actual frequency and level. Adjust the level to get values from 0.1 to < 1.
   Red levels indicate over-driven signals.
3. Start of measurement
4. Sweep Setup (see here)

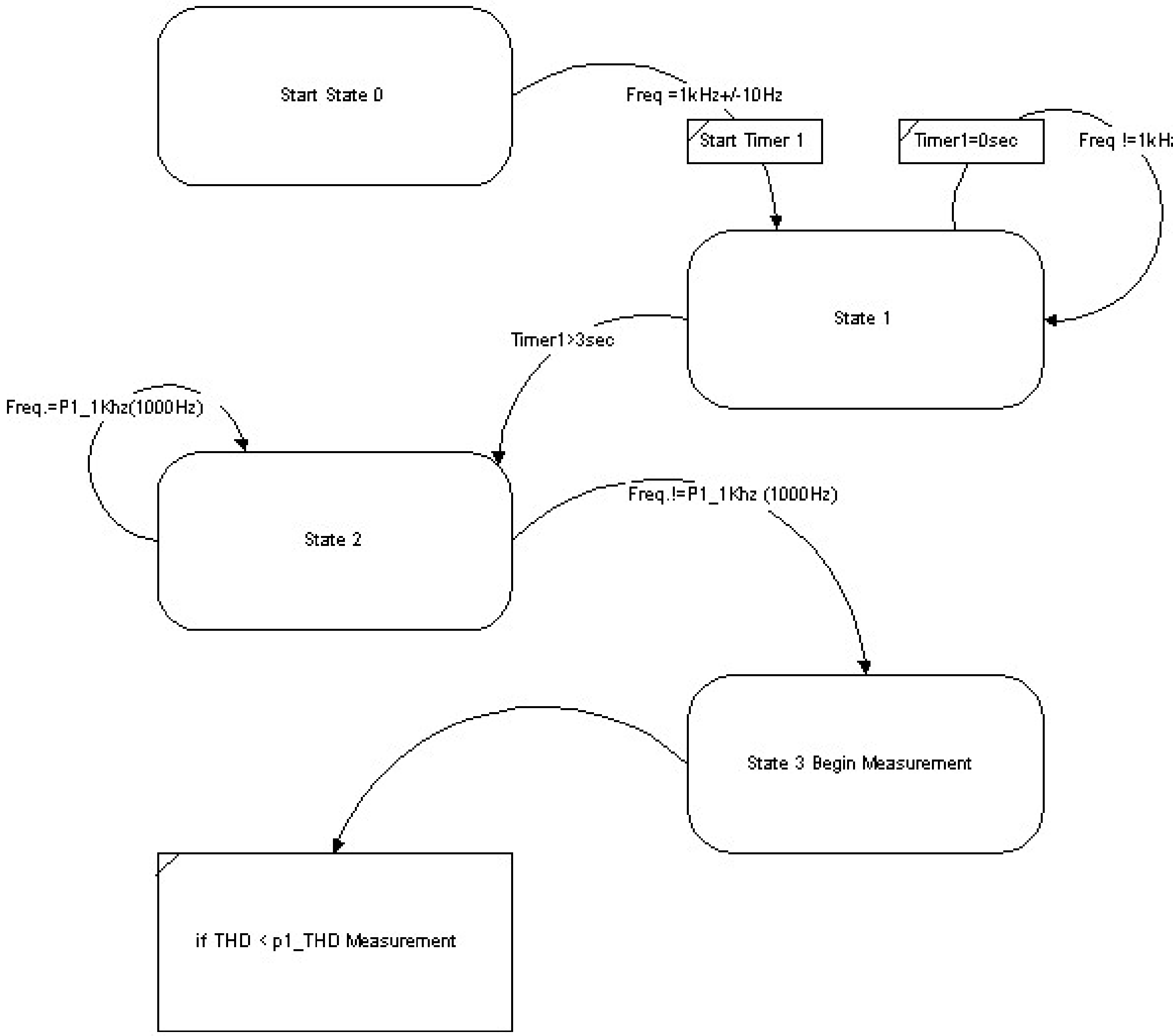
Asynchronous Measurement

You must use the asynchronous 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 to 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 has a distortion value under -6dB, will sorted, stored and displayed immediately.



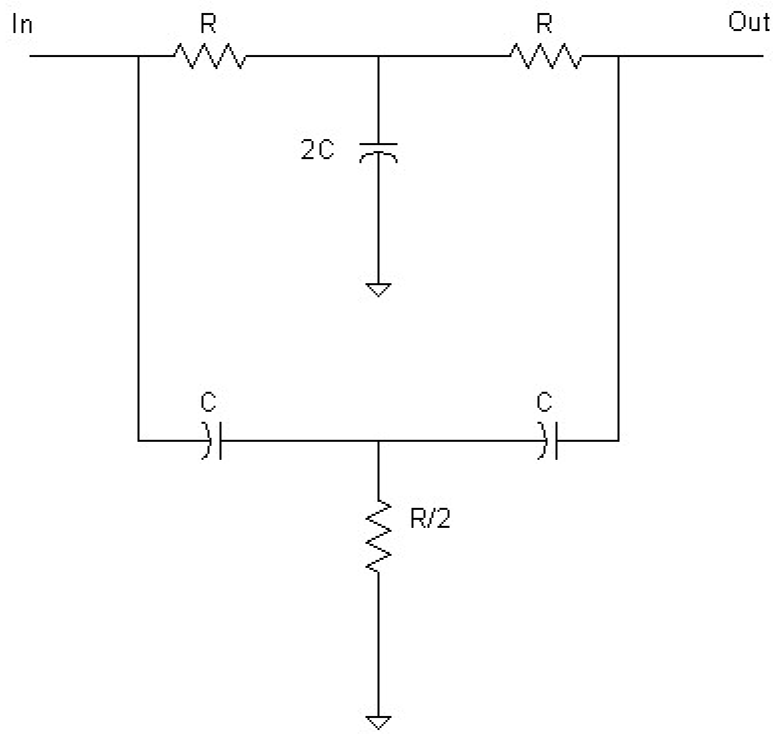
Measurement with pilot signal (see fig.): The required frequency measurement works with a threshold value of -50dB (default see fig.). It works with this state machine ->



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

Tips, Tricks
Often there are inconsistancies if cables are not correct, or the channels are flipped, or the setups are not ok, and so on ...
Please refer to this small helpful circuit and the correct diagram it should produce.

This Double-T-Filter:



C = 0.47µF
R = 1.5kΩ

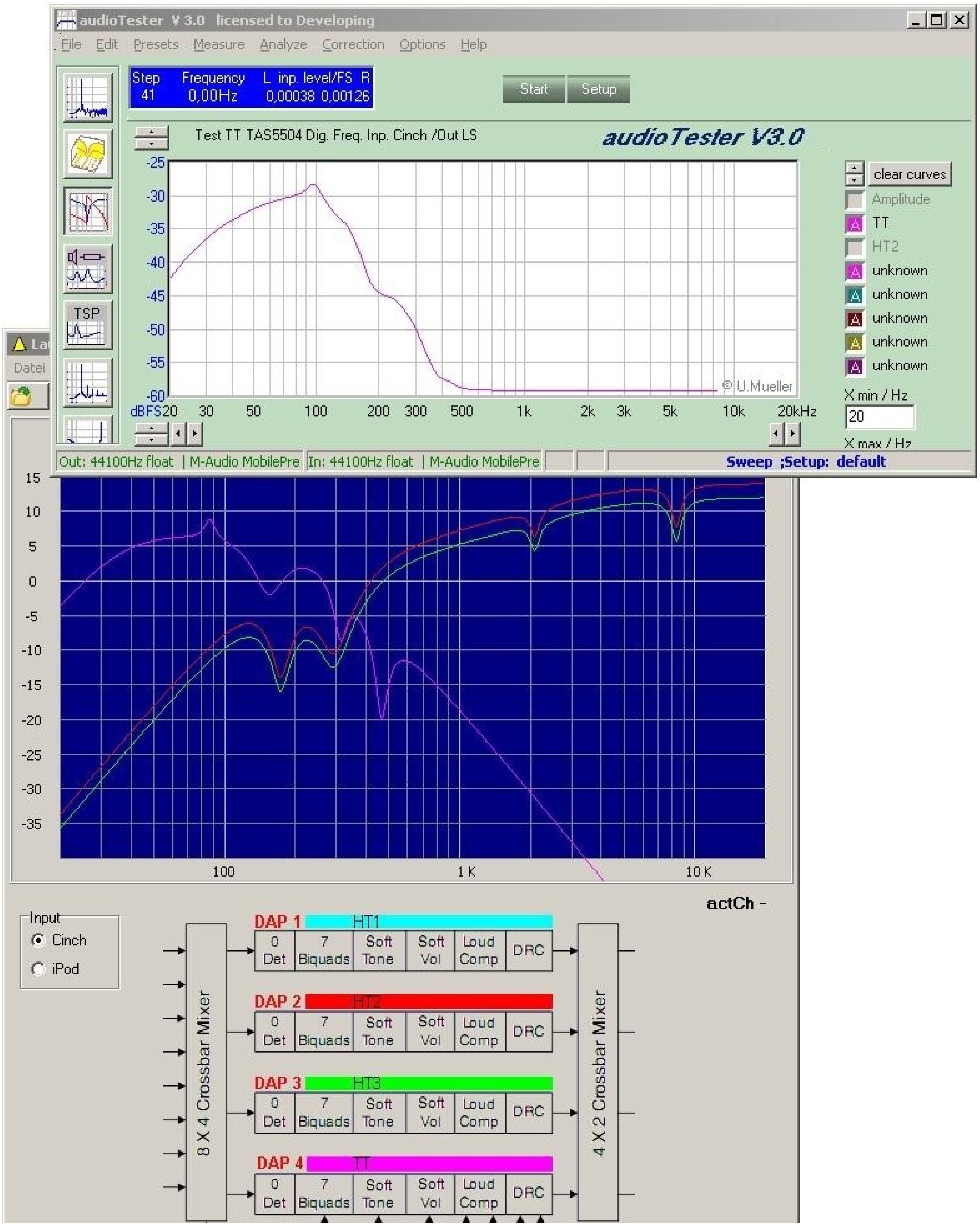
makes this waveform diagram:


USB
Stereo
Line
Output
Outputs
2
1
Stereo
Mic
Input
Inst/
Line
Ch. 2
Ch. 1

Example measurement: Frequency response phone preamplifier
Test device:; a self made RIAA network preamplifier.
sound card: M-Audio Mobile Pre.

A high pass filter to avoid subsonic noise is integrated

Measurement with a digital special speaker system
Test device: Zero series 4 channel load speaker System with DSP and Texas Instrument digital amplifier
Sound card: M-Audio Mobile Pre
Remark: Measurement at sub-woofer channel with DSP-Setup from PC-Config-Program



Sub woofer channel (TT) purple curve

Example:

Measurement Asynchronous Sweep iPhone with measurement microphone and headphone output
Test device: iPhone 3GS 16Gbyte
Sound card: M-Audio Mobile Pre
Remark: Sending Sweep file as MP3 via Email to the iPhone, playing linear
audioTester in Sweep Async Mode with 1kHz pilot tone
Sweep: 4sec. 1kHz -14dB and then 51sec. 20Hz-20kHz -20dB

Level pick up electrical from the earpiece output.

Measure microphone 20mm over iPhone Speaker (earpiece)
Repeat measurement 3 times (red,blue,green). Values below 300Hz not analyzable

Example:

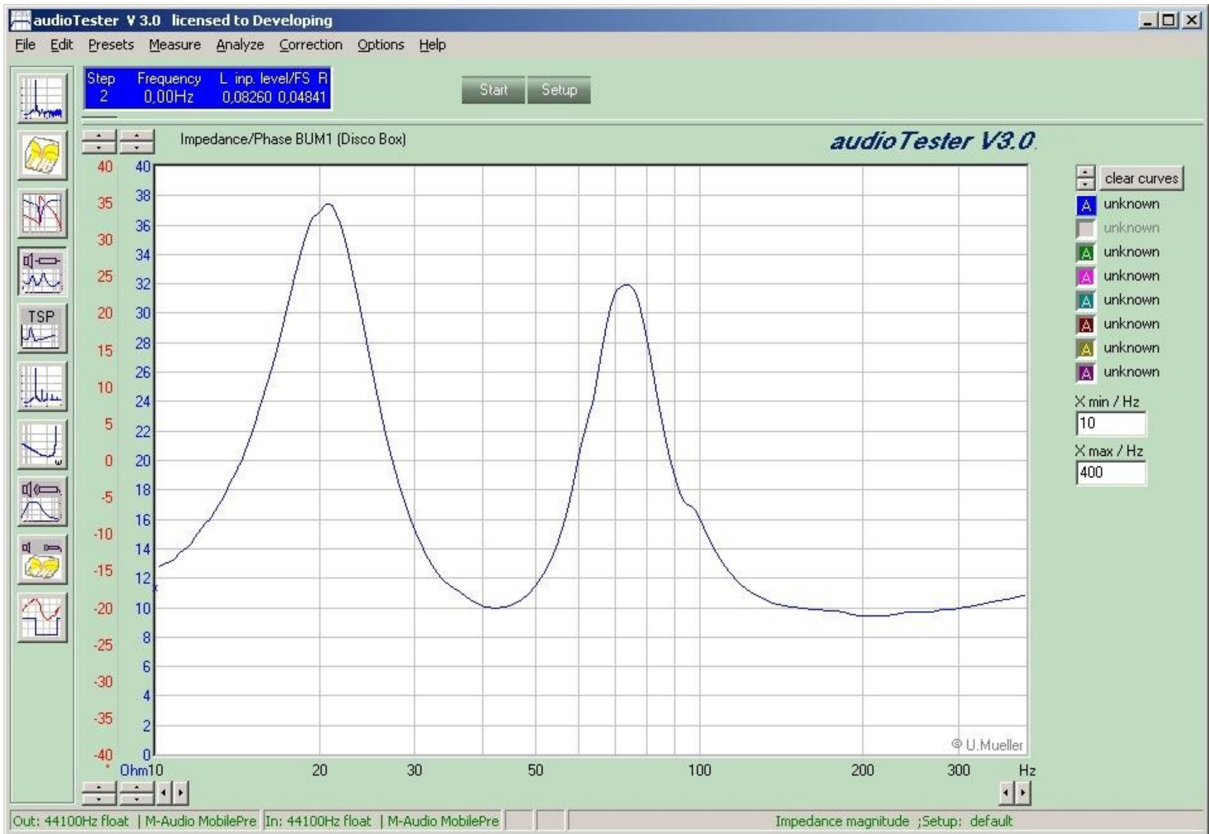
Sub-woofer measurement
A sweep signal from 10Hz to 220Hz was applied to the input of an active sub-woofer.
The signal was received with a measurement microphone at 1 meter distance.
The measurement was made also with the measuring modes 2D-Spectrum, Impulse measurement)

PC Speaker measurement:
A sweep signal from 10Hz - 22000Hz was applied to the input of a PC-Speaker..
The signal was received with a measurement microphone at 200mm distance.
The measurement with the PC -Speaker was made also with the measuring modes 2D-Spectrum, Impulse measurement


audioTester V 3.0 licensed to Developing
File Edit Presets Measure Analyze Correction Options Help
Step Frequency L inp. level/FS R
51 0,00Hz 0,00863 0,00010
LS-Out
Start
Setup
Direct Monitor
PC speaker sweep
audioTester V3.0
clear curves
unknown
X min / Hz
100
X max / Hz
20000
dBFS
© U.Mueller
Out: 44100Hz 16Bit | M-Audio MobilePre
In: 44100Hz 16Bit | M-Audio MobilePre
Sweep ;Setup: default

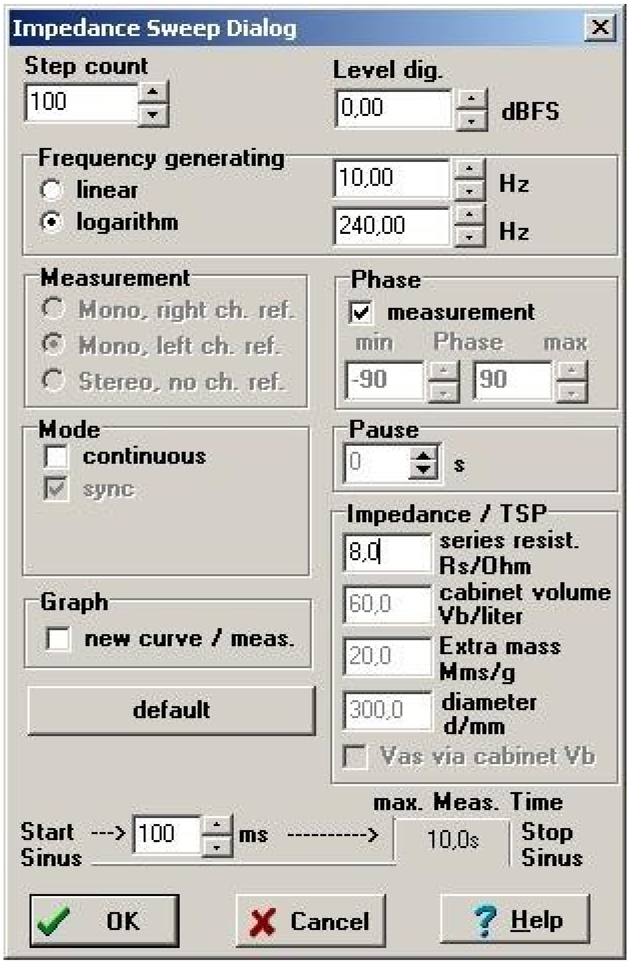
3.2 Impedance magnitude

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 in the dialog .(Here we used 8.0 Ω.) The reference resistor should have a similarly value as the expected impedance. In the *setup dialog* some settings are disabled, otherwise it is like the sweep dialog (see here).

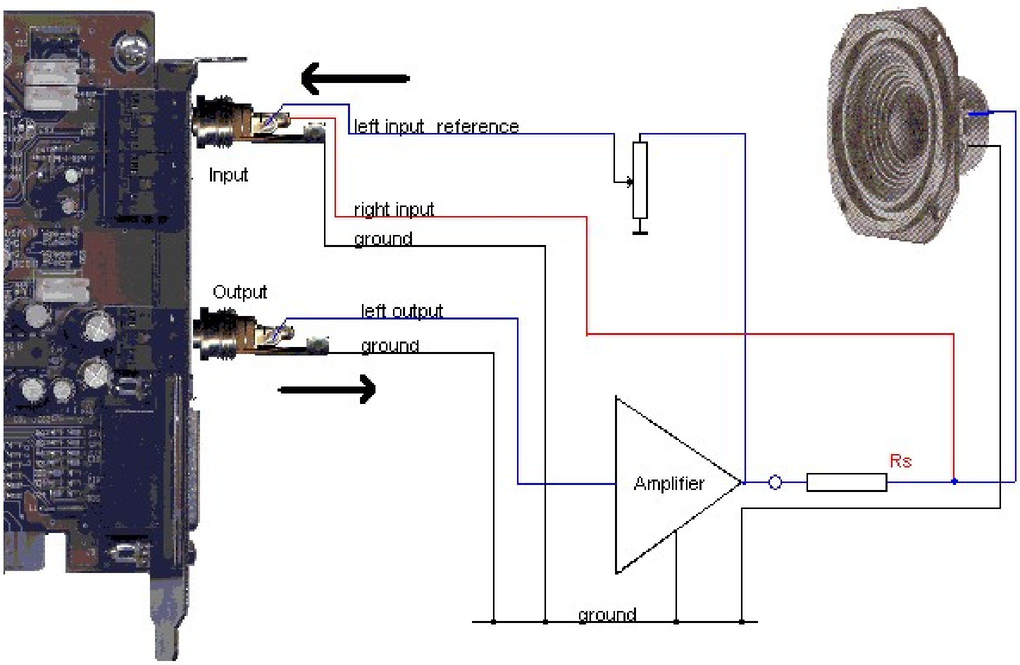


The bass reflex tube from the speaker measurement above is not exactly tuned. The two peaks at 20Hz and 75Hz should better be equal. You can optimize it very easily with this measuring mode. Here I think the tube is too short or the tube diameter is too big.

Impedance-Dialog



The wiring diagram for the impedance measurement mode:



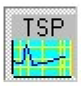
Remark:
Connect the right input (red wire) only if reference measurement is used. For influence of the reference measurement see here . Be careful not to exceed the max. input level of the sound card.
Increase output level slowly
Don't use a bridged amp.
Left and right sound card channels are equivalent.
Sound level at the speaker should be less than 1V.

3.3 Thiele Small Parameter

Measuring Thiele-Small Parameters

Introduction

The set of Thiele-Small parameters are an industry norm, a virtual standard. They were developed in the 1970's.They are used to assist in the design of low frequency loudspeaker-enclosure 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 these 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 steps.



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

The Thiele-Small-Parameters:

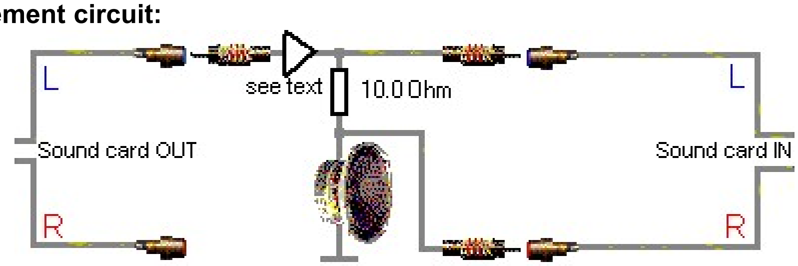
Assumend parameters:

Sd	is the area of the transducer diaphragm (cone)	unit: cm²
Mms	is the extra mass for the second measurement	unit: g
Vi	is the test cabinet volume for the second measurement. (self made only for the test)	unit: Liter

Calculated by ***audioTester***

fs	free air resonant frequency	unit: Hz
Zmax	impedance at resonant frequency	unit: Ω
Rdc	DC-resistance	unit: Ω
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

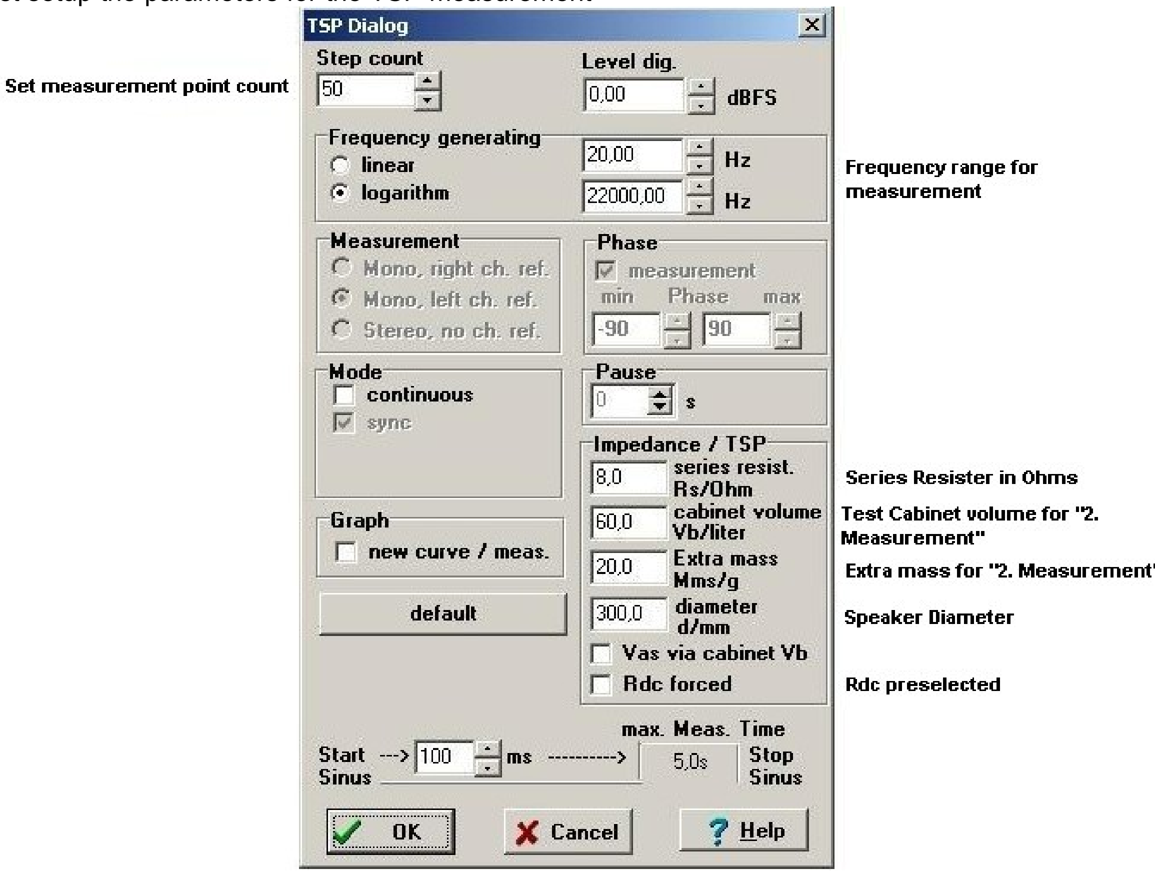
Measurement circuit:



Important Hints:
Depending on the sound card it is possible to connect an amplifier to the sound card output to increase the sound level.
Please check the max. possible input level of your sound card, with the input connected to the amplifier output!
Use a rectifier network to decrease the voltage level if necessary.
Place the loudspeaker on a cushioned under support, to prevent vibrations at diverse resonances.
Use wires with a large cross section (large gauge wires).
Please remove protector grids from the speaker (see here).
Do not use thin or worn bad cables! (see here).
The extra mass (modeling clay or similar) must be fastened secure at the speaker diaphragm.

A waveform like this results from an incorrectly fastened extra mass. (see here)

First setup the parameters for the TSP measurement



1.Measurement: Determining fs, Zmax, Rdc, Qms, Qel and Qts
Connect speaker as described above and the start measurement.

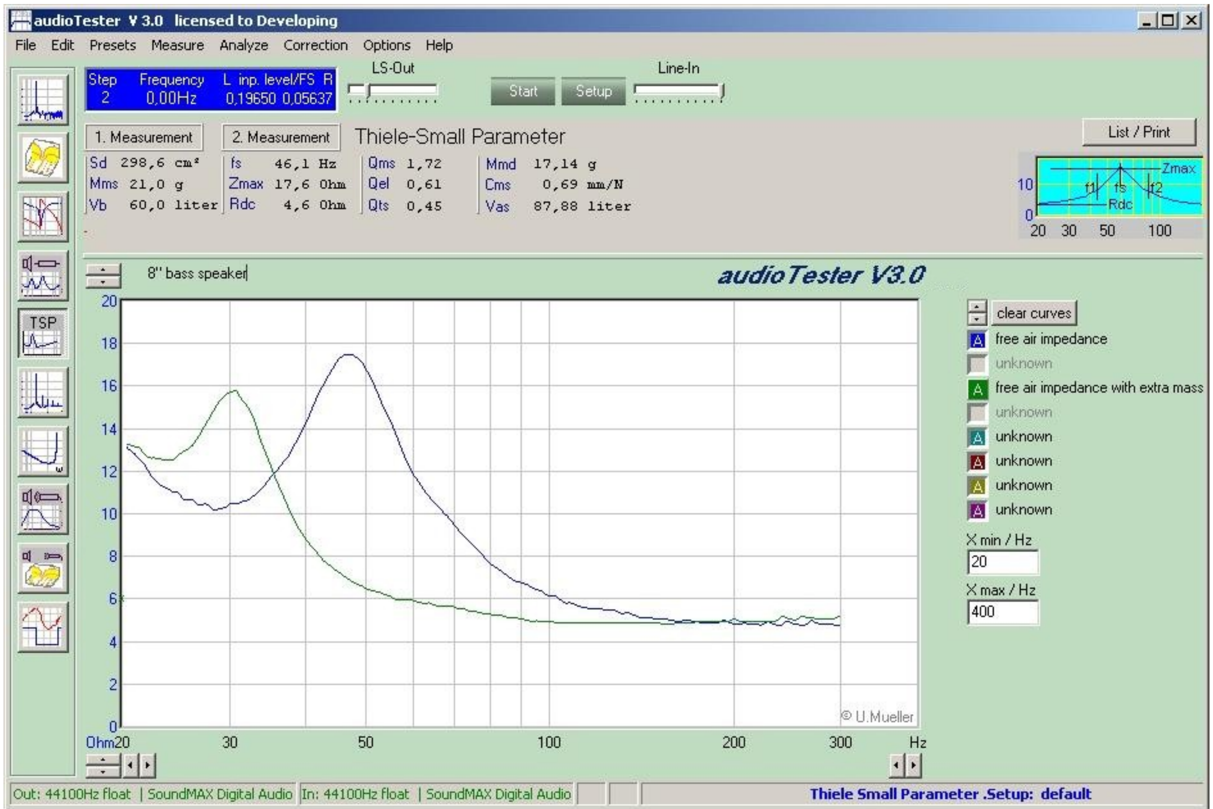
After the measurement drag the Free Air Impedance curve (in the example it is the blue one) to the panel labeled '1.Measurement' (see picture below).
The values fs=48.4Hz, Zmax=19.2Ω are immediately determined.

If you had checked the option 'Rdc forced' in the dialog, then now you are able to edit the value of Rdc in the main window and the calculation of the TSP parameters are done with
this value. Please enter the Rdc value before measurement.

If an error occurred, a message appears below the measurement values.
LIST of POSSIBLE ERRORS:

Error: f2-f1=0	Impedance could not determined
Error: division by zero	Problems while determining Zmax and Rdc
Error: floating point error	General error
Error: fs not found	Curve could not be determined (no resonance)
Error: Rdc = 0	Is the correct Resistor entered / connected ?
Error: fs not plausible	Curve could not be determined (no resonance)
Error: fs not plausible (f1, f2)	Error while calculating fs (curve to short, or more than one resonance ?)

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



2.Measurement: Determining of Mmd, CMS and Vas
To determine Mmd, CMS and Vas, the speaker must built into a test cabinet **or** an extra mass (20g of modeling clay) must fastened at the speaker, shown in the picture below
For a chassis (frame or basket) of 200-250mm (8"-10") use a 30 liter closed cabinet.
For a chassis (frame or basket) of 250-300mm (10"-12") use a 60 liter closed cabinet.
In general, the size of the test cabinet should be chosen, such 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.

The values are influenced by the temperature, moisture and so on.
If you decide to measure with a cabinet, choose 'Vas via cabinet' in the *Setup dialog*.

After the measurement drag the free air impedance curve (in the picture the green one) to the panel labeled '2.Measurement' and drop it there.
The calculated values are shown.

Hint:
When measuring with extra mass, the resonance freq. is lower than the free air resonance.
When measuring with a test cabinet, the resonance freq. is higher than the free air resonance.

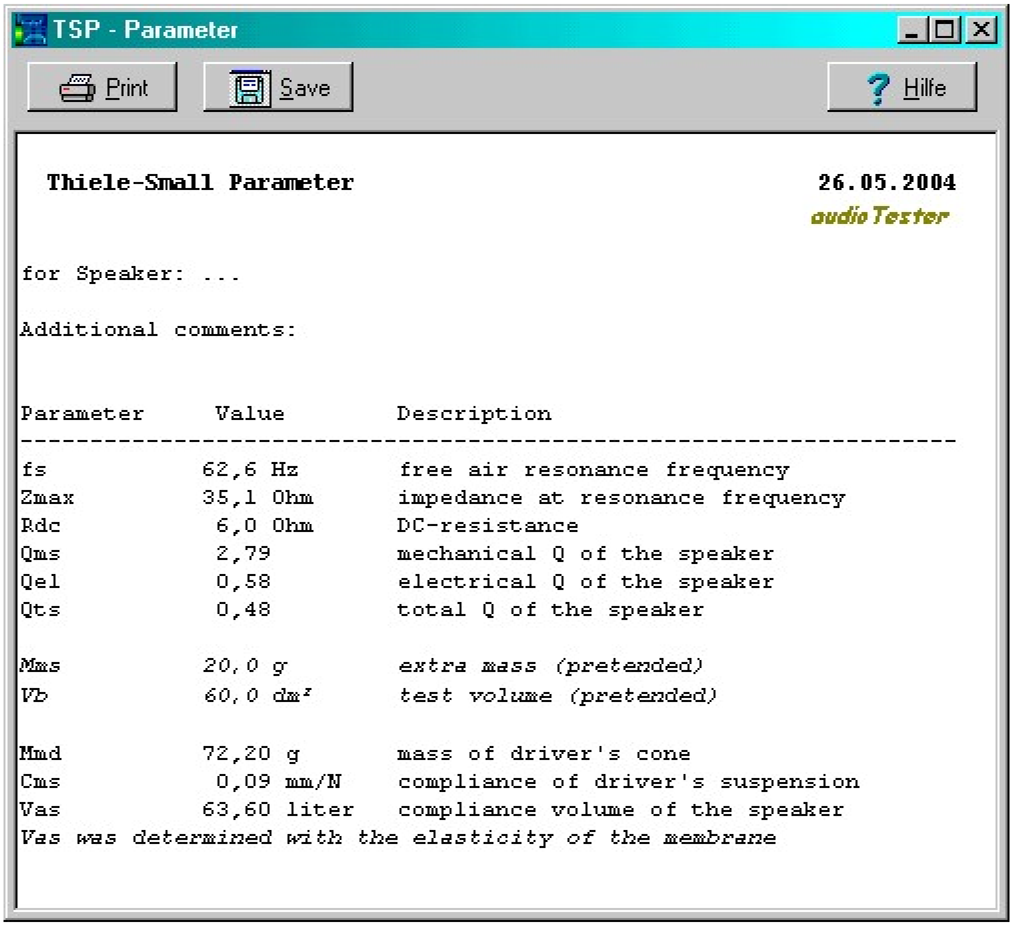
Error messages:

Error: Calc Mmd	Floating error when calculating Mmd
Error: Calc Cms	Floating error when calculating Cms
Error: Curves with and without mass are identical	you dropped the wrong curve

Error: Curves with and without extra cabinet are identical	you dropped the wrong curve
Error: please make first measurement	do it

SUMMARY The TSP measurement reacts very sensitively with changes in measurement parameters, wiring and environment.
Try to use the same cables and same speaker position, while comparing different speakers.

