User's Manual

DIGITAL AUDIO

ANALYSIS

SYSTEM

for Windows 9x

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Foreword

The Digital Audio Analysis System DAAS®32 was developed and produced by adm engineering, Germany.

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The following instructions are only binding if you are using the DAAS®32 inside of the European Community, otherwise you can ignore these statements.

> The Digital Audio Analysis System is in conformity with the actual required standards in accordance with 89/336/EWG in consideration of the following statements:
>
> • All unused input and output sockets must be applied with the protection-plug delivered with the DAAS®32
>
> • All in conjunction with the DAAS®32 used equipment such as power amplifier, measuring microphone, computer etc. must be marked with a CE-sign
>
> • Use only double shielded high quality cables with high quality cinch plugs (metal sleeve)
>
> • The inner conductor of each input- and output cable must not be accessible
>
> • The ferrite delivered with the DAAS®32 must be applied to the microphone cable, close to the DAAS®32 converter board
>
> • Each cable longer than 0.5 meters must be applied with a ferrite like the microphone cable. Ask your distributor to get more ferrites

Contents

1 General

1.1 Preface

The thinking behind the development of the Digital Audio Analysis System was to produce a reasonably priced, PC supported measuring system for the measurement of frequency response curves. From the original concept of a sampling system, analogue to digital converter and computerized evaluation techniques the system has developed to its current form with wide ranging functions and features; the DAAS$^{®}$32.

Despite the numerous analysis possible from one sampled signal it is the 'measurement' not the signal which takes priority as far as the user is concerned. That is to say that the loudspeaker or amplifier developer/tester don't need to have any knowledge of digital signal evaluation techniques as he will be closely guided through each measurement to the final result. All important parameters are optimally set for each measurement. For the user who wishes to delve deeper into the system the possibility exists to change the system parameters and use the system in other areas with the "Spectrum Analyzer" and "oscilloscope" functions.

This new version of DAAS$^{®}$32 includes many useful and important features:

- Operation under the modern operating system WINDOWS 95
- High quality by using 16 Bit Digital/Analog- and Analog/Digital-converter
- Sampling rates up to 64 kHz
- Possibility to control external devices via switching lines (optional)
- Measurement of frequency and impedance curves with variable frequency resolution
- Calculation of phase from frequency- and impedance-curves with automatic delay correction for acoustical measurements
- Minimal phase and excess phase
- Step response, impulse response and ETC
- Waterfall spectrum (cumulative spectral decay)
- Free field correction of acoustic frequency response curves
- Distortion measurement by virtually any frequency
- Evaluation of distortion curves versus frequency
- Intermodulation distortion with variable frequencies
- Difference tone distortion with variable frequencies
- Thiele Small parameters including equivalent volumes

- Digital storage oscilloscope function with trigger, time selective frequency analysis and waterfall diagram
- Spectrum analyzer function with linear or exponential averaging and smoothing
- Reverberation time analysis (RT60)
- Maximum frequency resolution approx. 0.1 Hz
- Interface to speaker CAD programs Audiocad, CALSOD and AkAbak®.

Working with the Digital Audio Analysis System DAAS®32 will give you a whole series of useful and simple to use features, for example the 'screen zoom' and the possibility to show frequency and impedance curves on top of each other.

1.2 Introduction to how the system works

The Principe how the system works should be explained by means of the frequency response measurement.
Principally there are 3 ways to measure frequency response. The simplest and eldest method is applying a "wobbled" signal to the object to be measured. A signal of known frequency and amplitude is fed to the input of the object being measured and the signal on the output is measured and compared. To obtain a complete response curve numerous individual measurements are required.

Measuring microphone and amplifier

Motor

Sine wave generator   **Object**

Motor
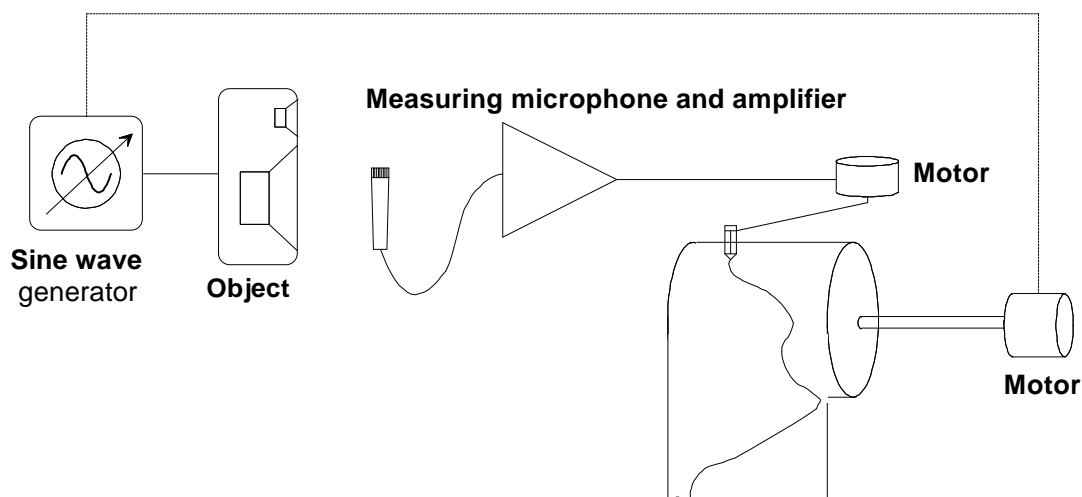
By using a plotter as shown above a continuous response curve can be obtained. If a bandpass filter, the middle frequency of which can be changed in synchronization with the generators frequency, is inserted between microphone and plotter, it is possible, within certain limits to remove room reflections, i.e. it is possible to achieve free field measurements within a sealed room. This principle is known by the name "TDS".

A far better and quicker method is to use banks of filters. The system comprises a number of filters connected in parallel, each with a different middle frequency. Each filter is connected to its own measuring instrument.

Input —— Amplifier ——

| Bandpass 1 — Detector 1 |
| Bandpass 2 — Detector 2 |
| ⋮ |
| Bandpass N — Detector N |
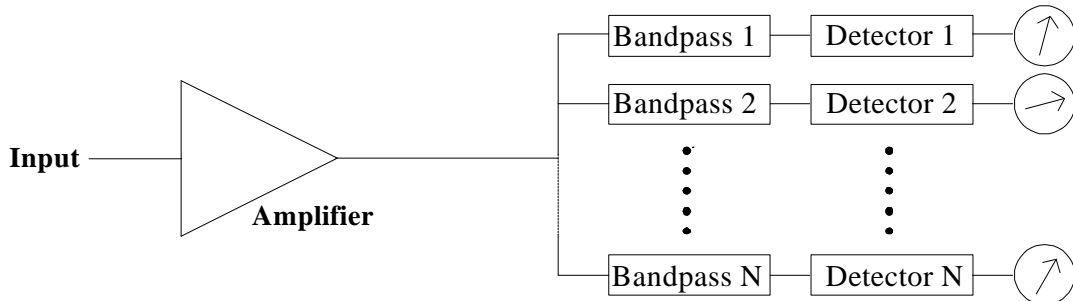
When the middle frequencies are so chosen that the filter bands border exactly with each other a complete spectrum can be achieved. With this system the entire frequency response curve can be obtained with only two individual measurements. The measured object (e.g. a loudspeaker) is fed with a signal that contains all the frequencies within the frequency range we are interested in. A suitable signal is the "Pink Noise" or "White Noise". Now the spectrum at the input of the measured object (reference measurement) and the spectrum at the output is measured. The frequency response is achieved by dividing the output spectrum by the input spectrum.

Further, selective measurements are also possible (i.e. single frequency bands) and using these, for example, distortion measurement is possible. Signal evaluation, for example dB(A) is also possible.
However for a high resolution a high number of filters are required. These filters, particularly the ones with very narrow frequency bands, are expensive.
Typical parallel filter analyzers contain 32 band filters, each 1/3 Octave = Terz.

The development in computer technology over recent years has enabled another technique to be used, that of the FFT Analyzer. This type of analysis is based on a mathematical algorithm which simulates a thousand parallel filters.

By using an analog to digital converter the signal can be read by the computer while simulating the filters. The hardware requirements are therefore drastically reduced.

| Input | Antialiasing Filter (low pass) | Sample and Hold | Analog Digital Converter | Fast Fourier Transform. | Display |
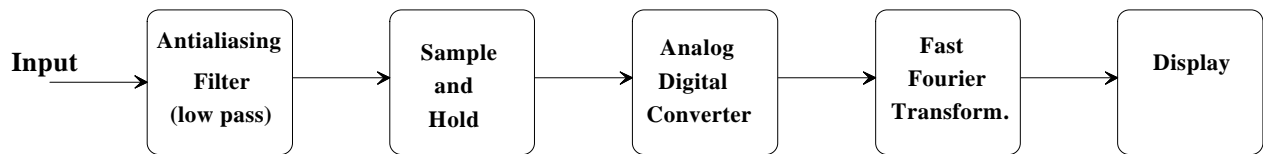
The measurement of frequency response is similar to that of the parallel filter Analyzer namely through evaluation of the signal at the input and output of the object under test.

Theoretically, if we use Periodic Noise, it should be possible to obtain a frequency response by measuring the signal at the output only. However because amplifiers with variable gain and filters are used it is necessary to measure the signal on the input also (reference measurement).

The frequency resolution, equivalent to the number of filters using the parallel filter analyzer, is only limited by the processing power and memory of the PC.

There is no need to use an external signal generator. All used signals are generated by DAAS®32 for the various measurements:

- Pink noise, white noise and a Maximum-Length-Sequence (MLS) for frequency response and impedance measurements
- Sweep-signal for impedance measurements at loudspeakers
- Sine wave of any frequency for distortion measurements
- Combinations of sine tones for intermodulation-factor measurements
- Tone series for harmonic distortion measurements
- Special PN-Sequence for frequency response measurements with the spectrum analyzer function

Further signals can be requested or generated by the user. Since the Digital Audio Analysis System uses the WAV file format you can modify or generate test signals simply by using a common sound editor such like GOLDWAVE.

1.3 Problems, questions and technical support

If occur problems or questions, please follow the steps below:

1. Refer this manual, section "Diagnostics and trouble shooting"
2. Check the README.DOC file on the DAAS®32 CD ROM
3. Contact your distributor or adm. Before contact your distributor or adm please have the following information handy:
   - DAAS®32 version (click "INFO" - "VERSION" in the main menu)
   - Operating system and version
   - Main board manufacturer
   - Speed and type of your computer (e.g. Pentium 200)
   - Contents of your AUTOEXEC.BAT and CONFIG.SYS
   - Have a list handy with all other peripherals you are using, as well as the I/O address information about this peripherals
   - If problems with measurement results are occuring, store all involved signals (frequency response or impedance measurements: store also the reference signals!) to FLOPPY disk and send them to us

2 Installation and Start

2.1 Installing the DAAS®32 converter board.

Switch off your computer and isolate from the main supply. Remove the cover, choose a free 16 bit slot and remove the cover plate associated with it. The slot should be as far away as possible from other cards and the power supply. Possible re-arrangement of the cards may be necessary. Before you remove the DAAS®32 converter board from its protective sleeve touch an earth point (e.g. a central heating radiator) to discharge any static that may have built up on you. Insert the card in the chosen slot ensuring it is fully located and retain it with the screw from the cover plate previously removed.
Mistakes during the installation may cause damage to the card and damage caused in this way is not covered by the guarantee. Remember the guarantee on your computer may also be affected. If you are in any doubt about installing the card consult an expert or your dealer.
Replace the computer cover, switch on and ensure that the machine boots correctly. If this is not the case check again that the card is correctly inserted in a 16 bit extension slot and that all cables and plugs are correctly fitted. Further information for special PC hardware can be found in the Readme files.
Every DAAS®32 converter board is tested before dispatch. It is possible that an address conflict has occurred with another extension card. The DAAS®32 converter board is supplied set to I/O address 358 Hex to 35F Hex. The address can be changed, the method for doing this is described in section 7.

2.2 Installing the DAAS®32 software

Use the following steps to install the Digital Audio Analysis System software:

1. Turn on your computer and start WINDOWS 95
2. Insert the DAAS®32 CD in your CD ROM drive
3. Choose RUN...from the start menu, the RUN dialog will appear
4. Type D:SETUP resp. E:SETUP (depends on your CD ROM drive letter), the SETUP program will start
5. Choose the hard disk and the directory to install DAAS®32

The DAAS®32 installation program will apply a new program group "Digital Audio Analysis System" in the start menu.

Please read the README.DOC file on the DAAS®32 installation CD!

2.3 Start DAAS®32

Choose PROGRAMS from the start menu, select DIGITAL AUDIO ANALYSIS SYSTEM and click DAAS32.
Or choose RUN...from the start menu (the RUN dialog will appear), and type C:\<DAAS®32-Directory>\DAAS32.

Refer the WINDOWS handbook for more variations to start a program and how to place a DAAS®32 icon on the desktop..

Note: If the DAAS®32 converter board is not installed, the jumper settings does not match the adjustments in system options or something else is wrong, the error message "No answer from converter board - Switch to DEMO mode?" appears. If you are going to configure the system click the "No" button. If you have not installed the DAAS®32 converter board in your PC, click the "Yes" button. The system will now ignore the converter board and pass all signal data direct from output to input.

2.4 Using the system

The graphic user interface of the DAAS®32 system can be controlled completely with mouse. But while doing some measurement sometimes the mouse control is not the optimum, so you can also control the DAAS®32 system via keyboard. Simply press the key marked with a short underline. Examples: To start a measurement move the mouse to the button "F1 Start" and click the left mouse button, or simply press F1. To adjust the sampling rate move the mouse to the button "kHz: 48" and click the left mouse button, or simply press the key K. To open the menu press and hold down the "Alt" key, then press the key signed with the short underline.

All keystrokes and keystroke combinations will be described in this manual like follows:

D	Press the key "D",
F1	Press the key "F1",
Ctrl + F2	Press and hold down the "Ctrl" key, then press the key "F2",
F8 - F2	Press and release the "F8" key, then press and release the "F2" key.

The following illustration shows the DAAS®32 main window with its various screen areas.
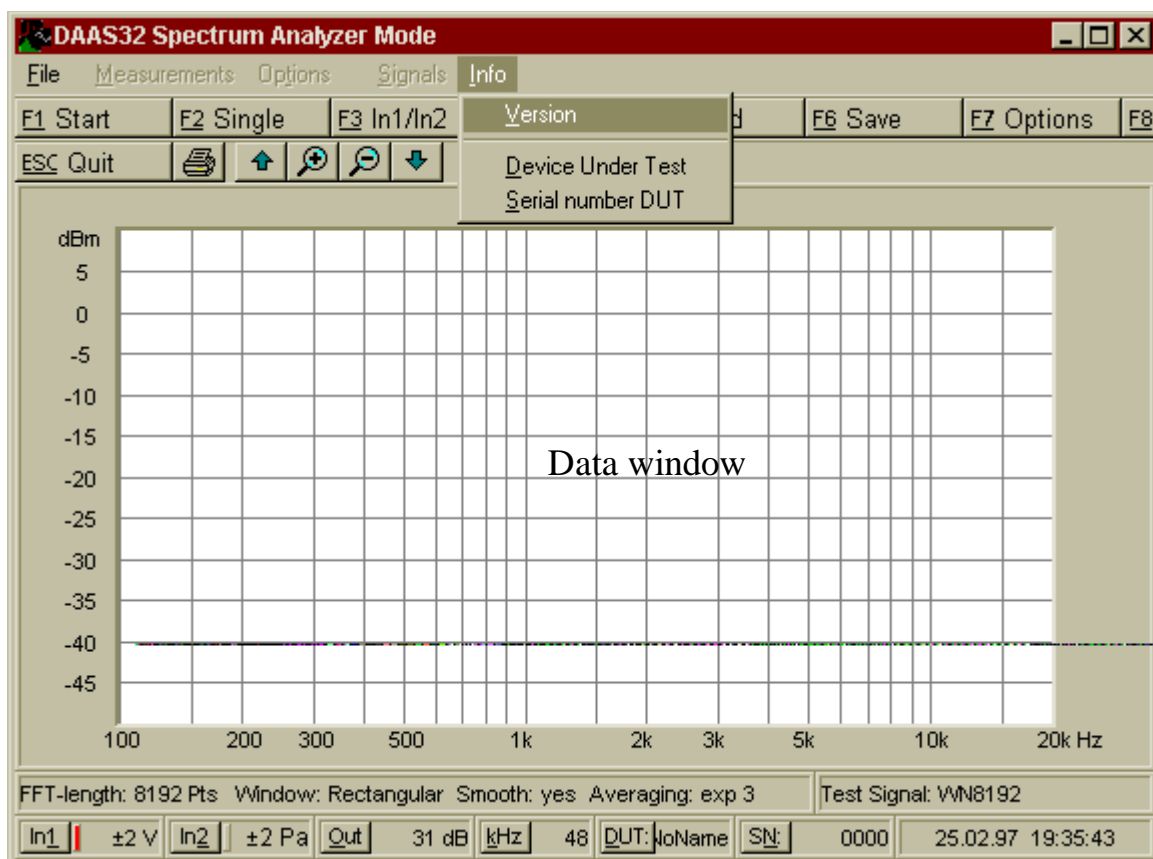
At the top of the window below the title bar you will find the menu bar. Here you can start the various measurements, setup the options, start the signal player and recorder, get the version info and enter the Device under Test infos.
With the buttons below the menu bar you can control the DAAS®32 completely. You can make a printout every time by pressing P or moving the mouse to the corresponding button and clicking the left mouse button.

All measured signals and calculated curves will be shown in the data window. Move the mouse in the data window and click with the left mouse button to insert a lettering. Press the right mouse button to readout signal or spectral data values. The frequency or time value and the corresponding ordinate value will be shown above the data window.

To change the visible data screen area by using the keyboard refer the following table:

Shift right   : key →
Zoom Out X-axis   : key Ctrl+ ←
Zoom In X-axis   : key Ctrl+ →
Shift left   : key ←

Shift down   : key ↓
Zoom Out Y-axis   : key Ctrl+ ↓
Zoom In Y-axis   : key Ctrl+ ↑
Shift up   : key ↑
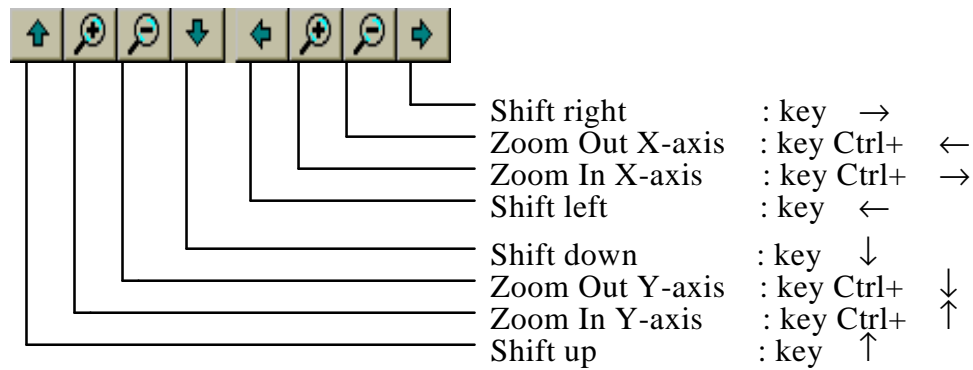
The lower section of the DAAS®32 window contains the status line and various control buttons. In the status line there are some actual measurement parameters shown such as FFT length, weighting window, averagings, test signal and so on. With the control buttons there are the following adjustments possible:

- "In 1": Sensitivity of the electrical input,
- "In 2": Sensitivity of the microphone input,
- "Out": Output attenuation (higher values means lower test signal level),
- "kHz": Sampling rate,
- "DUT" Name of the Device under Test,
- "SN":   Serial number of the Device under Test.

Note: Some control buttons can be blocked sometimes (depending on the actual measurement condition).
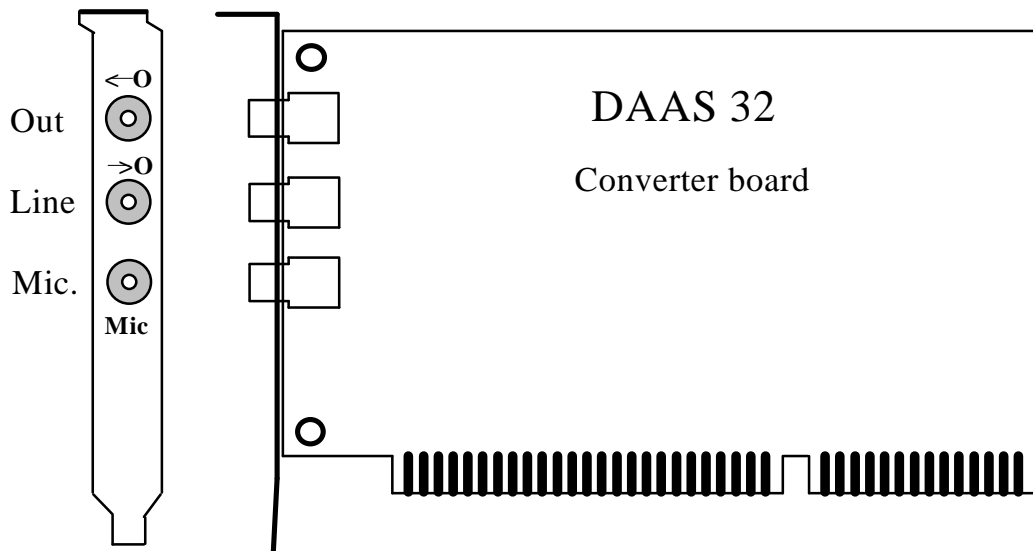
The DAAS®32 converter board has 2 inputs and one signal output. The microphone is connected to the "Mic" socket. Using the microphone supplied the maximum sound pressure is ±2 Pa and ±8 Pa.