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



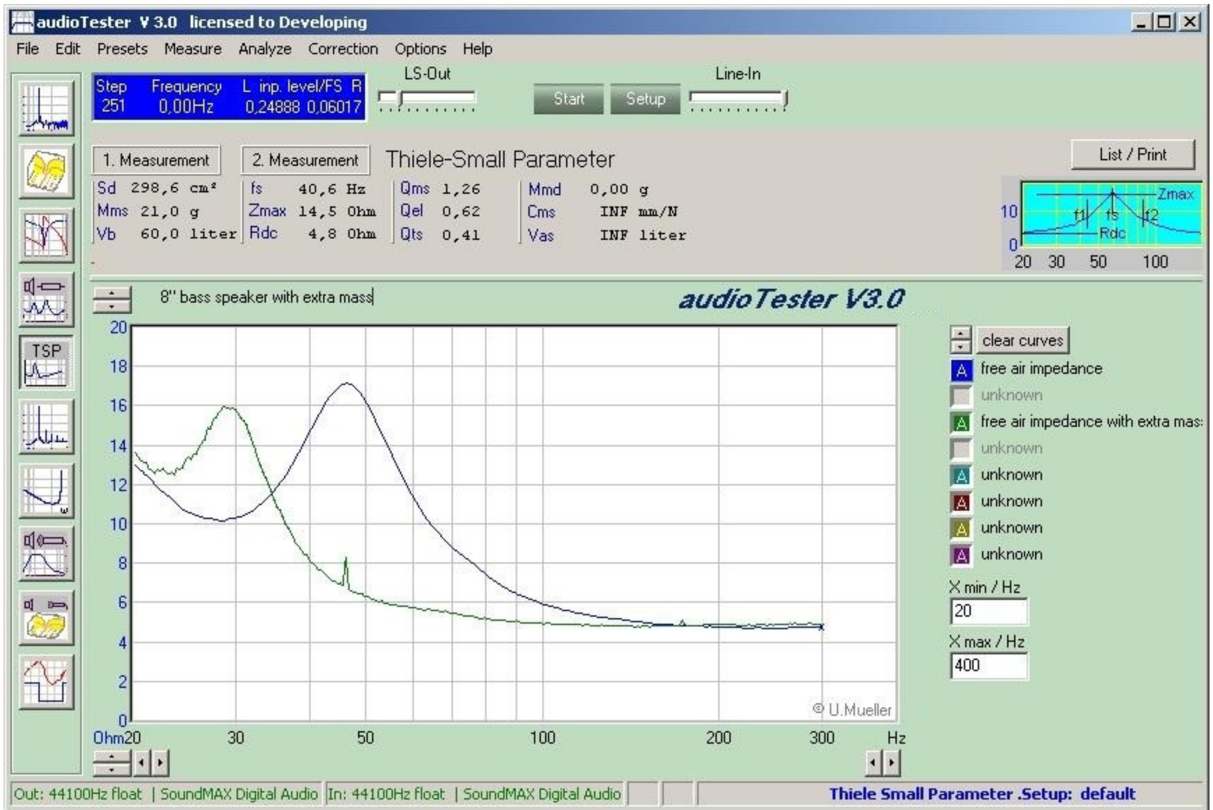
The speakers without an extra mass,...

... and with an extra mass

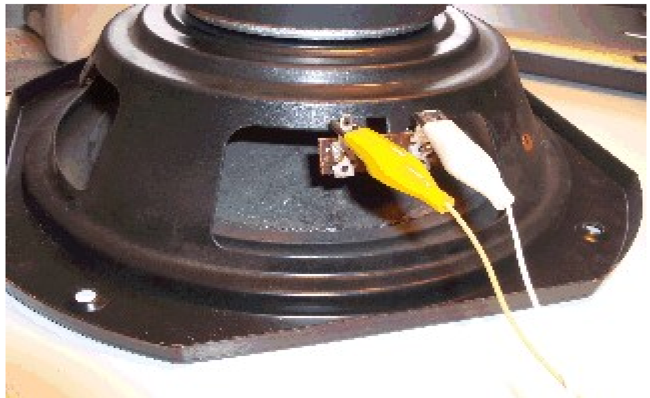
Examples for measurement errors:

A speaker grid influences the measurement

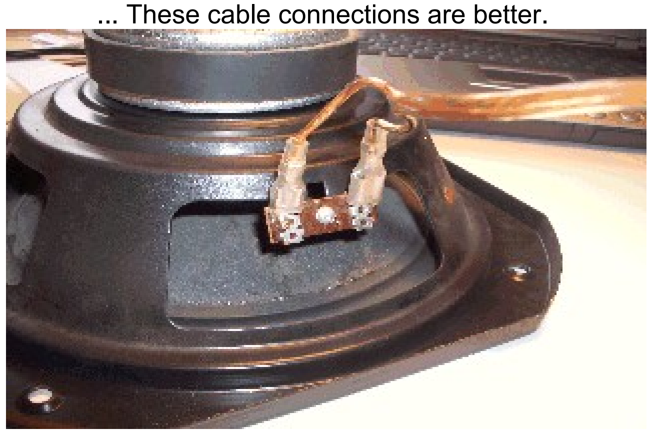
Not correctly fastened extra mass



These cables are thin and poorly connected, ...

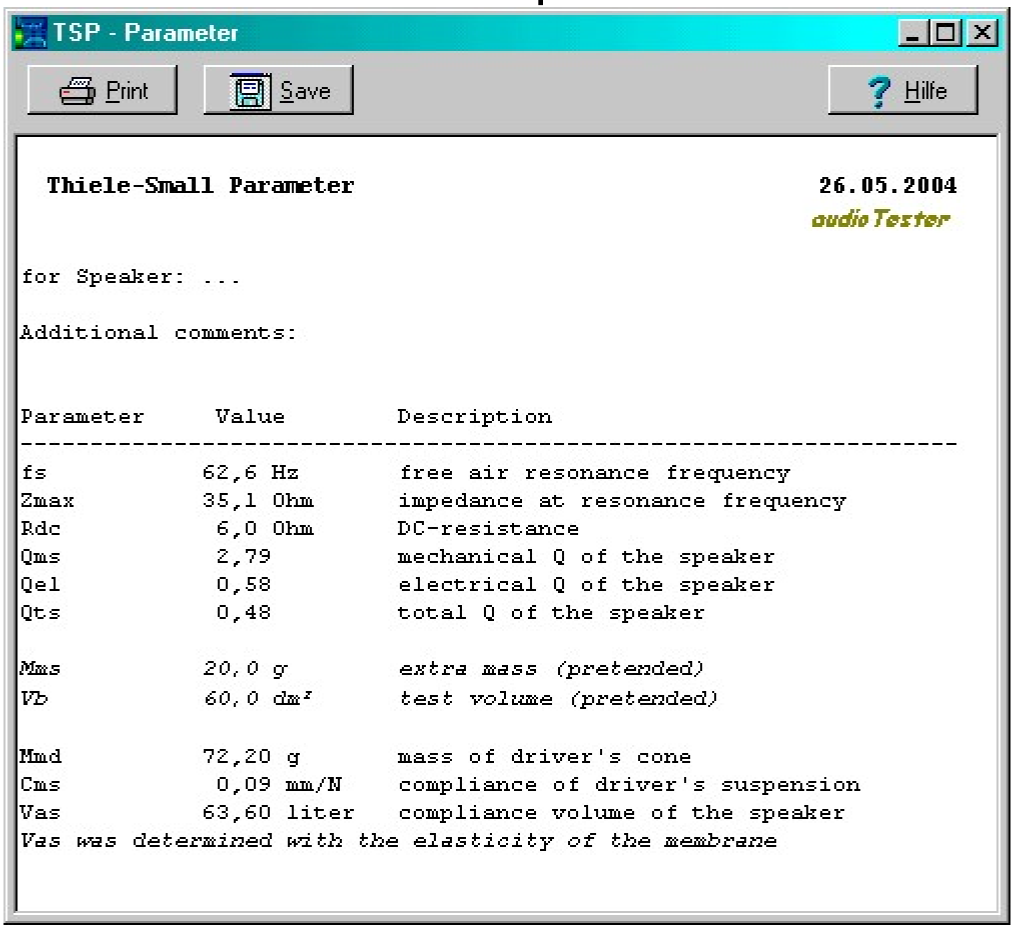


... These cable connections are better.



Attention: Cone membrane facing down - is ONLY for these photos. **The cone membrane must always face up while measuring!**

Thiele-Small print out



Thiele-Small Parameter 26.05.2004

audio Tester

for Speaker: ...

Additional comments:

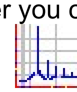
Parameter	Value	Description
fs	62,6 Hz	free air resonance frequency
Zmax	35,1 Ohm	impedance at resonance frequency
Rdc	6,0 Ohm	DC-resistance
Qms	2,79	mechanical Q of the speaker
Qel	0,58	electrical Q of the speaker
Qts	0,48	total Q of the speaker
Mms	*20,0 g*	*extra mass (pretended)*
Vb	*60,0 dm³*	*test volume (pretended)*
Mmd	72,20 g	mass of driver's cone
Cms	0,09 mm/N	compliance of driver's suspension
Vas	63,60 liter	compliance volume of the speaker

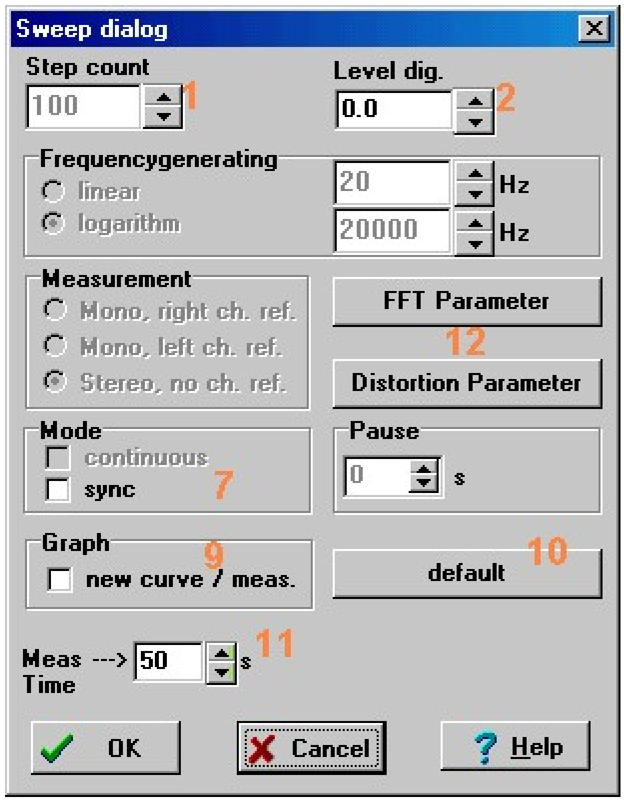
Vas was determined with the elasticity of the membrane

3.4 Distortion Measurement

Distortion Measurement in frequency domain

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

Therefor we have a tool button *distortion measurement* . The operation 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.

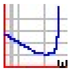


With the button *FFT Parameter (12)* you open the known FFT-Dialog . The FFT-Points should be greater than 4096. The FFT-Windows should be Rife-Vince 3 Windows (see here).
With the button *Distortion Parameter* (12) you open the extended Analyze Dialog .

Do not use the asynchronous measurement (External sweep, see Sweep Dialog point 7 above), it doesn't work here..

3.5 Distortion vs. Power

Distortion measurement vs. Power

 *Power Distortion Measurement.*

Please notice all the hints in the following text. With this measurement you could damage your sound card, the computer and the measured device (the amplifier)! Please read the text below and notice the hints for the test setup.

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 will measure self induced distortions instead of the distortions of the measured device.
For example: You must produce a output level from -40.5 to -0.5dB for a measurement from 1mW to 10W. You use only 9-10Bits for sine generating at low levels and that makes the signal even worse. Therefore, use a 24Bit sound card for best results.

The measurement:
First: Output a test tone, with an adjustable level (default -30dBFS) and an adjustable frequency (default 1kHz).
Measurement of the voltage at the output of the amplifier.
With a load resistor (R_L in wiring diagram) we calculate the test power.
With this result, we are able to calculate the test tone level range.
Please see the procedure, in the debug window. You open the debug window, in the setup dialog.

With the checkbox *'Gain Check'* in the setup dialog you are able to work without the test tone. The highest output level is 0dB, and the lowest is calculated over a voltage range or a power range. The absolute output level must be adjusted with the level attenuator of the sound card or the measured device.
For example :
Select a measurement range from 1mW to 10W, and a test tone level of -30dBFS.

Debug output:
1. 16:12:53 Expected voltage: 0.0894427V - 8.9442719V @ 0.001W - 10,000W / 8.00Ω
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

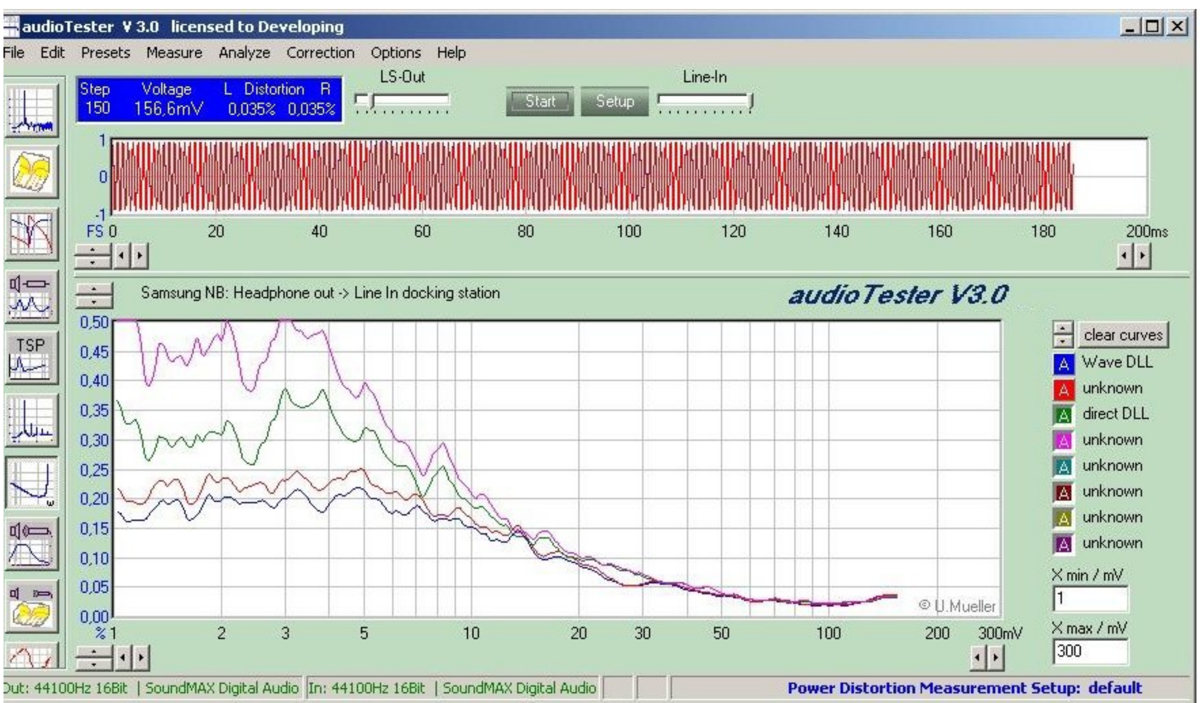
1. line: With the formula $P=V^2 / R_L$ we calculate the expected voltage at the input.
Here it is possible that you could receive one of these error messages:
The input voltage exceeds the calibrated voltage of the sound card. The sound card is overloaded and will be damaged !
The measurement stops immediately.
This is clear, if you see that the max. input voltage of a sound card is around 1V .
With 1V the max measured power is $P=1^2/8 = 0.125W$ @ 8Ω.
So use a voltage divider and connect it before the sound card input and recalibrate the ***audioTester***
.
You should 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 stops 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 by increasing of the gain of the measured device.

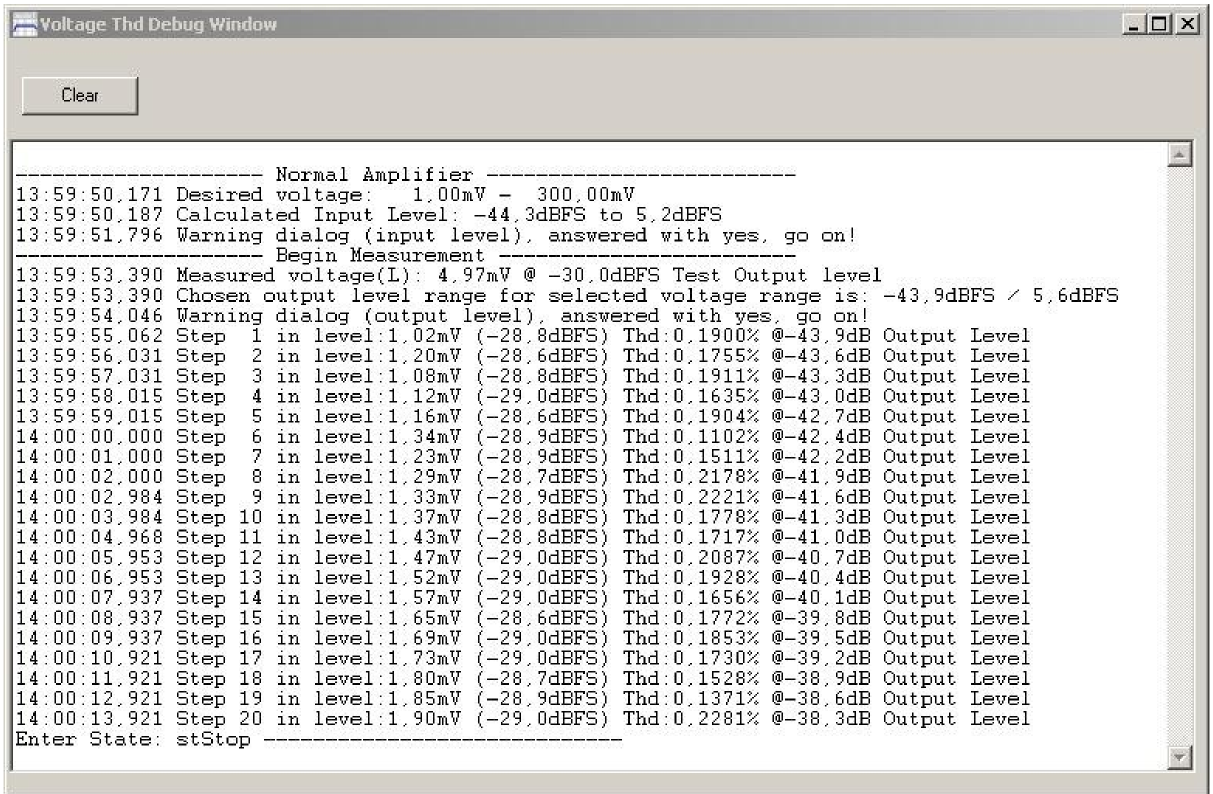
Now the measurement begins.

Example:

Distortion measure vs. level in dB

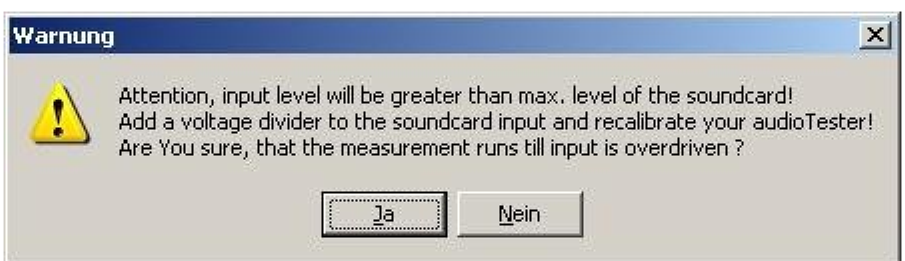


Distortion measure vs. level in %

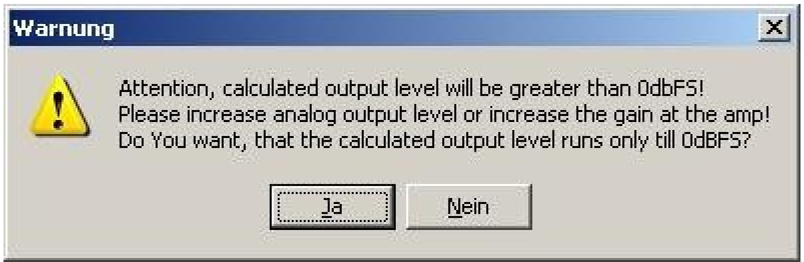


Debug window for the measurement above:
1. line: In Setup-Dialog selected level range
2. line: Calculated level range in dB. Calculated with the calibration value (here 165mV)
3. line: Warning dialog that the max. input level of the sound card will be overrun was discarded.
4. line: Test level with -30dB was applied and the output gives us 4.97mV at the sound card input.
5. line: So we calculate a output range from 1mV to 300mV as -43.9 to 5.6dB . Oh Oh
6. line: Warning dialog that output level above 0dB doesn't work . The warning was discarded.
7. Now the measurement runs Step x, level at input in mV and dBFS ... Distortion in %, at xx.x dBFS output level

In the example above there were two warning dialogs and both are discarded:

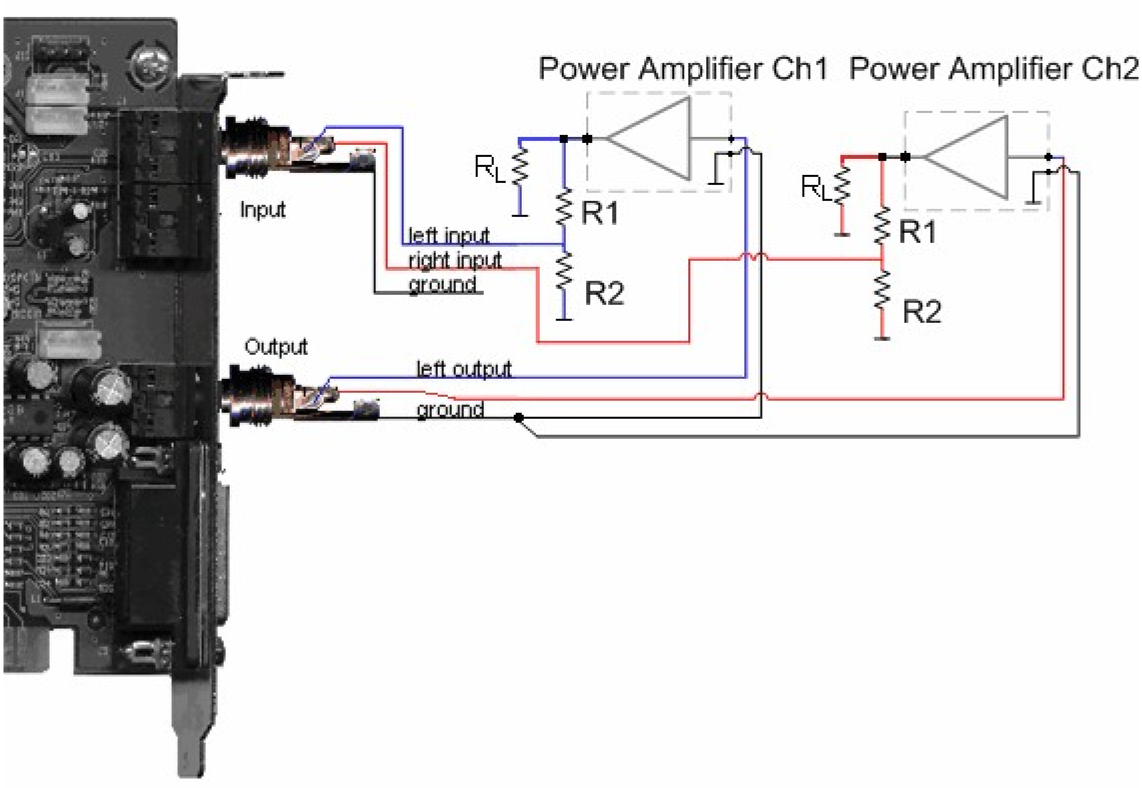


Click No and the measurement is stopped immediately.
Click Yes and the measurement runs until the sound card is overdriven.
Attention: If the first measured point is already overdriven, the measurement is stopped immediately.
Example: Calculated input level sound card: 6dB - 46dB. This doesn't ever work!
In the text of the dialog is a possible solution shown: Use a voltage divider before the sound card input.
See below or in the wiring diagram.



Click No and the measurement is stopped immediately.
Click Yes and the measurement continues until the output level becomes 0dB, or stops before that point.
In the text of the dialog a possible solution is shown: Increase the output level perhaps by using the sound card mixer, or better use the volume control on the test device.

Measurement schematic:



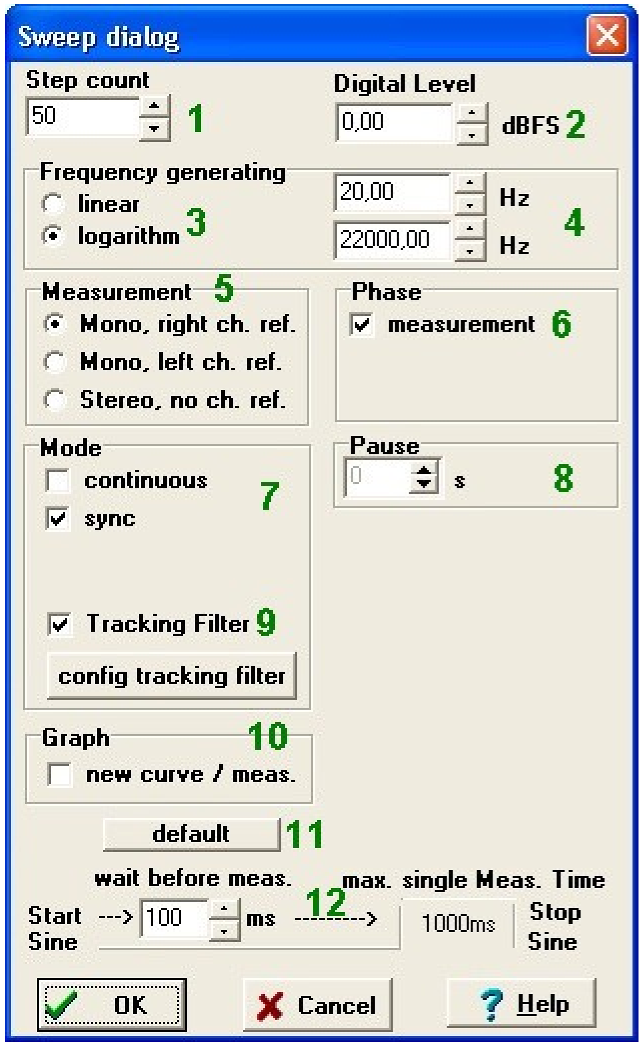
With this measurement schematic you can measured without any risk, even with a bridged amplifier.
In any case, avoid connecting the ground plug of the sound card input with any amplifier outlput!
If your measured device is a bridged amplifier, please activate the corresponding check box in the Power 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) 8Ω.
Max. voltage from the amp. is V=sqrt(PxR)= 28.3V The voltage must be divided down to 1V.
Therefore we choose R1 =3.3kΩ and R2=120Ω and get a ratio from 28.5
The input resistor of the resistor divider is 3.42kΩ in parallel to 50kΩ (of the sound card) = 3.2kΩ
This is no problem to measure at an amplifier output.

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

3.6 Sweep Adjustments

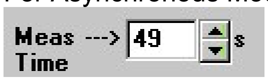
Sweep Setup



1. Count of the frequency steps
2. The digital level of the measurement signal in dBFS (below Full Scale).
3. Step variation of the measurement output signal: linear or logarithmic.
4. Setting of the and End frequencies. Range from 0.1Hz up to SF/2 (half of Sample Rate).
5. Measurement modes:
   Mono (+ phase) with right or left channel as reference.
   Compensates for nonlinear frequencies of the sound card.
   (Also activates Phase Measurement Block)
   Stereo measurement (without any phase measurement)
6. Selection of the phase measurement. You must use a different curve letter for the phase curve (see here).
7. Selection for continuous cycling of sweep measurement (Enables "Pause". See point 8.)
   Synchronous mode: ***audioTester*** applies the measurement signal.
   Asynchronous mode: Measurement signal is applied from an external source. Max. single measurement time. (See point 12.)
8. Pause (seconds) after each sweep cycle (only for continuous measurement). e.g. to set parameters on the measured device.
9. Tracking filter, a bandpass centered the actual sine frequency is applied over the time domain samples.
   Default a bessel bandpass 8th order filter with qualality of 8 is applied. Useful for measurement of

noise measured devices.
The tracking filter is switches ON/OFF with the checkbox "Tracking Filter".
Other filter with button "config tracking filter" . Dialog see here

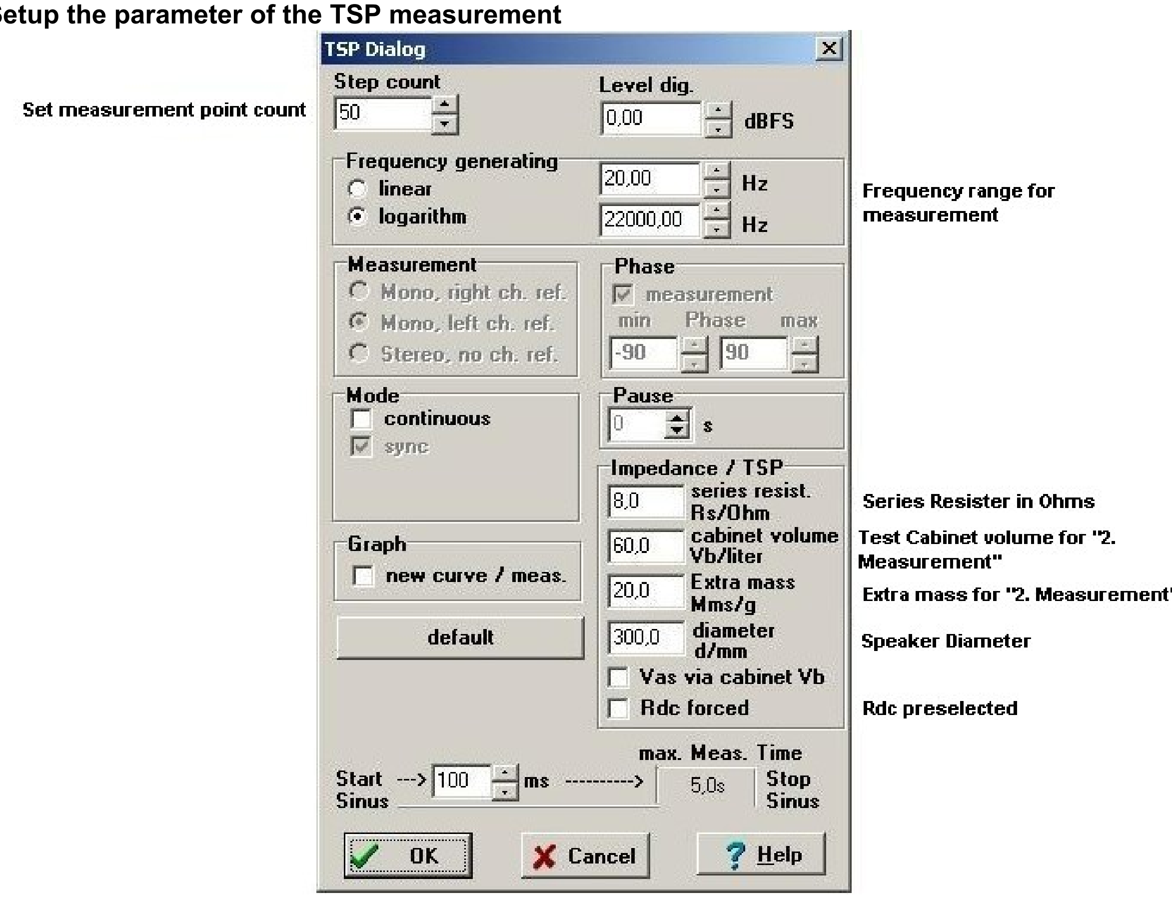
10. Graph a new pair of curves each measurement. (Only in Contnuous Mode)
11. Button to restore the default parameters.
12. For Syncronous Mode: Setting of the measurement delay, and display of the max. time for each measurement step. Max. Time. depends of samplerate and lowest tone frequency. The smallest "Min Time" available is 100ms. The largest "Max Time" available is 50000ms.
For Asynchronous Mode: Display of the whole measurement time is provided.