The input "Line" (Symbol →O) is the input for electrical measurements. Three ranges are available ±0.2 V, ±2 V and ±20 V.

All signals generated by DAAS®32 are available from the socket "Out" (Symbol ←O). The output has a maximum level of ±4V into 600 Ohm and can be reduced in 1 dB-steps from within the program. The available range is 90 dB.

For measurements on loudspeakers or other objects of low resistance a power amplifier must be connected between "Out" and the object. A normal hi-fi power amplifier will be suitable. Use the "Aux" or the "Tuner" input of the amplifier. Setup the output attenuation to around 30-35 dB.

CAUTION If you want to use a hi-fi power amplifier, first read chapter 12 to prevent from system damage!

3 Measurements

The most important and most commonly used measurements are pre-programmed into your DAAS$^{®}$32. The measurements are user controlled throughout, i.e. the user is guided through the measurement process and allowed to choose various alternatives as the measurement progresses. All the processes are self explanatory and can be carried out with little or no preparation. The parameters for the various measurements have already been optimized, they can however be changed for special measurements by the experienced user. In this way measurement error is greatly reduced.

3.1 Frequency response measurements

Using DAAS$^{®}$32's digital signal processing, with only two measurements and in only a few seconds it is possible to establish the frequency response from loudspeakers, crossovers, amplifiers etc. It is also possible to plot the response curves using a Terz (1/3 octave) scaling.
Further, it is also possible to calculate the phase frequency response and impose it on the frequency response diagram. During acoustical measurements, with the aid of the correlation's-analyzer, the delay is calculated and corrected, so that the delay corrected phase curve can be plotted. The minimal phase can be calculated, also a fine delay-correction using the excess phase.
From an object's (say loudspeaker's) frequency response curve, using DAAS$^{®}$32 it is possible to obtain more further important properties: the step response, the impulse response and the waterfall plot(Cumulative Spectral Decay).

Open the "Measurements" menu and choose "Frequency Response". It is also possible to start the frequency response measurement by pressing Ctrl + F.

Firstly you must carry out a reference measurement of the input signal of the Device under Test.There are 4 possibilities:

1. electric - acoustic (e.g. Loudspeakers),
2. electric - electric (e.g. amplifiers, crossovers),
3. acoustic - electric (e.g. Microphone),
4. acoustic - acoustic (Loudspeaker comparisons).

In case 3 and 4 an acoustical reference measurement is necessary. Connect the signal output "Out" with the amplifiers input (use "Aux" or "Tuner") and connect the speaker to the amplifiers output. Setup the output attenuation to around 30-35 dB using the "Out" button. Click "Test level" to check the sound level. If the

sound level is too high or too low this can be adjusted using the amplifiers volume control knob or the output attenuation.

Position the microphone on the required reference location and start the reference measurement.

In the first two cases (electric -electric and electric-acoustic) an electric reference measurement is necessary. Connect the output "Out" with the object and the input "Line". If the input impedance of the Device under Test is lower than 600 Ohms, use an amplifier like described before. In this case connect the output of the amplifier with the Device under Test and the input "Line".

The connections, using a speaker box as Device under Test is as follows:
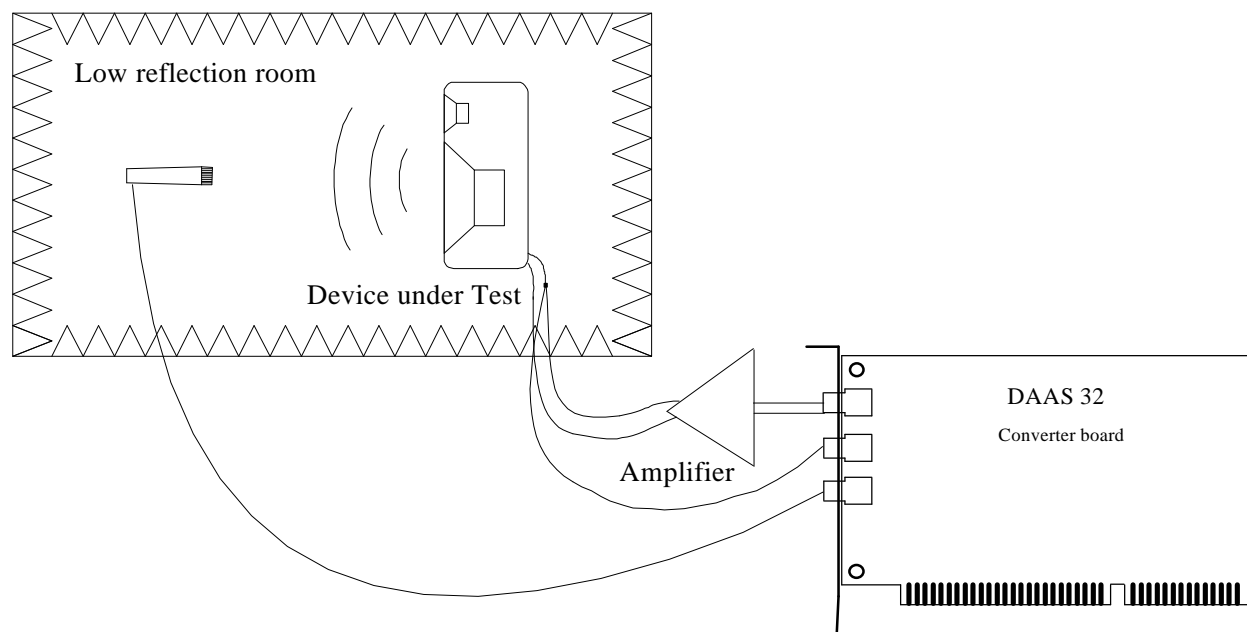
To start the reference measurement press F1 or the "F1 Start" button.

The best input voltage range depends on the Device under Test. If you require to test an amplifier or some other piece of hi-fi equipment set the measurement level using the "Out" button to the range 30 to 40 dB (the equipment must not be over driven.....clipping!) and choose the input range "2 V" or "0,2 V" (using the "In 1" button).

When measuring loudspeakers, the signal level is increased by the amplifier, so generally the 20 V input range should be selected. Level correction can again be achieved by adjusting the amplifiers volume level.

It is also possible to measure the test signal level. Press O or click on "Out.", then click on "Test Level". Now a short part of the test signal is played and measured,

and the level is displayed. Set the level with the amplifiers volume control or set the attenuation for the required level.

F2 and F5 are the "Load" keys. With this keys it is possible to load a previously measured signal for evaluation. It is therefore possible to measure and save a whole series of signals and evaluate them at a later date. It would be possible to load and use a previously measured reference signal, however this would of course only have any value if all aspects of the measuring set-up have not been changed in the meantime.
Using F2 you can quick load the last reference measurement. Every reference measurement will be automatically saved under a special file name.
Further details are contained in the section on "Loading and saving signals".

On completion of the reference measurement the following screen should appear:
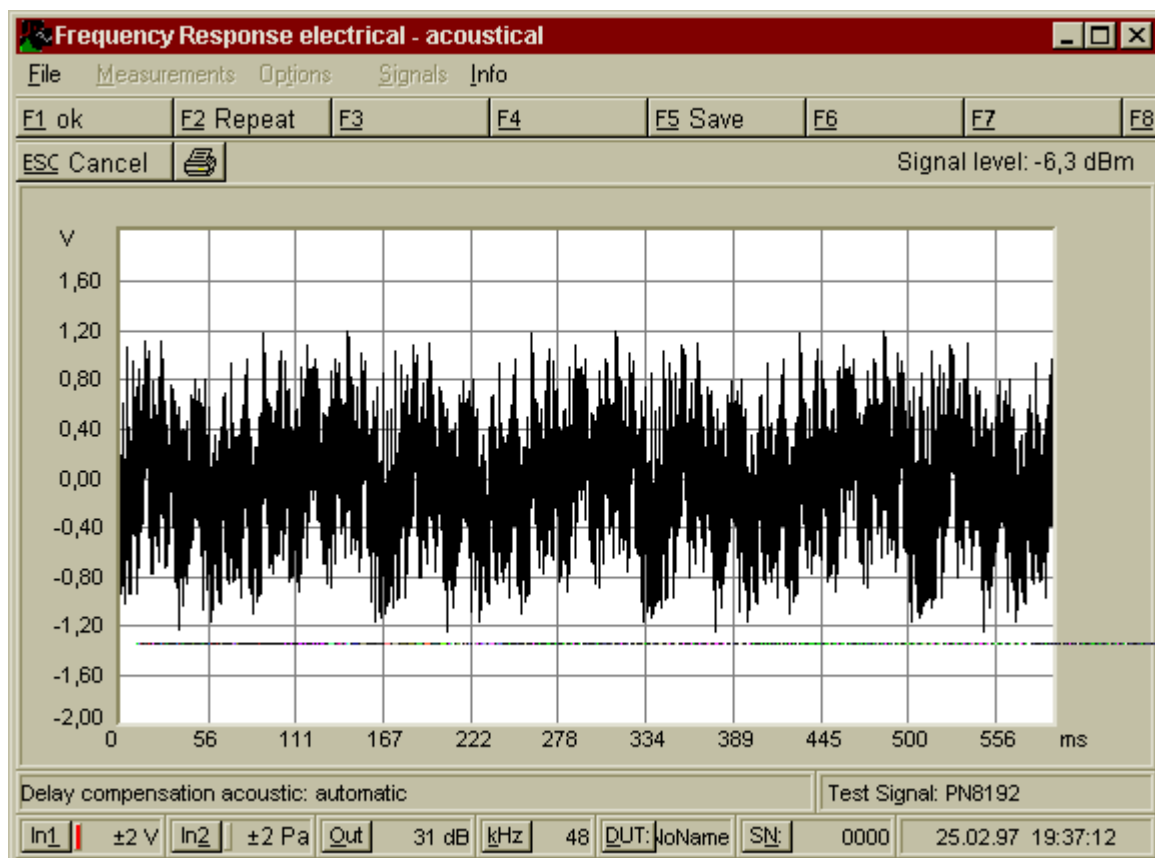
If the signal is too large a warning beep will be audible, and if it is too small a warning message will appear. If you ignore these warnings, the measurement accuracy will be reduced. In both cases the amplifier volume control or the input sensitivity should be adjusted and the operation repeated.

CAUTION The volume level should now of course not be changed for the duration of the measurement.

If required the signal can now be saved using F5.

Press F1, if the signal is OK. If the level or the output attenuation is changed the curve will still be correct except the absolute level will be different; i.e. the curve will have shifted along the amplitude axis. On completion of the evaluation connect the output of the Device under Test with "Line" (measuring amplifiers etc.) or position the microphone (measuring Loudspeakers etc.). Choose a voltage range so that the range is fully used but over driving (clipping) does not occur. When everything is set start the measurement by pressing F1. Alternatively load a signal from disk with F5 (see section "Loading and saving signals").

If the measurement is OK it can be saved using F5.

If in the "Options" menu you chose 'manual' delay compensation, you are asked during a loudspeaker measurement to enter the distance between loudspeaker and Mic. This information is most important if the distance is greater than 10 cm or if the phase response has to be calculated very accurately. Enter this information as accurately as possible rounded up to the nearest cm.

The normal setting for this option is delay compensation: automatic, i.e. no distance is requested as it is measured automatically. The manual setting is only for use when the automatic setting is producing obviously false results.

The calculated distance is shown in the status line at the lower section of the DAAS®32 window. Check that this distance is reasonable before continuing.

CAUTION Particularly with bass speakers it is normal for the automatic method to show a few cms more than the actual distance, this is due to the characteristics peculiar to bass units.

The following screen shows a frequency response measurement of a car speaker at a distance of 29 cms.
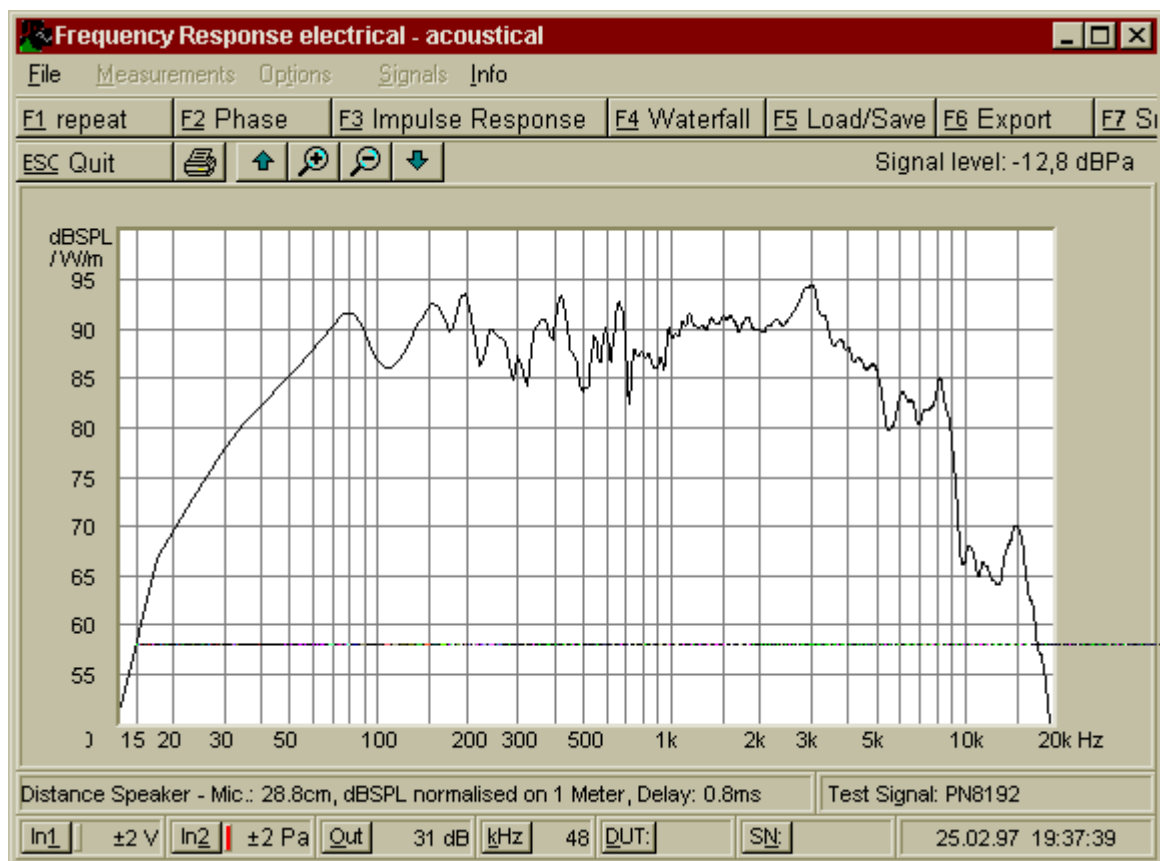
The scale can be altered under the "Options" menu. In this example dBSPL/2,83V normalized to 1 meter has been chosen. This means that the curve is shown corrected as if the measuring distance was 1 meter and with a signal level of 2,83 volt (i.e. 8 ohm at 1 watt). Assuming a distance of over 10 cm is chosen an exact level setting and distance is unnecessary as an automatic correction will be applied. The somewhat unusual scaling is necessary if the data is to be exported into loudspeaker CAD programs and for general comparison purposes.

As already stated by some loudspeakers it is possible that the calculated distance is a few cms too large. If this is the case a manual correction using F8 - F4 is possible. Here it is possible to enter the exact distance having established this with a tape measure, for example. It is also possible at this point to fix this distance for all future measurements during this session.

If only a limited frequency range is required, the limits can be set using F8 - F2. After entering the limits the curve will be redrawn using these limits. If you are testing a speaker which does not have the full range from 20 Hz to 20 kHz this feature should be used.

Using F8 - F3 you can set the switch "Shift Curve to 0 dB at 1 kHz" to "yes" or "no". Set to "yes" will cause that the actual curve and all following curves will be shifted until its value at 1 kHz is 0 dB.

Using F1 you can repeat the measurement like described before. Using Ctrl + F1 you can repeat the measurement very quickly without choosing the input range etc. The via Ctrl + F1 measured frequency curve will be inserted into the existing diagram using a new color. Note that if you use Ctrl + F1 no new distance measurement will be done, the old values will be used. If you change the distance between loudspeaker and microphone, the quick repeat function does not work proper.
To remove the old curves simply press the space bar.

3.1.1 Load and save curves

Frequency curves can also be saved and re-loaded. Press F5 to get into the Load/Save menu.
Choose 'Save frequency curve' to save the actual measured curve to disk. Choosing 'Load frequency curve' will be load a frequency curve on top of any current curves which during this process are not deleted. Any number of curves can be loaded and displayed this way. However take note of the current scaling being used as it is possible that a loaded curve cannot be seen as it lies outside the current display range.
It is also possible to load an impedance curve into the frequency response diagram (F3: 'Load impedance curve').
To remove the loaded curves simply press the space bar.

3.1.2 Smoothing
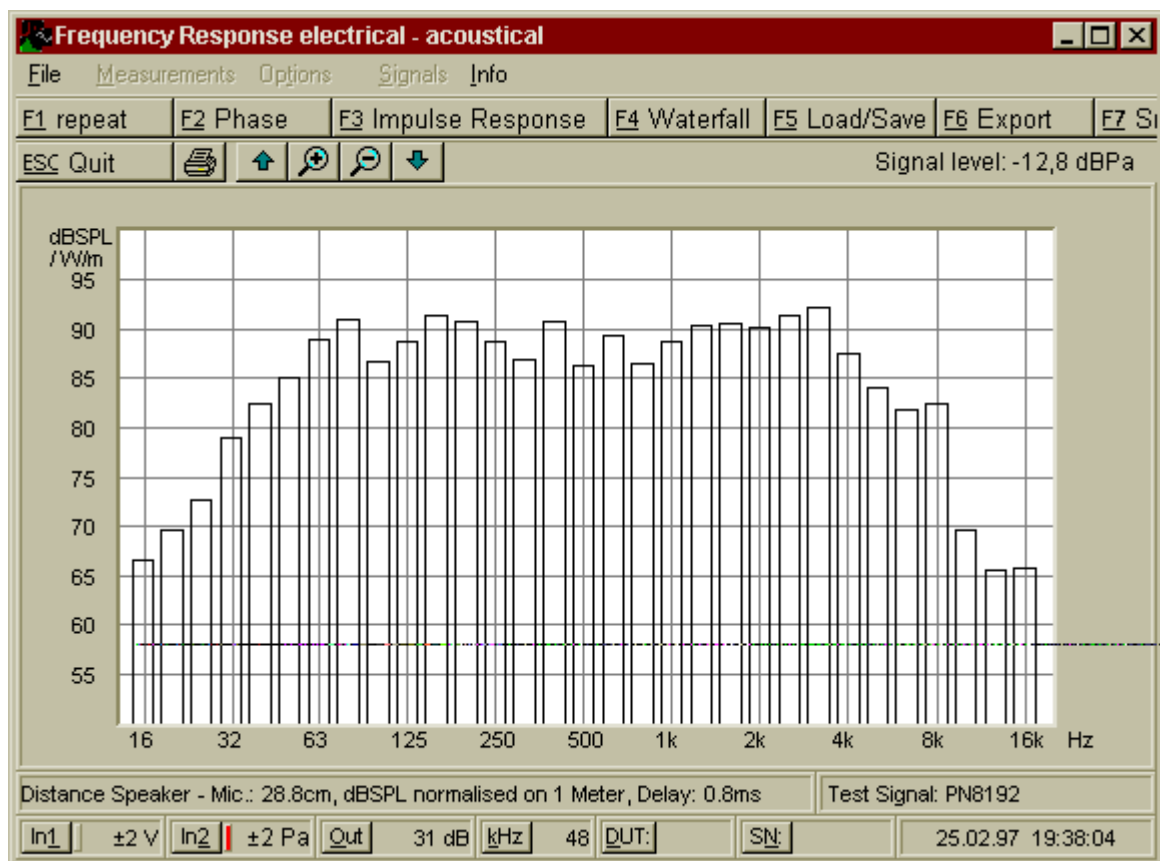
The frequency response curves, the phase curves and the group delay curves are smoothed with a special low pass filter, so that the relative frequency resolution is constant. The frequency resolution respectively smoothing factor is presetable in the "Options" menu and can be changed for the next few measurements (until you leave the frequency response measurements) by using the F7 button.

The frequency resolution respectively smoothing factor depends on the chosen FFT-length for the frequency response measurement:

FFT 65536 Points:	Octave /96	Octave /48	Octave /24
FFT 32768 Points:	Octave /48	Octave /24	Octave /12
FFT 16384 Points:	Octave /24	Octave /12	Octave /6
FFT 8192 Points:	Octave /12	Octave /6	Octave /3
FFT 4096 Points:	Octave /6	Octave /3	Octave /1.5
FFT 2048 Points:	Octave / 3	Octave /1.5	Octave /0.75

For some acoustic measurements the Octave/3 bar diagram is recommended. If the Octave/3 baroption is active the curve will appear as follows:
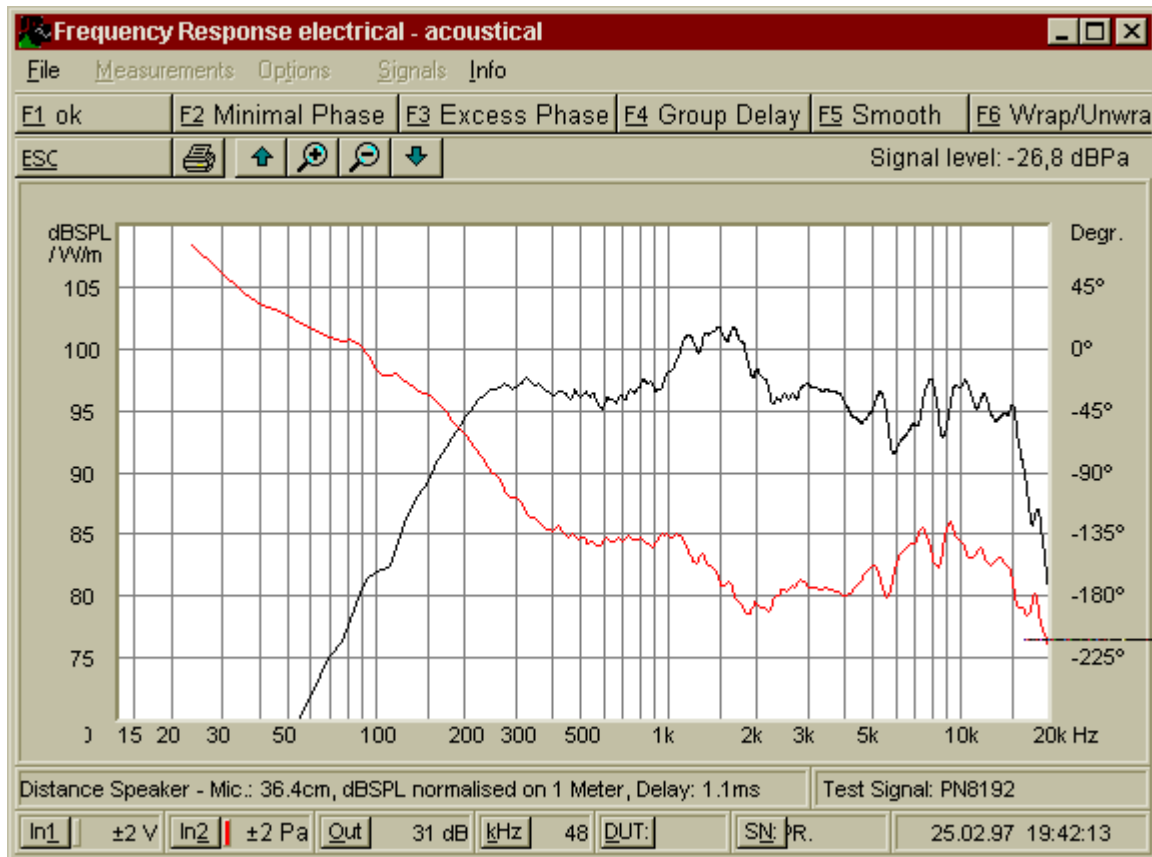

3.1.3 Phase response curve

From the frequency response curve other properties of the object can be calculated.
By pressing F2 the phase response curve will be calculated and displayed.

If the automatic delay correction is activated (default) or the distance from loudspeaker - mic is entered the following is shown:

The phase will be unwrapped shown. This means if the phase curve will achieve e.g. 180° it will not jump to -180°. You can change this option using F6.

The scaling of the phase response curve can be changed via the keys

between 5 Grad/DIV and 405 Grad/DIV.

3.1.4 Minimum phase and excess phase

Using the F2 key it is possible to calculate the minimum phase curve of a minimum phase system from the frequency response curve (magnitude response) using the HILBERT transform. This calculation can take, depending on your system and on the chosen FFT-length for the frequency response measurement, a couple of seconds.

The calculated distance between speaker and microphone will be corrected using the delay error derived from the difference between measured phase and minimum phase (sometimes called "excess phase").
If you want to calculate and display the excess phase curve press the F3 key instead of the F2 key. Very useful for evaluation of loudspeaker systems is the calculation of the group delay from the excess phase curve (see below).

CAUTION Do not calculate the excess phase curve after the minimum phase curve! If you want to calculate the excess phase curve after calculation of the minimum phase curve, first go back to the menu "repeat/phase/impulse response ...". Press the space bar to redraw, then press F2 to calculate the measured phase. Now you can calculate the excess phase curv**or** the minimum phase curve again.

Note that the minimal phase curve is possibly slightly in error (at the upper and lower frequency limit), because the HILBERT transform extends over an infinite frequency range, whereas the measured data have only a finite frequency range.

3.1.5 Group delay

From the phase response or the excess phase curve we can determine the group delay curve. The shown group delay curve
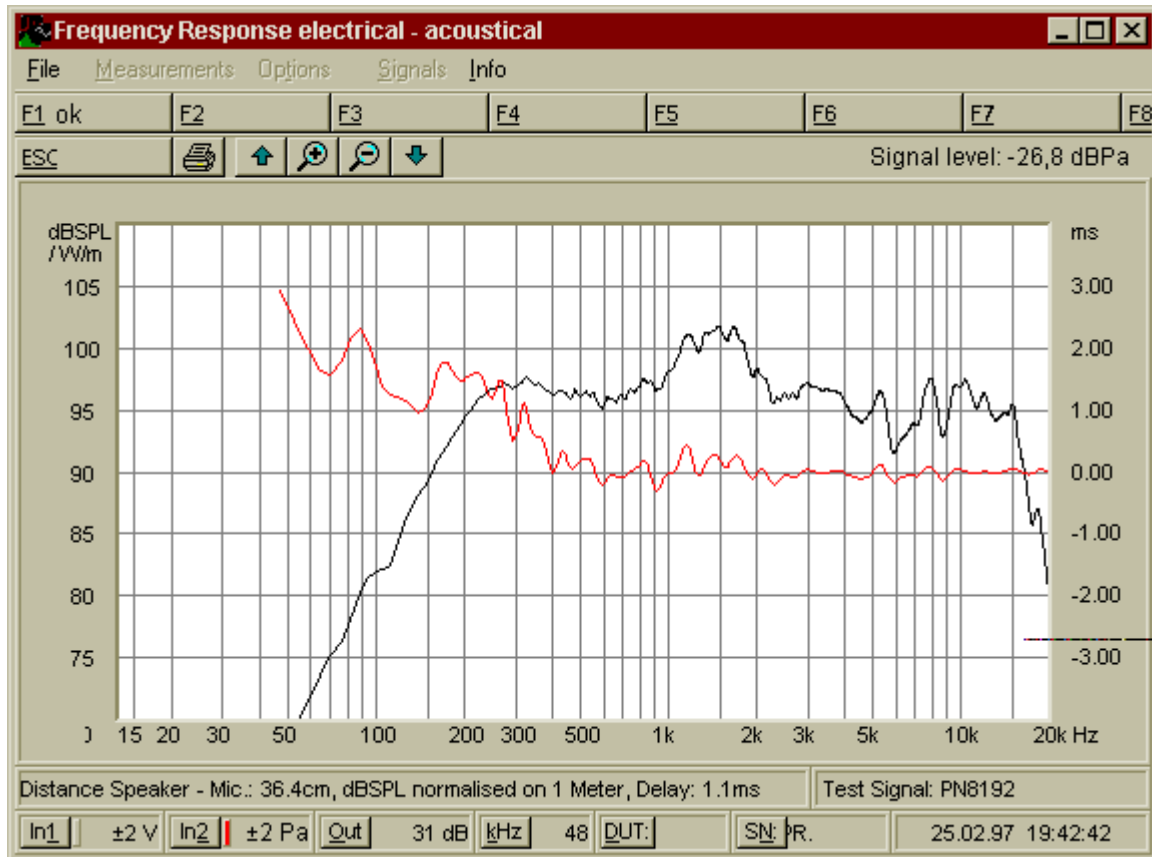
is the deviation from the via automatic delay compensation determined group delay.

It is recommendable to smooth the phase curve using F5 before calculating the group delay curve.

Hint: if you want to readout some values from the group delay curve press the right mouse button and move the mouse cursor to the interesting position. The frequency point and the delay value at mouse position and the corresponding distance in cm are shown in the upper left corner of the main window.

3.1.6 Impulse response and step response

From the measured complex frequency response curve it is possible, by using an inverse Fourier transformation, to calculate the impulse response. The impulse

response is the sound pressure level response a loudspeaker produces when driven by a very small needle impulse.

Press F3 to get the impulse response. Or activate the appropriate option in the "Options" menu to get automatically the impulse response after each measurement (see section).
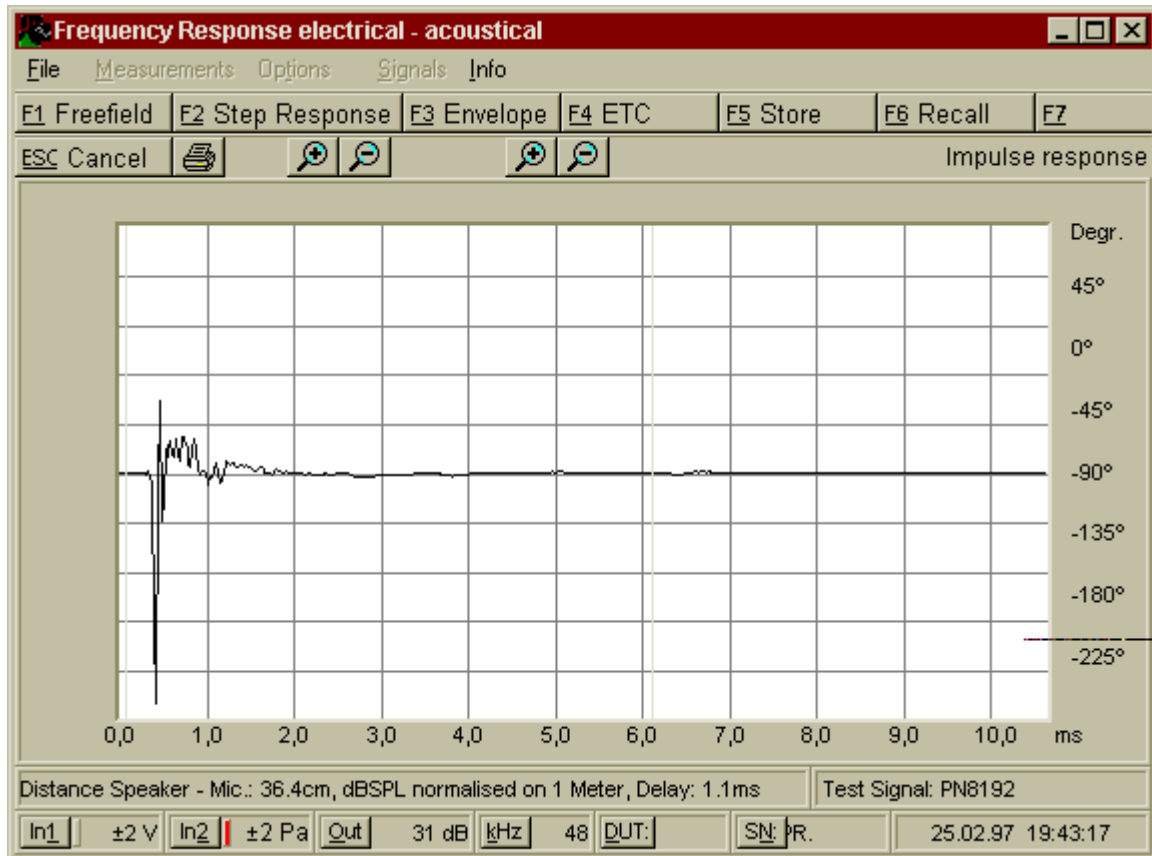
Press F2 to get the step response. The step response is the sound pressure level response a loudspeaker produces when driven by an on-switched direct-current-level.

Press F3 to get the envelope of the shown curve, and press F4 to calculate the energy-time-curve (ETC).

Using F5 and F6 you can save and reload impulse response curves, step response curves and ETC's.

To return to the frequency response curve without freefield correction press ESC.

3.1.7 Free field correction

For the successful measurement of frequency response curves various physical aspects of a loudspeaker need to be considered.

The 'real' frequency response can only be obtained in a free field environment or in an appropriately sized anechoic acoustic chamber. This is due to the reflections from floor, ceiling, walls etc. in all other situations.

An estimation of the lowest useable frequency in an anechoic acoustic chamber is a function of its size:
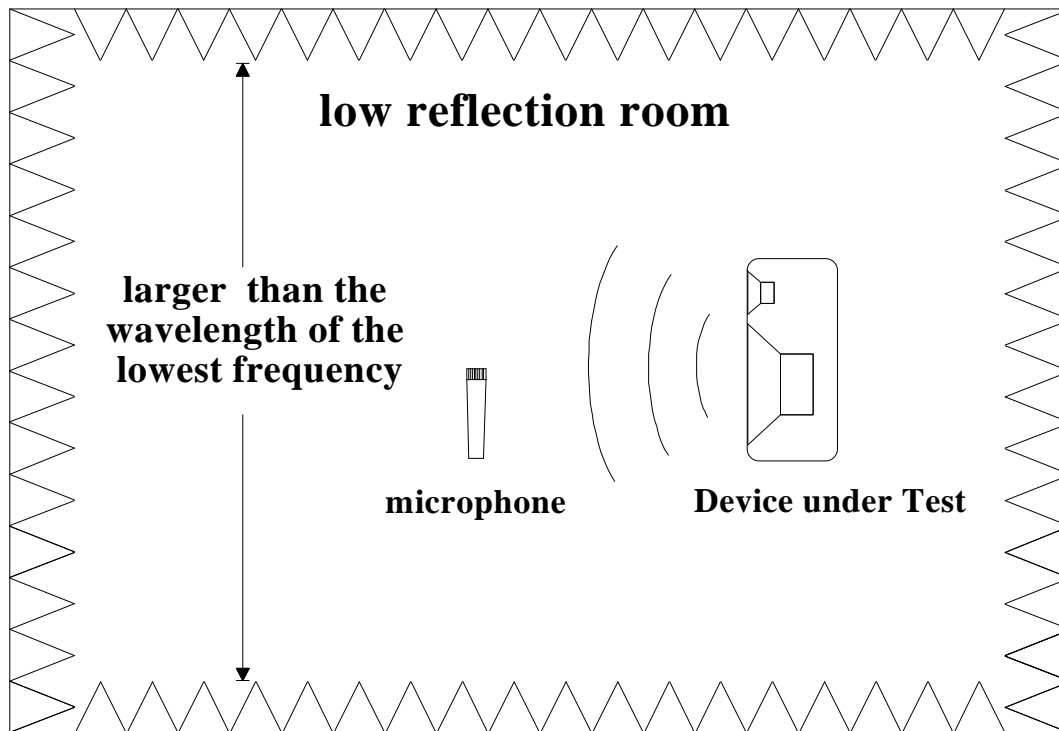
For a measurement down to 50 Hz the minimum size is 6,8 meter (!) net. The absorbent wedges must be 0,25 * wavelength i.e. 1.7 m. This gives a total of around 10 m !!!

If no special room is available you can also take measurements in a normal room. Choose the largest room available and keep the distance between the speaker and microphone small, about 10 - 30 cms. The room should not exhibit any 'hall' effects (echo).

microphone

Device under Test
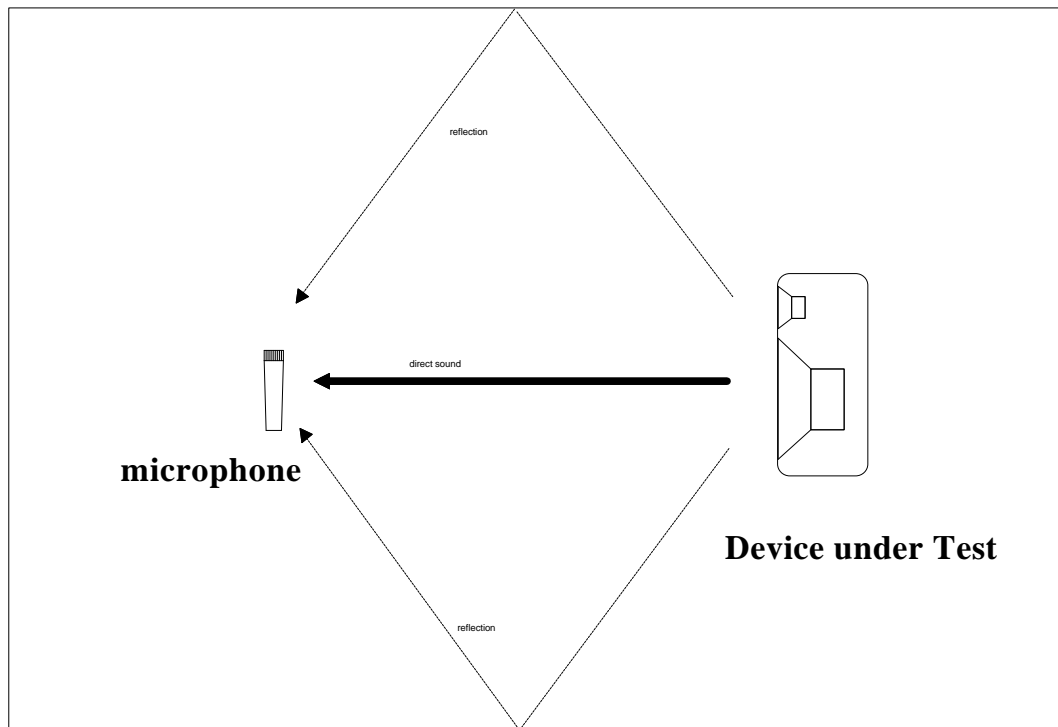
The undesirable reflections can be removed by using the impulse response.
In the waveform of the impulse response it is possible to see the echoes resp. the reflections. Search in the area of 3 to 10 milliseconds for a copy of the main impulse of the signal which appears between 0 and 1 milliseconds. For better detection of echoes you can change the display range using the

buttons, and you can calculate the envelope of the impulse response or the energy-time-curve (ETC).

In this example (see below) you can find an echo at about 6.5 milliseconds.
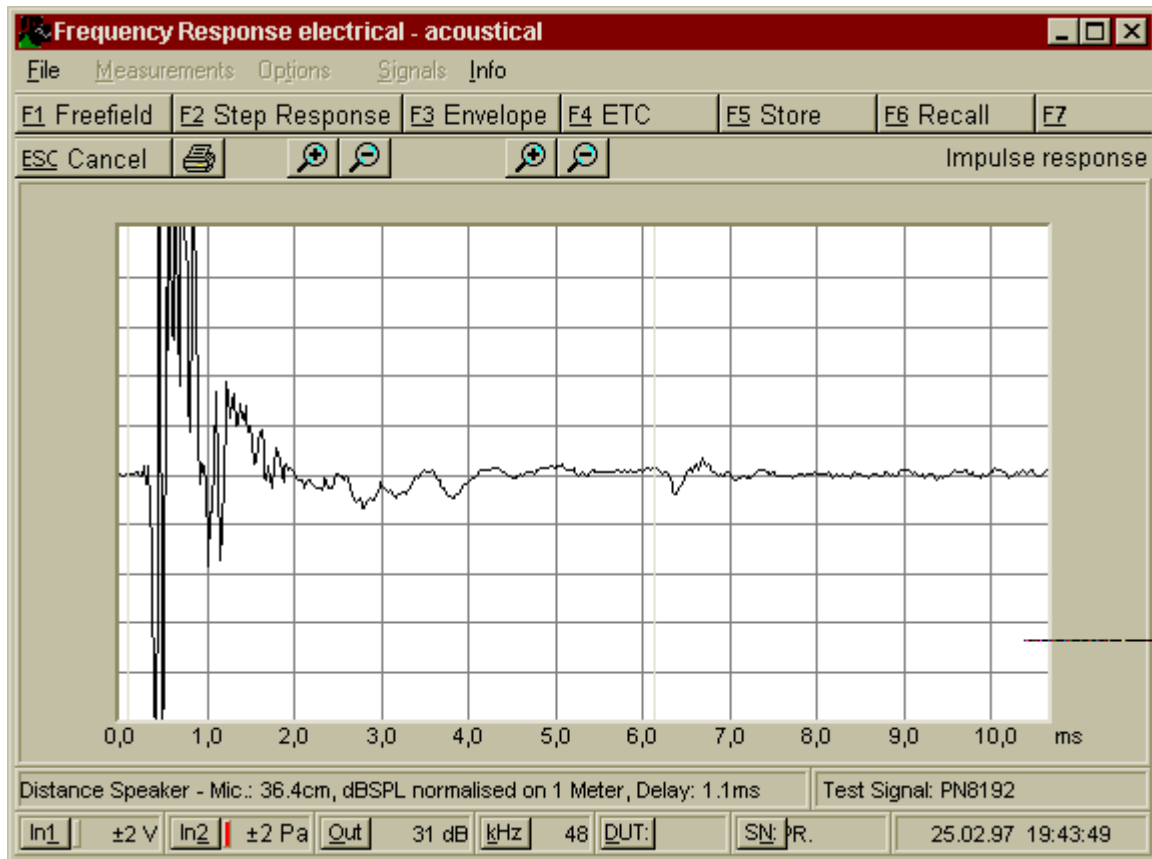
All signal parts to the left of the left marker (small vertical line) and to the right of the right marker line will be removed.