Select the total duration time of your external sweep signal
When measuring with a pilot tone: Only only the duration of the sweep (without the duration of the pilot tone).

3.7 TSP Adjustments

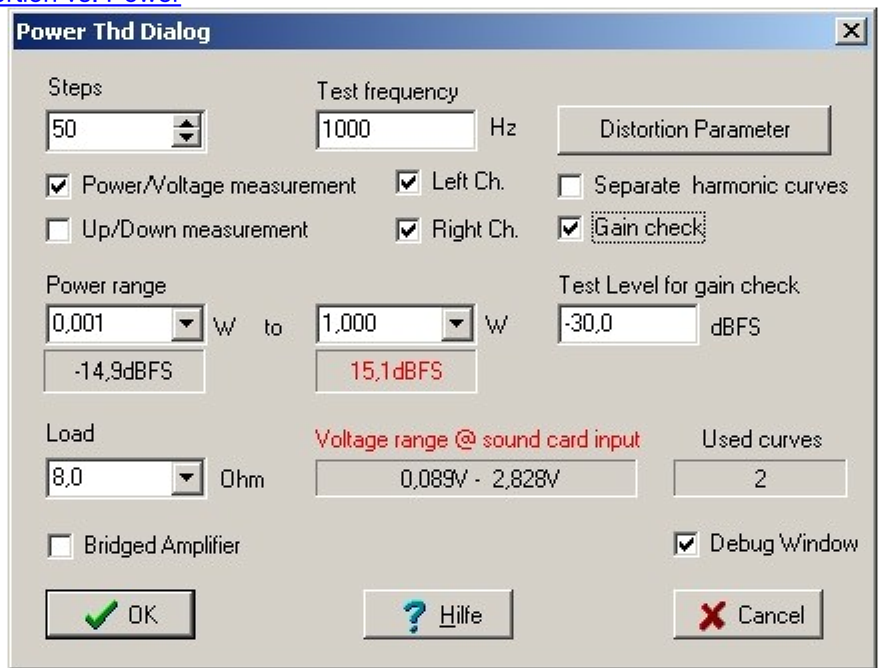
Setup the parameter of the TSP measurement



If you select "Rdc forced" you are able edit self-measured resistance in the main window. A measurement with an accurate multi meter is often better than a measurement with the sound card.

3.8 Power THD Dialog

Power THD Dialog
see also Distortion vs. Power

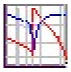


Steps	Count of the measurement steps
Test frequency	Frequency of the measurement tone
Distortion Parameters	Dialog for the distortion measurement parameters
Power/Voltage range	Power range or Voltage range to determine the distortion
Power/Voltage Measurement	Select for Distortion over Power, otherwise Distortion over Voltage
Up/Down Measurement	If selected it is measures 1 curve while running up to the max. level and then a second curve while running down to the min level. Per channel there are two curves consumed (see *used curves*).
Left/Right Ch.	You can select the left or the right channel or both. It influences the number of *used curves* .
Separate harmonic curves	The harmonics (H2-H9) from the Distortion Parameter Dialog are measured separately, each in a separate curve (see *used curves)* Example: A stereo measurement with up/down-run and the harmonics H2+H3 consumes 8 curves, where the first 4 curves must be selected first. Then before the down-run, the next 4 curves will be switched automatically.

	1.Measurement	
	1.curve: left channel	H2 run-up
	2.curve: right channel	H2 run-up
	3.curve: left channel	H3 run-up
	4.curve: right channel	H3 run-up
	2. Measurement	
	5.curve: left channel	H2 run-down
	6.curve: right channel	H2 run-down
	7.curve: left channel	H3 run-down
	8.curve: right channel	H3 run-down

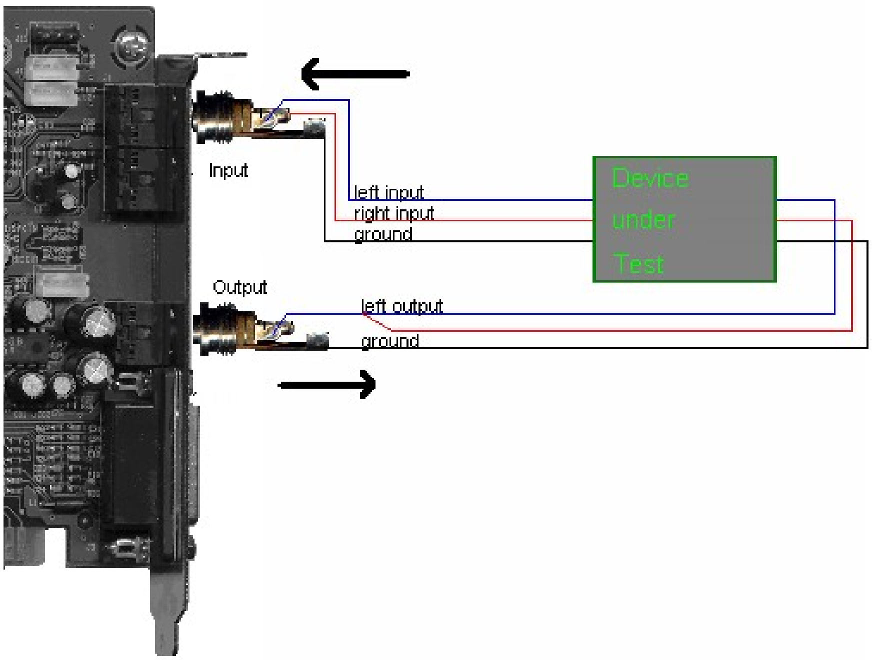
Gain Check	Test tone output for check the output levels or test tone in the range up to 0dB
Test Level for gain check	Digitally test the tone level for a trial measurement check to determine and test all the parameters for the measurement. Never use a loudspeaker during this trial parameter check, as it may be damaged.
Voltage range	Calculated range of voltage expected at the sound card input during the measurement
Bridged Amplifier	If selected, the power will be corrected (Px4). If the measurement schematic is like *see here*, we measure at the bridged amplifier only half of the voltage, and so only a quarter of the power. This will be corrected. To get absolutely correct results, especially H2, you better use a sound card with balanced inputs.
Debug Window	A debug window appears. You can see all terms while measurement. It also is a debug help, if there are problems while measuring
Load	Load resistor for the amplifier under test, *see here* R_L.

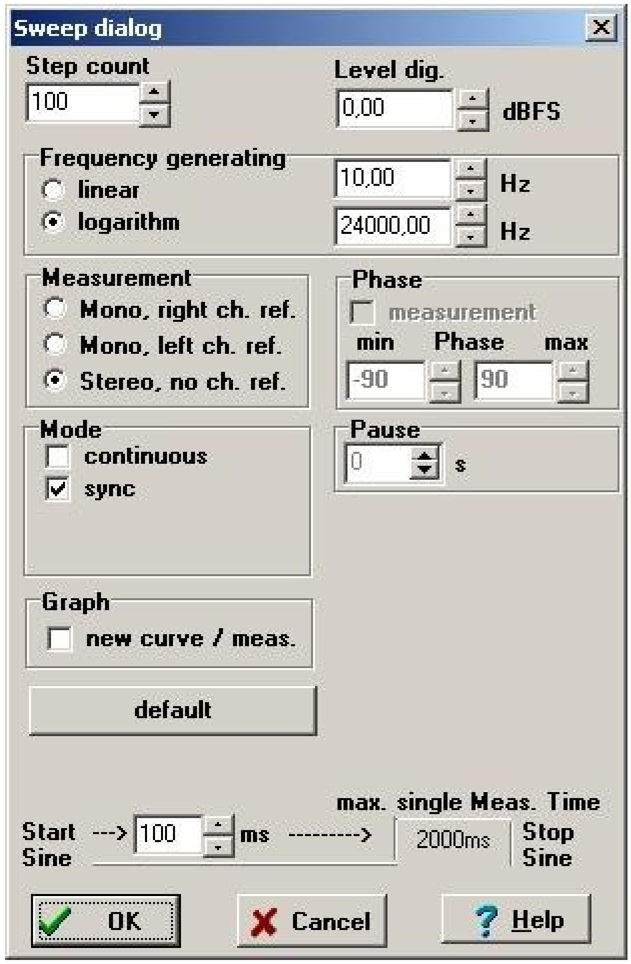
3.9 Sweep Wiring Diagram

 **Help**

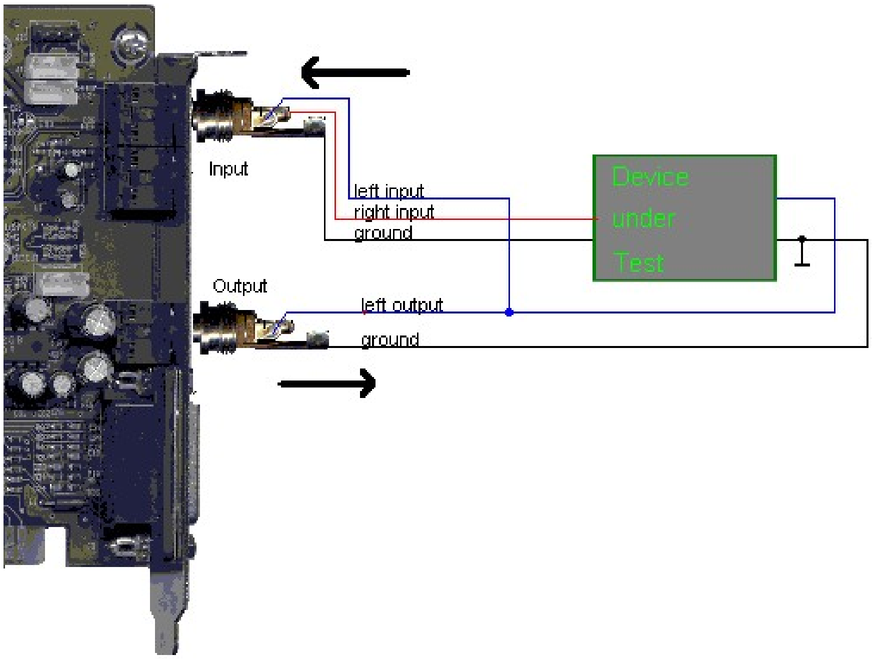
Typical wiring

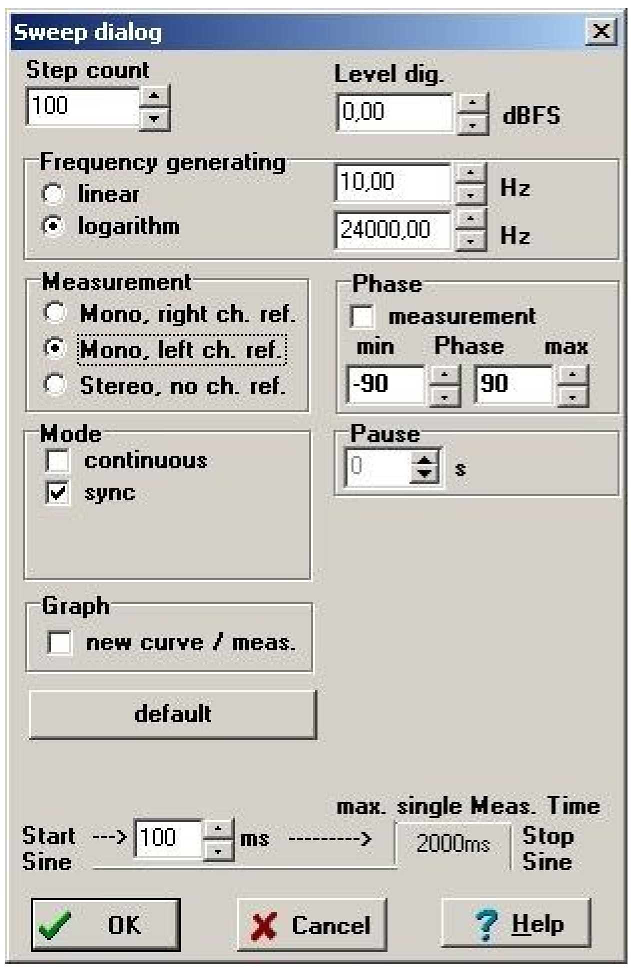
Stereo measurement:



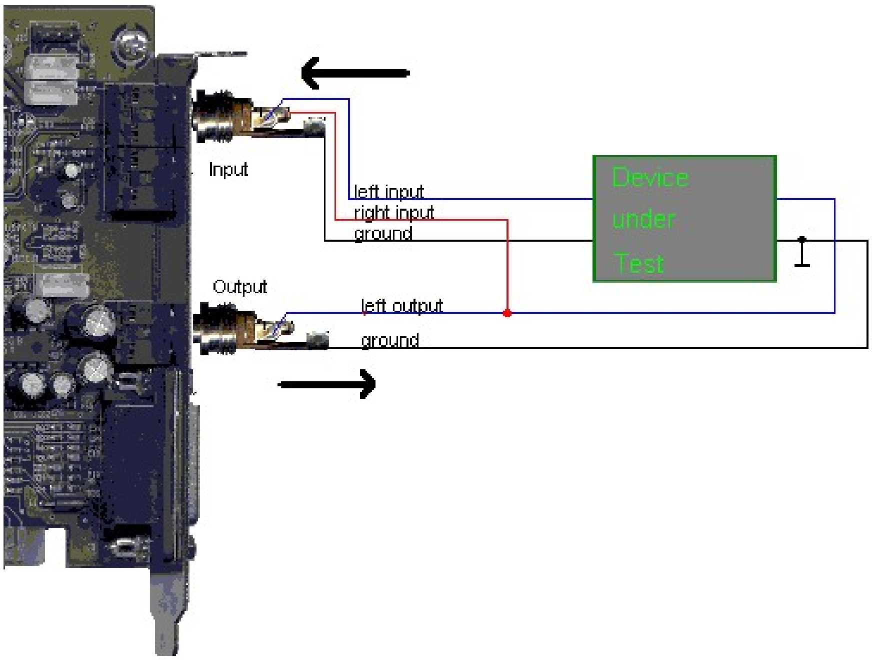


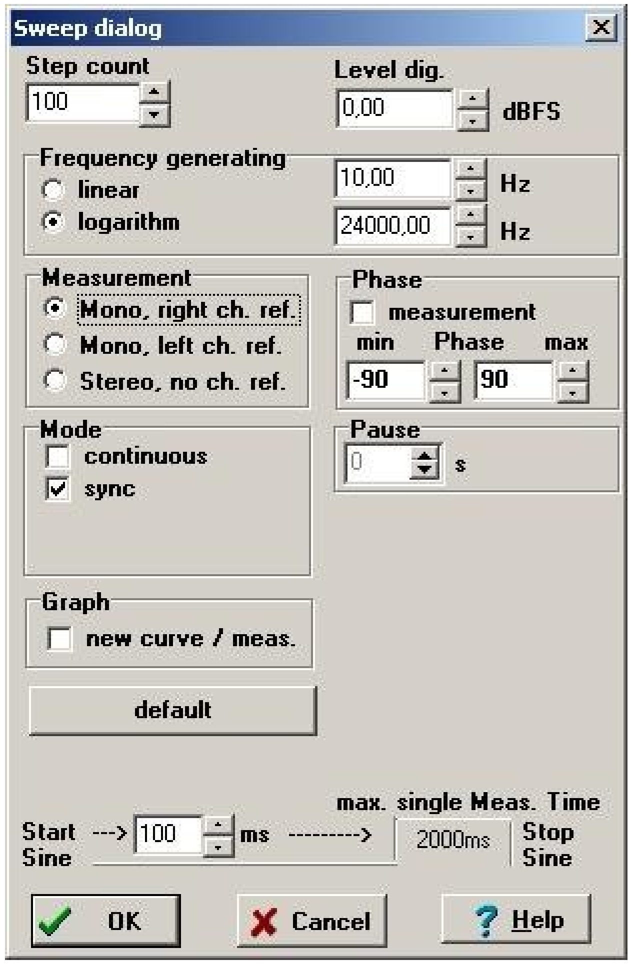
Measure Right, with Left as reference channel:


Input
left input
right input
ground
Device
under
Test
Output
left output
ground


Sweep dialog
Step count
100
Level dig.
0,00
dBFS
Frequency generating
linear
logarithm
10,00
Hz
24000,00
Hz
Measurement
Mono, right ch. ref.
Mono, left ch. ref.
Stereo, no ch. ref.
Phase
measurement
min
Phase
max
-90
90
Mode
continuous
sync
Pause
0
s
Graph
new curve / meas.
default
max. single Meas. Time
Start
Sine
--->
100
ms
--------->
2000ms
Stop
Sine
OK
Cancel
Help

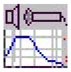
Measure Left, with Right as reference channel:



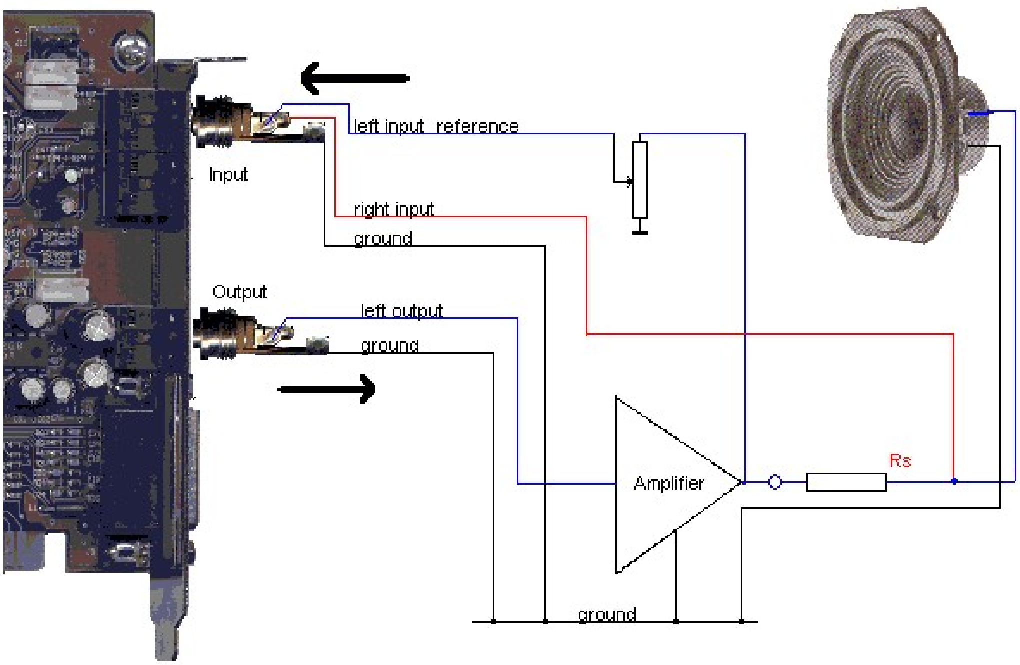


Remark:
For the stereo measurement (see above), the left channel output is used for both measured device inputs, this guarantees the same conditions for both inputs.
Of course you can alternately use the right channel to both inputs.

3.10 Impedance Wiring Diagram

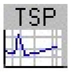
 Help

Typical wiring:



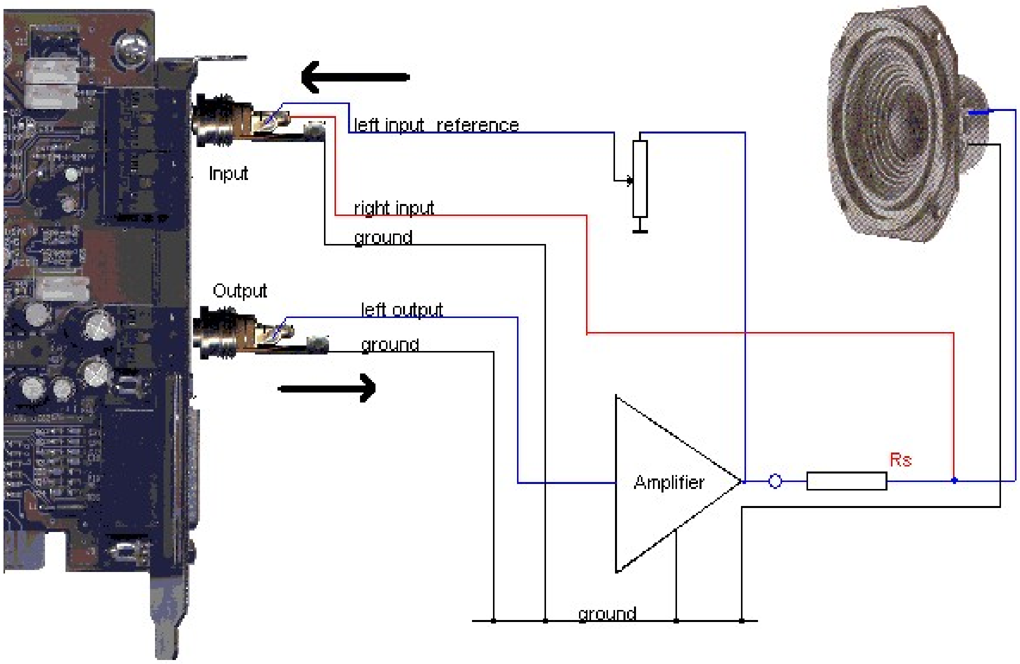
Remarks:
Wiring of the Left input (blue wire) is used as a reference for measurement.
Take precaution not to exceed the max. input level of the sound card.
Increase the output level slowly.
Don't use a bridged amplifier.

3.11 TSP Wiring Diagram



Help

Typical wiring:



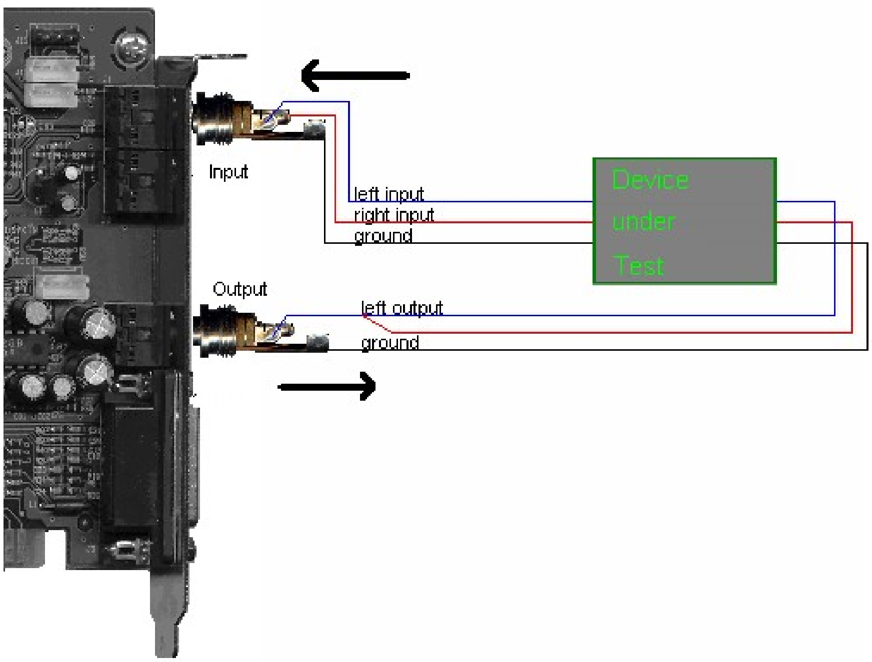
Remark:
The resistor Rs should have a similar value as the impedance of the loudspeaker (4-20Ω). The potentiometer protects the sound card input. Use a cable with adequate heavy guage, not just any test cables.
Place the test loudspeaker chassis loosly on a flat surface and the membrane facing upward (see photo).

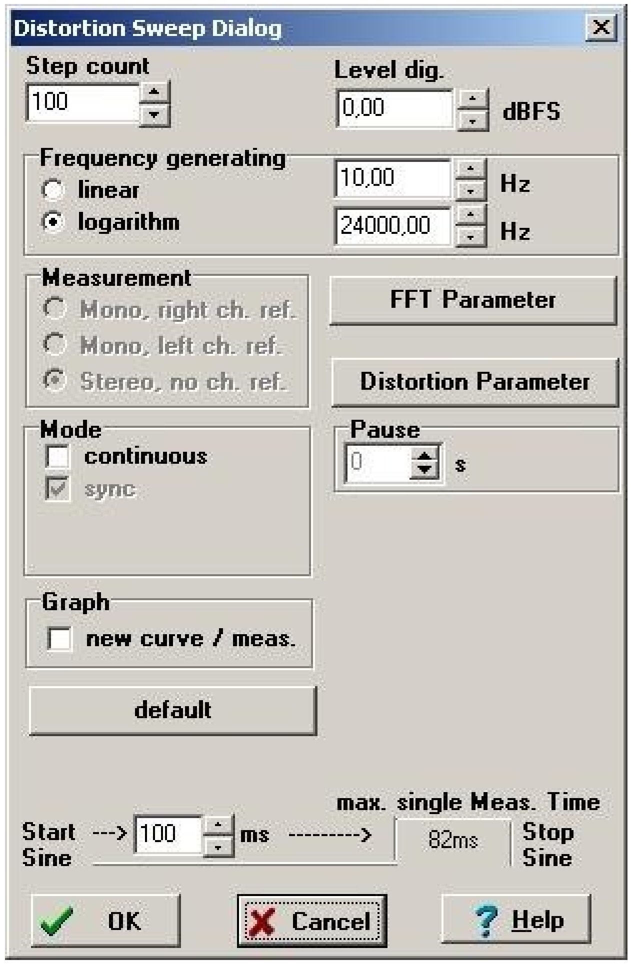
It is a small signal measurement, therefore low levels, low volume.
CAUTION: Because of working in the 'free air', the loadspeaker could be damaged. Be carefull to follw the diagram and the parameter settings.

3.12 THD Wiring Diagram

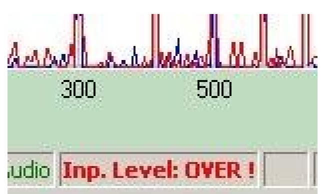
Help

Typical wiring:
Stereo measurement:





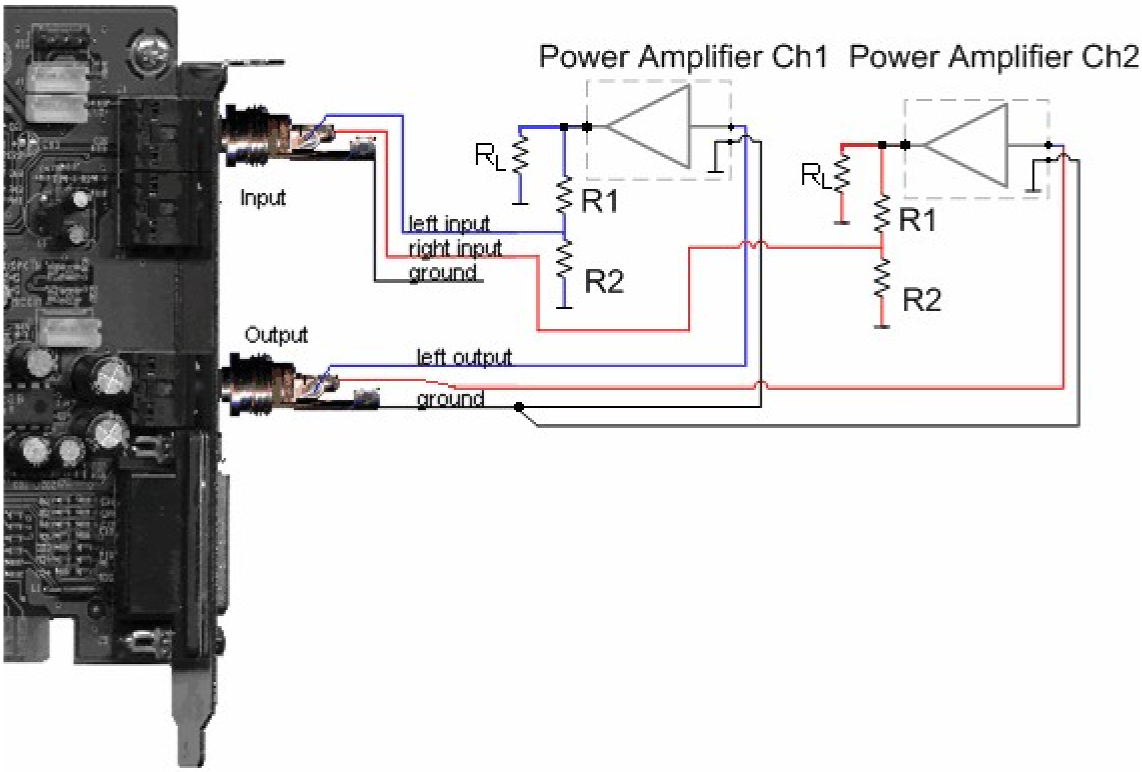
Remark:
As discussed in the help, please beware of over driving the input of the test device.
Over driving is not shown in the ***audioTester***, and it is not easy to detect. Decreasing the level with the windows mixer is not often the best idea. Possibly you could increase the distortion of the outgoing signal of the sound card. This effect happens very strongly with the distortion vs. power measurement.
Over driving of the sound card is shown by the ***audioTester***, as below.



3.13 Power THD Wiring Diagram

Help

Typical wiring:



Remarks:
With this circuit diagram it is possible to measure bridged amplifiers. In this case you must avoid connecting input ground to any amp output. And if your measured object is a bridged amplifier, you must selected "Bridged Amplifier" in the Power THD Dialog.

How to choose the Resistors R1 and R2?
Example:
Input sensitivity of the sound card is 1V.
Max. power of the amplifier is (P) 100W at (R) 8Ω.
Max. voltage at the amp output is V=sqrt(P x R) = 28.3V .
This voltage must divided down by the resitor network to 1V.
The amplifier voltage must be divided by (R1+R2)/R2
Therefore choose R1 as 3.3k and R2 as 120Ω --> divides by 28.5.

The input resistor of the sound card is much higher than the impedance of the voltage divider, so that there are no problems. The load of the 3.4kΩ (3.3+0.12) of the voltage divider for a power amplifier no problem. Now you must calibrate the sound card with the new voltage divider.

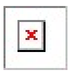
If you measure bridged amplifiers you **do not** connect the ground input connector of the sound card to any amp output, the amp output would be short-circuited. Here you must use only one bridged amp's output and apply the ground connector of the sound card to the amp case. In this case you measure only the half of the voltage and a quarter of the power.

4 Impulse measurement

4.1 2D impulse measurement

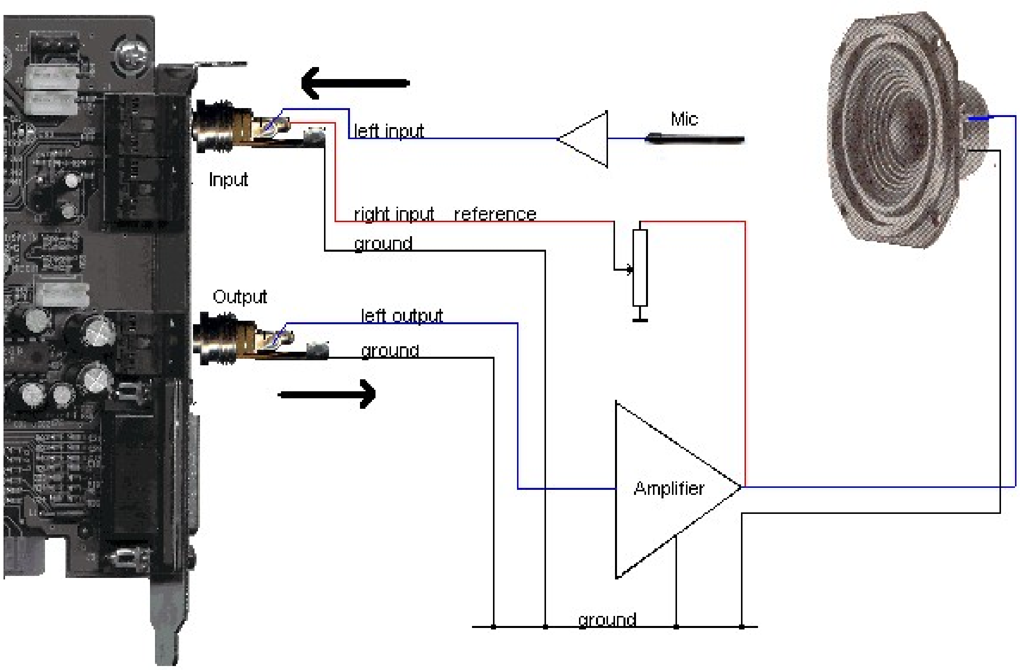
With this measure mode you are able to measure impulse responses of loudspeakers and filter systems. As a stimulus you can select a Dirac impulse, an MLS-Impulse (Maximum Length Sequence) or a sine wave burst. To select, click the setup button on the left, it opens the Sound-Setup.(see here)
You can also apply the impulse from external source via CD for example. In Setup Dialog you can select it with *external Impulse*.