To remove the detected echo at 6.5 milliseconds simply move the right marker using the right mouse button to the appropriate position at 6 milliseconds.

For special applications you can change the position of the left marker by using the left mouse button.

Note that cutting out the echoes reduces the frequency resolution. The frequency resolution is a result of 1/measuring time. If the time window between the left marker and the right marker is about 5 milliseconds, the frequency resolution will be about 200 Hz.

By pressing F1 the frequency response will be calculated from the impulse answer and cleared of errors due to reflections.

Press ESC if you don't want to cut the impulse response and go back to the measured frequency response curve.
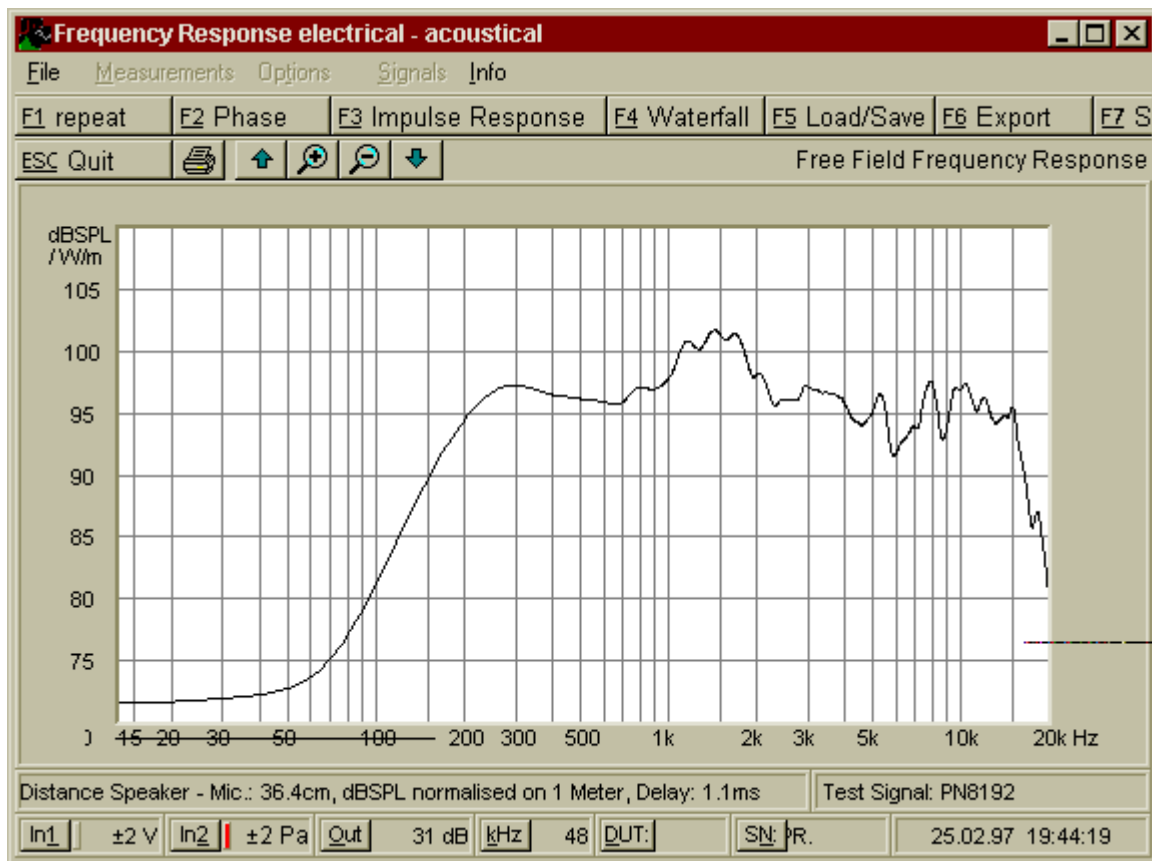
The area which is likely to be of little use due to the reduced frequency resolution is marked by a line on the frequency axis scale.

Note for DAAS 3L+ and DSA user: Unlike the DAAS 3L+ and the DSA the DAAS®32 will modify the originally data. This means if you cut the impulse response and calculate the freefield frequency response, the originally frequency response data are lost.

3.1.8 Combine curves

The disadvantage of the free field correction via impulse response is the reduction of the frequency resolution. This is especially problematic in the lower frequency range. The free field curve contains only few details of the true frequency response here.

A possibility to avoid this disadvantage is to combine a free field frequency response curve with a near field curve. So you can get a complete frequency response curve without the problems we mentioned before.

To combine curves do as follows:
1. Carry out the frequency response measurement of the speaker under test
2. If necessary calculate the free field frequency response curve
3. Position the microphone near to the speaker (near field measurement)
4. Press F8 - F1 and enter the chosen combine point.

The measurement then will be repeated and the frequency response curve values below the chosen combine point will be replaced by the new measured values. The combined curve can be saved and loaded like each other curve.

Note that the phase will not be combined, only the magnitude.

3.1.9 Cumulative spectral decay (waterfall diagram)

From frequency response curve it is possible to calculate the waterfall diagram. This shows, for all frequencies, the speakers cumulative spectral decay. It gives indications of undesirable resonances in enclosures and drive units.
Altogether between 48 and 72 curves are calculated. The first few curves are the frequency response with the test signal at the input of the speaker, all others are without. An ideal loudspeaker would have the first curve after switch of the input signal under the frequency axis. It would radiate no sound as no signal is present on the input. On a real loudspeaker this is however not the case.

Distance Speaker - Mic.: 36.4cm, dBSPL normalised on 1 Meter, Delay: 1.1ms — Test Signal: PN8192
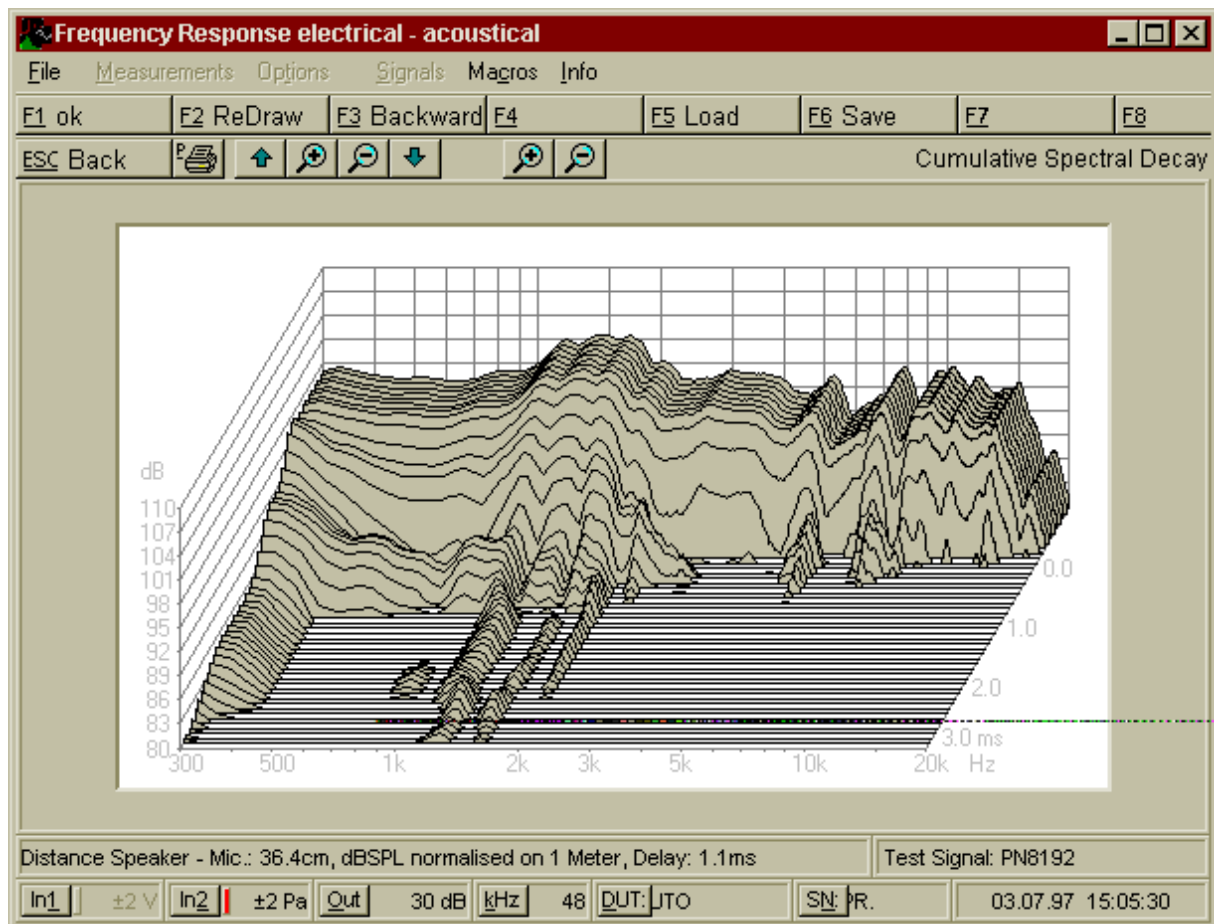
Here too the time axis can be altered with 'Zoom In t' and 'Zoom Out t'. However by using the zoom function all the curves are recalculated. The default setting for the time axis can be changed (see "Options").

The lowest and highest frequency shown (in this example 300 Hz to 20 kHz) can be changed, when, for example, measuring tweeters.

The time slices of the waterfall spectrum can be smoothed. This is useful because the FFT analysis in conjunction with the RECTANULAR weighting window causes leakage effects. This leakage effects can be reduced by smoothing the rising flank of the weighting window. But the disadvantage is the reduction of time resolution. This means a steep drop off appears smoothed.

Adjust the smoothing factor in the "Options" menu. One point means no smoothing, forty points means maximum smoothing. Use small values for tweeters and middling values for all other speakers. Default is 20 points.

The waterfall diagram can also be rotated. This means the rising slope will be shown instead of the falling slope. Use "F3 Backward" to get the rotated waterfall diagram. To see details of the rising slope use very small smoothing factors (one to five points).

Using F5 and F6 the waterfall diagram can be saved and loaded. When a waterfall diagram is loaded the current screen is deleted. The current screen is redrawn by pressing F2.

CAUTION: by removing the echoes during the free field response calculation all signal parts beyond the marker will be cut off. This corresponds then to the end of the waterfall.

3.1.10 Important notes for acoustic frequency response measurements

❑ A correct frequency and phase curve can only be obtained if the signal level is larger than the noise level.
If your results are obviously wrong check the signal to noise ratio. You can do this by using the spectrum analyzer. Choose the same test signal like the frequency response measurement. Measure the spectrum level without changing the set-up, then disconnect the speaker and measure again. Check the level. If you notice in one or more bands very little difference you must reduce the noise level, change room or increase the measurement signal level. The noise level in every band should be more then 20 dB under the signal level (the larger the better). If necessary change the test signal: pink noise by low frequencies has more energy than white noise or MLS. If the noise ratio problem is at high frequencies try white noise or MLS. See section "options for frequency response measurements" for more details.

The phase response in particular is very sensitive to unsuitable rooms or high noise levels. This problem often occurs with 'full range' measurement on speakers or speaker systems.

❑ The phase response can therefore only be calculated if an adequate signal to noise ratio is present. This is often not the case when measuring a single speaker through the full frequency range. A tweeter in the 100 Hz range will have a very low sound level resulting in the swamping from room noise. The phase response can be seen but is wrong due to the strong influence of the room. In this case the frequency range should be reduced.

❑ If you require to repeat earlier measurements it is important that the set-up is identical to what was used before.

• Type and size of room,
• Loudspeaker mounting (Box or sound wall of defined size),

- Angle, direction and distance between Mic and speaker.
- Type of measuring signal and the level of the measuring signal
- Type of measuring microphone

❑ If you wish to carry out several measurements for future evaluation it is essential to save both signals i.e. on the input and output of the object. For numerous measurements with unchanged set-up (e.g. several speakers in the same conditions) the input signal (reference measurement) need only be measured and saved once. Do not change any settings after carrying out the reference measurement!

3.1.11 Export

Frequency and phase curves can be saved for future use by other user programs, databanks, speaker CAD programs etc. Available formats are "AudioCad", "AkAbak®" and "CALSOD" (see section 9.2). The files can be read by any text editor.
Press F6 to activate the export function.
If you have set the upper or lower frequency ranges only the defined area will be exported.
The export or speakers frequency response (electrical-acoustical) will always be done in dBSPL/W/m.

3.1.12 Measurement with sinusoidal signals

Frequency response measurements can also be made with a special sweep signal. This sweep signal (PNSWEEP.WAV) contains a short periodic noise sequence at the begin of the signal for determining the distance by acoustic measurements. To use this sweep signal for frequency response measurement simply choose the test signal 'PNSWEEP.WAV'. All parameters e.g. FFT-length, number of averages, window and so on will be correct set automatically.
See section "Options for frequency response measurements" for more details.

3.2 Impedance measurement

Using the DAAS®32 it is possible to measure the impedance curve and the associated phase curve of a speaker or crossover. As with the frequency response

measurement this is carried out in two stages: one reference measurement and then the impedance measurements.
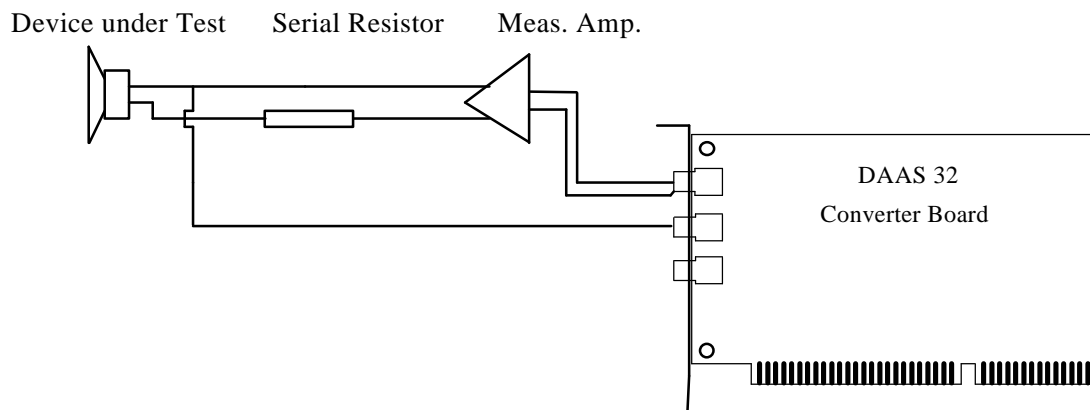
Open the "Measurements" menu and choose "Impedance Curve". It is also possible to start the impedance curve measurement by pressing Ctrl + I.

Depending on the Device under Test and the measuring condition there are two different measuring circuits used:

1. The first circuit uses a serial resistor in the range 100Ω - 4 kΩ. Set-up as follows:

Device under Test   Serial Resistor   Meas. Amp.

DAAS 32
Converter Board
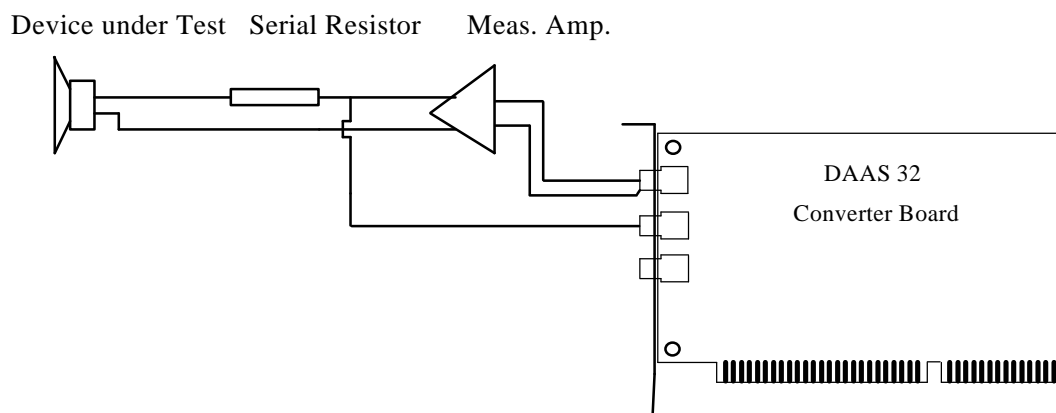
1. The second circuit uses a serial resistor in the range 0.1 Ω - 10 Ω. Set-up as follows:

Device under Test   Serial Resistor   Meas. Amp.

DAAS 32
Converter Board

Measuring normally speakers use circuit one in conjunction with a serial resistor in the range 150Ω.
The second circuit is useful to measure high power woofers under high current conditions. (use an amplifier with sufficient power capability)

If you are using a serial resistor greater than 600 Ω you can measure without using a measuring amplifier. In this case set the output attenuation "Out" to 0 dB.

Take care that the serial resistor used is identical with the corresponding entry in the status line. If not, change the entry using F6.

Set the measurement level using the volume control knob to the desired level. Start the reference measurement via F1. Or press F2 to load the last reference measurement. This is meaningful if you left the impedance measurement only for a short moment without changed anything. Or press F5 to load an earlier measured reference signal. This is advisable if you only want to load earlier measured signals.

The measured signal should be as follows:
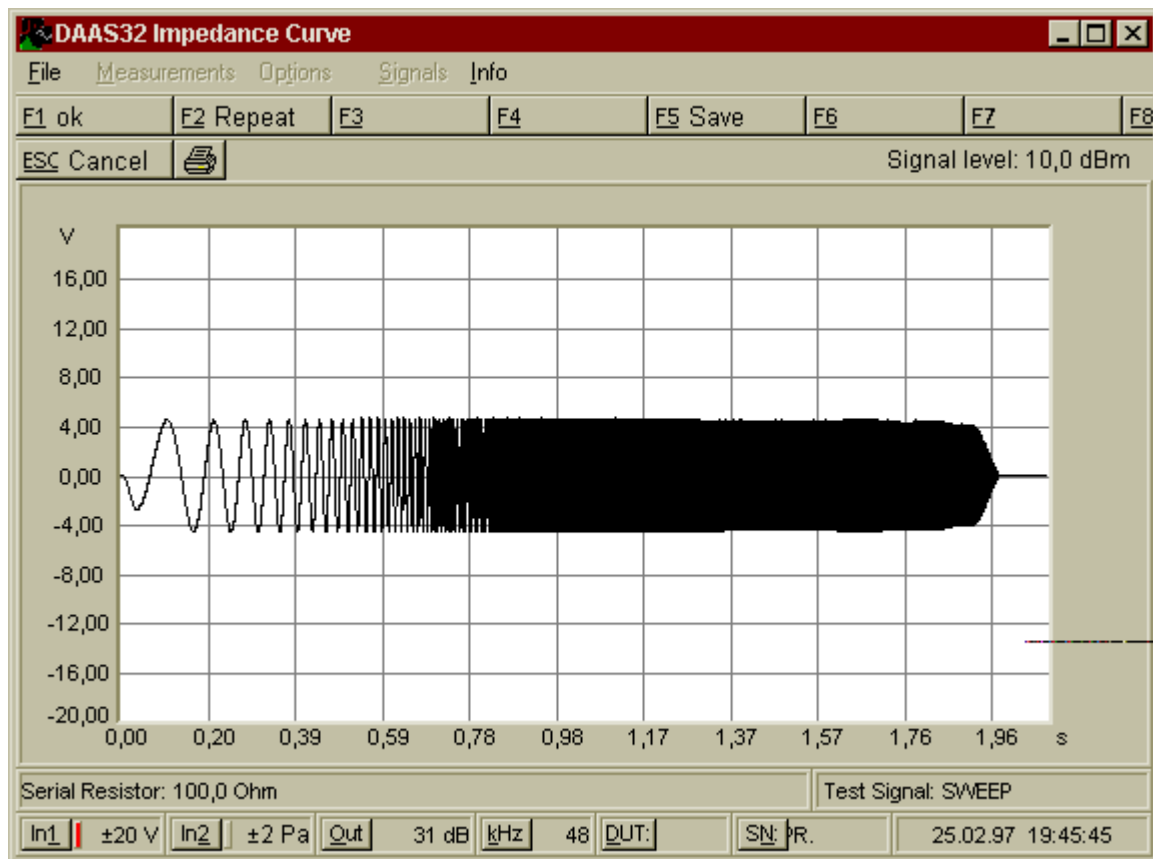
If something is wrong, check the connections or change the measurement level and repeat the reference measurement.
Press F5 to save this reference signal for future use.
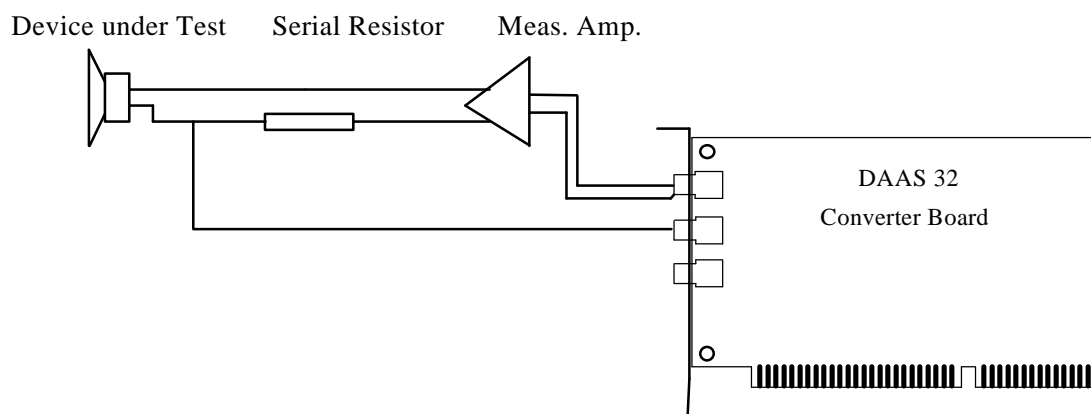If everything is OK press F1 to start the spectrum analysis.

CAUTION: Do not change the measurement level during the following measurements!

Now set-up the connections for the impedance measurement. The set-up depends on the Device under Test and the measuring condition like before.

1. Using the first circuit (serial resistor in the range 1Ω - 4 kΩ):

Device under Test   Serial Resistor   Meas. Amp.

DAAS 32

Converter Board
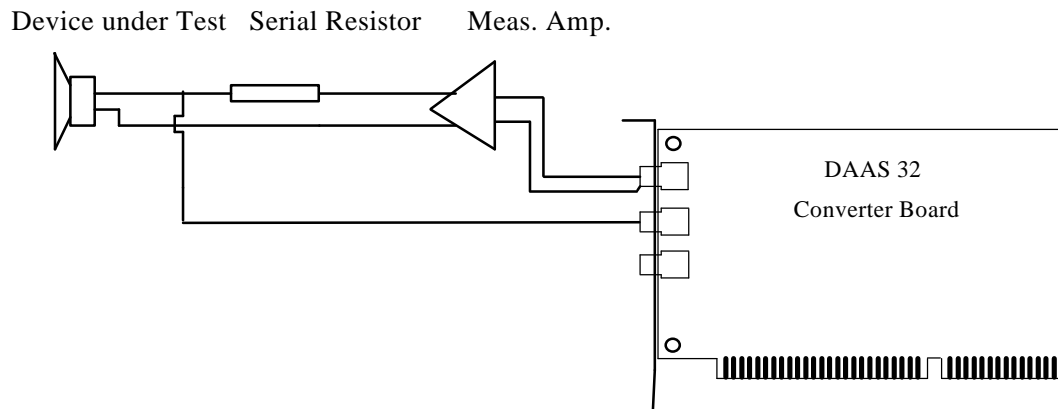
2. Using high current measurement set-up:

Device under Test   Serial Resistor   Meas. Amp.

DAAS 32

Converter Board

All connections should be very short and should have low impedance to avoid measuring errors. The cable to the input 'Line' should be connected close to the Device under Test.
Now start with the actual measurement.
Or press F5 to load an earlier measured signal. To load earlier measured signals you must consider the following aspects:
- Be sure that the serial resistor used by the signal to load is identical with the corresponding entry in the status line. If not, the results will be false.
- You should also load the reference signal (see before) to be sure that all conditions like signal level and so on are recovered.
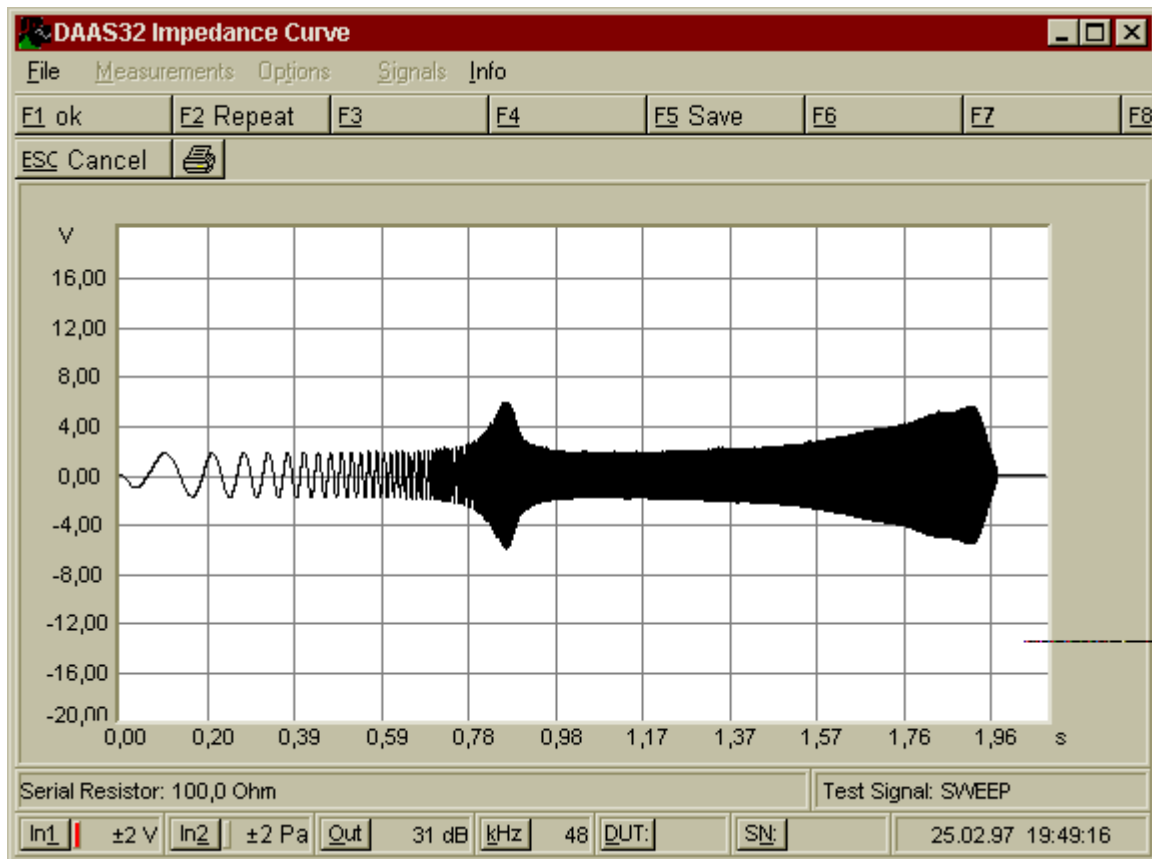
As a rule the input range is 2 V. Except by using serial resistors greater than 1000 Ω or measure very low impedance values the input range should be set to 0.2 V.

If the signal is of the right magnitude press F5 to save the signal if you want or press F1 to start the calculation and display the impedance curve.
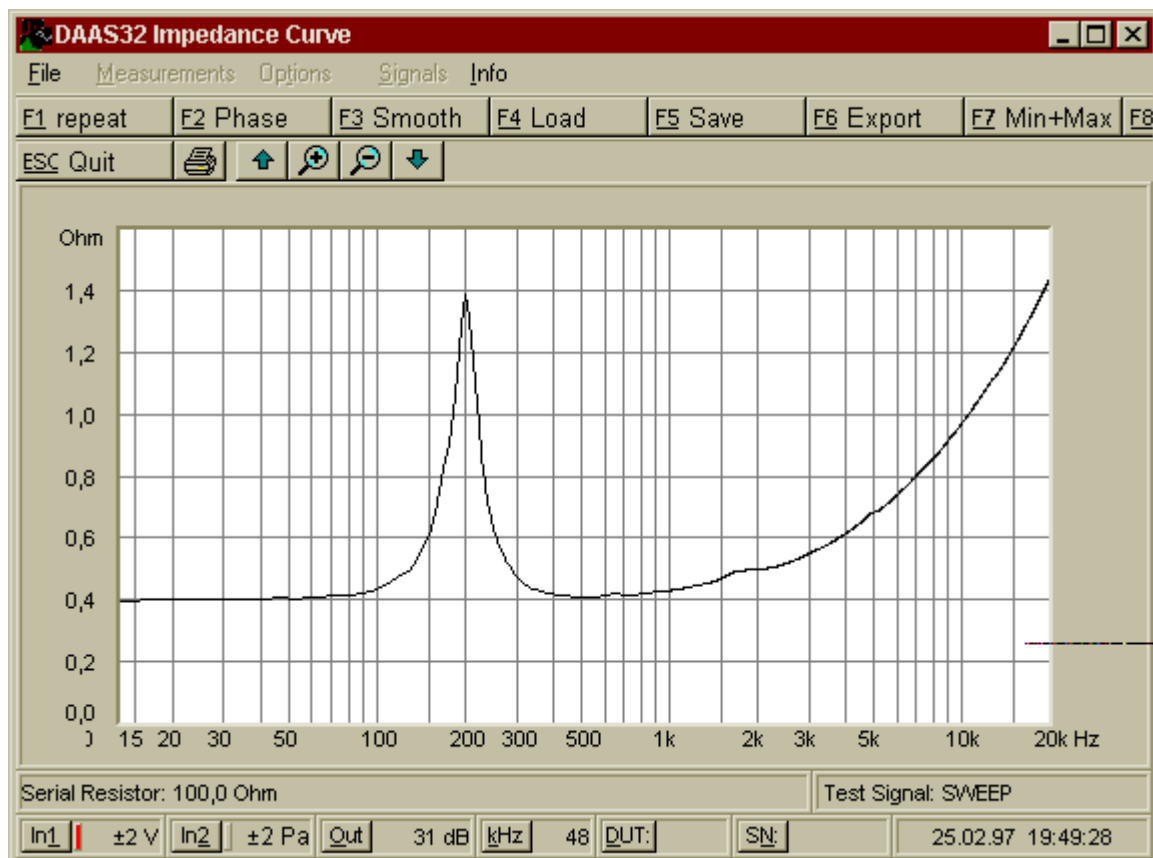
Using the 'Zoom In' and 'Zoom Out' keys the scaling can be changed from 0.2 Ω/Div to 300Ω/Div. Use UP and DOWN keys to move the display.
Press F2 to calculate and show the phase curve.
Move the mouse into the data window and press the right mouse button to read out impedance values.
You can smooth the impedance curve by pressing F3.
Using F4 and F5 the impedance curve can be loaded and saved. A curve load at this point will be shown ontop of the current curve i.e. the existing curve/s will not be deleted. To remove the loaded curves simply press the space bar.
A saved impedance curve can also be loaded into a frequency response diagram.

Using F1 you can repeat the measurement like described before. Using Ctrl + F1 you can repeat the measurement very quickly without choosing the input range etc. The via Ctrl + F1 measured impedance curve will be inserted into the existing diagram using a new color. To remove the old curves simply press the space bar.

CAUTION The default test signal is 'SWEEP.WAV'. Measuring speakers or other non linear objects it is strongly recommended to use 'SWEEP.WAV', because the impedance of these objects depends on the amplitude of the test signal (the

objects are non-linear). Using other test signals like Pink Noise or White Noise the impedance curve may come distorted or contains noise components.

The sine wave sweep signal 'SWEEP.WAV' ensures a constant magnitude during the whole measurement.

To use this sweep signal simply choose the test signal 'SWEEP.WAV' (see section "Options impedance measurement"). All parameters e.g. FFT-length, number of averages, window and so on will be correct set automatically.

3.2.1 Inductance and capacitance.

From an impedance- and phase curve capacitance and inductance of capacitors and coils can be calculated.

Press [F8] to start the calculation. First DAAS®32 looks for a capacitance. If the phase at the lower frequency range is lower than -60°, DAAS®32 calculates the capacitance from the complex resistance at the lowest visible frequency. The range for capacitance is approx. 1 µF to 100 µF. For larger values the sampling rate must be reduced.

If the phase at the lower frequency range is greater than -60°, DAAS®32 looks for an inductance. If the phase at the highest visible frequency is greater than 45°, DAAS®32 calculates the inductance from the complex resistance at the highest visible frequency. The range for inductance is approx. µH to about 10 mH.

3.2.2 Export

Impedance curves and phase curves can be saved for future use by other user programs, databanks, speaker CAD programs etc. Available formats are "AudioCad", "AkAbak®" and "CALSOD" (see section 9.2). The files can be read by any text editor.

Press F6 to activate the export function.

3.3 Thiele Small Parameters

The Thiele Small Parameters will be determined in DAAS$^{®}$32 from the complex impedance curve.

To start the Thiele Small Parameter measurement open the "Measurements" menu and choose " Thiele Small Parameter". It is also possible to start the Thiele Small Parametermeasurement by pressing Ctrl + T.

Now carry out a complete impedance curve measurement like described in section "Impedance measurement".

After successful measurement the DC voice coil resistance of the driver is requested. If the resistance in the lower frequency range is very even the value can be obtained from the diagram, otherwise the resistance must be measured using an Ohm-Meter. The various Q factors (Qms, Qes and Qts) plus the resonant frequency are then calculated.
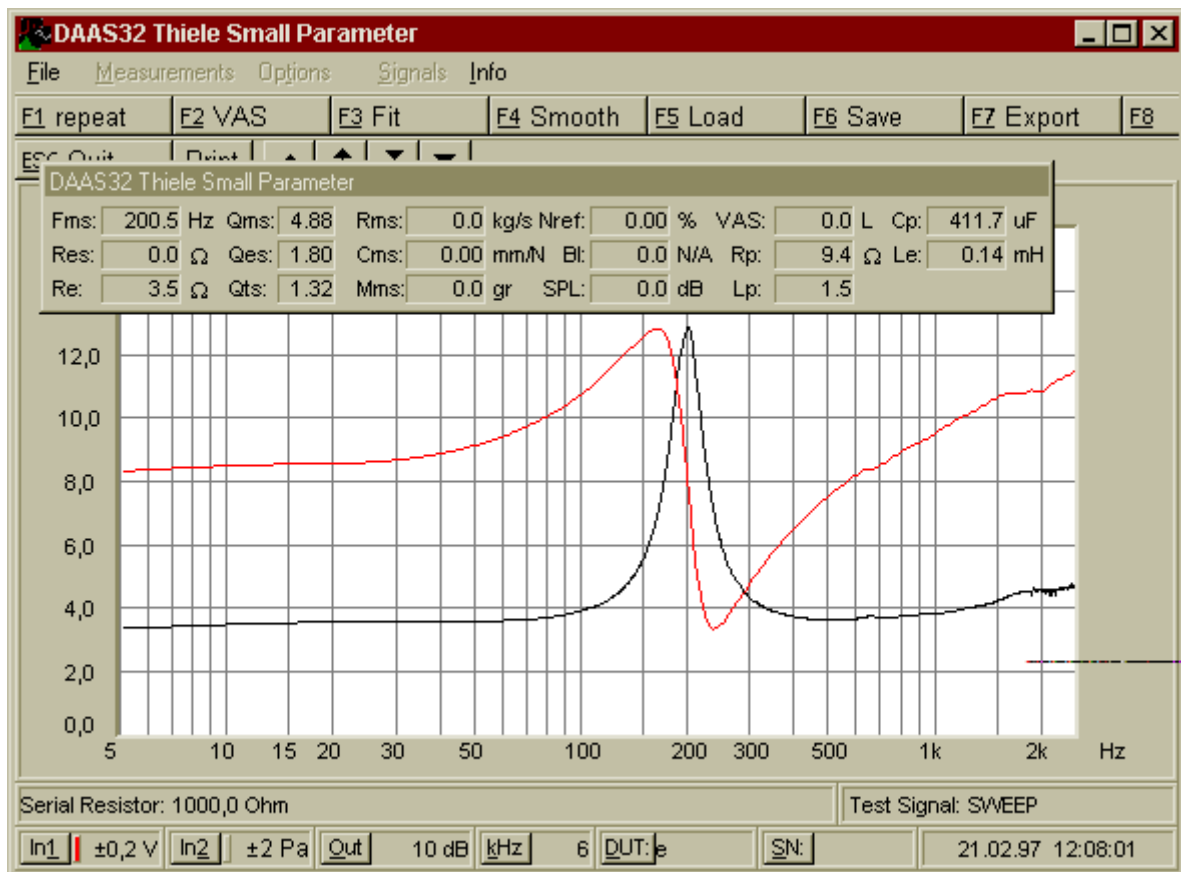
Use 'Load' and 'Save' like described in chapter "Impedance measurement". Use the 'Smooth' function cautious, because the impedance maximum peak will be reduced.

The key "F8 New Calc." lets you change the DC voice coil resistance for calculation. All results will be calculated again under consideration of the new DC voice coil resistance.

HINT Except for tweeters, it is recommended to reduce the sampling rate to 6 kHz (6 kHz sampling rate is the default value).

HINT Measuring woofers it is recommended to use serial resistors around 150 Ω. Before measuring, give the speaker a *„burn in"* time of app. 12 ... 24 hours with a sine - wave at e.g. 15 Hz; but do not overload the speaker!

During Thiele Small Parameter measurements the following warnings and error messages may appear:

• "Phase zero crosspoint / impedance maximum not found...".
Reason:
1. The maximum of the impedance curve is over 1000 Hz at 6 kHz sampling rate, or over 3000 Hz at 48 kHz .
2. The phase zero crosspoint is outside the acceptable range of the maximum impedance. (further than 2 * Resonance freq. / sampling rate * FFT-Length from the Resonant freq. (Impedance maximum)). This appears mainly when a speaker shows two resonances very close together.

• "The frequency resolution is too low. . . ".
Reason: The frequency of the impedance maximum is too low for the frequency resolution of the system. The impedance maximum may not detected accurate. Recommendation: Reduce the sampling rate or increase the FFT length for impedance measurements (see chapter "Options Impedance measurement").

• "Resonance frequency not equal', 'Square Root f1xf2...".
Reason: The resonant frequency is not in the geometrical center of the frequencies f_1 and f_2. Theoretical this should not appear, but it may however be the case.

3.3.1 Compliance Equivalent Volume VAS

To calculate the Compliance Equivalent Volume VAS and other characteristics such as power factor, Bl, SPL, Cms, Mms etc. the normal parameter set must first be obtained (see section 3.3).
Then press F2 = 'VAS' and the measurement is then repeated using a small weight on the speaker or build in the speaker on a closed box. Using a small additional weight, modeling clay is recommended. This should be formed in a ring and

pressed around the dust cover, e.g. 10 - 15 gram for small speakers and 30 gram for larger units. Weigh the modeling clay very accurately before fixing it to the membrane. Measure the speaker in an upright (normal installation) position. After measurement enter the speakers diameter in mm and the additional weight used or the volume of the closed box.

The key "F8 New Calc." can be used to change the previous entered entries. All results will then be calculated again under consideration of the entries.

3.3.2 Export

Thiele Small Parameters can also be saved for future use by other user programs, databanks, speaker CAD programs etc. Using the "AudioCad" or "AkAbak®" export format only the values will be written in a text file. Using the "CALSOD" format additional descriptions are applied e.g. "VAS: 600 L".
The files can be read by any text editor.

3.4 Oscilloscope

With the oscilloscope function, time signals up to 27 kHz and transient events up to a length of approx. 87 secs. can be displayed and printed. The chosen test signal is played synchronized with the oscilloscope.
Start the oscilloscope by pressing Ctrl + O or via opening the "Measurements" menu. Now the start screen appears, and you can set up the various parameters. Use F8 to select the test signal. If you don't want to play a test signal, press F8, then choose "None" or press ESC.
Adjust the measure time (number of points) with F6 and F7. The value will be shown in the status line.
Press F4 to set a trigger value.
To load a previous measured signal press F5.