Also on this page measurement of delays and measurement with limiters



Measurement of a wire !

Wiring Diagram



Important hints:
Input (right, red) is only necessary if you use the reference measurement. Effect of the reference measurement you can see here
Please pay attention to the maximum input voltage applied to the sound card.
Perhaps use an 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 should be set equally.

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

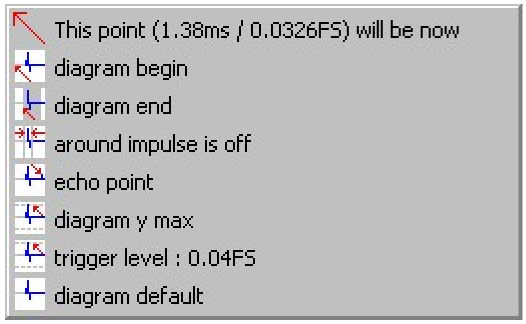
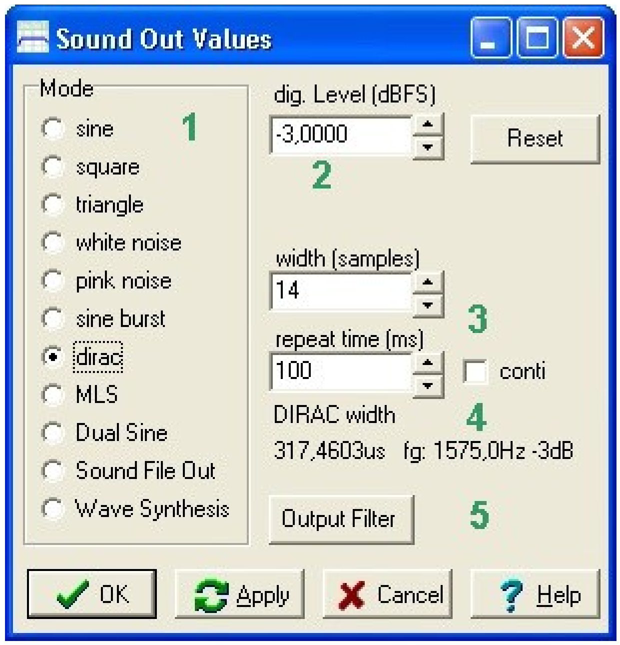


diagram begin	New beginning of the time diagram at the clicked position (here at 1.38 ms)
diagram end	New end of the time diagram at the clicked position (here at 1.38ms)
around pips	The displayed time range is set to new values, which makes it possible to see the environment around the impulse.
echo point	Setting of an echo point while impulse measurement, only till this point the FFT works (shown by a red, vertical line)
diagram y max	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

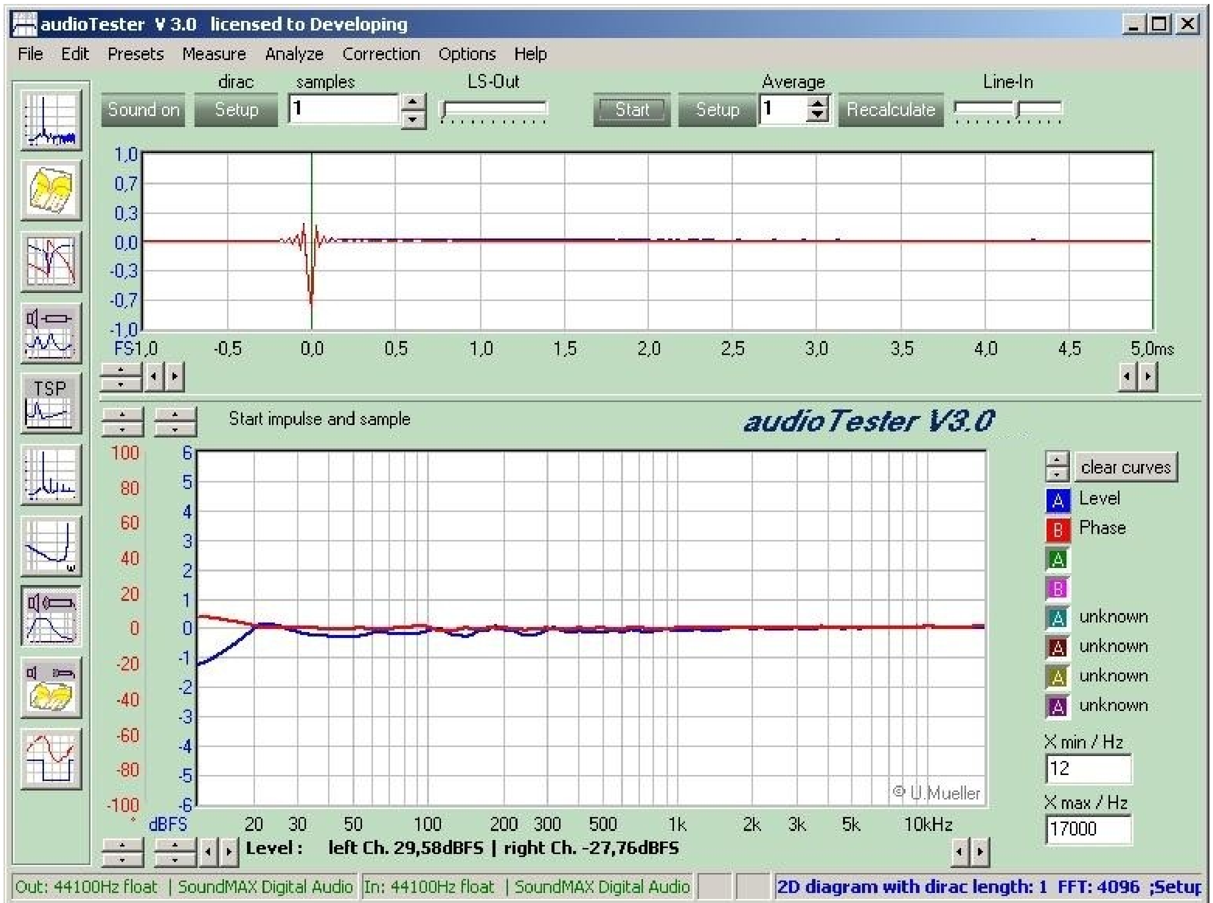
If you click the first row (at "This point ..."), you leave the menu without any changes !

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 level will be set to -9dBFS .
3. Selection dependents from the selected wave form, here you adjust the width of the Dirac impulse
4. Display of some calculated values, it dependents 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) -> 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
5. Opens the output filter dialog (see here)

Reference measurement of a piece of 'wire'



blue graph: originate frequency response of the 'wire'
red graph: phase
The small differences from an ideal graph, are explained by differences in the I/O E-CAPs of the sound card.

Example Measurement:

Sub woofer measurement:

Remarks:
For the first measurement a Dirac impulse with a width of 226µs is used (blue curve).
Then the start of the FFT was set at the point to -10ms manually in the time domain graph (by left click in the time domain diagram and click on *FFTbegin* in the Popup-Menu)
and with the button "Recalculate", it is recalculated (red curve). Both curves worked with the display filter "smooth 1/4 oct.".

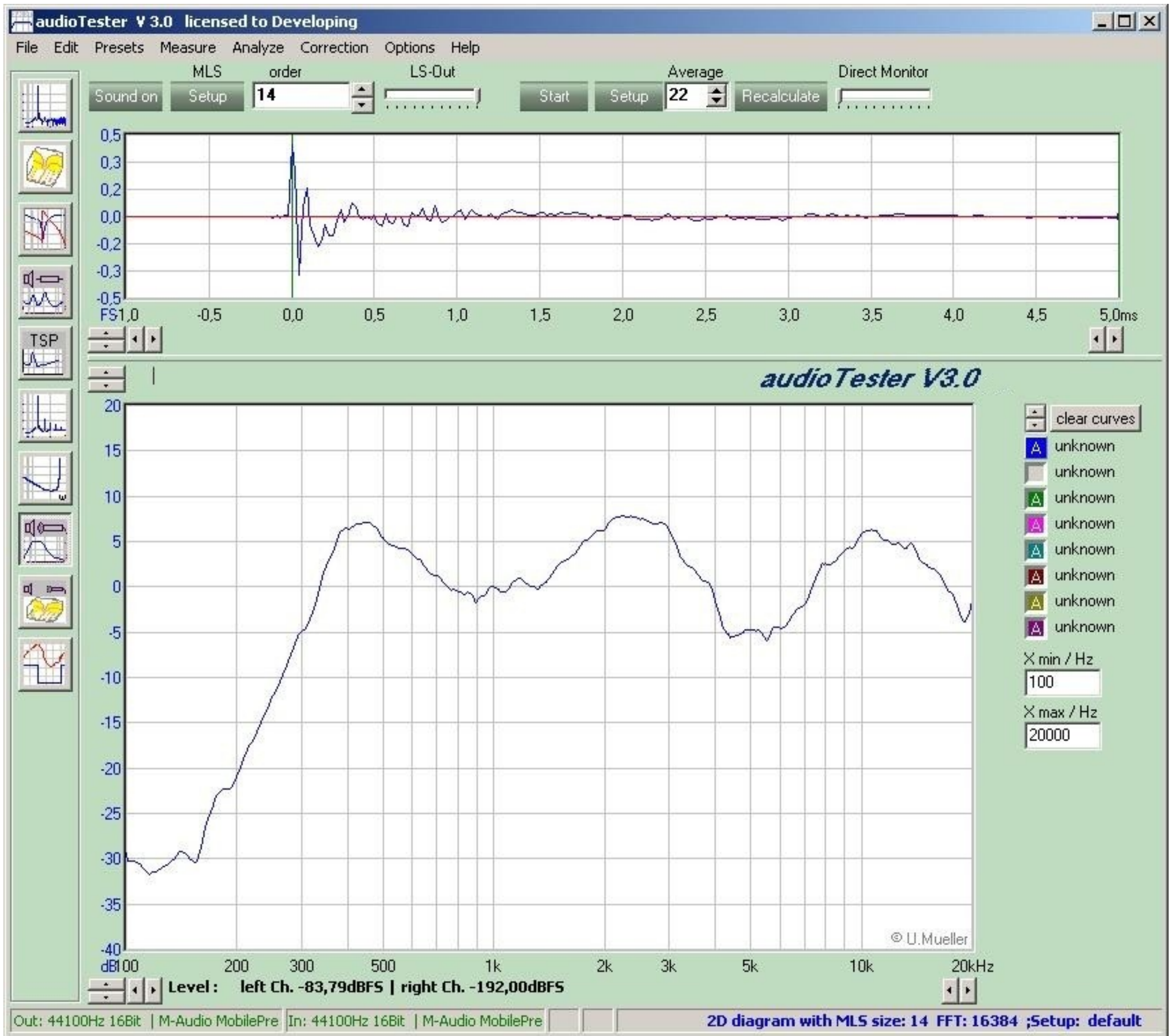
Measurement with Dirac 10

The second measurement was done with a MLS impulse order 14 (blue curve).
To eliminate echo of the room we set two echo points (left click in the time domain diagram and click on "*Echo point"* in the Popup-Menu) (green and pink curve).
Furthermore the FFT starts at -10ms (Popup-Menu FFTbegin).
All curves are smoothed with a display filter 'smooth 1/4 oct.' and paint with a line width of two see curve dialog.
Comparing this measurement with a pink noise spectrum or a sweep measurement, the Impulse measurement is faster and more flexible (with echo elmination point)

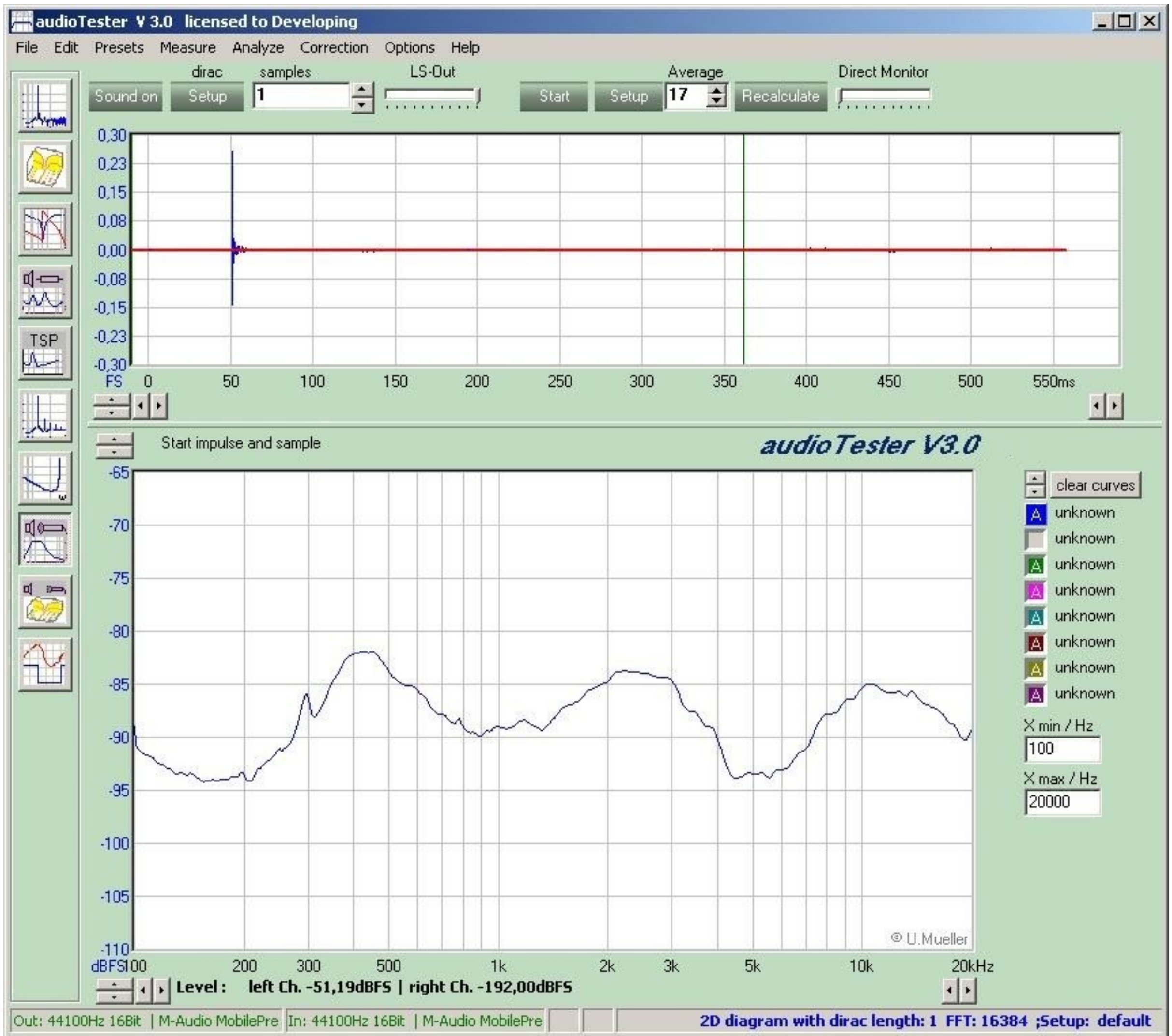
Measurement with MLS 14 and different echo points

PC Speaker measurement

1. Measurement with MLS 14, averaging 22 times, microphone distance 30cm.
2. Measurement with Dirac impulse, averaging 17 times, microphone distance 30cm.

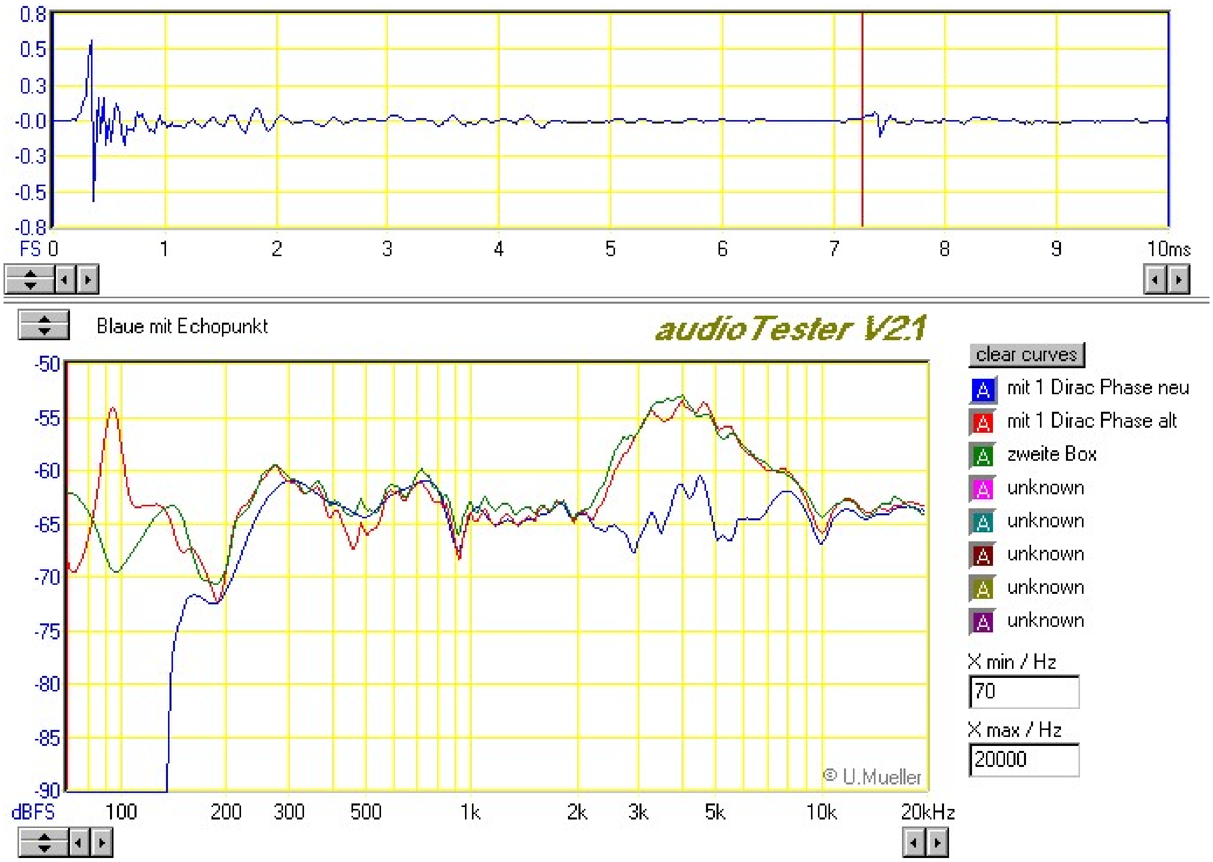


MLS 14



Dirac Impulse

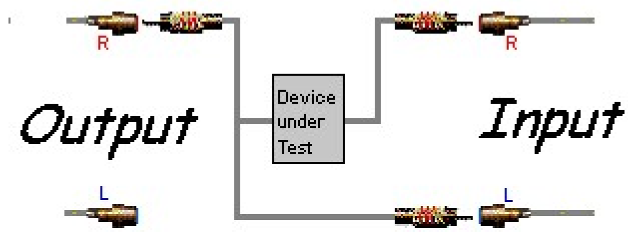
Echo point example



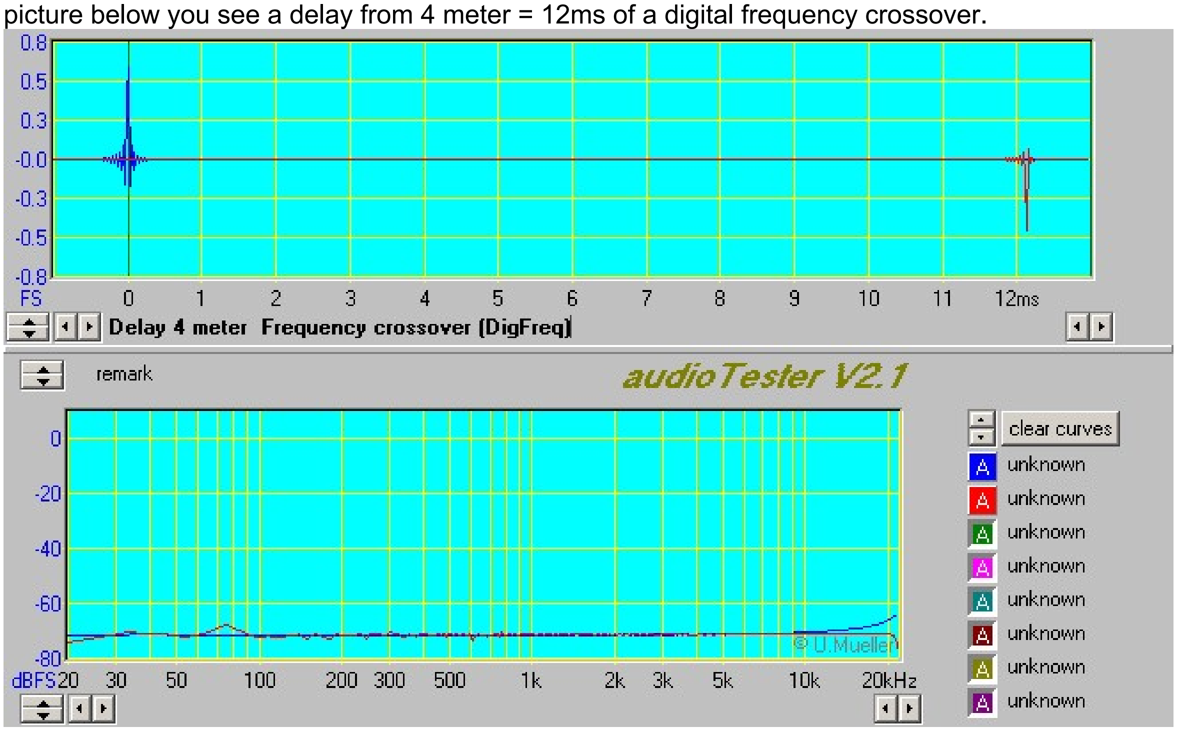
The blue line is calculated with the echo point (see red line in time domain diagram) and a click on the "Recalculate" button.
You see the missing low frequencies, depend on the decrease of the FFT-lines. The red and green line are calculated without an echo point, but with the same real echo. You see 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, which looks like the wiring of a reference measurement. But you don't select the reference measurement in the setup dialog!

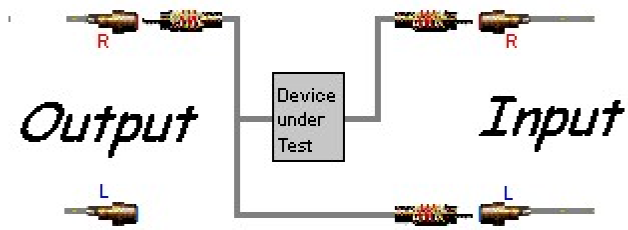


We use the time domain diagram, which is now extend to 2 curves. The audio device is stimulated with Dirac impulse, we see then the undelayed and the delayed impulse in the diagram. In the picture below you see a delay from 4 meter = 12ms of a digital frequency crossover.

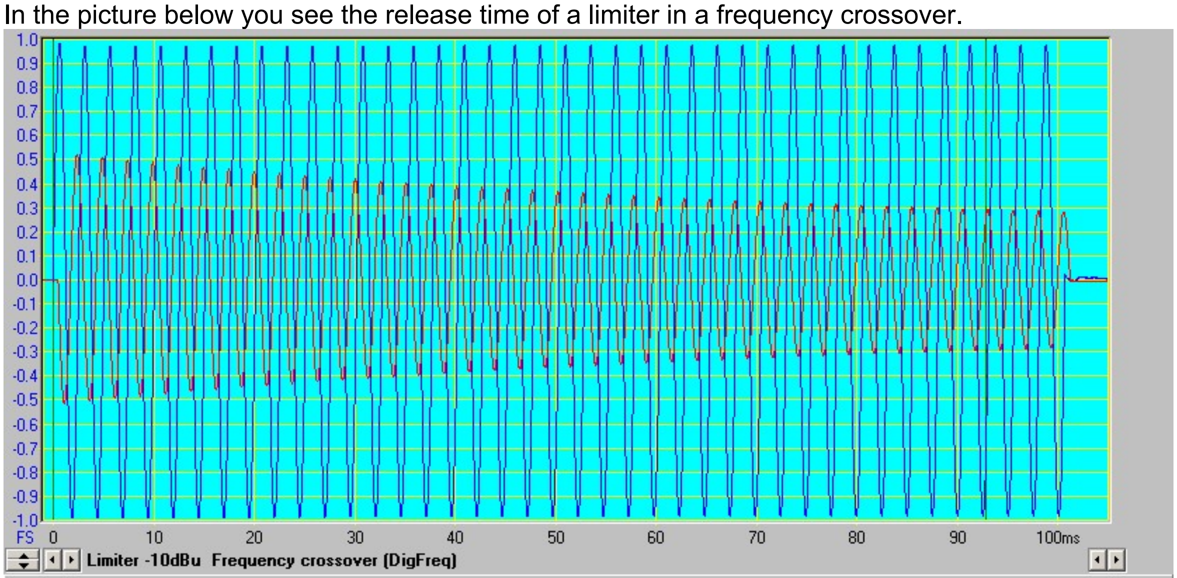


Limiter measurement

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 reference measurement !

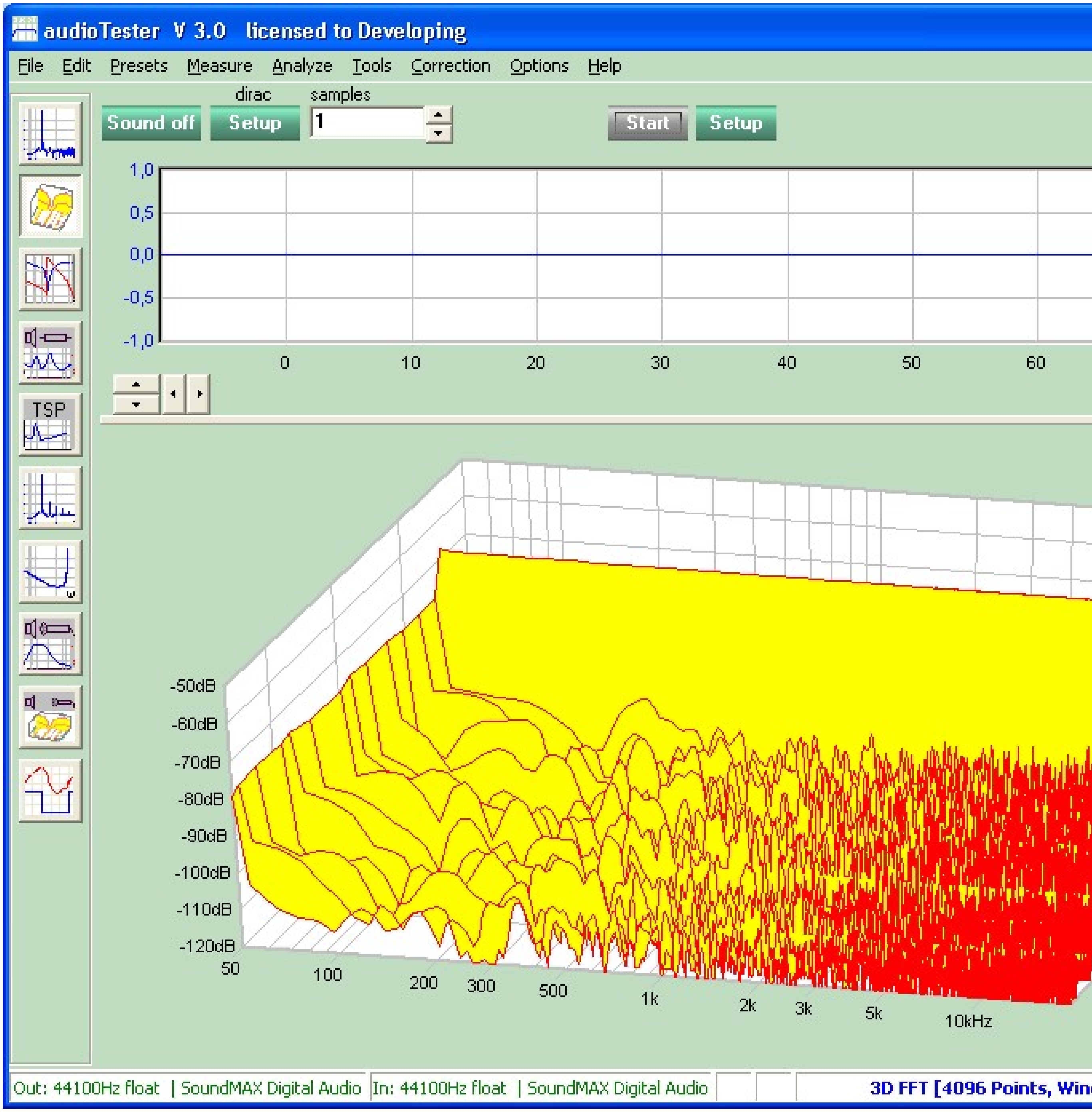


We use the time domain diagram, which is now extended to 2 curves. The audio device is stimulated with a burst impulse (!). We then see the attack and release progression in the time domain diagram.
In the picture below you see the release time of a limiter in a frequency crossover.



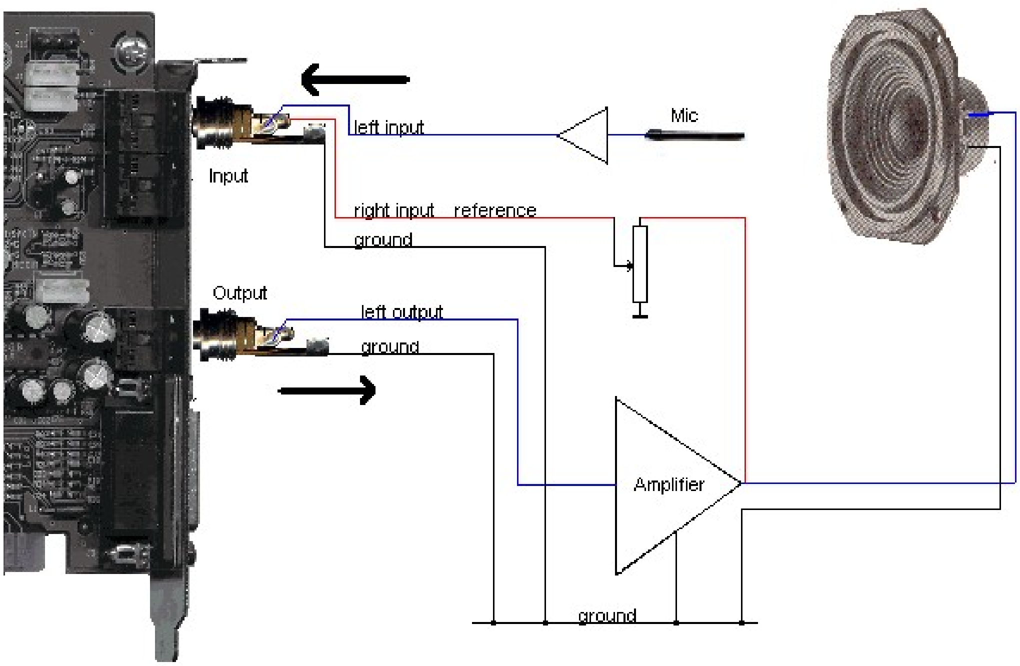
4.2 3D waterfall plot

With this measuring mode you are able to measure impulse responses of loudspeakers and filter systems and its breakdown over 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 opens the Sound-Setup.(see here)



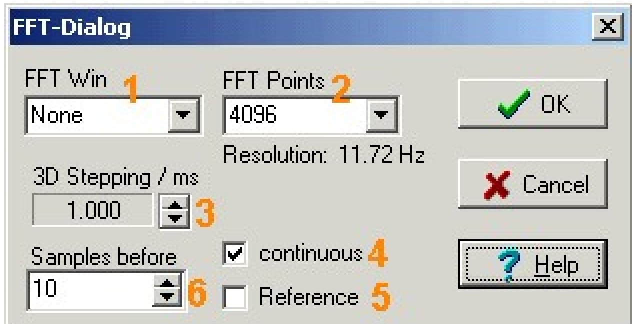
1. Impulse response in the time domain. Display options for the time diagram see here
2. FFT-Window, additional parameters see here
3. Impulse response in the frequency domain, see parameters for the waterfall plot see here

Measurement schematic



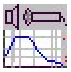
Important hints:
Reference input (right) only necessary if you use reference measurement.
Please use caution not to exceed the maximum input voltage of the sound card.
For sound card input protection use an attenuator before the reference input.
Please increase output level slowly.
Do not use a bridged power amplifier.
The left and the right output of the sound card are typically equal.