Start the measurement either with F1 or F2. Start with F1 if you want to repeat the measurement until ESC is pressed. The ESC button may be clicked with the left mouse button, however if DAAS®32 is busy measuring, a slight delay may be experienced.
Start with [F2] if you want to make a single measurement.
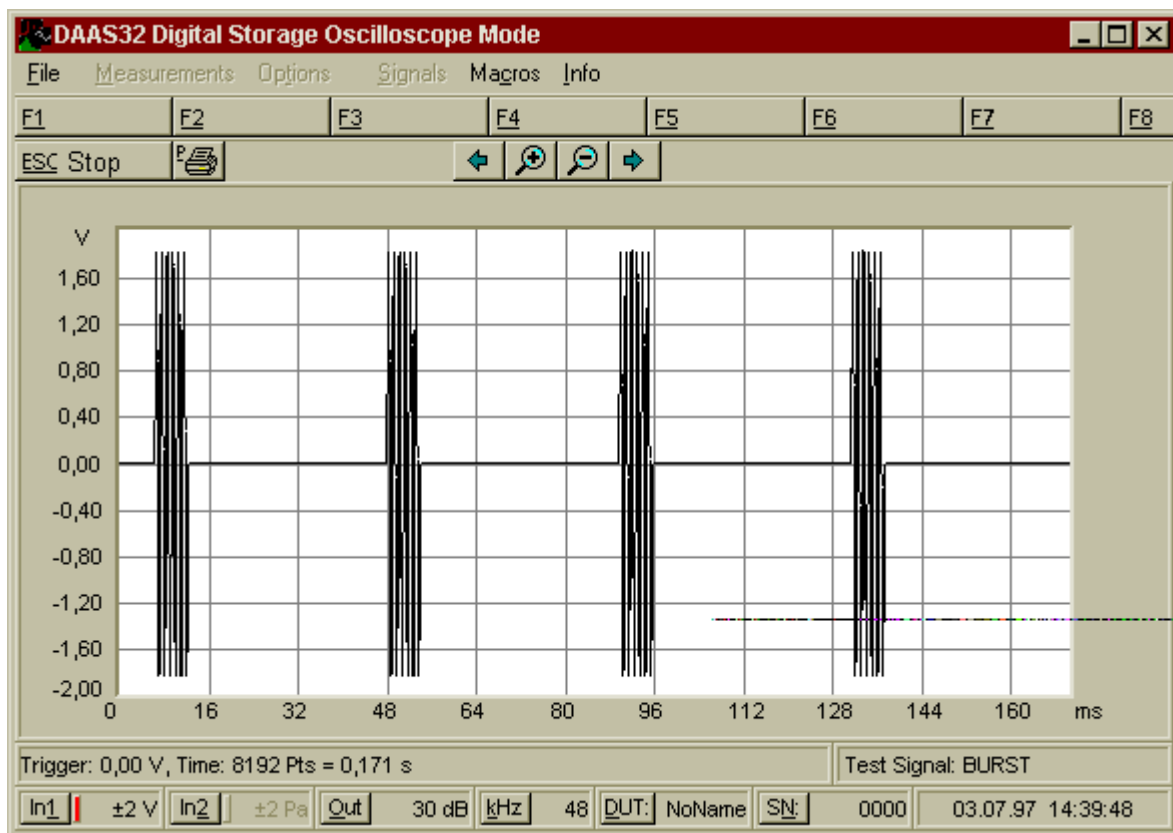
After a single measurement or when you have stopped the measurement using the
ESC key the system goes into the pause mode.
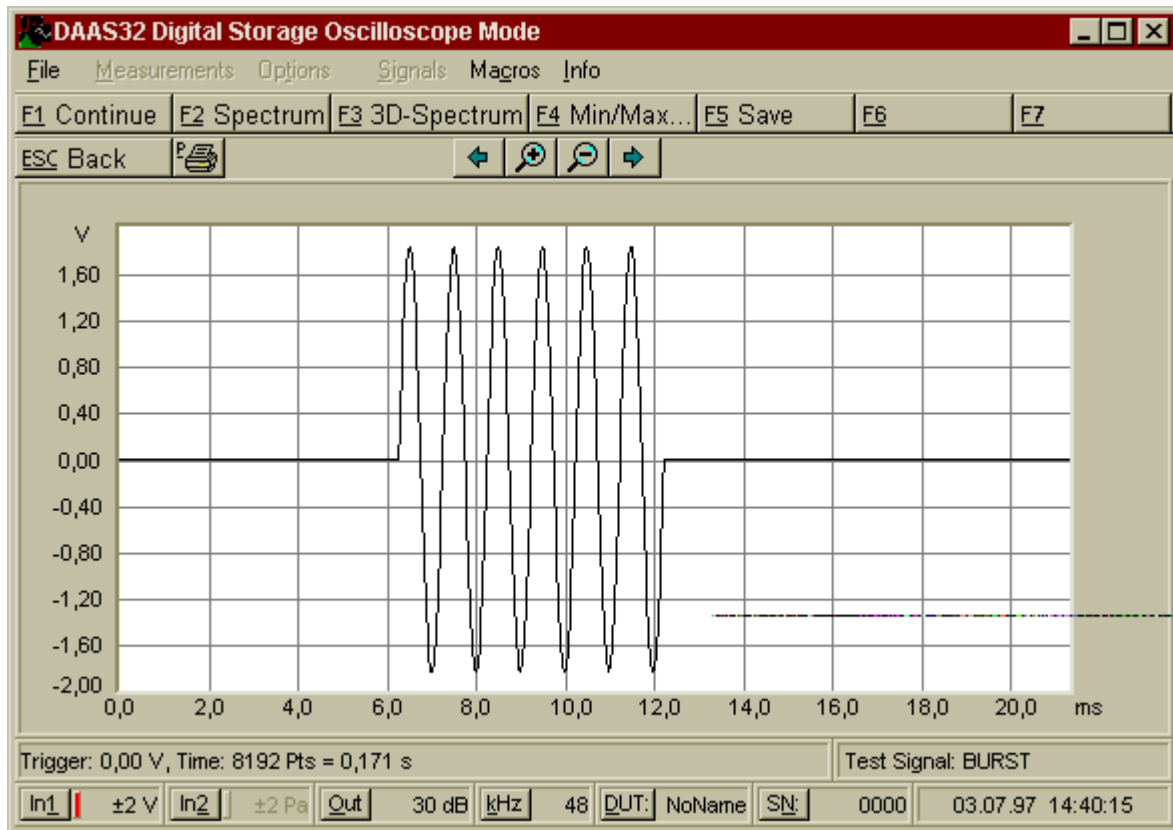
When the system is paused, use F1 to continue.
In pause mode the signal can be printed using the key P. Interesting signal
sections can be expanded and compressed using the Zoom keys and the left arrow
resp. right arrow keys. Supplementary, you can use the arrow keys in conjunction
with the shift key to fine adjust the visible section of the signal.
Move the mouse into the data window and press the right mouse button to read
out time and signal values.

3.4.1 Spectrum analysis

Using F2 a spectral analysis can be carried out on the shown section of the signal.
First you must enter the FFT length for calculation. Is the shown section of the
signal equal to the chosen FFT length, the signal will be analyzed with one FFT,
otherwise an averaged spectrum is calculated.
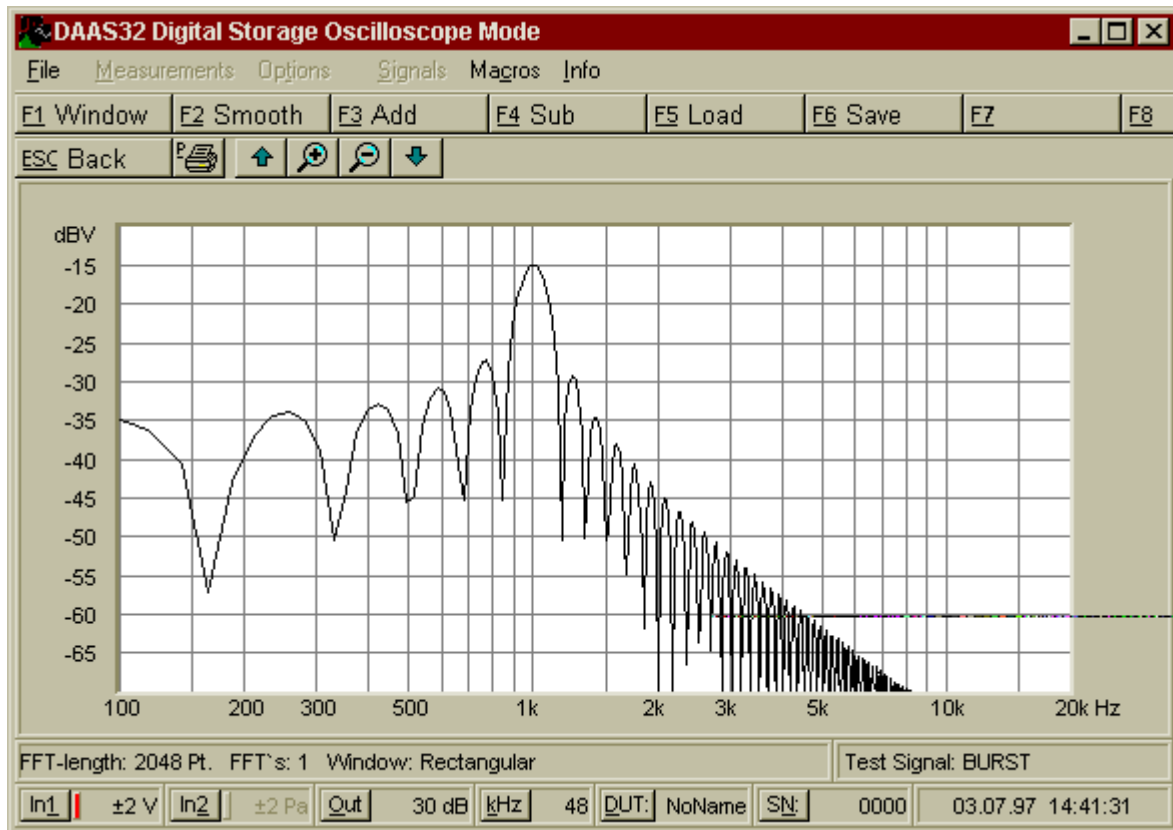
With F1 the weighting window is selected. Use F2 to smooth the curve. Use the Zoom keys and the arrow keys to adjust the display range.

Move the mouse into the data window and press the right mouse button to read out frequency and magnitude values.

Curves can also be loaded and saved. Note that for add and subtract curves all parameter such like FFT length and sampling rate must be identical.

For further explanations refer section 3.5.

3.4.2 3D-Spectrum

To analyze the spectrum in dependence on the time you can calculate a 3D-waterfall-spectrum. This means the shown section of the signal will be split into signal parts with a length equal to the FFT length. Each signal part then will be analyzed and its spectrum shown in a waterfall diagram.

In the oscilloscopes pause mode press F3, then enter the desired FFT length to get the 3D-waterfall diagram.
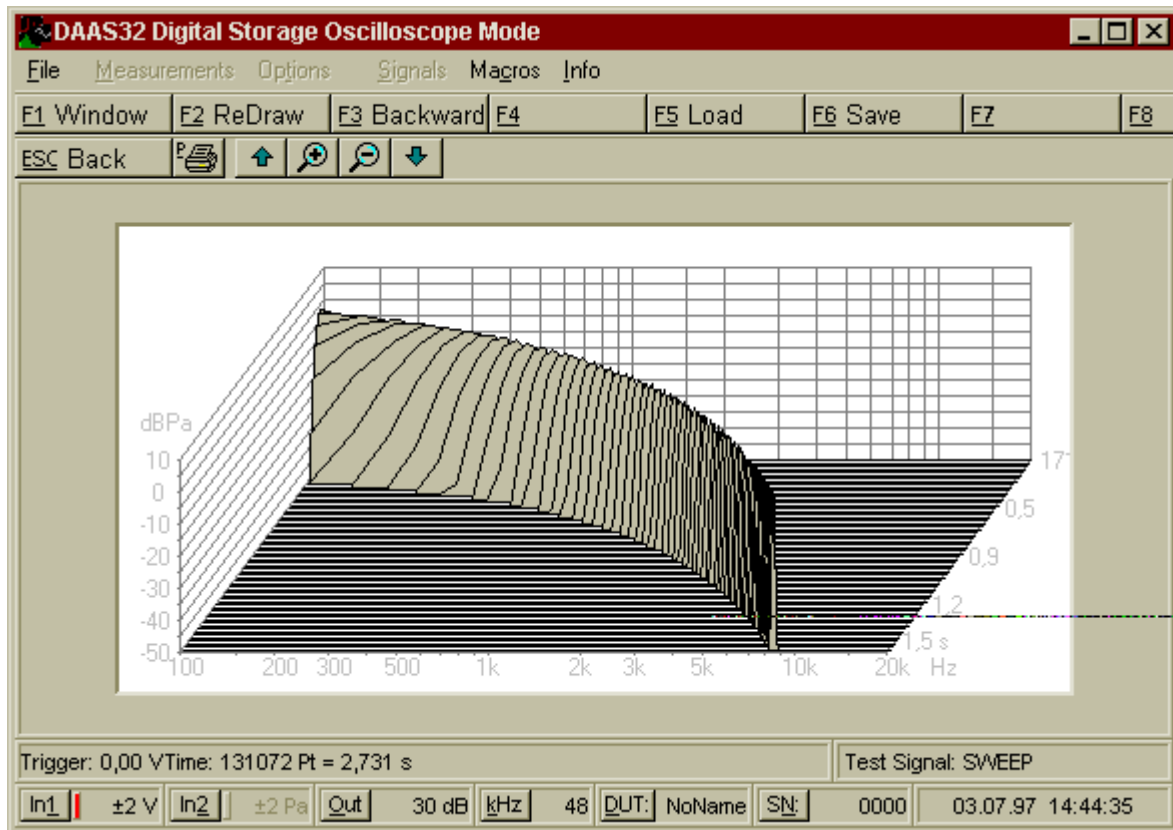
The waterfall diagram can also be rotated. Use "F3 Backward" to get the rotated waterfall diagram.

Using F5 and F6 the waterfall diagram can be saved and loaded. When a waterfall diagram is loaded the current screen is deleted. The current screen will be redrawn by pressing F2.

3.5 Spectrum analyzer

With the spectrum analyzer, time signal spectrums in the audio range up to 28 kHz and transient events (up to a length of 65536 points) can be displayed and printed. The chosen test signal is played out synchronized with the measurement cycle. By using a special test signal and special set-up, quick frequency response measurements can be carried out without time delay problems and without a reference measurement.

Start the spectrum analyzer by pressing Ctrl + or via opening the "Measurements" menu. Now the start screen appears, and you can set-up the input and output options and the sampling rate. Press F7 to set-up the analysis parameters.
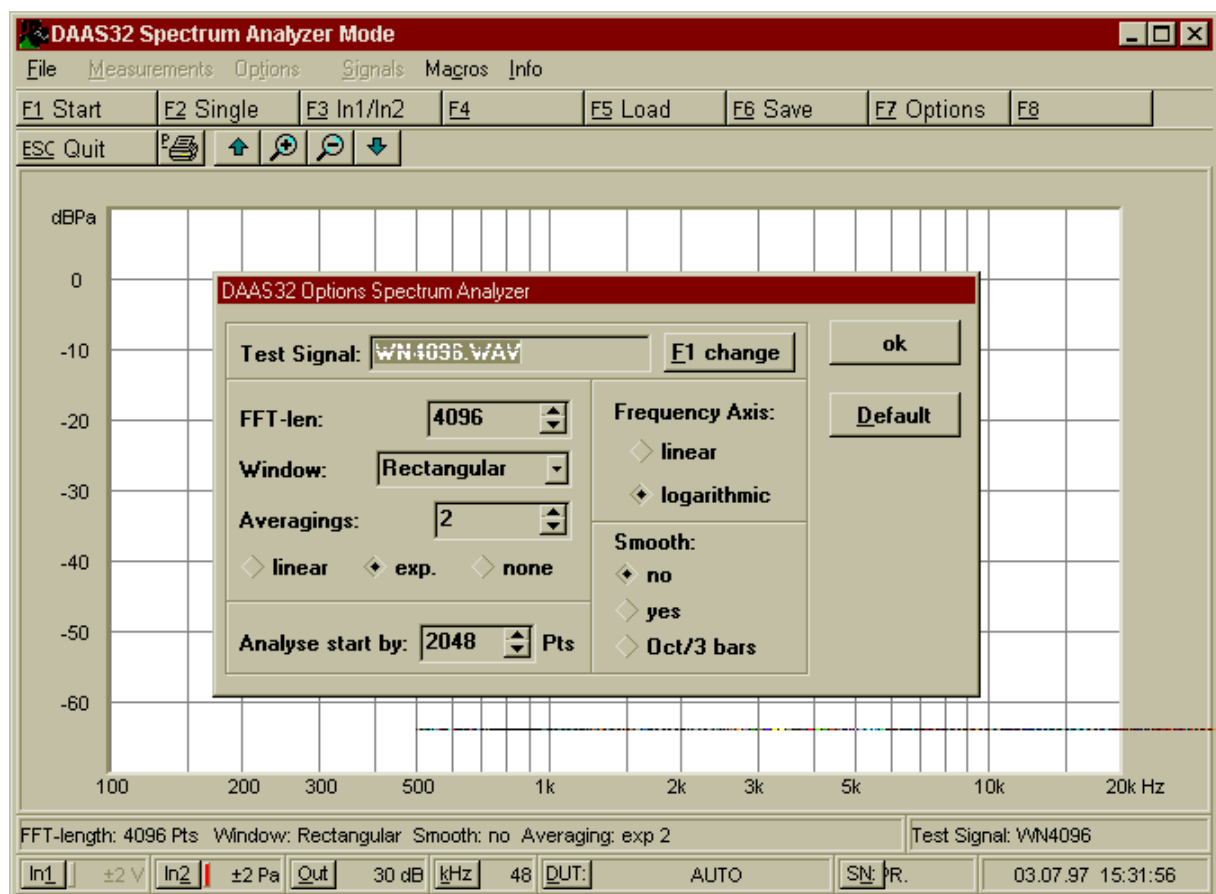
Choose the test signal using F1. The FFT length (number of points) can be changed in the range 512 points to 65536 points. The FFT length in conjunction with the sampling rate will designate the frequency resolution:

frequency resolution = sampling rate / FFT length.

The following weighting windows are available:

- Rectangular (for transients),
- Hann (for general use),
- Blackman,
- Kaiser (for harmonic analysis, Side-Lobe-attenuation approx. 100 dB),
- Flat Top (for exact amplitude setting).

The default window is the 'Rectangular' window.

The smoothing option is only useful in conjunction with noise signals (see 3.5.1). Set the smoothing option to "no" if you are analyzing harmonic signals! Note that the "Terz"-smoothing is not really a smoothing, the result is an Octave/3-spectrum.

Select the option "averaging: none" to make single measurements or to make continuous measurements without averaging. Select the option "averaging: linear" to make the desired number of measurements. Each spectrum will be averaged using the same weighting factor. If the adjusted number of averagings are reached, the system goes into the pause mode. Select the option "averaging: exp." to make continuous measurements with an exponential averaging under consideration of the adjusted number of averagings.

The option "Analyse start by" determines the waiting time between test signal start and analysis start. To set this value greater than zero is meaningful by acoustical measurements to prevent errors due to acoustic delay or to prevent errors caused by the transient period. Default: 2048 points = 43 ms (using 48000 Hz sampling rate).

To get the default parameter press the "Default" button.

After close the option window you can start the measurement either with F1 or F2. Start with F1 if you want to repeat the measurement until ESC is pressed or the number of averages are achieved (linear averagings only). The ESC button may be clicked with the left mouse button, however if DAAS®32 is busy calculating, a slight delay may be experienced. Start with F2 if you want to make a single measurement.
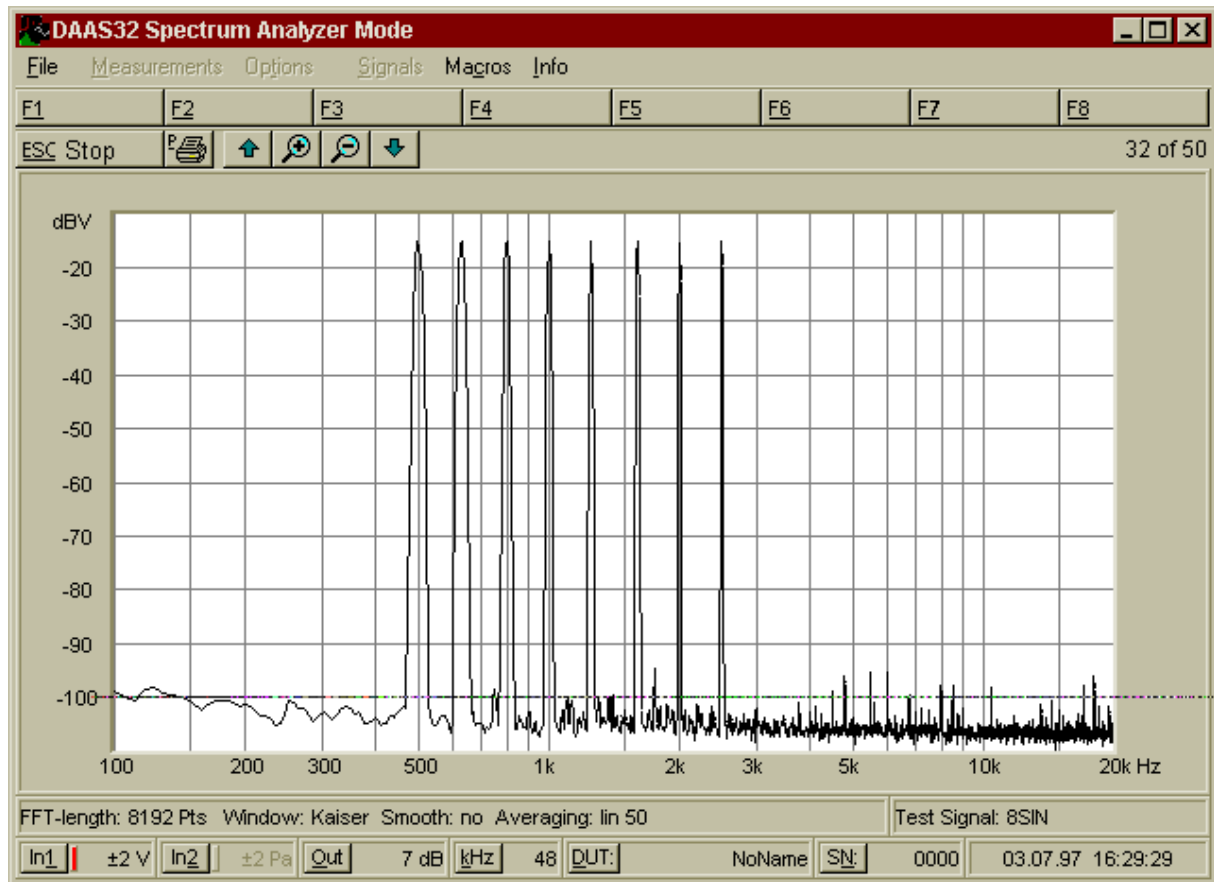
If the system is in pause mode, you can adjust the visible screen range or read out frequency and magnitude values like described before.

3.5.1 Frequency response measurement 'virtually' in real time

The DAAS®32 software package contains a number of special test signals which were created with help from the DAAS®32 FFT and therefore contain all the properties to fit exactly in the analysis. These signals 'WNxxxx.WAV' are particularly suitable for the measurement of frequency response curves in the spectrum analyzer.