3D FFT Dialog

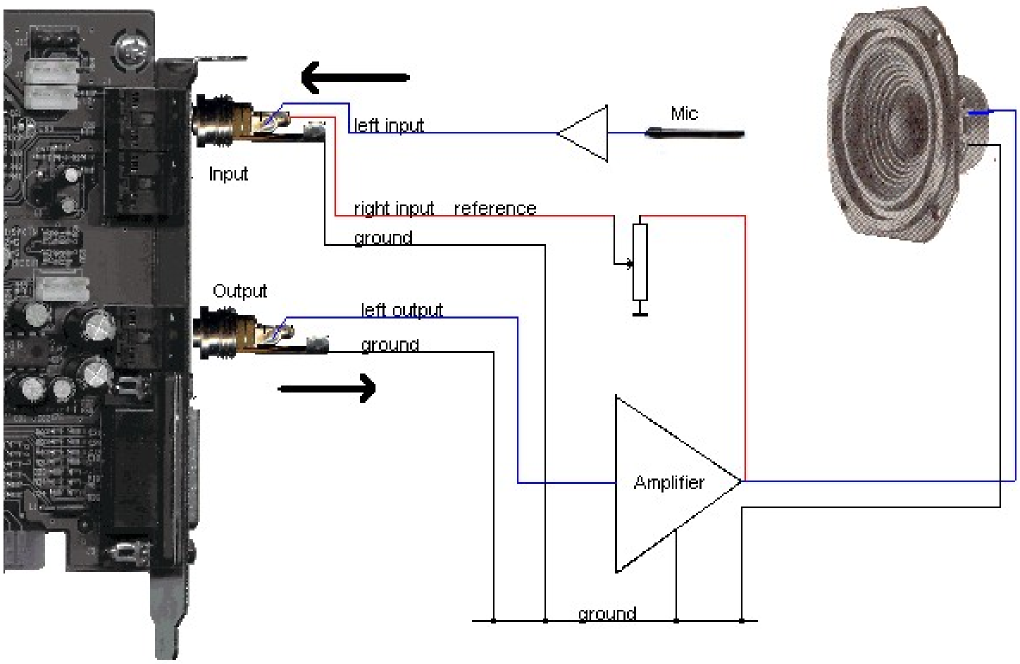


1. *FFT-Windows*, here "none" is better.
2. Count of *FFT-Points*, from 64 to 32768. See below for the steps in Hz, which depends on the sample frequency and the FFT-Points
3. Steps of the frequency ribbons. The step size depends on the sample frequency
4. Choice between single or *continuous* measurement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

4.3 Impulse Wiring Diagram

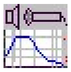
 Help

Typical wiring:

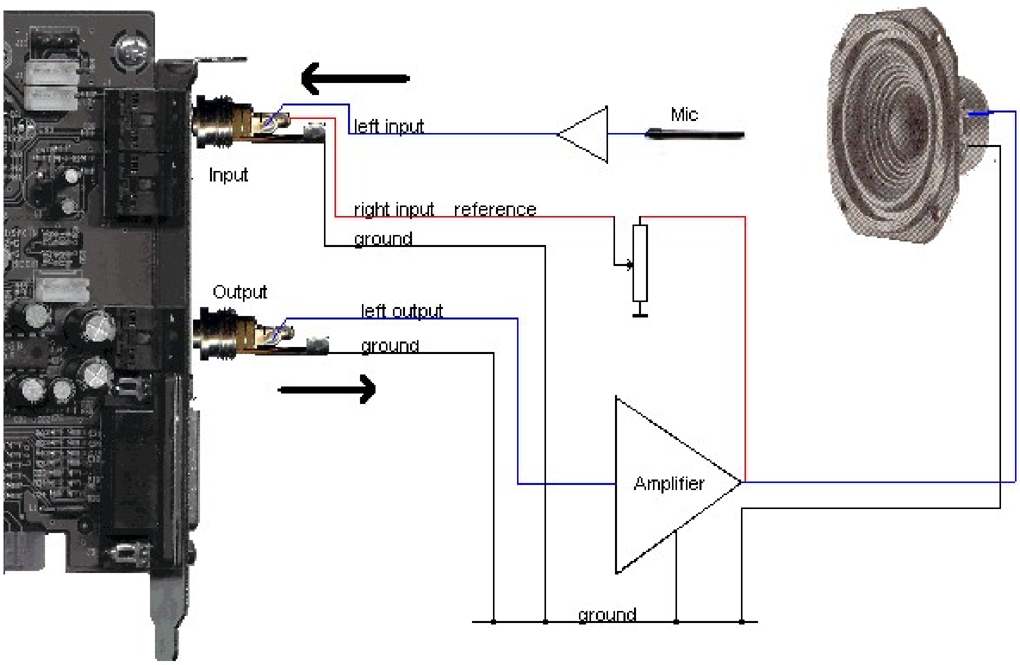


Remarks:
Wring of the right input (red wire) used only if reference measurement is used. See influence of the reference measurement here
CAUTION: Be carefullnot to exceed the max. input level of the sound card.
Use a potentiometer in front of the right input.
Increase the output level slowly.

4.4 Waterfall Wiring Diagram

 **Help**

Wiring Diagram:



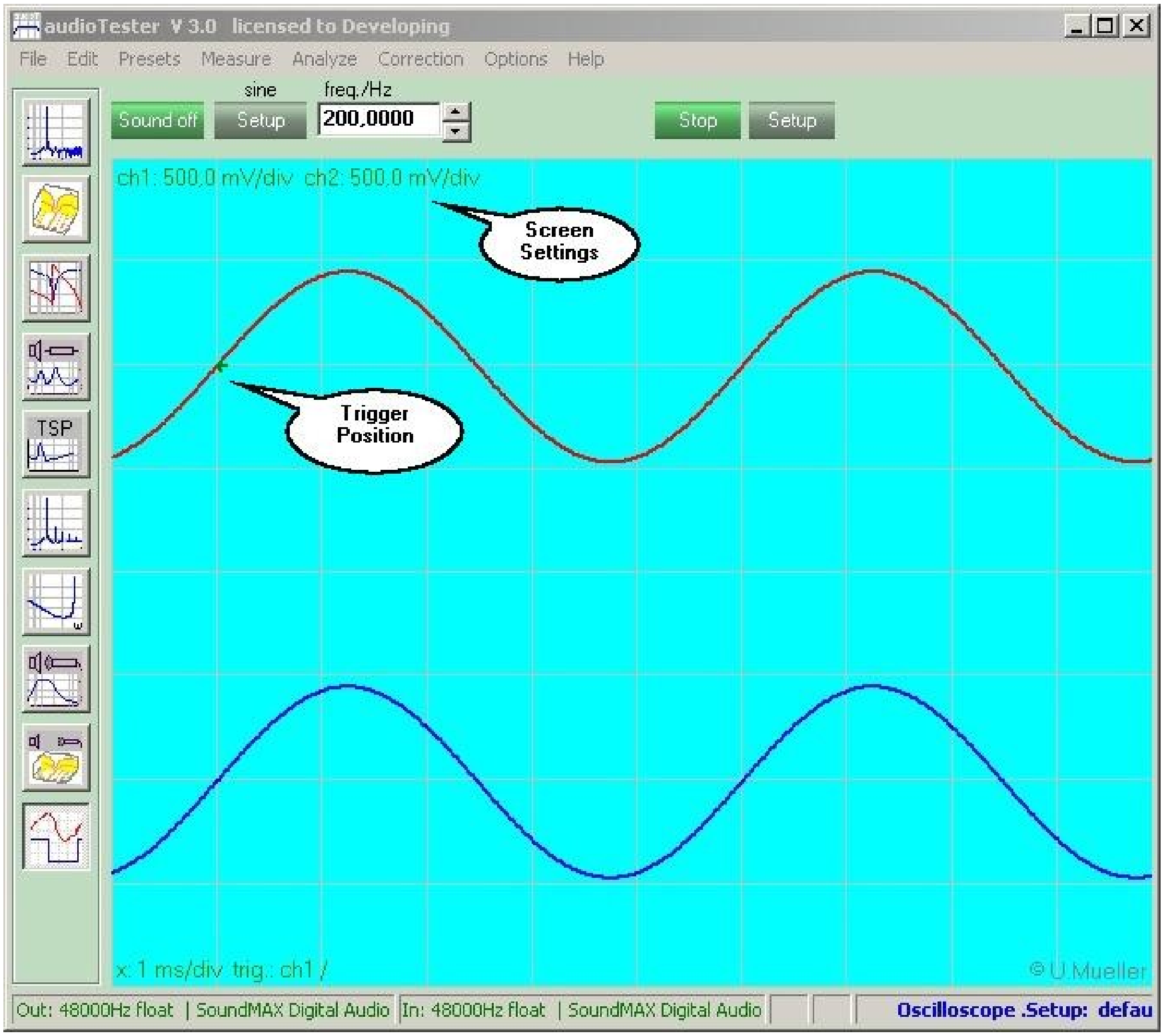
Remarks:
Wiring of the right input (red wire) used only if reference measurement is used.See influence of the reference measurement here
CAUTION: Be carefullnot to exceed the max. input level of the sound card.
Use a potentiometer in front of the right input.
Increase the output level slowly.

5 Oscilloscope

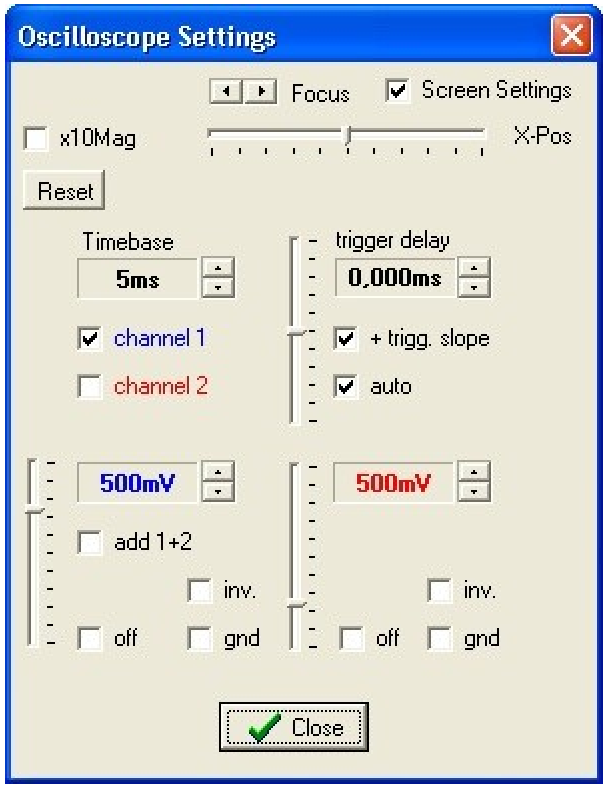
5.1 Oscilloscope

Features:

- Time base: 20µs - 200ms/div
- X/Y Mode
- Trigger channel selectable
- Trigger slope selectable
- Trigger auto/manual
- Trigger Delay adjustable
- Y-Mode: 200µV - 2V/div (calculated after calibration)
- inverse channels
- adding channels
- Simulated beam focus
- x-Pos adjustable
- x-beam 10x magnification



Settings Dialog



Time base:
The time base has a range from 20µs to 200ms/div in a 1-2-5 sequence and of course an x/y mode.

Voltage settings V/div:
100µV to 2V/div in a 1-2-5 sequence. Attention, this value is calculated, there are no physical resistors or amplifiers inside or outside

Illum, Focus and Screen Settings:
Illumination: Now replaced by a color setting dialog, right mouse click to the display the Diagram Options dialog.
Focus: Focus of the beams, simulated by line width
Screen settings: Display of x and y settings

Trigger delay
With the trigger delay you can see signals before the trigger event. The delay time adjust works in steps of the sample frequency. At 44100Hz the steps are 22.67µs .

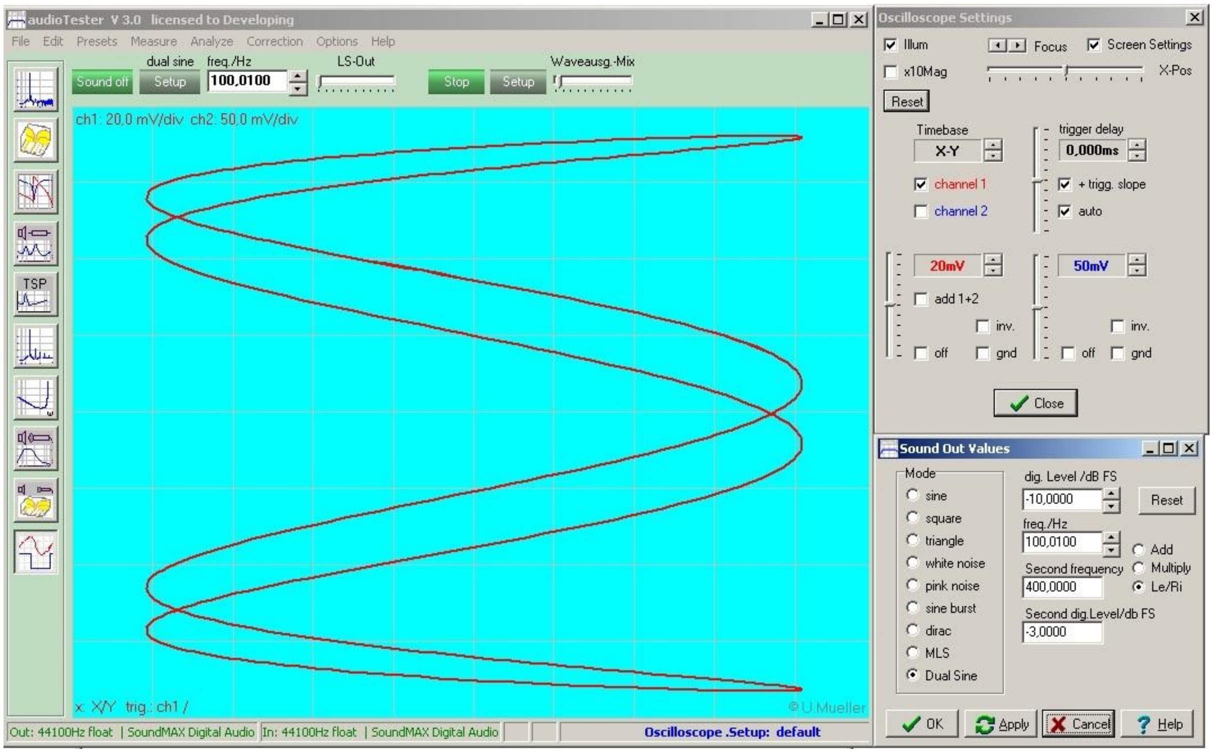
trigg. slope +/-
pos. or neg. slope to trigger.

Reset Button:
Set all settings to a default state.

Restrictions:
There is no AC/DC switching, because of e-caps in every sound card.

Wiring Diagram here

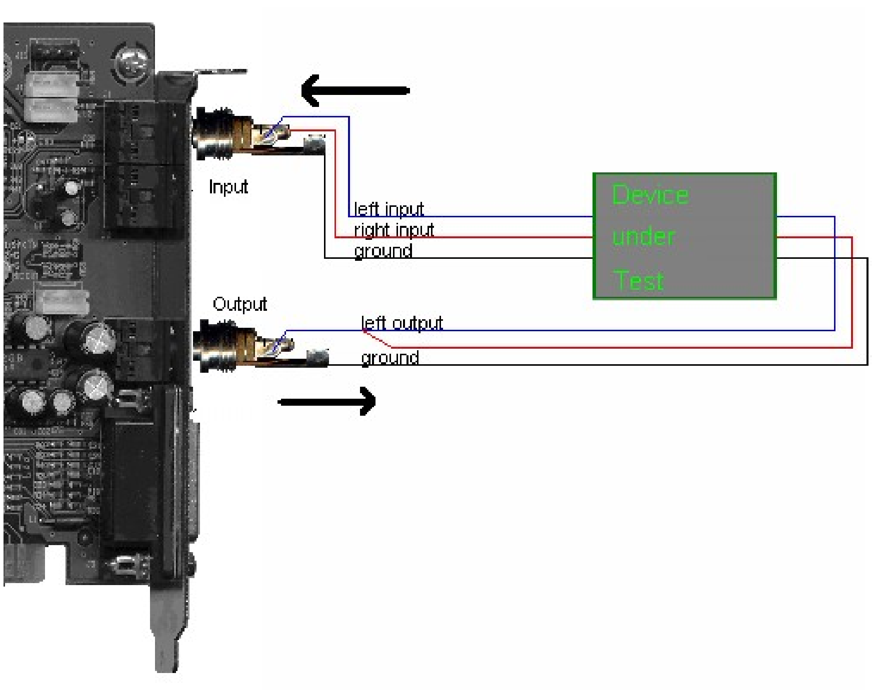
Example: Lissajous-figure



5.2 Oscilloscope Wiring Diagram

[Help](#)

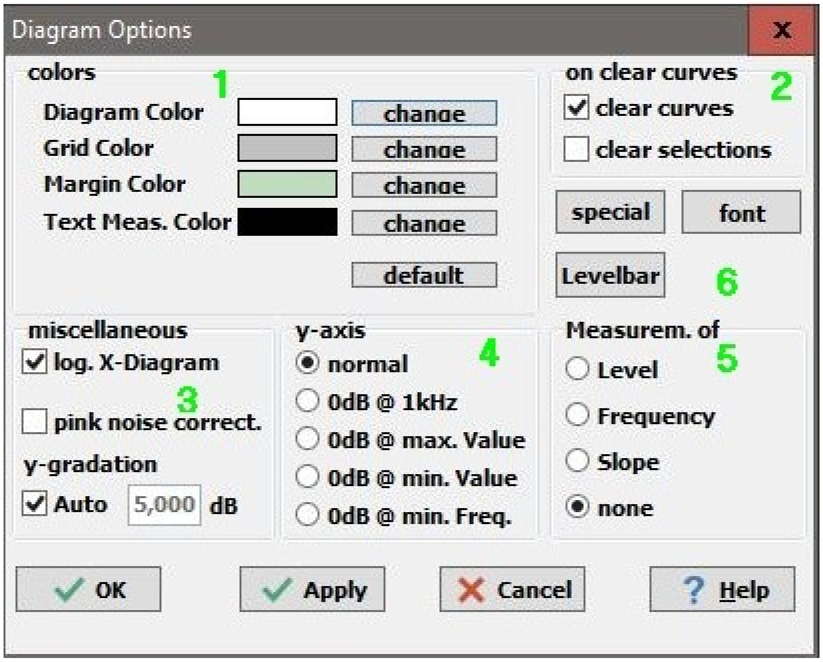
Typical wiring:



6 Dialogs

6.1 Diagram Dialog

Diagram Dialog Box
Enter this dialog box with a right mouse click anywhere in the main waveform diagram.



1. A click on the "Change" buttons starts the color dialog for the Diagram-Grid and the Margin (diagram surface/background).

2. Here you can define the function of the "*clear curves"* button in the diagram. The button can be set to "clear all curves" and/or "reset all selections", that means the first two curves are now selected and visible.

Button "special": Specifies the X-Axis Unit (normally Hz) and specifies a factor to multiply the X-Axis values.
For example: the diagram shows a measurement of a 1kHz sine wave, if you use a factor of 4 the diagram shows a sine wave with '4kHz' .
Another example: If you want to show revolutions per minute, set X-Axis unit to 'rpm' and set the X-factor to 60 .

3. Here you can switch between a linear and a log scale X-axis.
"pink noise corr.": Apply a pink noise filter to linearize a curve measured with pink noise signal.
"pink noise corr." deselect

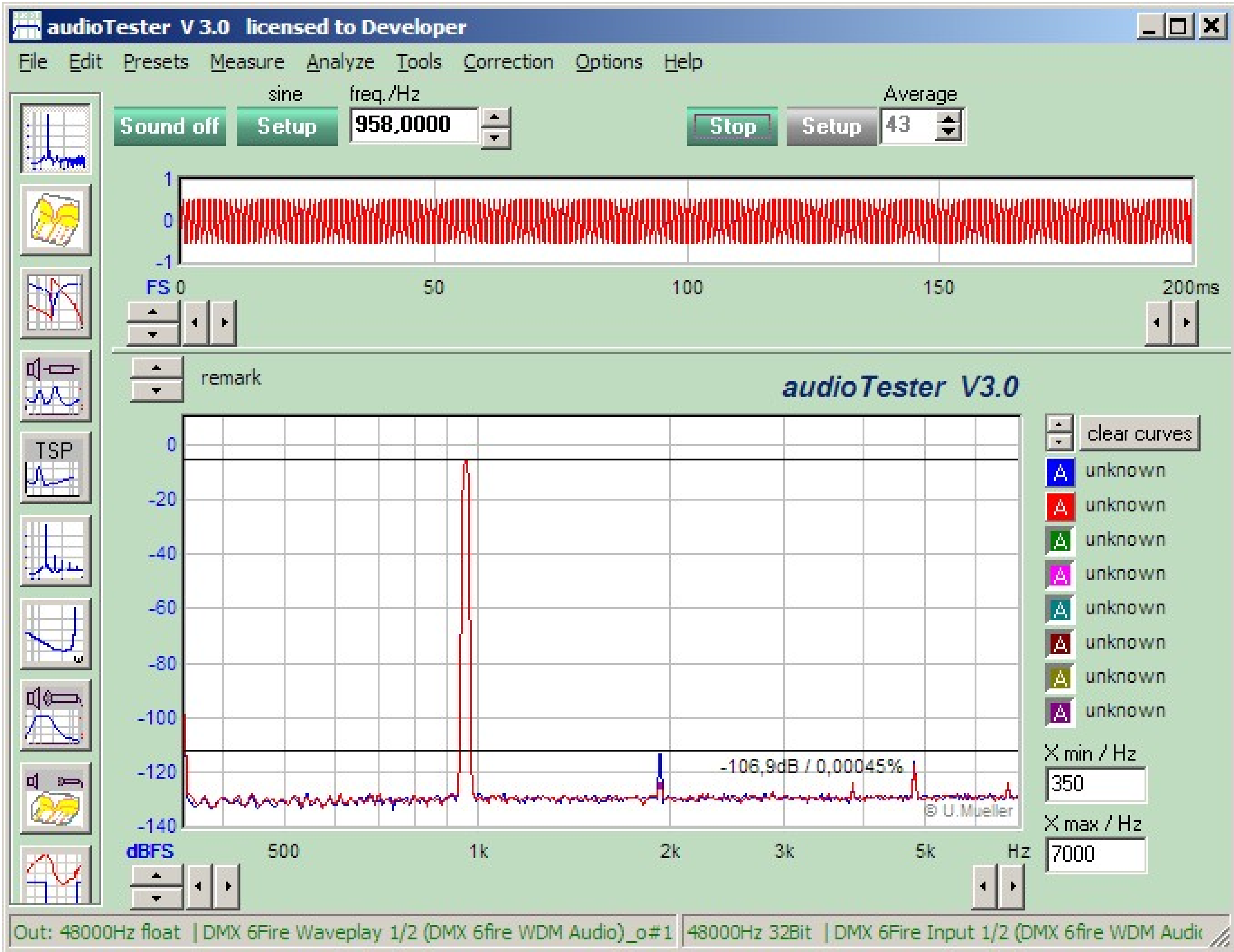
"pink noise corr." select

"y-gradtion": Switch on/off auto gradation. Off enables the textbox for manually Y-scale settings

4. The scaling of the curves can be set to normalize the Yaxis at 0dB. This is selectable for 0dB at 1kHz or 0dB at max. value
or 0dB at min. value or 0dB alt min. frequency.
The first curve of an Y-axis group sets the Y-Offset value. The values are set for each Y-axis group individual.

5. Here can you define the function of a click with the left mouse button. You can set Left-Click to measure Levels, Frequency or Slope of the curves.

Left-Click: Y-Axis Level measurement



Left-Click: X-Axis Frequency measuremet


audioTester V 3.0 licensed to Developer
File Edit Presets Measure Analyze Tools Correction Options Help
sine
freq./Hz
Sound off
Setup
958,0000
Stop
Setup
Average
2
1
0
-1
FS 0
50
100
150
200ms
remark
audioTester V3.0
0
-20
-40
-60
-80
-100
-120
-140
957,7Hz
dBFS
500
1k
2k
3kHz
clear curves
unknown
unknown
unknown
unknown
unknown
unknown
unknown
unknown
X min / Hz
350
X max / Hz
3000
Out: 48000Hz float | DMX 6Fire Waveplay 1/2 (DMX 6fire WDM Audio)_o#1
File

Left-Click: Slope measurement

6. Opens the dialog for level bar settings, (level meter, bar graph) see below

Special Diagram Dialog

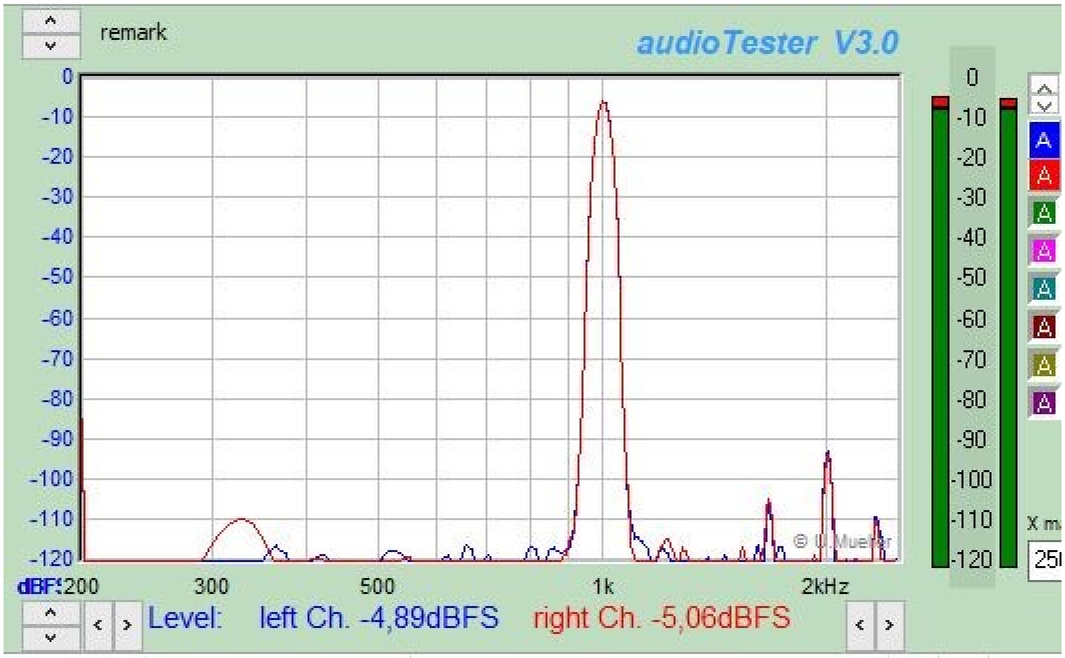
With this diagram you can influence the values in the diagram. The values have no influence to the numeric measurement values!
With the *x-data-factor* you influence the x-axis.
With the *x-axis unit* you can change the standard unit *Hz* to another one.
Example: change the unit 'Hz' to revolutions per minute, *x-data factor* is than 60 and *x-axis unit* 'rpm''

With the *y-data factor* you can multiply all (!) y-values with a factor.
With the *y-data offset* you can add all (!) y-axis with a offset.

6. Level Meter Dialog

1. Switch on/off (Level Meter, Levelbar)

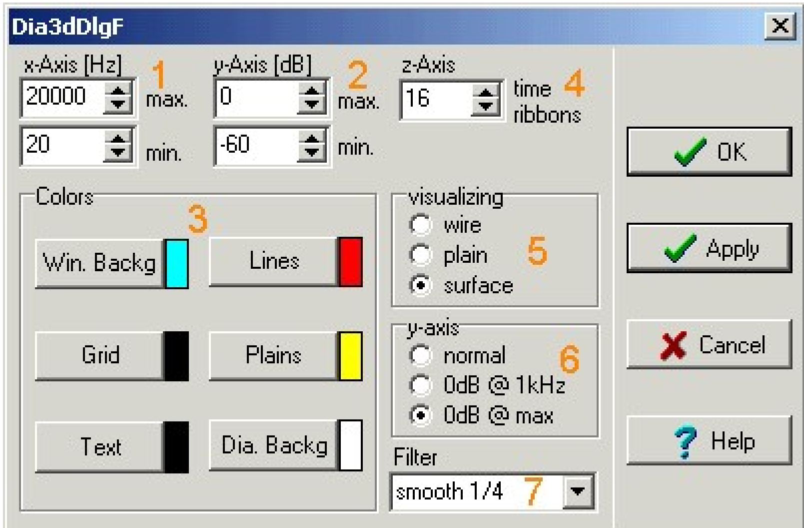
   The levelbar is on the right side of the diagram (see below)
   The level values will be taken from the left side of the diagram (here 0dB -120dB)



2. The level for the clipping color is set here -10dB, and red display)
3. The colors of the sound levels are selected with the buttons

6.2 3D Diagram Dialog

3D-Diagram Dialog

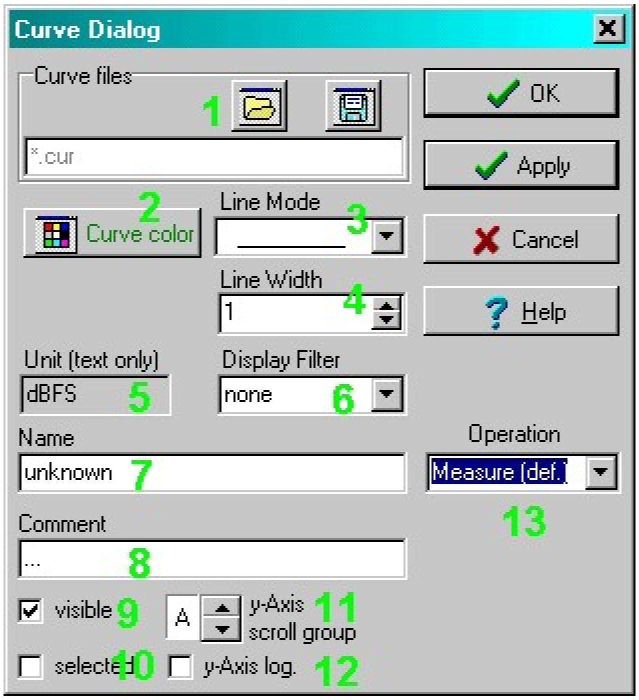


1. Limits of the x-axis - frequency in Hz
2. Limits of the y-axis - level in dB
3. Selection of the diagram colors, window background, the lines, the grid, the plains, the text and the diagram background
4. Count of the number of time ribbons
5. Visualization of the time ribbons as grid model, plains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

6.3 Curve Dialog

Curve Dialog

On the right side of the mainform diagram are buttons for up to 8 namend curves (or waveforms).
Right-Click on these buttons to show the Curve Dialog.
Her 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.

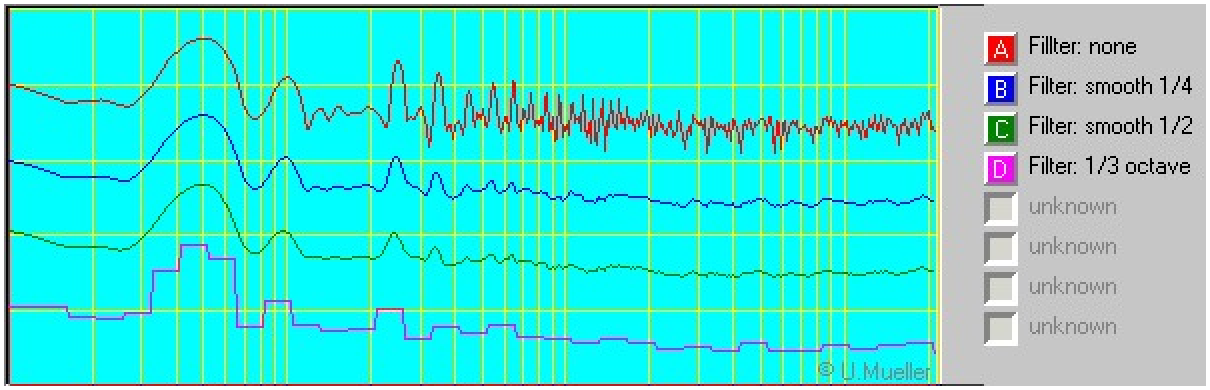
2. Dialog for the curve color.