The spectrum comprises (when used the square window and a FFT length tuned to the signal, for example 'WN8192.WAV' in conjunction with a FFT length of 8192 points) a line without the statistical variations which normally occur when measuring a noise signal:
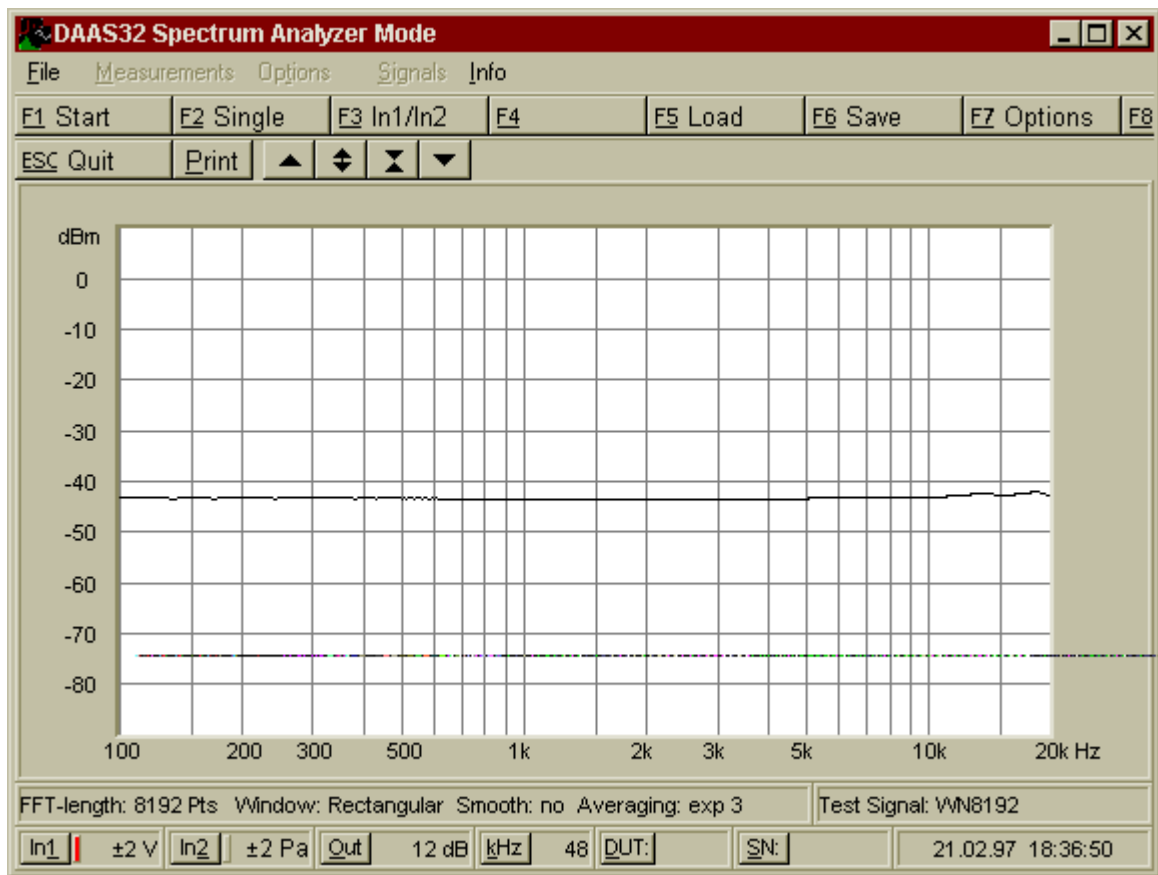
Due to the measurements being continuous, this method is particularly suitable for, for example, fine tuning of crossovers and speaker enclosures. The following sketch shows a typical set-up.

Set the amp volume control or the output attenuation to a level well above the surrounding noise level. Select the acoustical input and position the microphone. After starting the measurement, set the amp volume level so that the overload display does not quite appear.
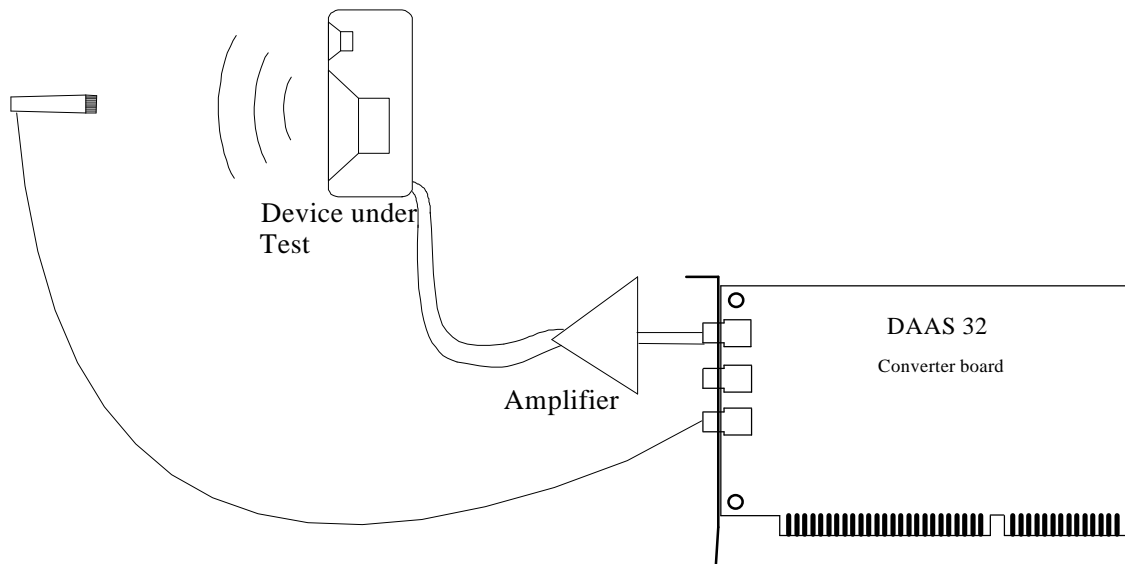
The spectrum now shown is directly the frequency response curve of the object (speaker). Note that the curve shown is a relative curve, the absolute values (e.g. 2 Pa/V) cannot be read directly from here.
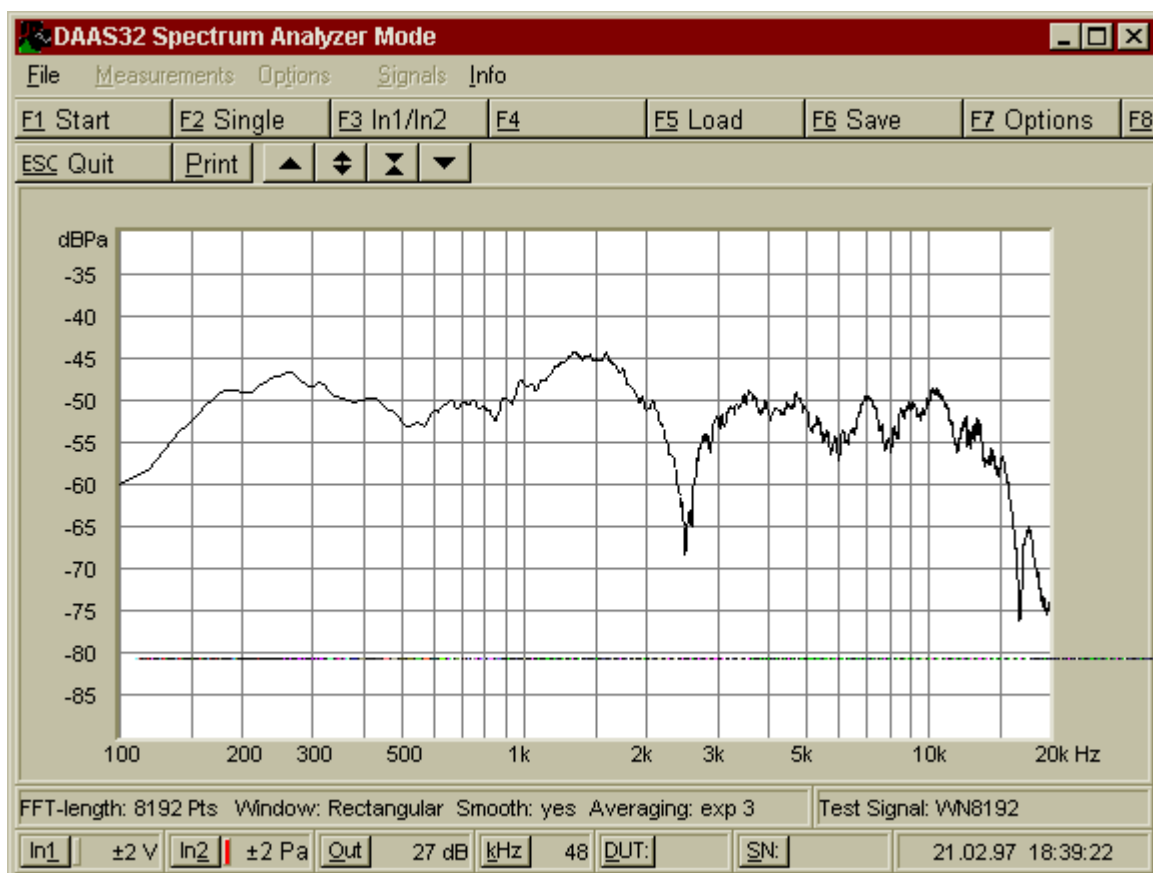
For this frequency response measurement the smoothing option mentioned above is very useful. Set the smoothing option to 'yes' to get a soft curve. If you want to

get the frequency response curve as octave/3 bars, you must change the options like follows:

- Set smoothing option to "Oct./3 bars", and
- set the test signal to the appropriate "PNxxxx.WAV" instead of "WNxxxx.WAV". This is important!

This frequency response measurement with the spectrum analyzer can also be carried out with reduced sampling rate so the higher frequency resolution (measurement range 12 Hz to 2.5 kHz) can also be used.

3.6 Distortion

Distortion, or more accurately non linear distortion, are changes to a signal due to amplitude related amplification or attenuation during passage through an amplifier or speaker. The transfer gain characteristic of such a signal is not straight but curved. A difference is made primarily between quadratic and cubic gain characteristics:

Output   **Output**   **Output**

Input   **Input**   **Input**

linear characteristic   **quadratic characteristic**   **cubic characteristic**
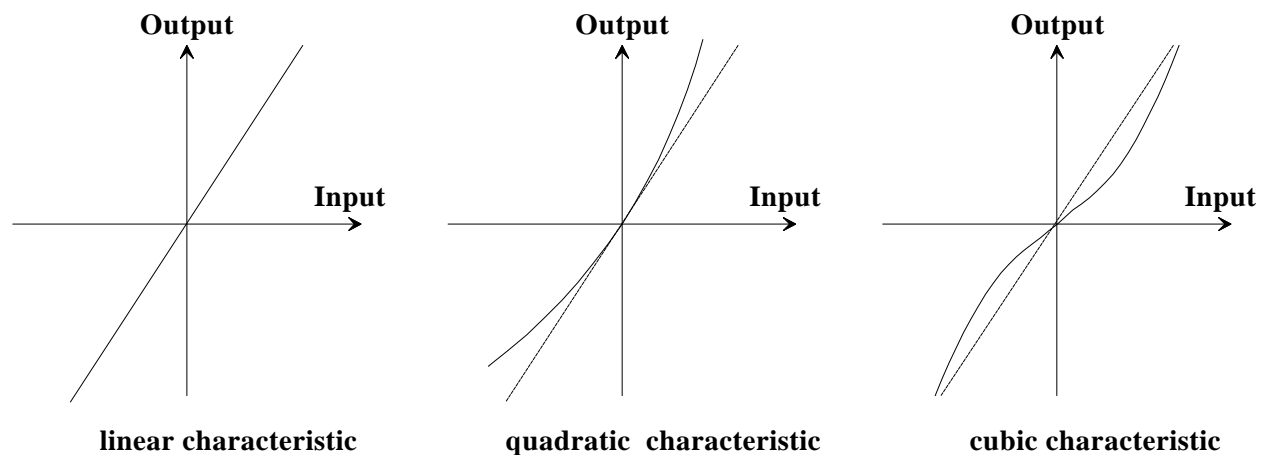
Generally gain characteristics are made up of elements of linear, quadratic and cubic characteristics.
When a sine wave of frequency f is distorted, overtones, or harmonics, are created which were not present in the original signal. Quadratic characteristics create only even (2*f, 4*f etc.) and cubic characteristics only odd harmonics (3*f, 5*f etc.). Linear gain characteristics create no distortion.

Non linear distortion will be usually measured by one of three methods: using the harmonic distortion, the intermodulation factor or the difference tone factor.

3.6.1 Harmonic distortion

Harmonic Distortion is the most common method of measuring non linear distortion. It calculates the strength of the harmonics created by distorting a sine wave signal, compared with the total signal. Harmonic distortion measurements are useful in the lower and middle frequency up to approx. 12 kHz. Above 9 kHz the third harmonic (and the most particularly damaging harmonic) lies outside the audio frequency range.

The DAAS®32 has a sine wave generator of variable frequency in the range 20 Hz to 12 kHz for the measurement of harmonic distortion. The set-up for measuring a speaker is as follows:
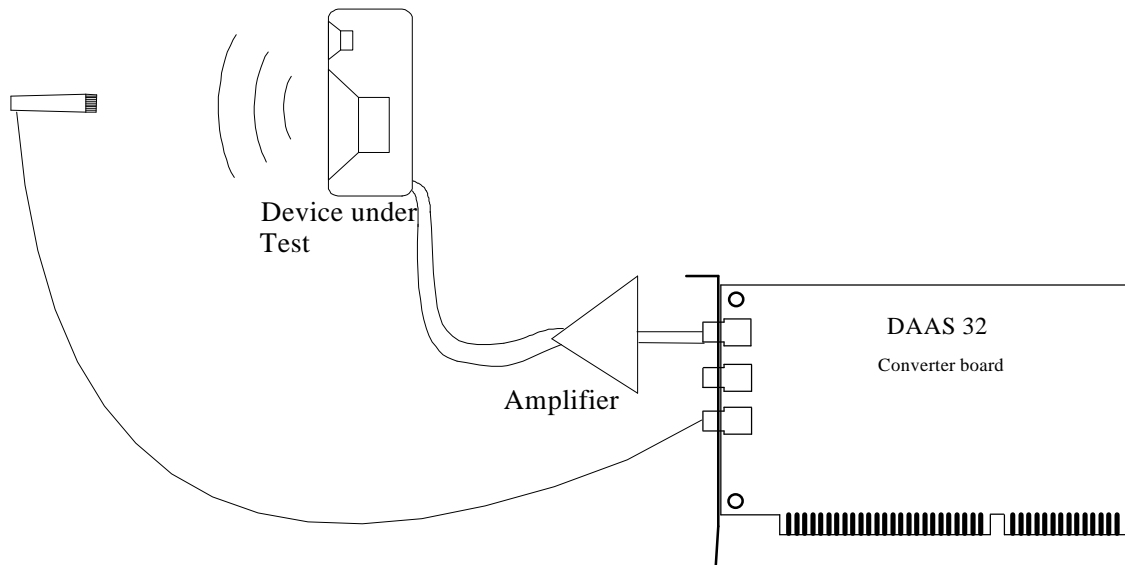
Position the microphone to the desired location and click or press the "In 2" button to select the acoustical input. Click or press the "Out" button and set the output attenuation to around 30-35 dB (the amplifier must not be over driven.....clipping!). Click "Test level" to check the sound level. If the sound level is too high or too low this can be adjusted using the amplifiers volume control knob or the output attenuation.

If everything is OK press F1 to start the measurement. After the measurement (short sine tone) the measured signal is displayed. Change the output level and re-measure if required. Note that the input sensitivity range should be set so that the input level range occupies approx. 1/3 - 1/2 of the display. An ideal setting is as follows:
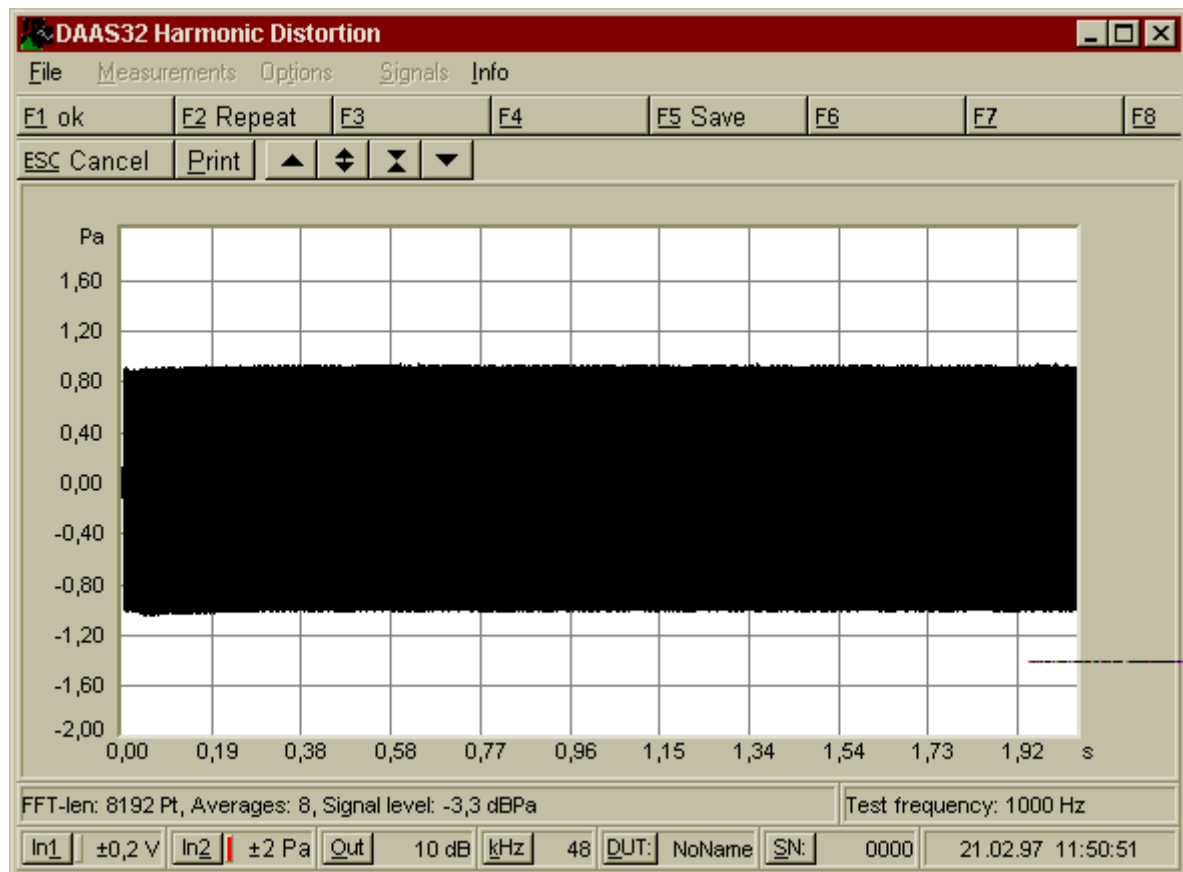
If the signal is too large a warning appears, if too low accuracy in the lower range will be lost.
Use F5 to save the signal at this point.
Press F1 for 'measurement OK', and the harmonic distortion will be calculated.
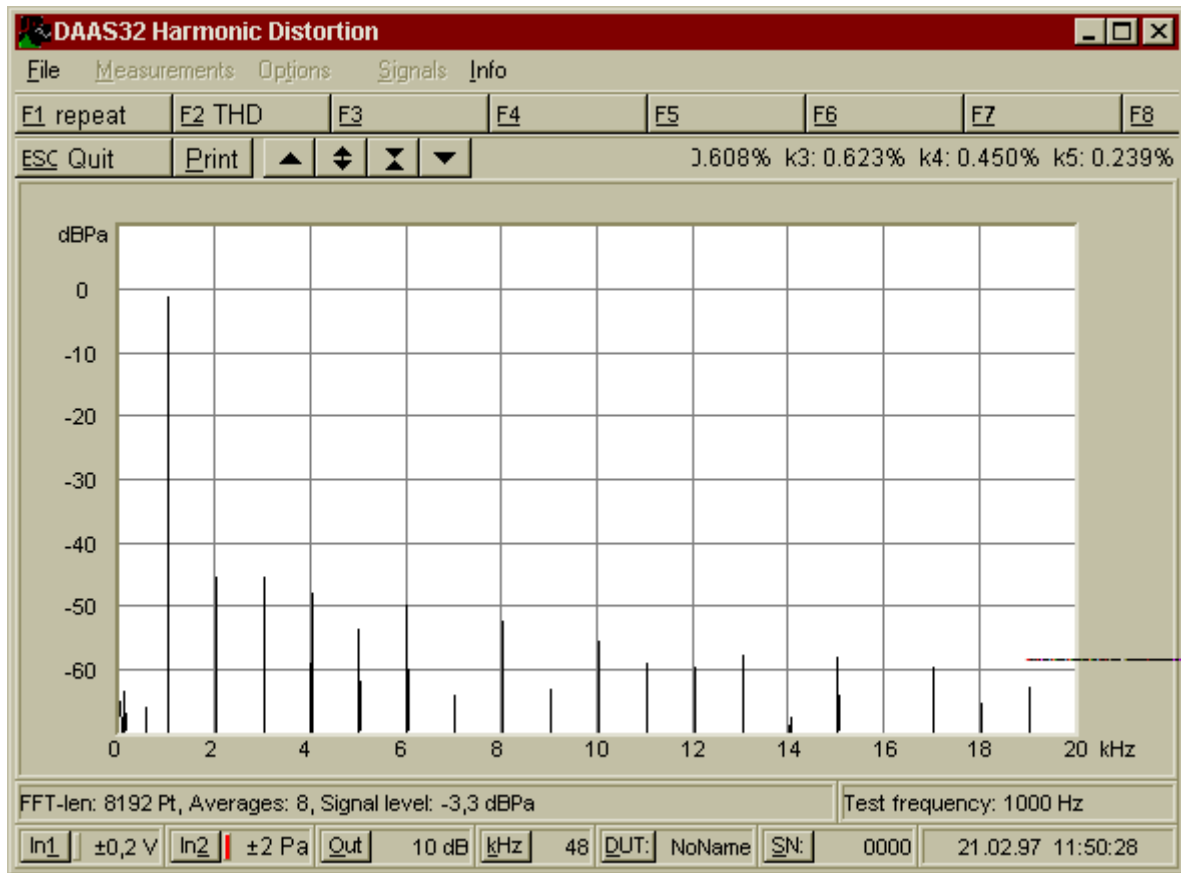
The following important factors will be displayed:-

- k2: 2nd Harmonic distortion.
- k3: 3rd Harmonic distortion
- k4: 4th Harmonic distortion
- k5: 5th Harmonic distortion

K2 is relatively non-critical. Values up to a few percent are not considered serious. K3 is however very critical and values around 0.1% can be noticed. Sum harmonic distortion is calculated as follows:

$$k_{total} = \sqrt{k_2{}^2 + k_3{}^2 + k_4{}^2 \ldots .}\ .$$

If you need more harmonic components, press F2 to get k6 - k9 and THD (Total Harmonic Distortion).