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 document generation only.

6. In the drop box *Display Filter* you select the curve filter. You can select between none , a smooth ¼ octave , a smooth ½ octave and a terce (1/3 octave) filter. The filters are only for the purpose of displaying the curve. The filters have no influence to the measured numerical values.



7. The text field *Name* applies a name to 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. The comment is visible only in this dialog and it is stored in a *.cur file.

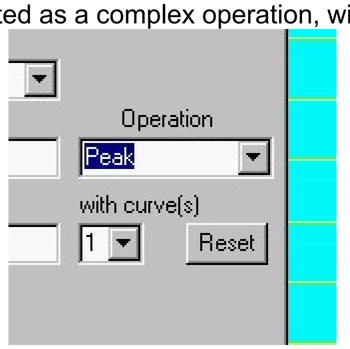
9. A curve can be visible or invisible. An invisible curve is also deselected.

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 a max. 2 curves to receive measurement values. If there are more than 2 curves selected, only the upper two curves will receiveing values. All the selected curves in the frequency diagram are stored.

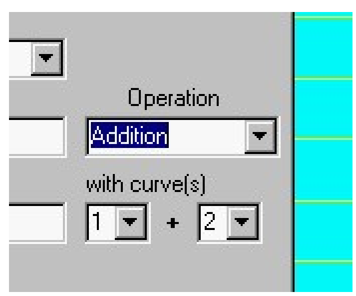
11. A curve belongs to one of 8 Y-axis groups (A-I). Each group has its own Y-axis, which is shown at the left margin of the diagram. The color of the scale is the color of the first curve of this group.

12. Set the y-axis to a logarithmic scale, to measure voltages or resistances in a wide range

13. Curves with Operation set to "Measure" (default) will receive measured values. It is also possible for a curve to show the peak value of another curve or the sum or difference of two other curves. It is calculated as a complex operation, with the phase value. See picture below:

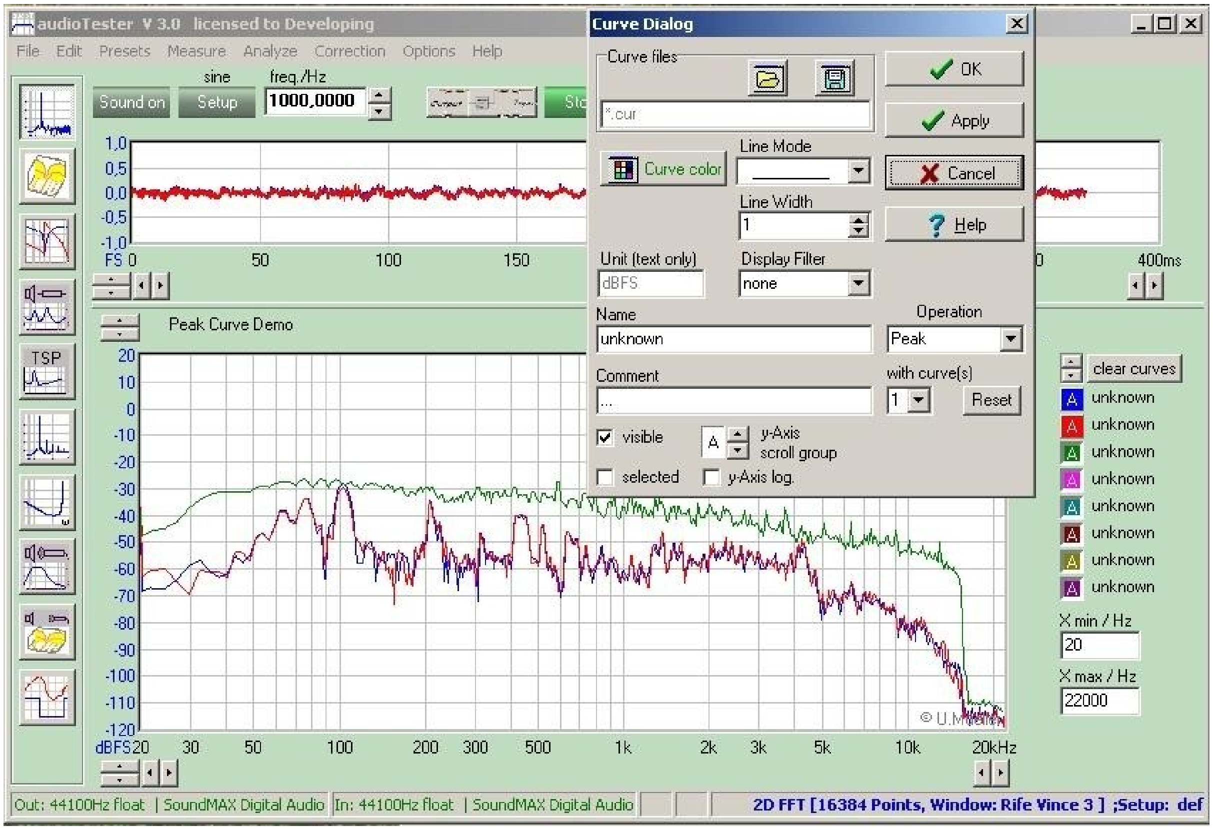


curve x shows the peak level of curve 1



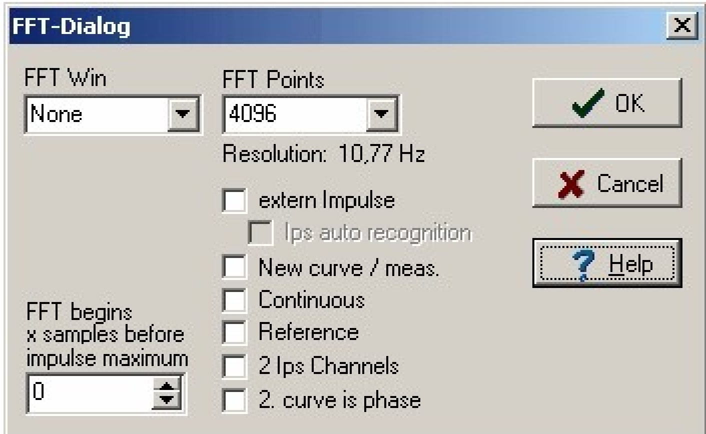
curve x shows the sum of curve 1 and 2

Operation Peak: The green colored curve is the peak level of the blue one (channel 1)



6.4 FFT Dialog

Setup FFT



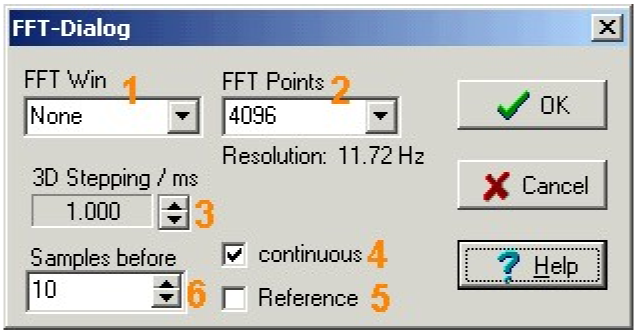
FFT window	Selection of the FFT-Window (None, Hamming, Blackman, Rife Vince ...)
FFT Points	Count of the FFT-Points (64 - 256k)
FFT begins	Samples before, for impulse measurement with Dirac-Impulse. No function here at the 2D spectrum analyzer .

<u>**Checkboxes:**</u>

extern Ips	Impulse is not produced by the ***audioTester***, but the impulse comes from a CD or other sources. How to make an impulse CD see here at Soundfile.dll
Ips auto reco...	Recognized at external Ips automatically the MLS size or a Dirac Impulse
New Curve ...	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.
Continuous	Continuous measurement
Reference.	Reference Measurement
2Ips Channels	Two Channels selected, only for impulse measurement. No function here at the 2D spectrum analyzer .
2 curve is phase	The second curve is the FFT-Phase
2 curve is group delay	The second curve is the group delay

6.5 FFT Dialog 3D Measurement

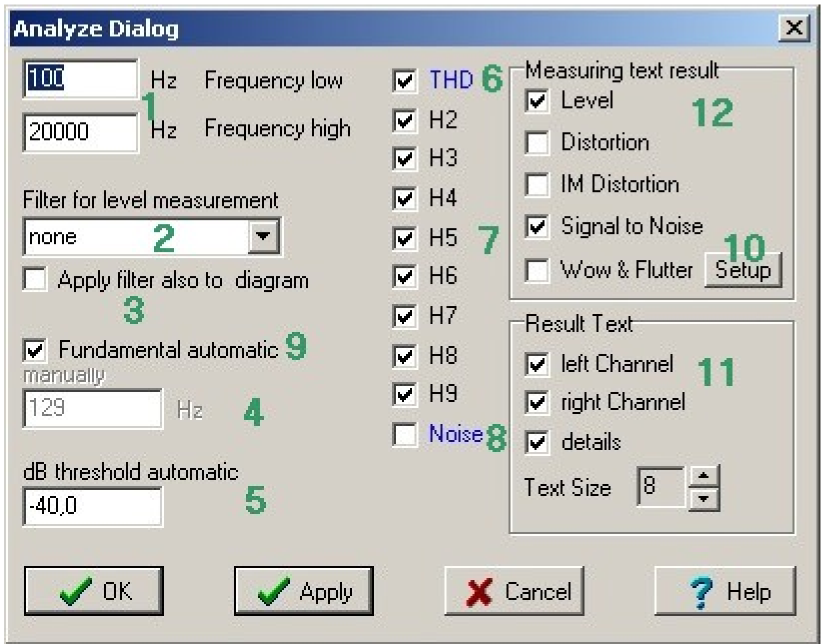
3D FFT Dialog



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 beginning of the FFT

6.6 Analyse Dialog

Analyse Dialog



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

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

2. Filter
You can apply the 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 button opens the wow & flutter dialog

11. Result text: Here you can format the text of the measurement results.
With the scroller you can enlarge the font size of the result string.

12. Measurement selection: You are able to select more than one measurement result.
The text results are listed below the diagram.

Measurement method THD+N
The fundamental wave is removed from the frequency spectrum, then the effective voltage value over harmonics H2-H9 and the noise between the frequencies 'low' and 'high' are summed. Then this value is divided by the total effective voltage value (that means, with the fundamental wave) now the result is THD+N.

Measurement method Inter modulations Distortions
Please select in the sound dialog **Dual Sine** and enter at **freq/Hz** the **main frequency** (eg. 7kHz) and a **Second frequency** the interferency frequency
(eg. 60Hz). The IEC 268 Part 3 says that the interference frequency should be 12db louder than the main frequency.
Eg. : dig Level = -15dB second dig. Level = -3dB

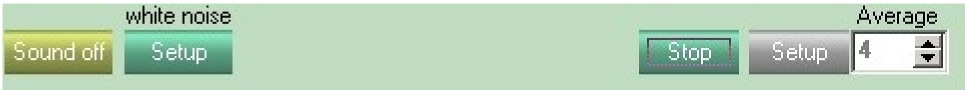
6.7 Time Domain Filter

With the function *Time Domain Filter* you are able to combine several filters for data in- and output.
It is possible to apply filters for both channels separately.

Input: Time Domain Filters are applied to the sampled data immediately. (see here)
The dialog for the input time domain filter is reachable with the menu point Analyse/Time Domain Filter

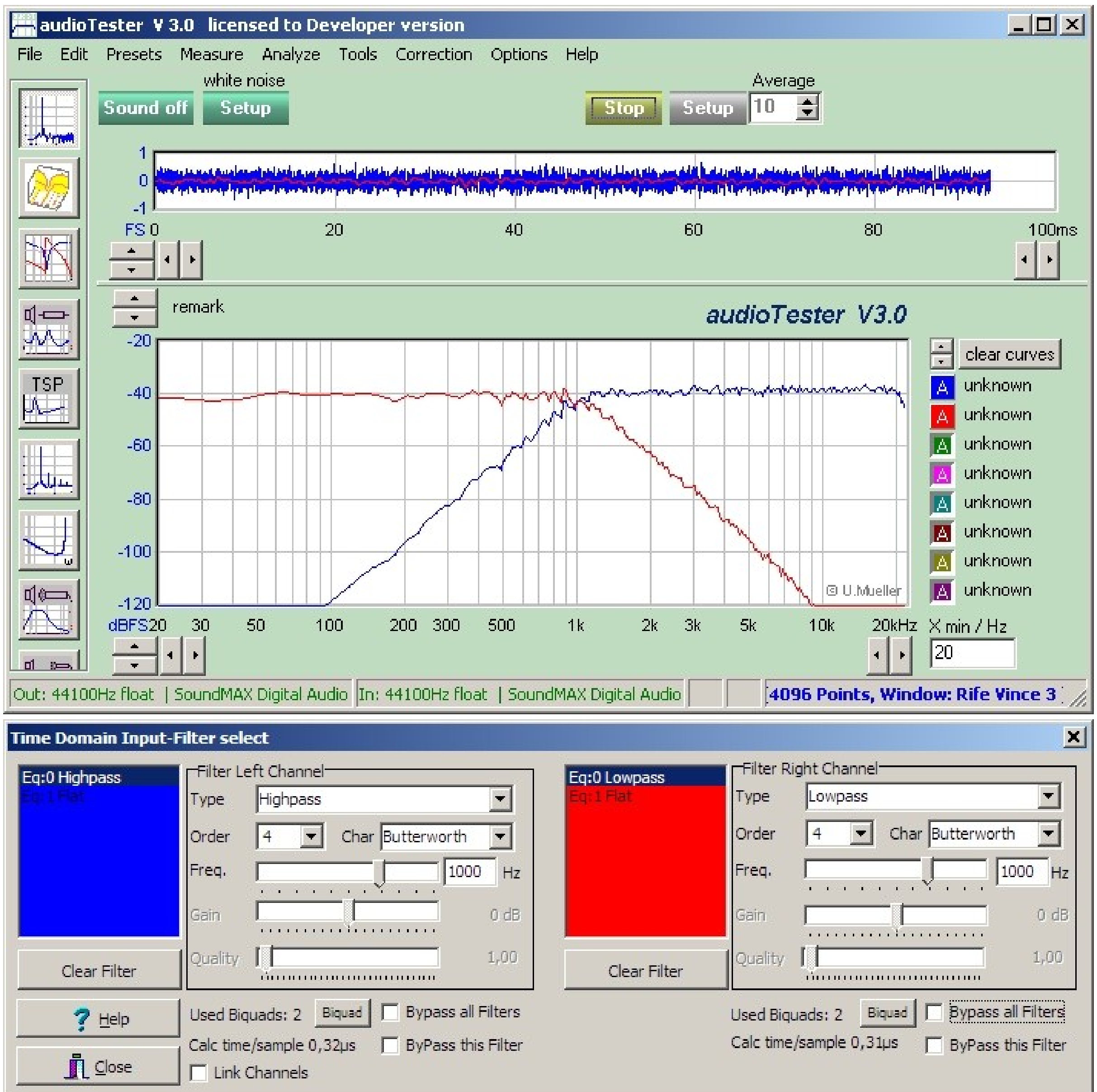
Output: The Time Domain Filter works just before the delivery to sound card driver.
The dialog for the output time domain filter is reachable with the *out filter* button in the Sound Setup dialog.

Activated filter you see in the color of the Sound Out/Sample In -Buttons.
Without any filters the buttons are green, if a filter is applied the button is yellow.
In the picture below the sound out has one or more filters, the sound in button has no filter.



The input filter works in 2D and 3D Measurement, in impulse measurement and in the Sweep Mode (sync and async).
The filters do not work in the oscilloscope, TSP Measurement, Impedance Measurement, Distortion-Sweep and Power THD.
The filters have influence to the diagram and the numeric measurement results !

The output filters works in all sound out modes.
The output filter option while playing sound files is only possible with the SoundDirect.DLL



The available filters are: Highpass, Lowpass, Shelving-High, Shelving-Low and Peak-Filter.
In High- and Lowpass filter you can adjust the frequency and the order (2- 8).

In High- and Lowpass filter you can adjust the characteristic (Butterworth, Bessel(-3dB), Tschebischeff)
In Shelving-High and Low you can adjust the frequency and the gain (+/-40dB)
In Peak-Filter you can adjust the frequency, the gain (+/-40dB) and the quality (0-32).

Handling:
From the list box on the left select a filter to modify it. To insert a new filter select the last filter, a Flat-Filter and modify it
The **frequency** is adjustable with the slider bar or the input field. In the input field you must end the input with the enter key.

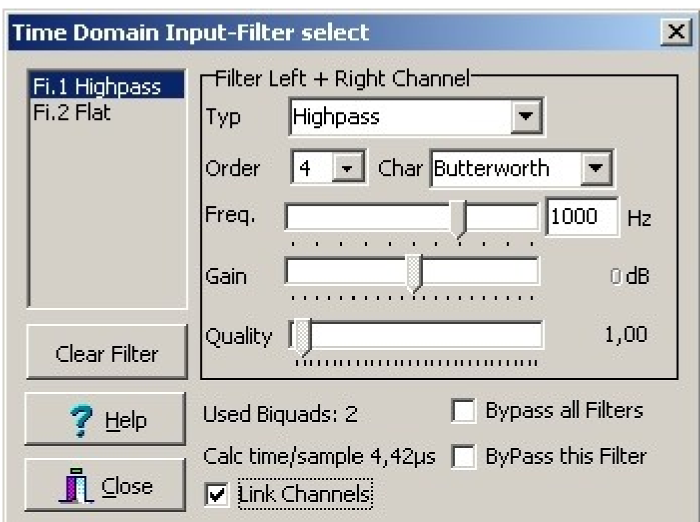
Below the filter modify group you can see the actual used biquads (max. 20)
and the time for calculation of all the filters. The time should be less than 4-5µs (1 sample at SF 48kHz is ~20µs)

All filter settings are stored and reload at program start, but not the bypass setting.

Bypass all filters is the default setting
To activate the filter deselect *Bypass all filter* (see figure above)

Changes in filters are applied immediately.

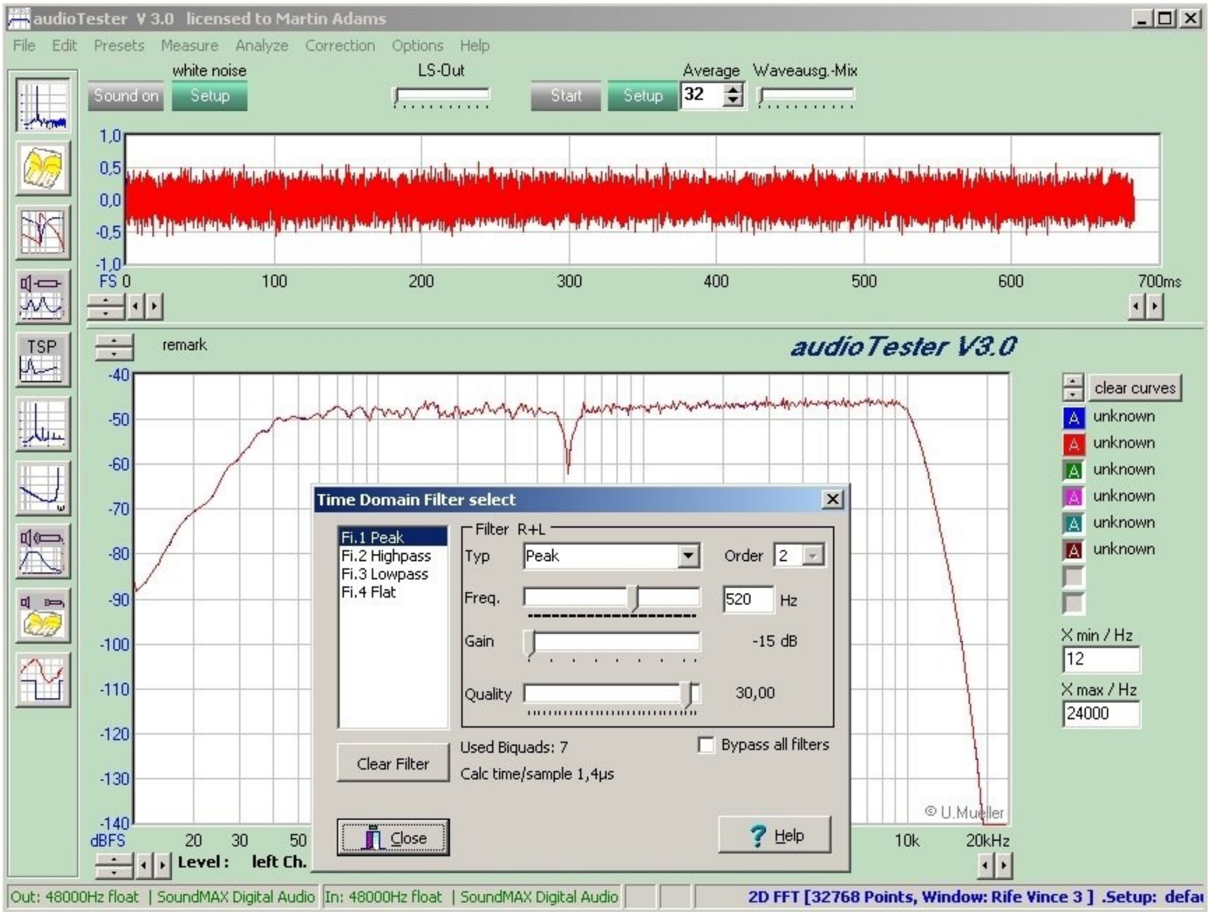
Button *Clear Filter* clears the selected filter from the list.



Link Channel, if activated (see above image) the same setups works on both channels (Stereo)

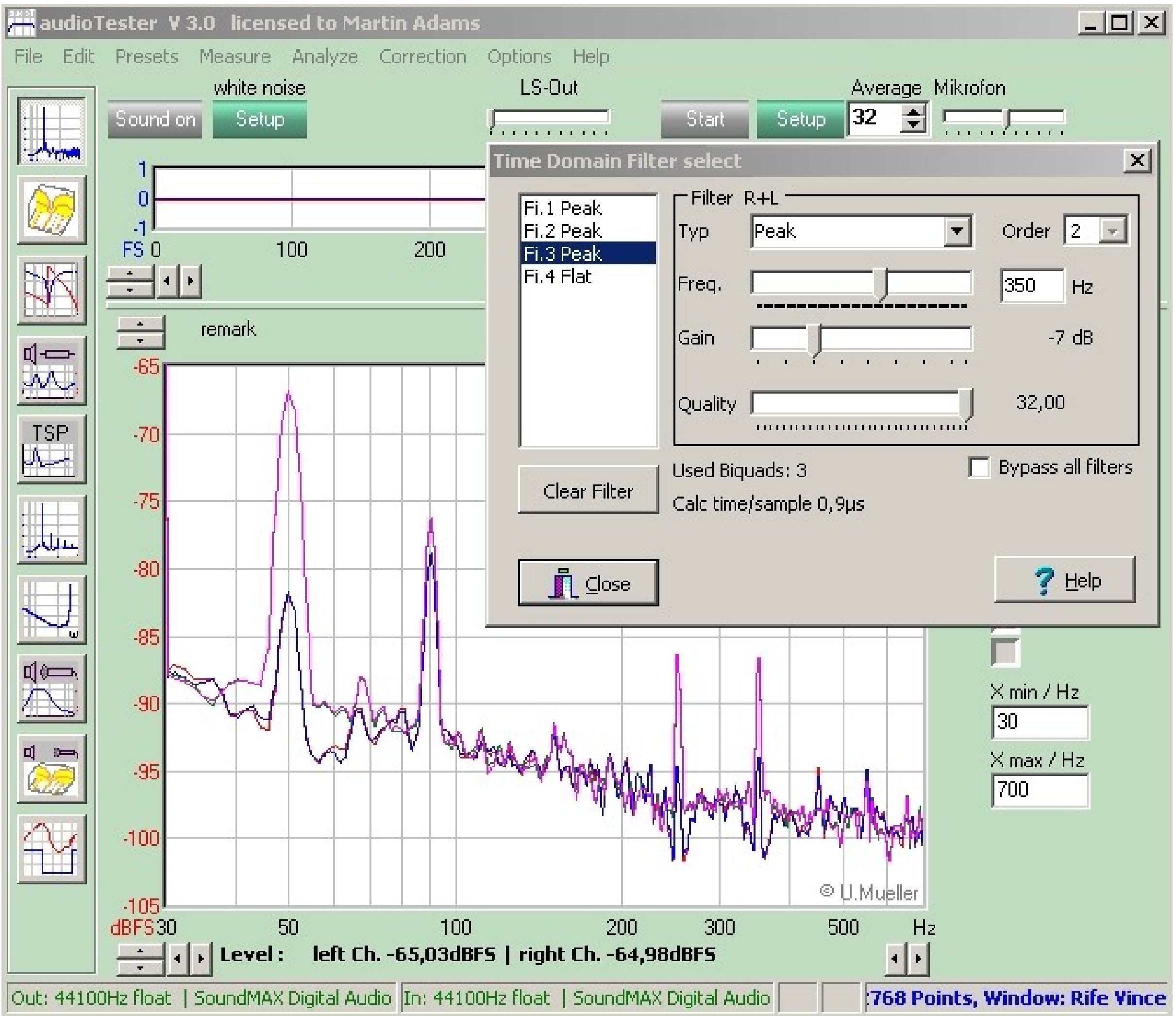
Example:
3 input filters:
1. Peak, 520Hz, Gain -15dB, Quality 30
2. Highpass, 40Hz, order 4
3. Lowpass, 10kHz, order 8

A white noise with a level of -6dB is applied and a Loop Back to the input with an averaging of 32 and a 32k point FFT
The sample time is ~1.4µs, 7 biquads are used.



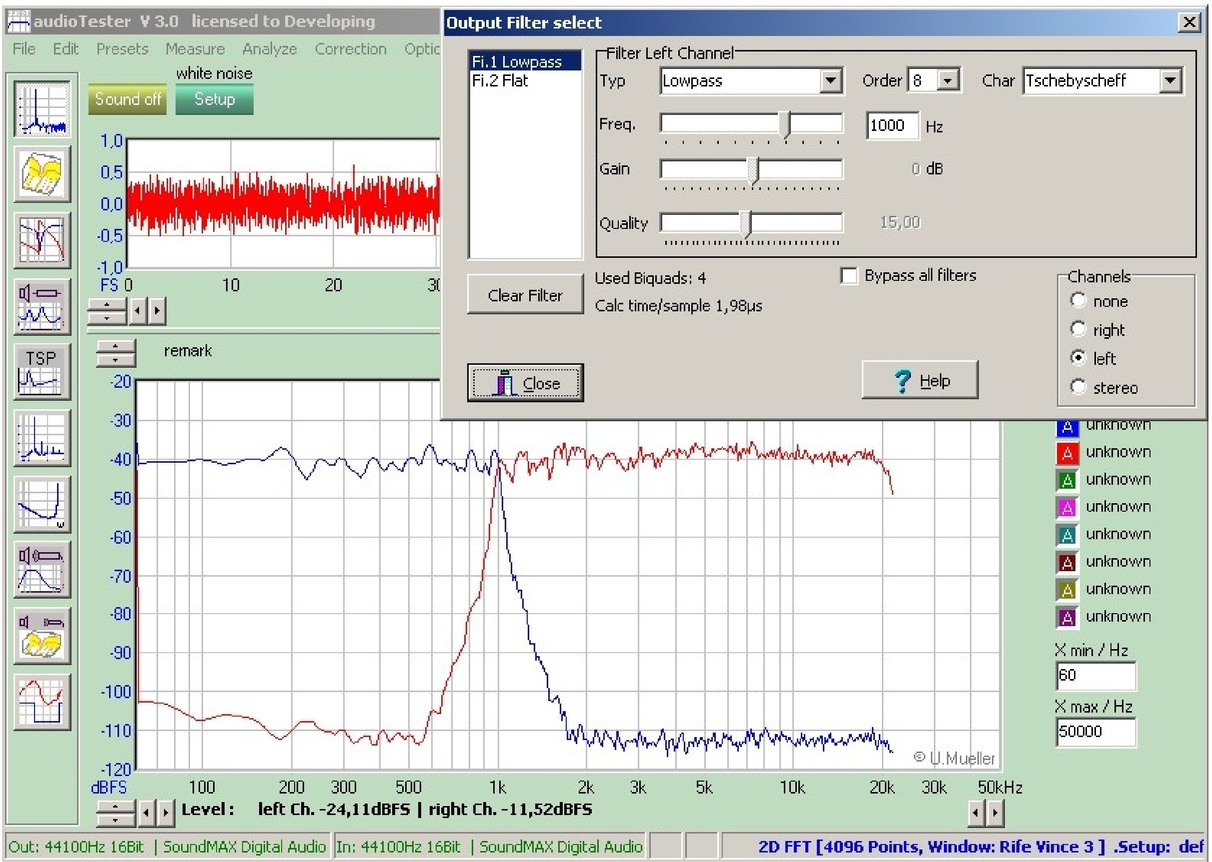
Example:
3 Peak Filters to eliminate 50/60Hz buzzing at input
1. Peak, 50Hz, gain -15dB, quality 30
2. Peak, 250Hz, gain -8dB, quality 32
3. Peak, 350Hz, gain -7dB, quality 32

Violet curve before, blue curve after.
The buzzing was simulated by an open microphone input



Example in impulse input mode:
Different Filter Characteristics
Butterworth
Bessel, Bessel -3dB
Tschebyscheff

Example Output Filter:
In the picture below there are different filters for the right and left channel applied.
Left channel: lowpass with Tschebischeff characteristic and 8th order at 1kHz
Right channel: highpass with Tschebischeff characteristic and 8th order at 1kHz
For example you are able to simulate a crossover network for loudspeakers.
Connect a woofer to the left channel and a tweeter to the channel and simulate with the filters a real network
and measure it with a microphone or hear it with an audio file (in this case you must play a mono sound file !).

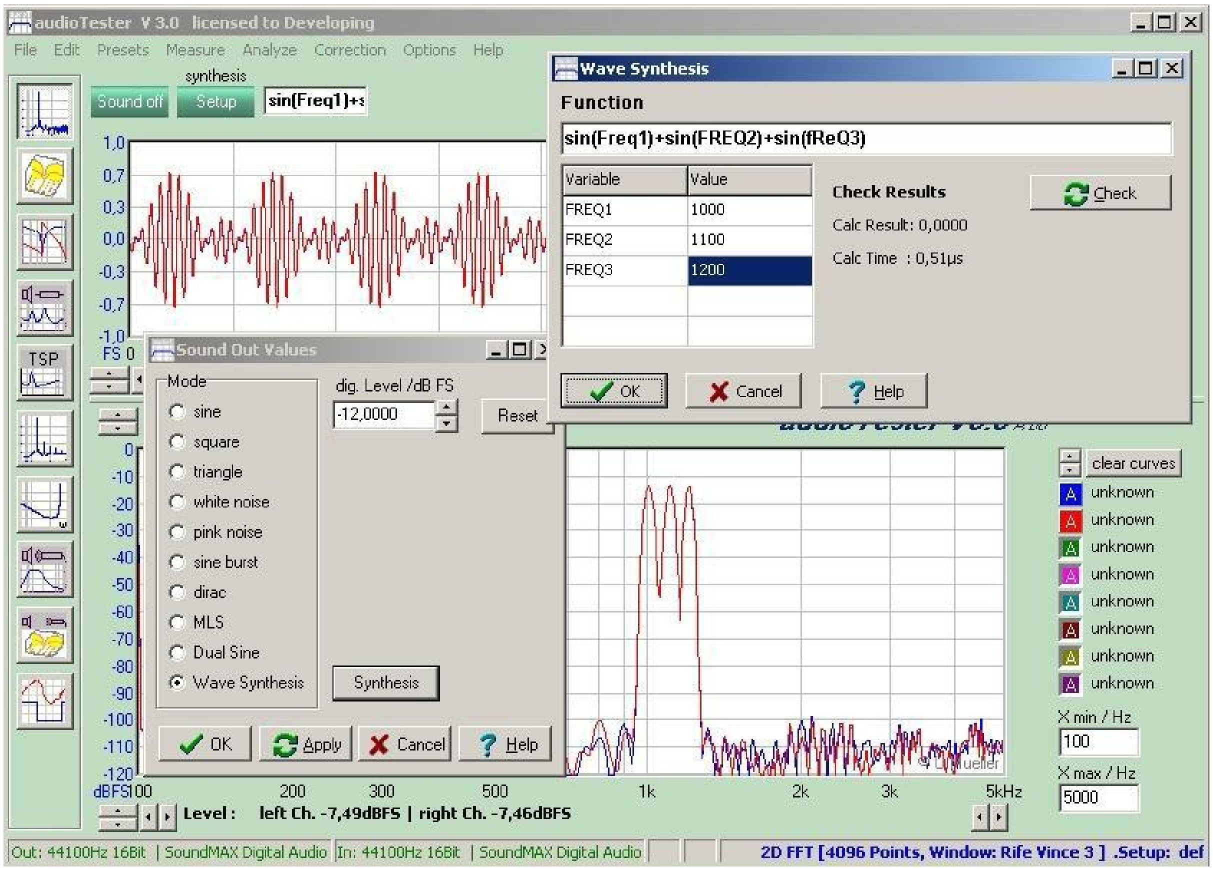


6.8 Wave synthesis Dialog

With the Wave Synthesis Dialog it is possible to create an output wave with a free synthesized function.
This dialog is opened from the Sound Out Setup button (upper-left) by Clickimg the Synthesis button
The dialog provides different math. functions and variables.

With the *Check Button* you can test your formula, and it will check the calculation time (*Calc Time*).
That is important, because the function will be calculated at every sample. E.g. at a sample frequency of 48000Hz every 20.8µs.
The calculation time of the formula must be less than the sample time. A good time is 1/3 of the sample time e.g.~7µs at 48kHz SF.
Formula errors are shown in an error check box.

The formula input text is not case sensitive (see below)

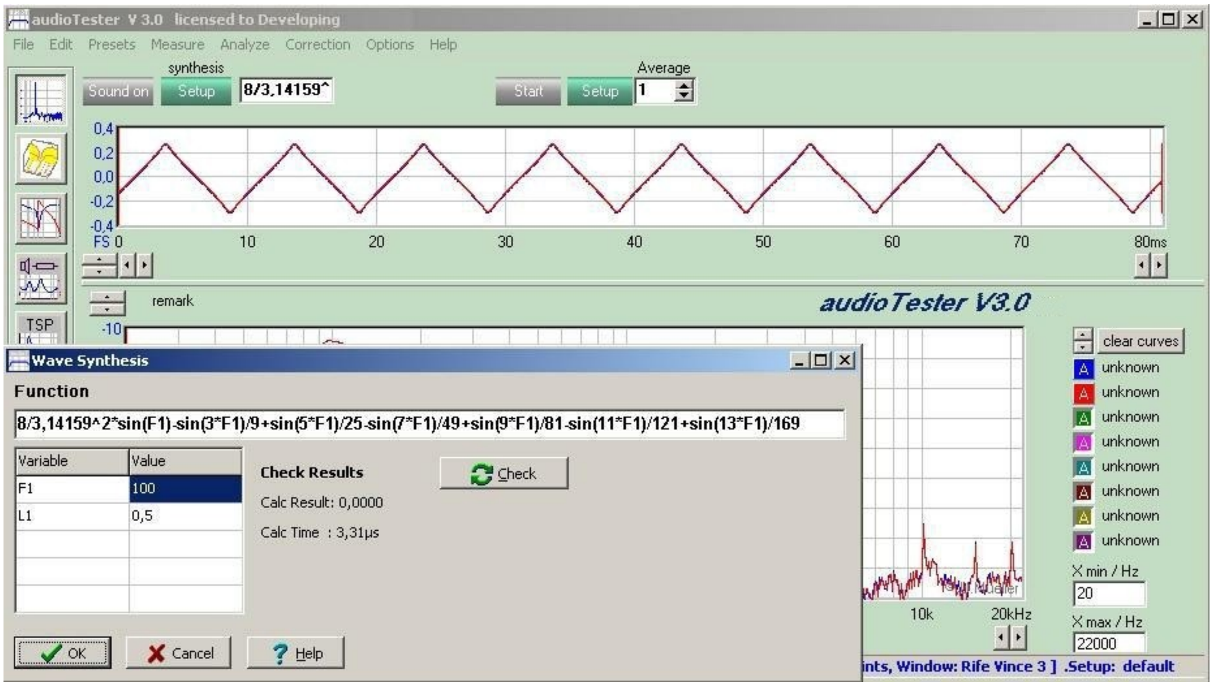


All calculated samples will multiplied finally with the digital level input field.

A maximum of 5 symbolic names and constants can be used.
A special set of variables is which begin with the character '**F**' like **f**requency (F2 or F or Freq...) are provided. These variables will be modified at every sample time according to
the formula [2 * PI * F / SF] where F is one of the special variables beginning with "F".

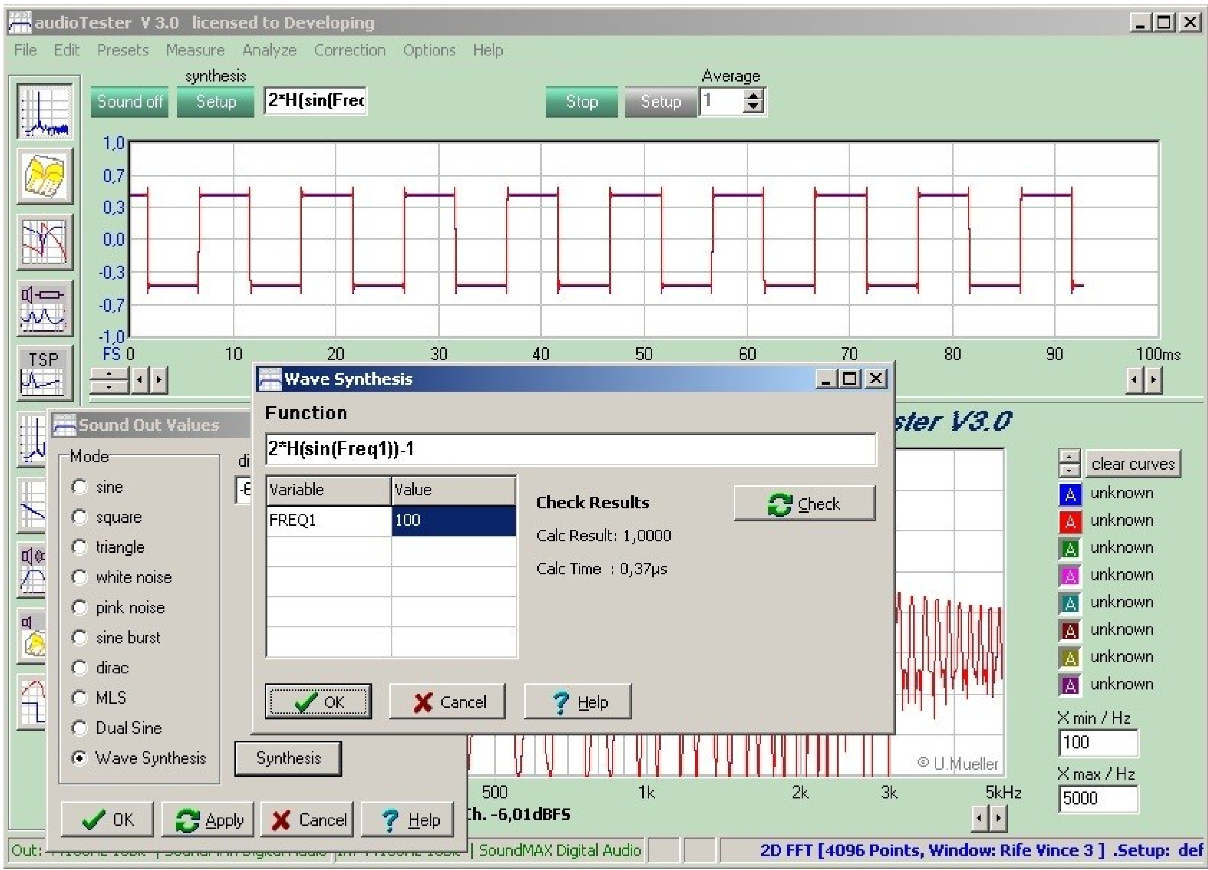
The following functions are available for use:
sin, cos, tan, cotan,
abs, ln, log, sgn, sqrt, exp,
arcsin, arccos, arctan, argcotan,
sinh, cosh, tanh, coth,
^, for power e.g. 3^2=9,
h, for heaviside: This function is 0 for every negative value and 1 for all others.
Also we have the operators + - * /

Please observe the examples below:



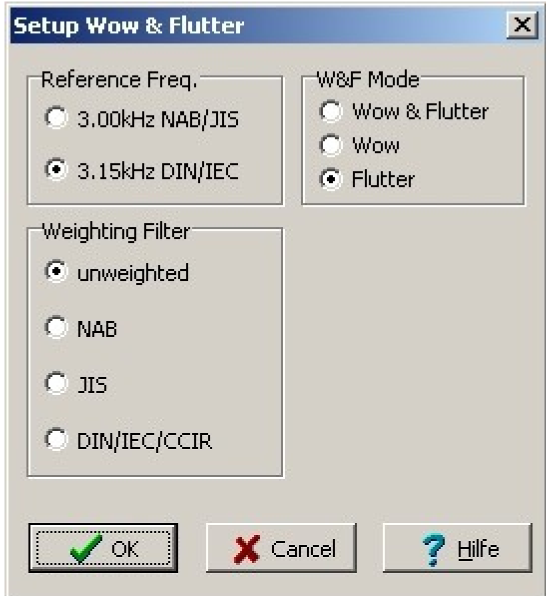
A triangle wave (the variable L1 is not used)

A square wave, with the Heaviside function, it runs faster.



6.9 Wow + Flutter Dialog

There is a special dialog for the Wow & Flutter Measurement.
This dialog is opened from Analyse Dialog with Wow & Flutter Setup button.
The Setup is only selected available, if no measurement is running.
You must stop a runnuing measurement before setup Wow & Flutter.



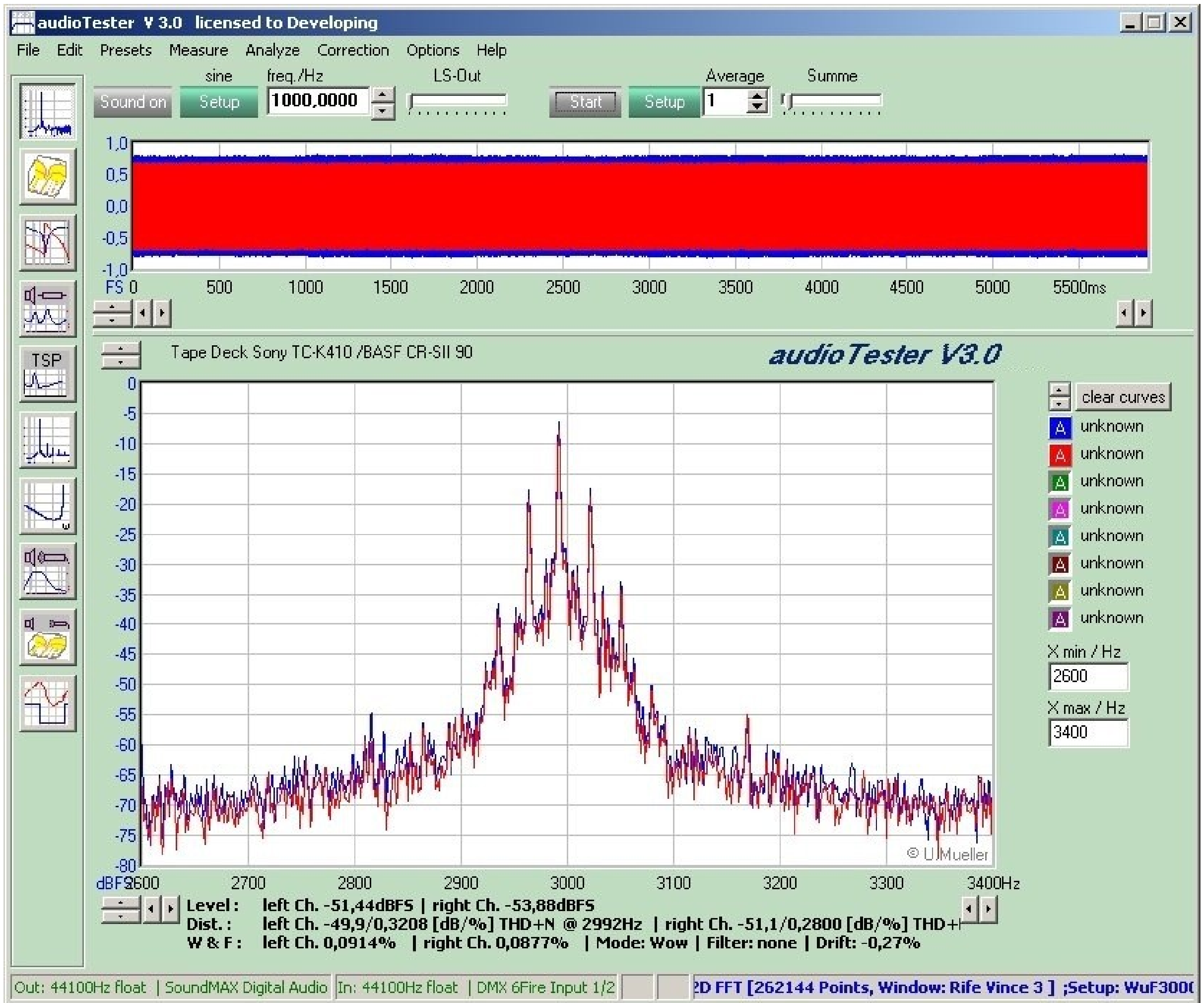
In the group *Reference Freq.* you choose the fundamental frequency of the test tape (test cassette).
3kHz is normal for measurement of the standards NAB and JIS
3.15kHz for DIN45507, IEC 386 and CCIR 409-2 standards.

In the group *Weighting Filter* you choose the filter to use for measurement
- unweighted - no filter
- NAB - measurement standard NAB Rec.
- JIS - measurement standard Japan Industry Standard
- DIN/IEC/CCIR - measurement standard DIN45507, IEC 386 und CCIR 409-2

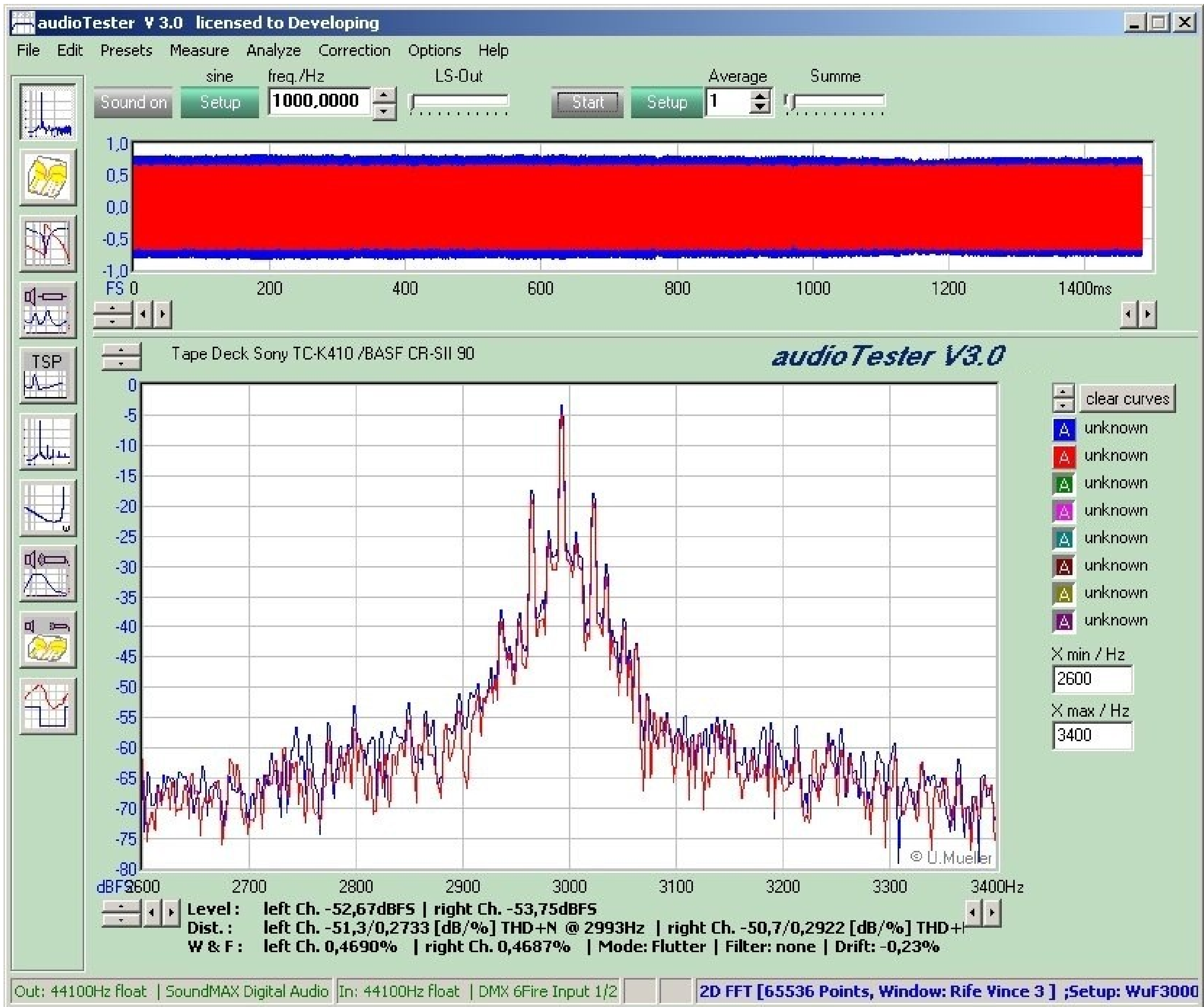
In the group *W&F Mode* you choose the mode for Wow or Flutter, both Wow and Flutter.
- *Wow*: is measured over a lowpass of 10Hz 8th order for the frequencies below 10Hz.(FFT 256k automatically)
- *Flutter*: is measured over a highpass of 10Hz 8th order for the frequencies above 10Hz.(FFT 64k automatically)
- *W&F*: is measured without any special filter except the *Weighting Filter* (see above) (FFT 256k automatically)

For Wow & Flutter Measurement there are Presets available with usefull entries for the diagram and the FFT Size.
Minimum requirement is CPUs faster than 1.6GHz

Example:



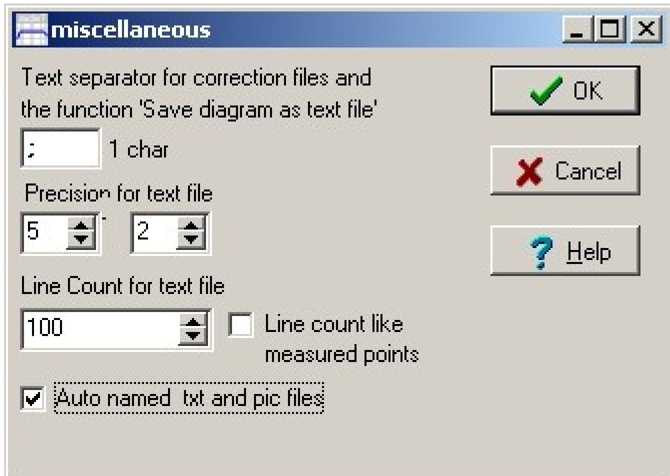
Wow Measurement old Tape Deck



Flutter Measurement old Tape Deck

6.10 Miscellaneous Dialog

Miscellaneous Dialog
In this dialog are several settings, mainly properties *saving as text file* .



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

The **line count** is the count of lines in the text file. If there are more or less measurement values than line count, then the frequency and the level are interpolated over the line count range. Only visible values in the diagram are written. That means, if you decrease the diagram bounds, only the actual visible values are written. If at the lower diagram bound and the upper diagram bound there are no measurement values (as is possible with sweep measurement) then the new boundaries are the reported upper and lower values.
All the selected curves in the frequency diagram are stored.

If you select **Line count like measured points** the text file is filled with the count of all the measured points (e.g. 2048 lines at a FFT with 4096 Points)
There is no interpolation necessary by the program. If at the lower and the upper diagram bounds are no measurement values (as is possible with sweep measurement) then the new boundaries are the valid upper and lower values.

Auto named txt and pic files:
If this checkbox is selected, the text and picture files are stored without the file dialog and with an automaticly generated file name.

Examples:
Menu point: *File/Save diagram to BMP/JPG* or ***F2*** as
FreqDiaAsPic_2010_06_21_14-27-23.jpg
Menu point: *File/Save freq. diagram as text file* or ***Strg+F2*** as
FreqDiaAsTxt_2010_06_21_14-27-16.txt
Menu point: *File/Save time diagram as text file* as
TimeDiaAsTxt_2010_06_21_14-27-19.txt

The path is the path which is first used with the normal file dialog.
The picture format (bmp/jpg) is the format which is first used with the normal file dialog.
Format and path are stored after closing the program.

7 Sound

7.1 Soundcard selection

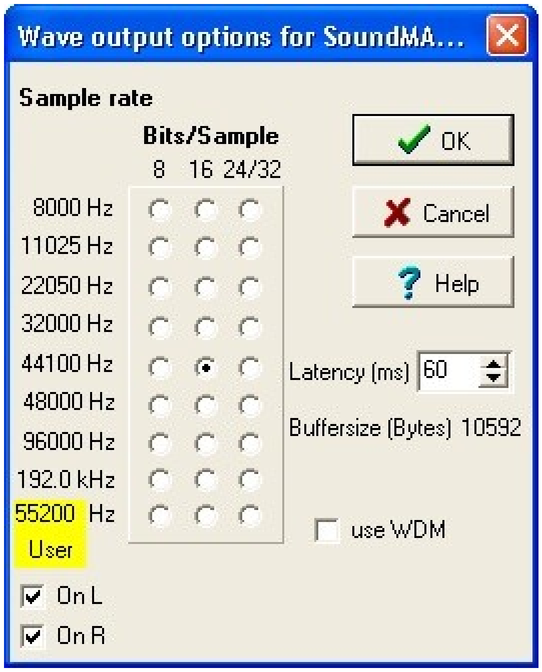
Selection of the sound card

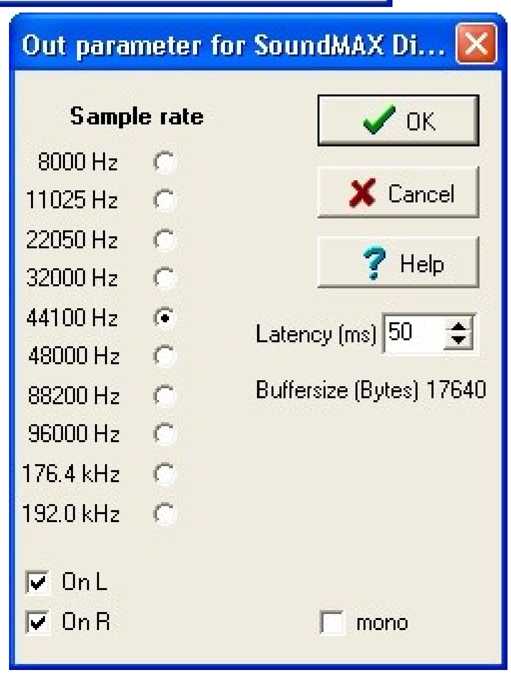
You can select a different sound card for sound in and sound out.
There is a maximum of 8 different sound cards.
It is recommended not remove or add a sound card while audioTester is running (USB Sound Cards !).

This dialog can also activate with a click on the status bar line of the program.

7.2 Sound Out Parameter

Changing of output sound parameter





Sound Out Parameters Dialog for Sound Wave DLL

Sound Out Parameter Dialog for the SoundDrirect DLL

In the **matrix** you can select the possible **sample rate/data width** pairs
The new SoundDirect DLL has only float values, therefor no other data widths.

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'** (only wave sound) 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.

With the checkboxes **OnL** and **OnR** the output of the two channels can be switched on and off separately.
In rare cases this may not lead to the expected success,
Then, in **Windows**, select / check the **Start** button, then **Settings**> **Ease of Access**> **Audio**, then turn off the toggle under **Mono Audio**

MONO-Audio activate

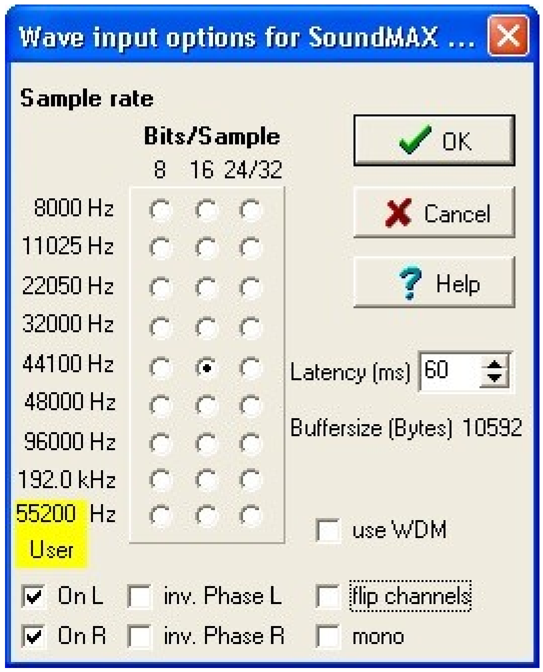
 **Off**

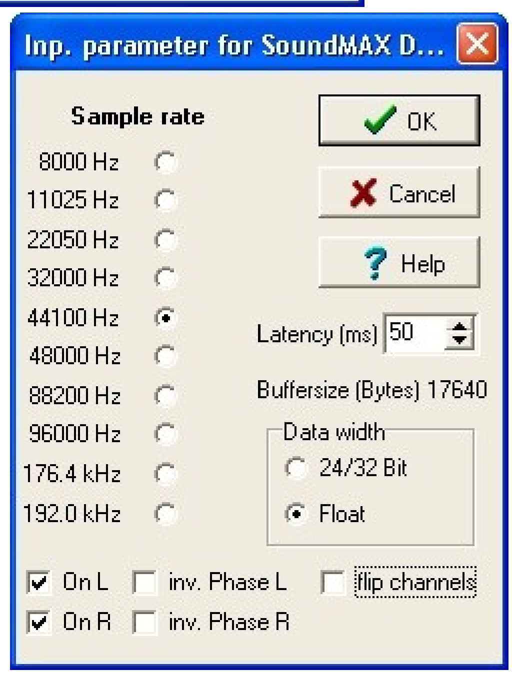
So it is correct!

This dialog can actived also with a click on the status bar line of the program.

7.3 Sound In Parameter

Changing of input sound parameter





Sound in Parameters for Sound Wave DLL

Sound In Parameters for Sound Direct DLL

In the **matrix** you can select the possible **sample rate/data width** pairs
The new SoundDirect DLL has only float values, therefor no other data widths.

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'** (only wave sound) 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

In the **Sound In Parameter Dialog for the Sound Direct DLL** you choose between 24/32Bit and Float data format, default is 24/32Bit.
There is no quality difference between these formats. It is only made for the compability for some sound card drivers.

This dialog can actived also with a click on the status bar line of the program.

7.4 Sound menu

Menu Options

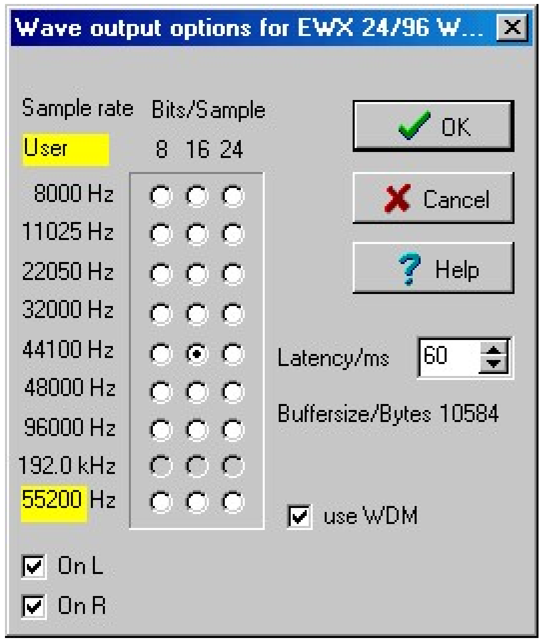
In the menu point *Options* you are able to reach the dialogs to adjust the sound card parameter, to switch on/off the mixer support and get the calibrations dialog.

Miscellaneous	Dialog for various setup values e.g. saving diagrams as text file
Alternative sound DLL	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
Link Audio.In/Out-Dialogs	Only one dialog for devices and parameters.
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

Selection of the sound card

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

Changing of output sound parameter

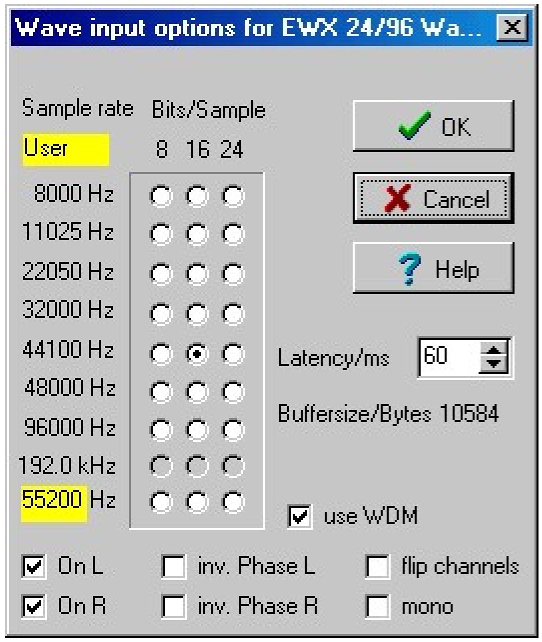


In the **matrix** you can select the possible **sample rate/data width** pairs

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 **sample rate/data width** pairs

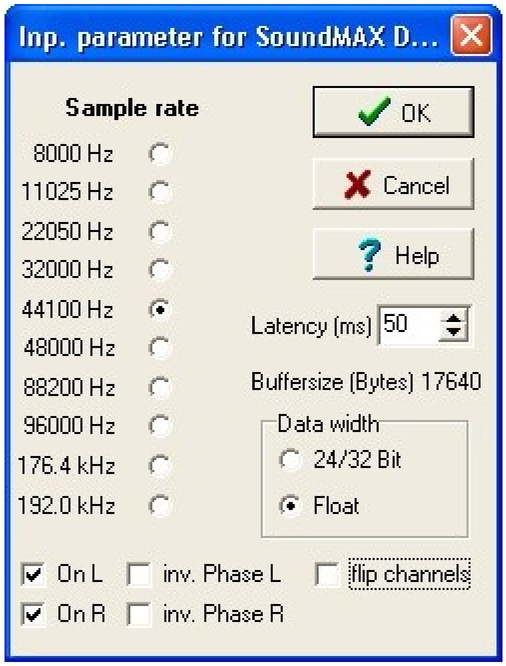
The **latency time** must be less then possible, but without any interruptions in the time domain diagram.
A good value is 60ms, this value is preselected. The value depends on the computer performance, the sound card and the quality of the card driver.

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 Sound DLL 'SoundDirect.DLL' is loaded by default, since Version 3.0. In the dialog for the

'SoundDirect.DLL' you can almost chanced the sample frequency. The data width is only float

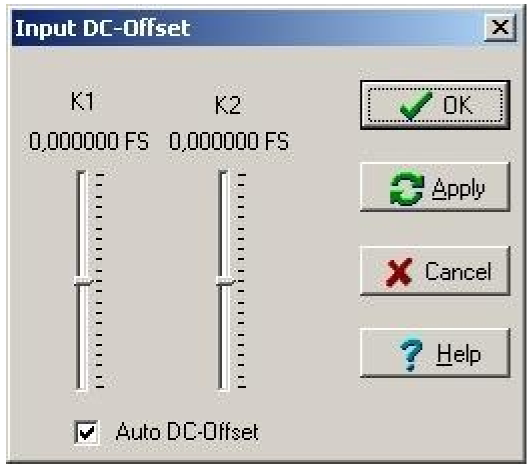


Mixer Support
Mixer Support On -> the controller on the main window are linked to the Windows Mixer. The controlled input channel is selectable at the menu point *Wave In Channel* . The controlled output channel is always the *Volume Control*, at which only the channel *Wave* is selected.
Mixer Support 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'

7.5 DC Offset

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..



DC Offset Auto ON

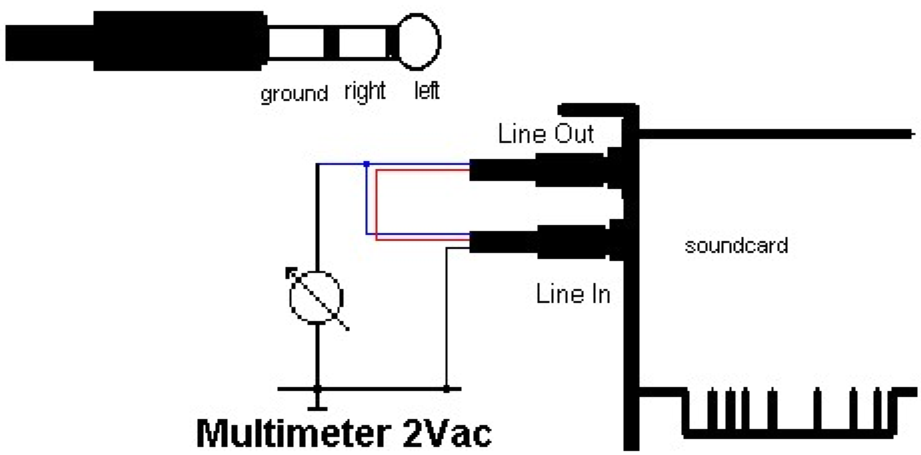
DC Offset Auto OFF

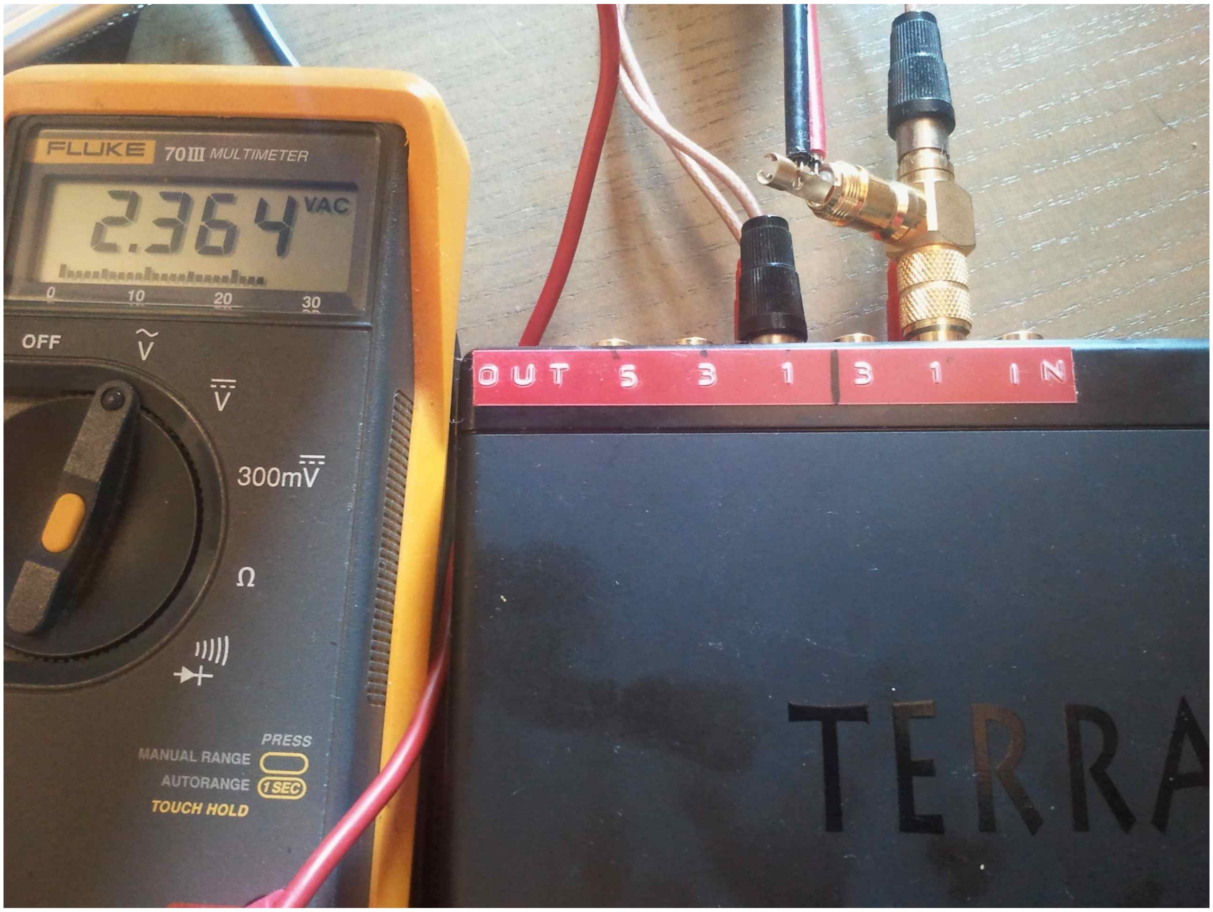
7.6 Calibration

Calibration

Before working with the ***audioTester***, the Line - Input should be calibrated. Specially for the Power THD/Level measurement the input must be calibrated and also if you use the absolute level units like dBV, dBm, dBu ... (*Menu: Analyze/Level Units/ dbV,dbu,dbm,dBSPL* ...).
To calibrate the Line-Input you need a multi meter with an AC Range of 2 Vac and a AC signal source... the sine wave generator of the ***audioTester***.
Please use a low frequency (like 50-60Hz) with the sine wave generator, because this is where the multi meter is most accurate.

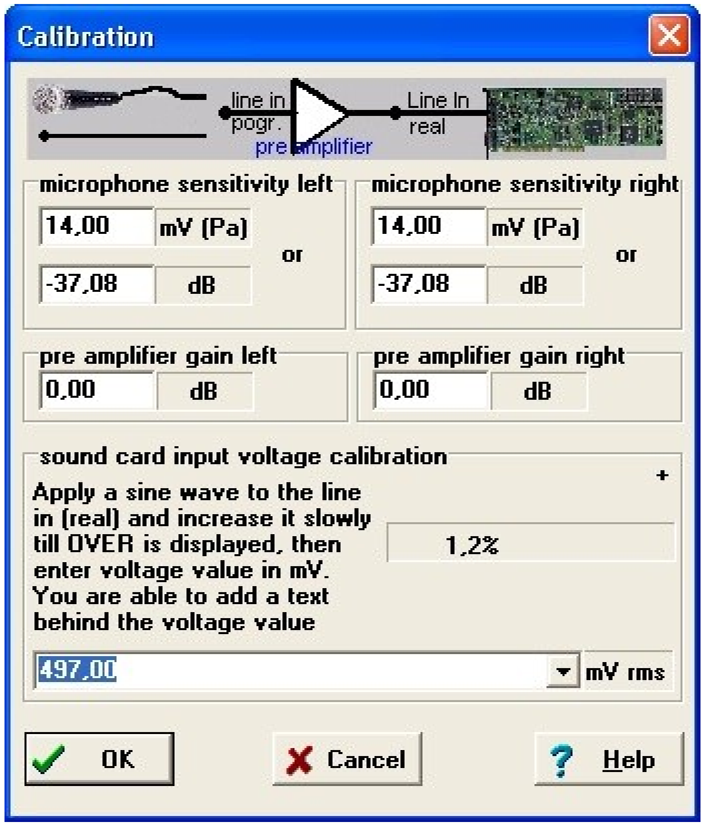
The signal has to be wired to the right and left channel of the Line-Input. After that, adjust the AC output for a voltage level of about 100mV . Then push the Line-Input Fader (Windows-Mixer) to the maximum . After the calibration this is the ONLY POINT that is actually calibrated. This is indicated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





- Now start the sound out with a sine wave of 50-60Hz and than open the Calibration dialog

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

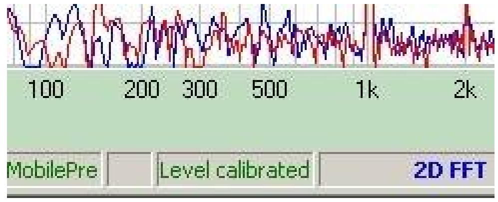


- A moment later a percent value is visible, but don't concern yourself with this number, it is used internally by the program as a calibration constant.
- Then increase the voltage with the windows mixer until you see the text **OVER**. OVER means, that the A /D Converter of the sound card is overdriven

(Beyond the maximum signal it can handle). The digits after OVER shows the number of overdriven samples.
When the display changes between the percent values and OVER, note this value on your multimeter.
Next enter the value you noticed in the input field (in mV). In most cases it is a value between 200-1400mV

Over is visible, if the level reached the last bit, at 16Bit resolution it is 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 stop to increase 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***. You can insert a remark text behind the voltage level to identify different sound card or resistor networks.



If you select an absolute voltage unit (i.e. dBV), its shown like above.

Hint:
You have to enter the rms value, which is shown by the voltmeter. Do not measure the voltage with a oscilloscope which will be peak to peak.

If You will use a measurement for measure absolute sound levels, You are able to input the microphone sensitivity (**2**) in the unit mV/Pa or dB.
You take the value from your microphone data sheet.
At (**3**) you must input the gain of the microphone preamplifier. 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 $2x10^{-5}$ N/m²= 20µPa (Pascal) - 20Pa corresponds to 120dBSPL that very loud.

Internally calculates the ***audioTester*** as follows:
PreAmpGain (dB) + 20xlog(INP (mV) / MicSens (mV/Pa) / 2x10^-5 (Pa))

7.7 Correction Files

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. You have to distinguish between the Input and the Output correction. There is 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).
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 of your device. 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.

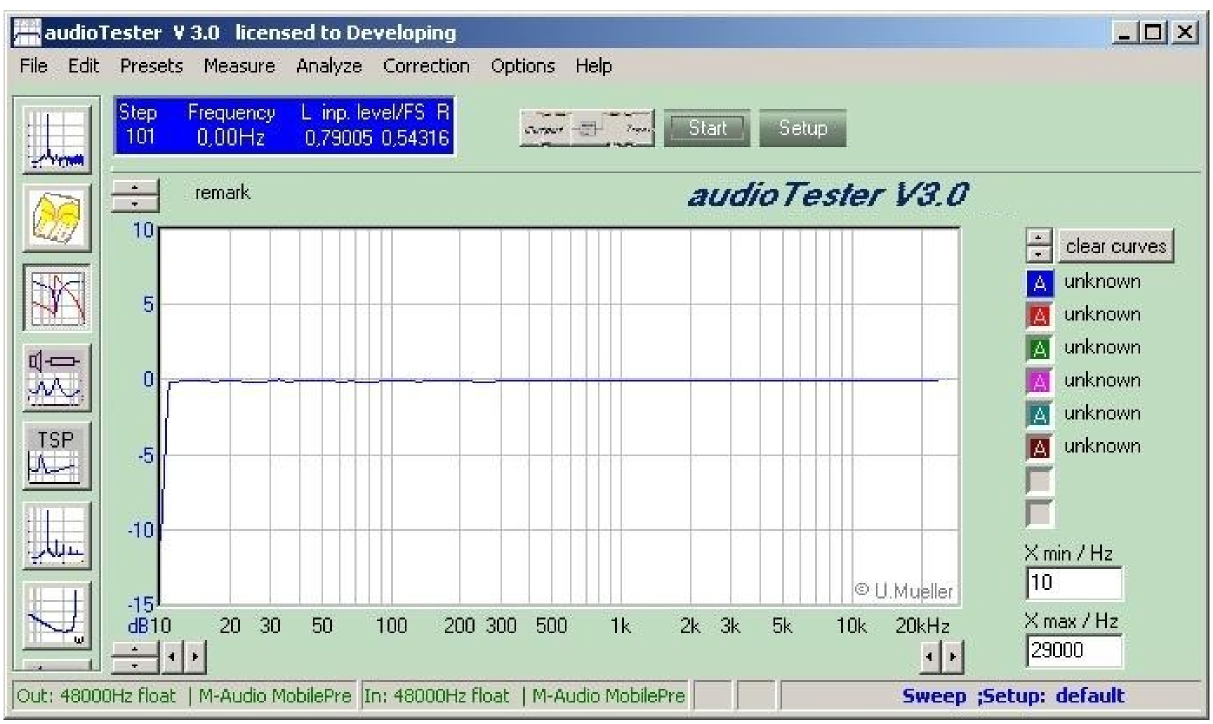
How to load a correction data:
You must distinguish between the loading of an Input or an Output correction data file. With the corresponding menu you have the option to load a correction file.
The extension *.cor is already automatically added. After loading a correction file, it is immediately active, 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 colour of the left channel; the Output correction file is presented in the colour of the right channel. If you do a

second editing or loading you have the option to do changes in a very easy way.

Not corrected sweep of a sound card



Now the its corrected

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.

7.8 Soundxxxx.DLL

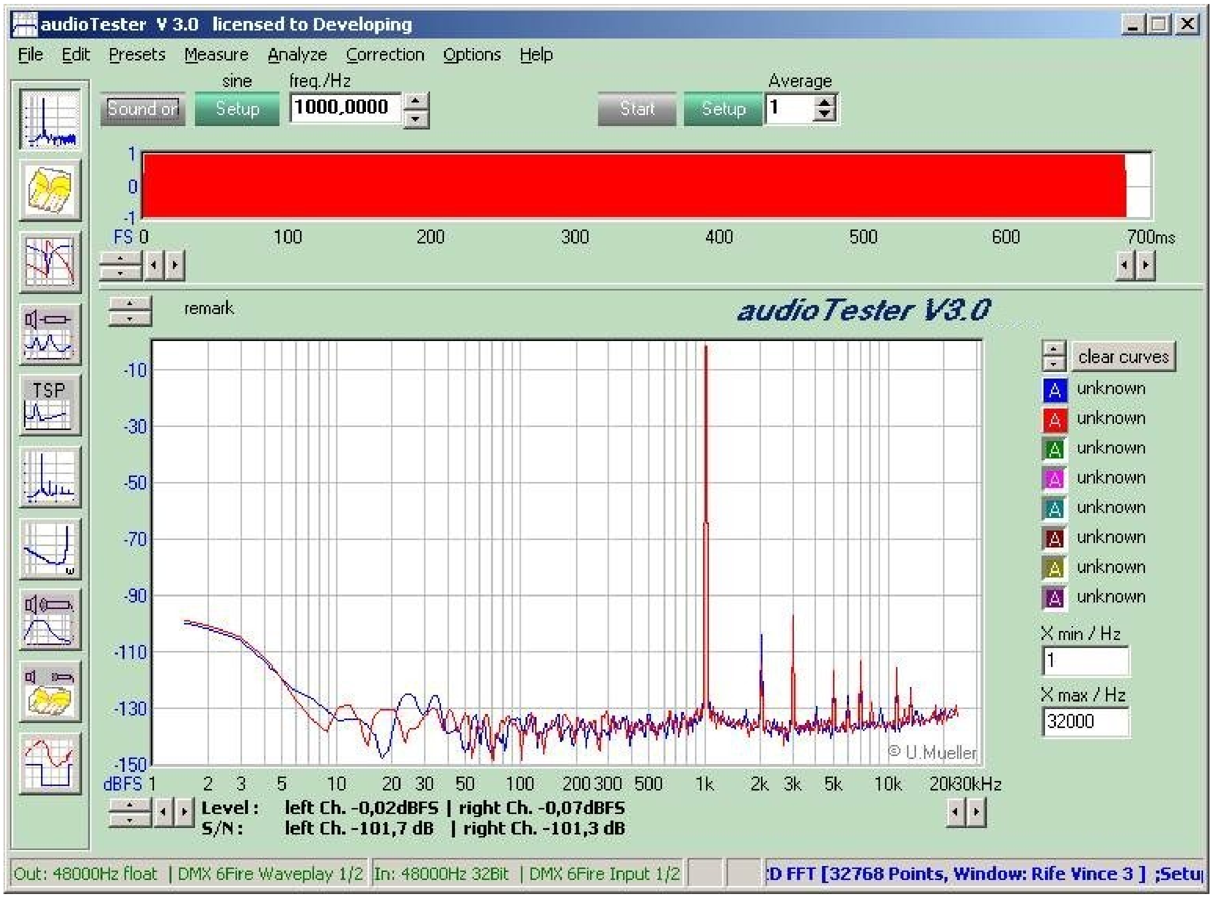
SoundDirect access with DirectX	default setting (used the bass.dll by Un4Seen Developments) the samples comes as float values, if this doesn't work with your sound card drivers it is possible to switch to 32Bit integer.
SoundWave access with mmsystem	for older Windows versions and for system with bad sound DirectX driver
SoundAsio access with ASIO driver	for short latency time. (used the openasio.dll http://www.martinfay.com)

USB-Sound Cards
USB sound cards are especially ideal for use with notebooks, because the build in hardware is seldom good for measurement.
It is nice if USB sound cards are working without an external power supply (CAR-Hifi).
There are basic USB-sound cards with USB 1.1 support and max 48kHz sample frequency, often like USB-Stick. They are often not better than the inbuilt hardware.
And than we have the complex systems with USB 2.0, with this cards it is no problem to transfer sound data's at 192kHz and 24Bit.
It is often so, that the sample frequency with the audioTester is not transfers to the sound card, then you must additional use the mixer application of this card
Sometimes it works without any further actions, but attention in some cases there is a sample rate conversion in the card driver, this is not good for best measuring results.

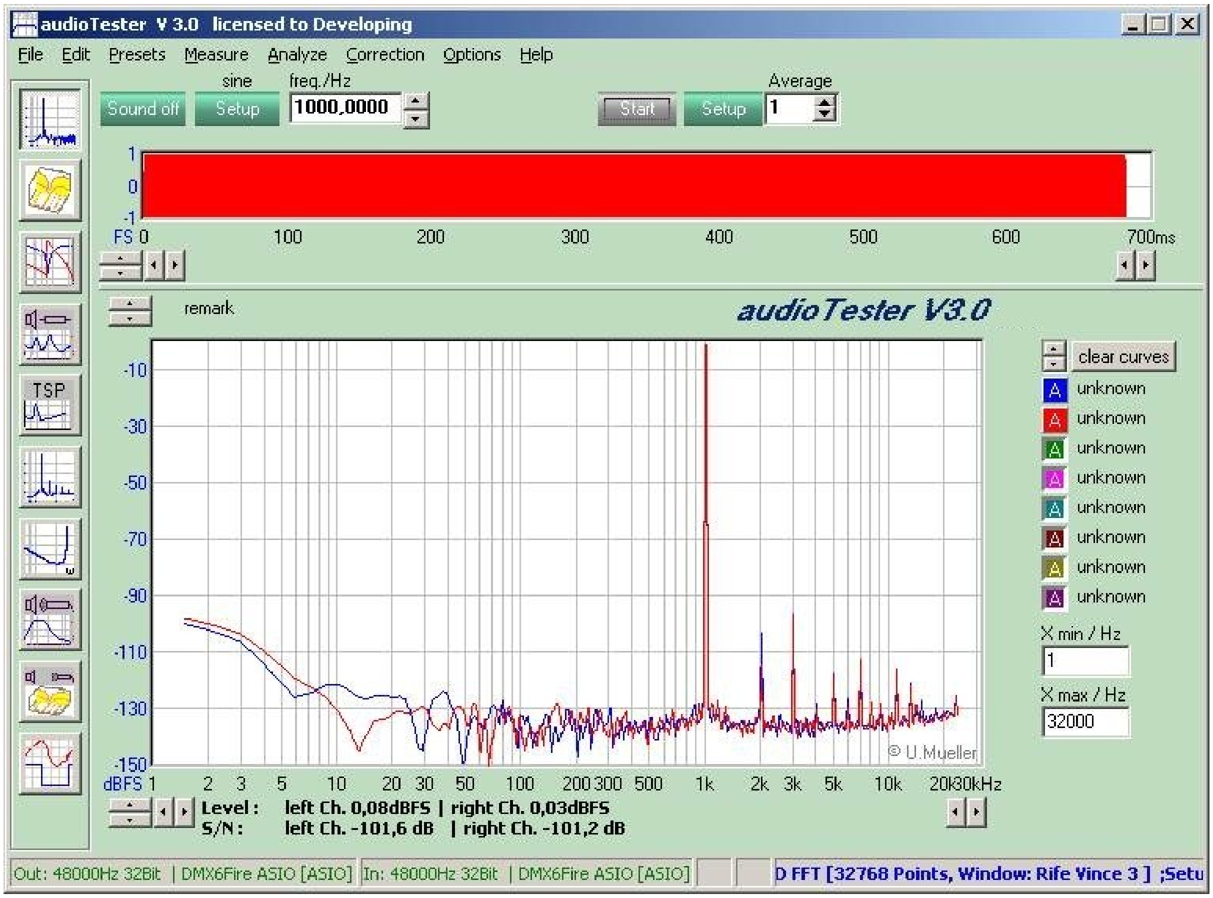
If you see at start sampling the error dialog: 'Sound direct unknown ' then please change the data width to float or 24/32Bit in the 'Options/Sound-Para-In' Dialog.
Some sound cards show only a flat line at -192dB, but no error dialog, do the same.

Please don't remove or plug in a USB sound card while measuring.

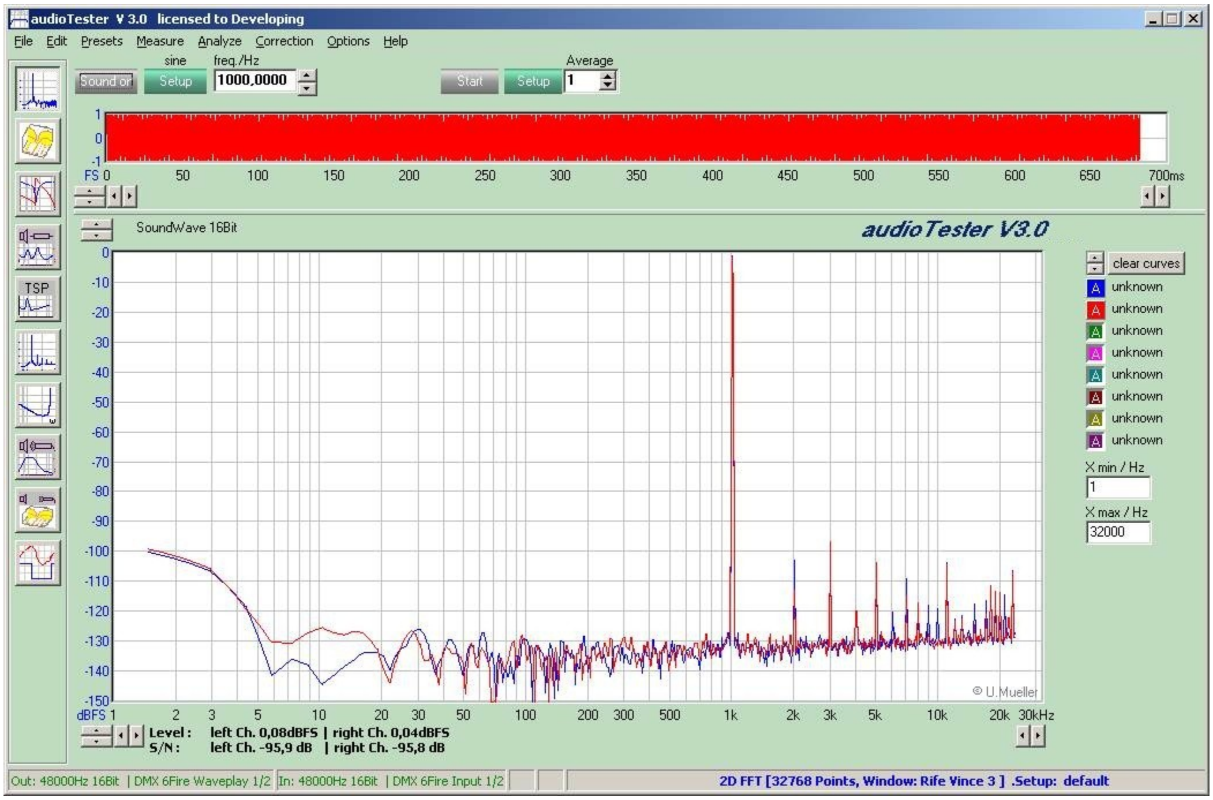
Examples with a Terratec DMX 6fire (Output directly to Input)



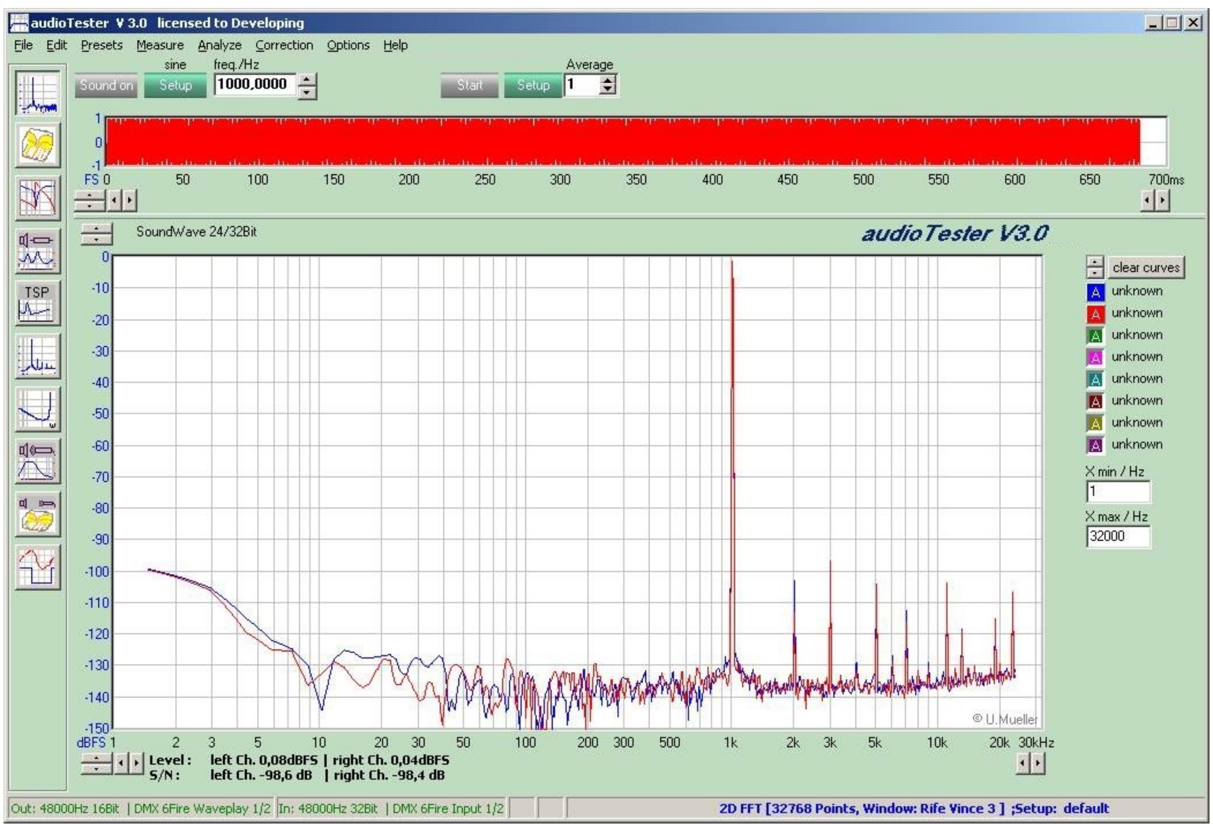
In this case we must switch from data width Float to 24/32Bit (Audio-In-Parameter Dialog)



ASIO exactly the same values

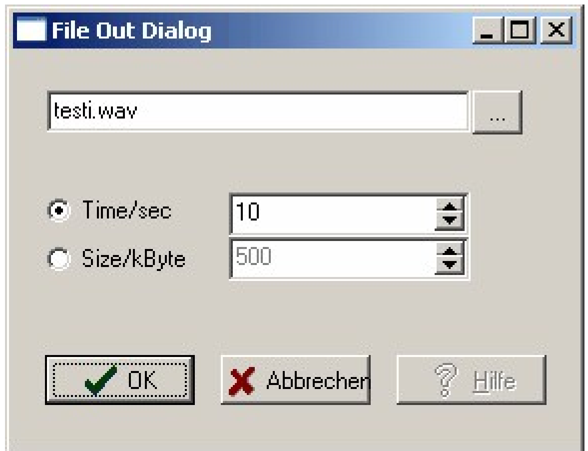


Soundwave with 16Bit data In/Out no so perfect



Soundwave with 24Bit Input and tone output with 16Bit, a little bit better.

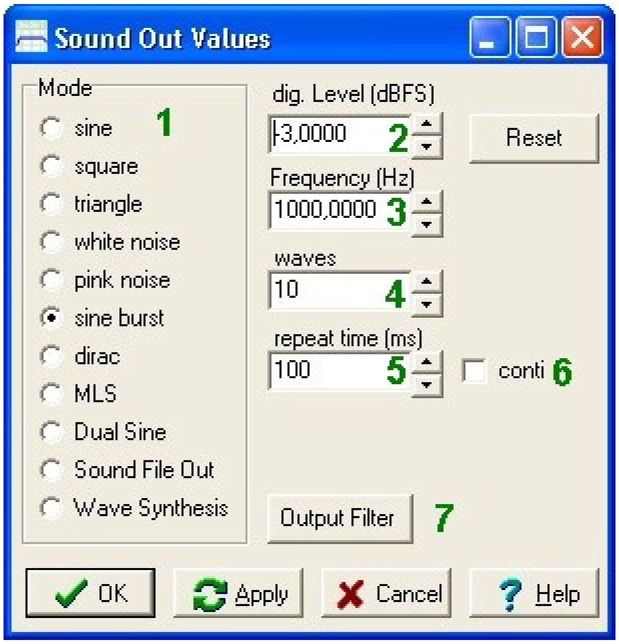
SoundFile
The SoundFile.DLL writes the tones or impulses into a wave file and not to the sound card.
The wave file format (sample freq, data width) is adjustable with the normal *options/audio-out-parameter* dialog.
This is very interesting to produce a CD for the asynchronous impulse measurement.
The tone you choose normally with the Sound Setup Button and click onto the *Sound on* Button, then the following dialog opens:



Via a File Dialog choose the file name of the wave file
The size of the wave file is adjustable over the time in seconds (max. 300sec.) or over the size in kBytes (max. 50Mbyte) .
With a click on OK the sound will be written into the wave file.

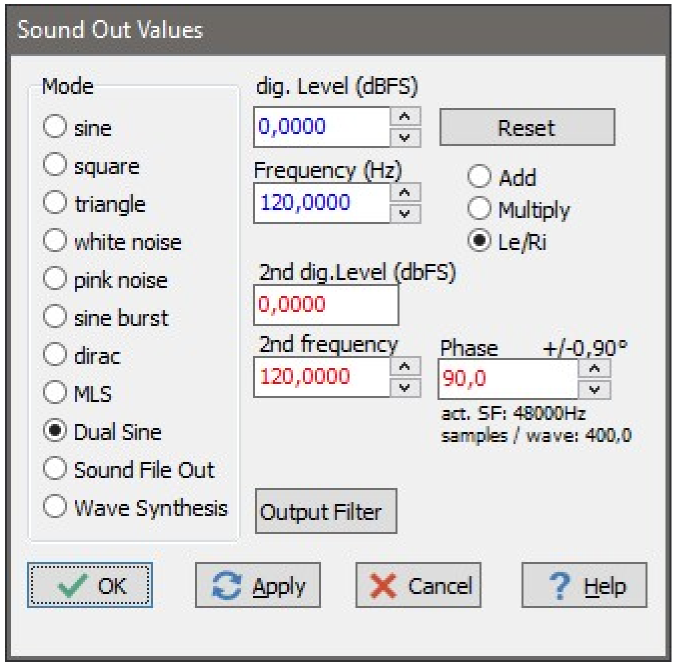
8 Sound Out Setup

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- and MLS Impulses you are able to force a continuously output.
7. With the "Out Filter" Button can you apply a time domain filter.

Dual Sine:
With the radio buttons 'add, multiply and Le/Ri' you are able to change tone modulation
add: the sine waves will be added (see **Inter modulations Distortions**)
multiply: the sine waves will be multiplied
Le/Ri: the first sine wave is applied to the left channel and the secondary sine wave is applied to the right channel
The second frequency (red in the picture) can also be provided with a phase offset to the first frequency.
The phase can be changed with the up/down button with a fixed increment and/or with direct value input.
The increment is calculated from 360(SF/Freq.) in the example on the left 360°/(48000Hz/120Hz)=0.9° per angular step
Two values are shown below the phase input field, the current sample rate and the number of samples for one full wave.
Here in the picture on the left you can see that at 120Hz, for example, the full wave consists of 400 samples (at SF of 48kHz). With a SF of 192000Hz there are 1600 samples (picture on the right)



With the **Output Filter** button, the output signal can be assigned a filter, see also Time Domain Filter

Wave Synthesis:
Additional selection **Wave Synthesis**

Sound File Out:
If you using the SoundWave.dll or the SoundDirect.dll it is possible to play sound files.
The SoundWave.dll accept wave files and the SoundDirect.dll accept WAV, MP3, OGG and AIFF files.
The sound files are playing endless until switch off by the 'Sound-Off' button.
If a sound files anohter sample rate as selected in the audioTester, Windows will convert it,
in this case it may be possible that there are some sound degradation.

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