IMPORTANT: If you want to use external test signals for the harmonic distortion measurement e.g. from a sine wave generator it is necessary to set the measuring frequency using F4 to zero! Using external signals only k2 - k5 are available.

3.6.2 Harmonic distortion curve vs. frequency

Often, just using one frequency for determining harmonic distortion is not enough. DAAS®32 enables up to 259 points to be used in the range 20 Hz to 12 kHz to establish a distortion curve.

The method used for measuring a distortion curve is very similar to that for a single measurement.

Use "F4 Test frequency" to set the frequency limits as opposed to the measurement frequency and the number of points.

On starting the measurement the following screen is shown:
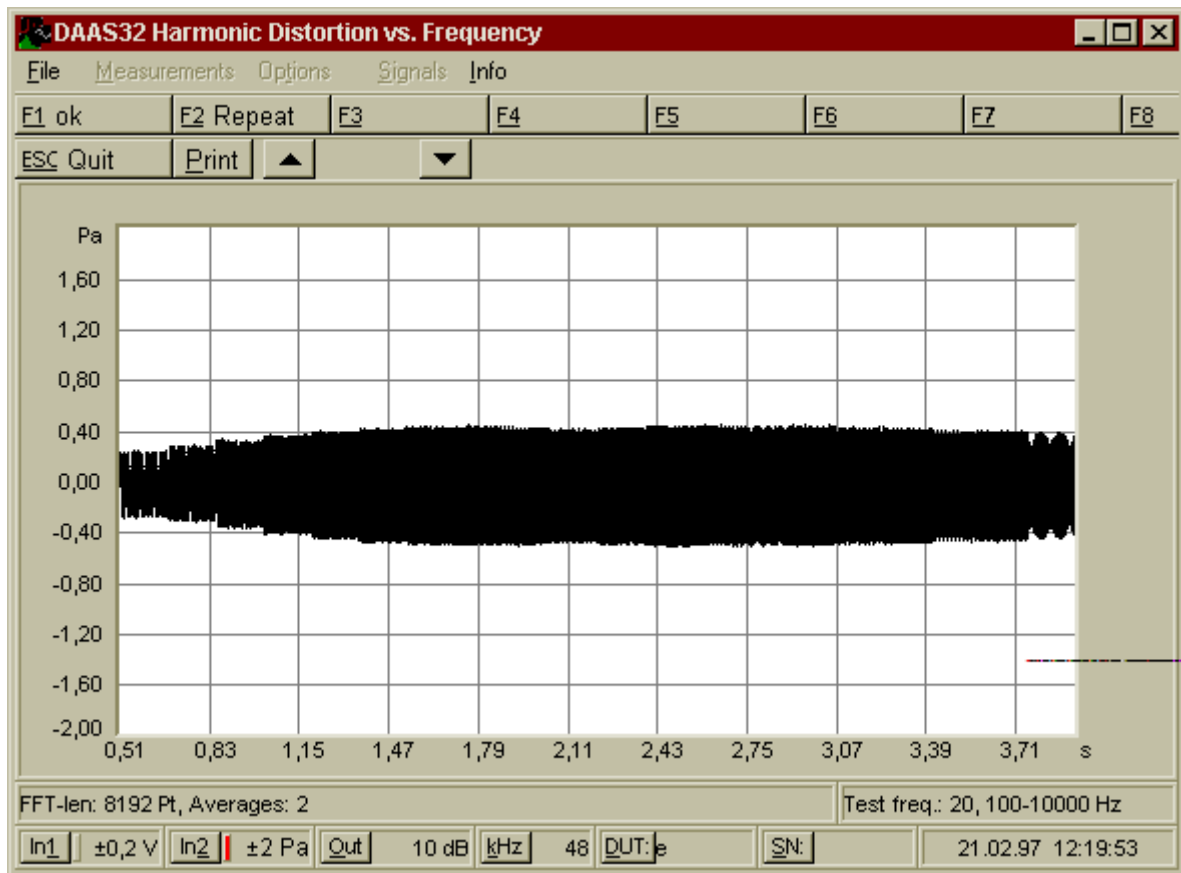
The signal tones are combined to form a single picture. Left are the low frequencies and right the highest. Press F1 to calculate the harmonic distortion curve:
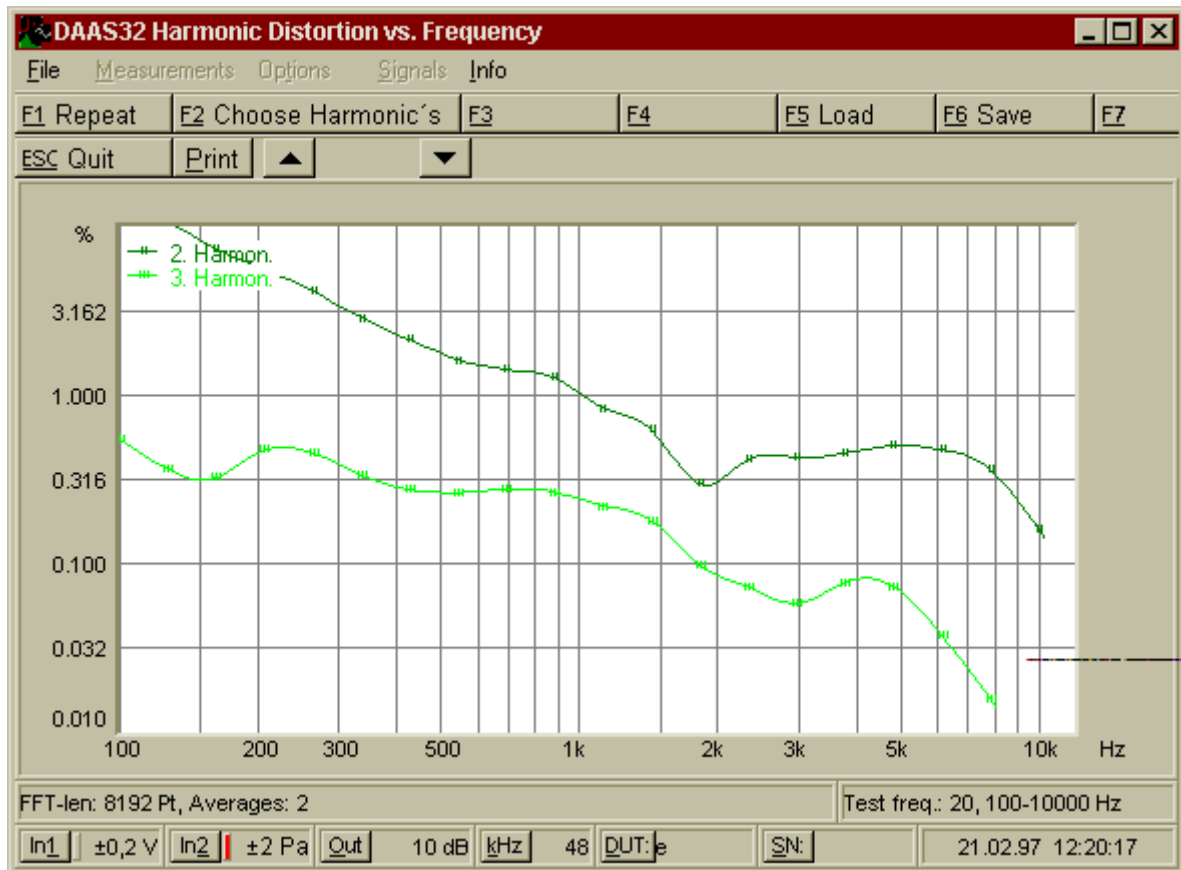
Press or click "F2 Choose Harmonics" to select the visible harmonic components.
Use F5 and F6 to save and load distortion curves.

3.6.3 Intermodulation factor measurements

To determine the quality of a transfer system a two tone process like the
intermodulation factor measurement is a proper instrument, because real signals
e.g. speech or music contains a lot of various frequencies at the same time.
The handling of the intermodulation factor measurement is identical to distortion
measurement.
The DAAS®32 provides a composite signal for the intermodulation factor
measurement.
The signal comprises two tones with an amplitude relation of 4:1, 2:1 or 1:1. The
tone with the lower amplitude can be selected within the range 800 Hz to 8000
Hz. The second tone can be selected within the range 100 Hz to 0.4 x the
frequency of the tone with the higher frequency.
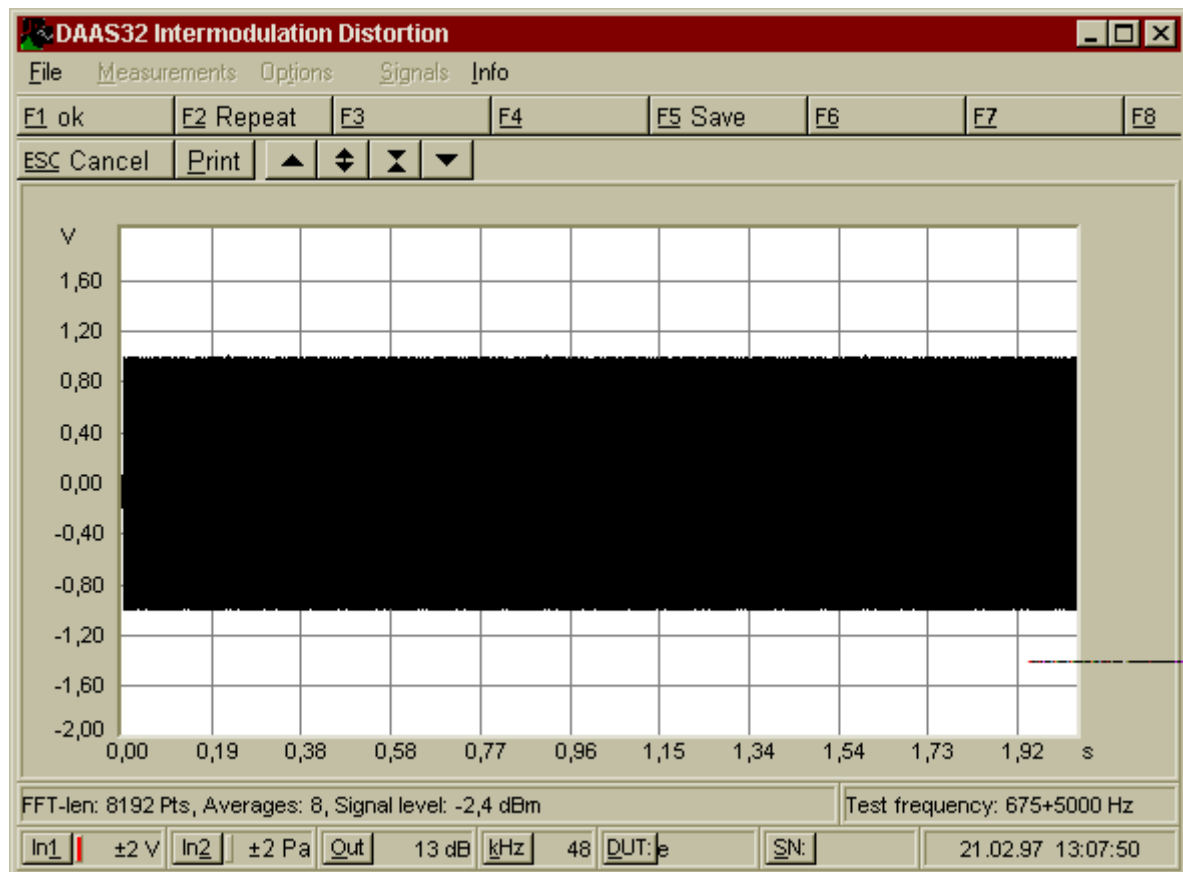
Pay attention that the tone with the higher frequency does not coincide with any harmonic wave of the tone with the lower frequency!
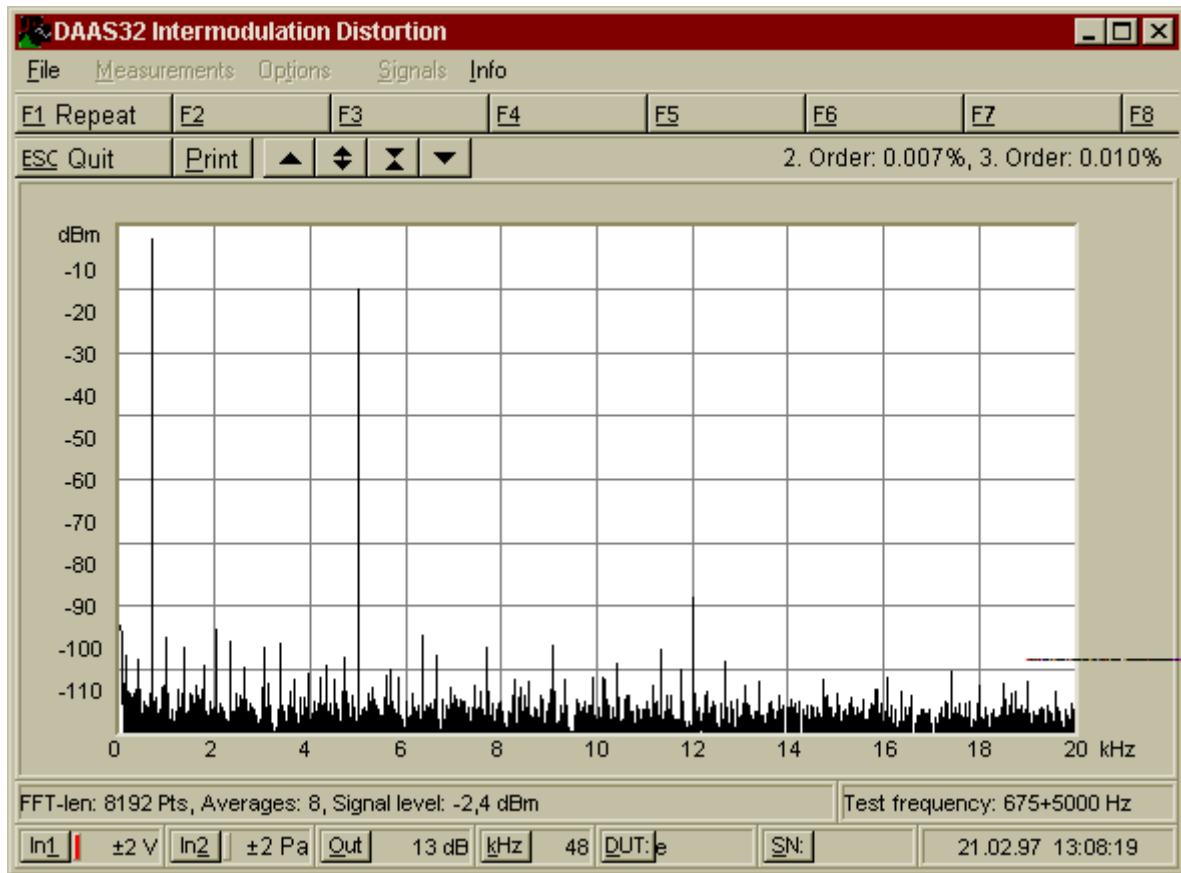
The second and third order distortion components are found at

$$m_2 = [S(f_2+f_1) + S(f_2-f_1)] / S(f_2) \qquad \text{second order}$$

$$m_3 = [S(f_2+2f_1) + S(f_2-2f_1)] / S(f_2) \qquad \text{third order.}$$

This evaluation is carried out as recommended in DIN 45403.4.

Using other amplitude relations than 1:1 causes the lower measuring limit in comparison with distortion measurement is slightly higher. This is because the reference signal magnitude U is about 6 to 12 dB lower.

The use of external test signals for the intermodulation factor measurement is not possible.

3.6.4 Difference tone factor

The difference tone factor measurement is also a two tone procedure using two sine waves with frequencies close together. It is a proper instrument to determine the quality of transfer systems in the upper frequency range. The harmonic

distortion measurement is not qualified here because all harmonics lies outside the audio frequency range.

The handling of the difference tone factor measurement is identical to distortion measurement and intermodulation measurement.

The DAAS®32 provides a composite signal for the difference tone factor measurement.

The signal comprises two tones in the frequency range 5000 Hz to 15900 Hz with a difference in the range 70 Hz to 1000 Hz.
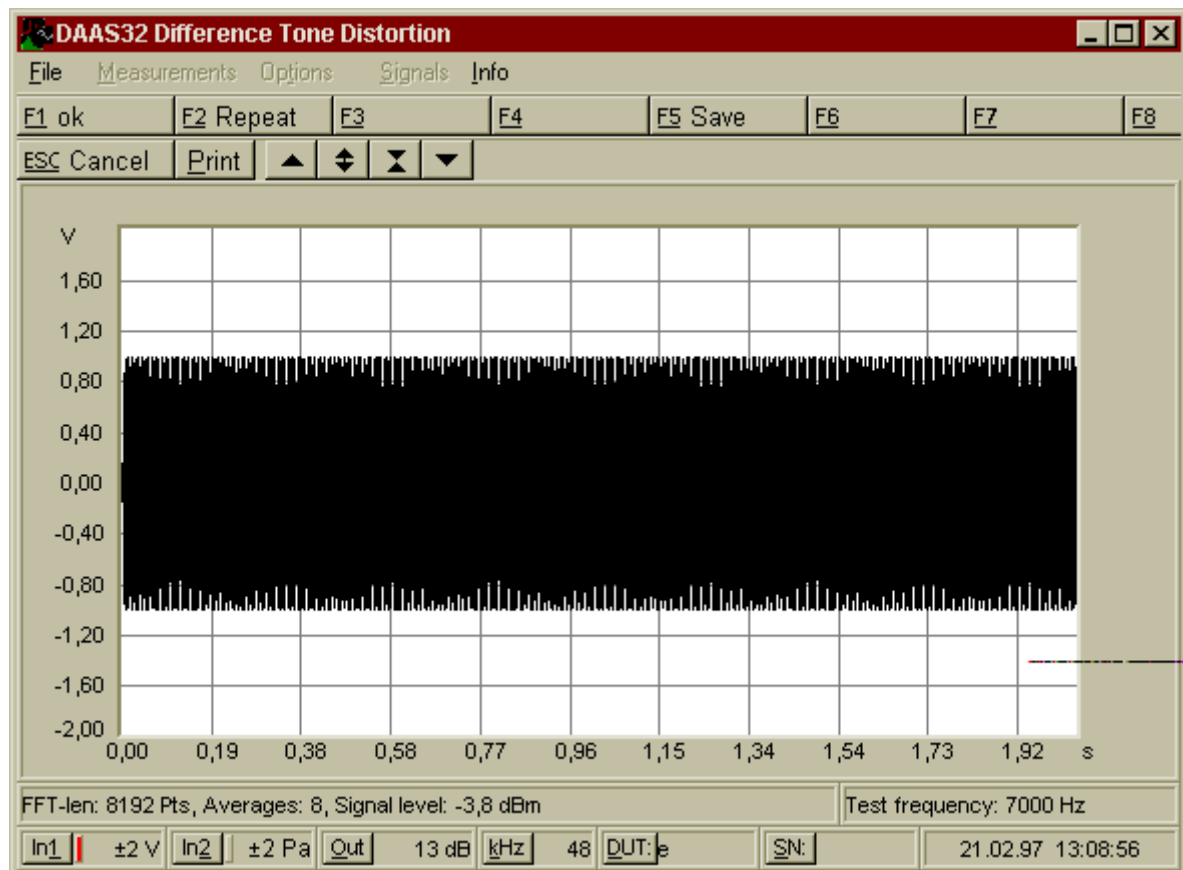
The distortion components are found at

    (f1-f2)   second order

    (2xf1-f2) + (2xf2-f1)   third order.

The evaluation is carried out as recommended in DIN 45403.3.
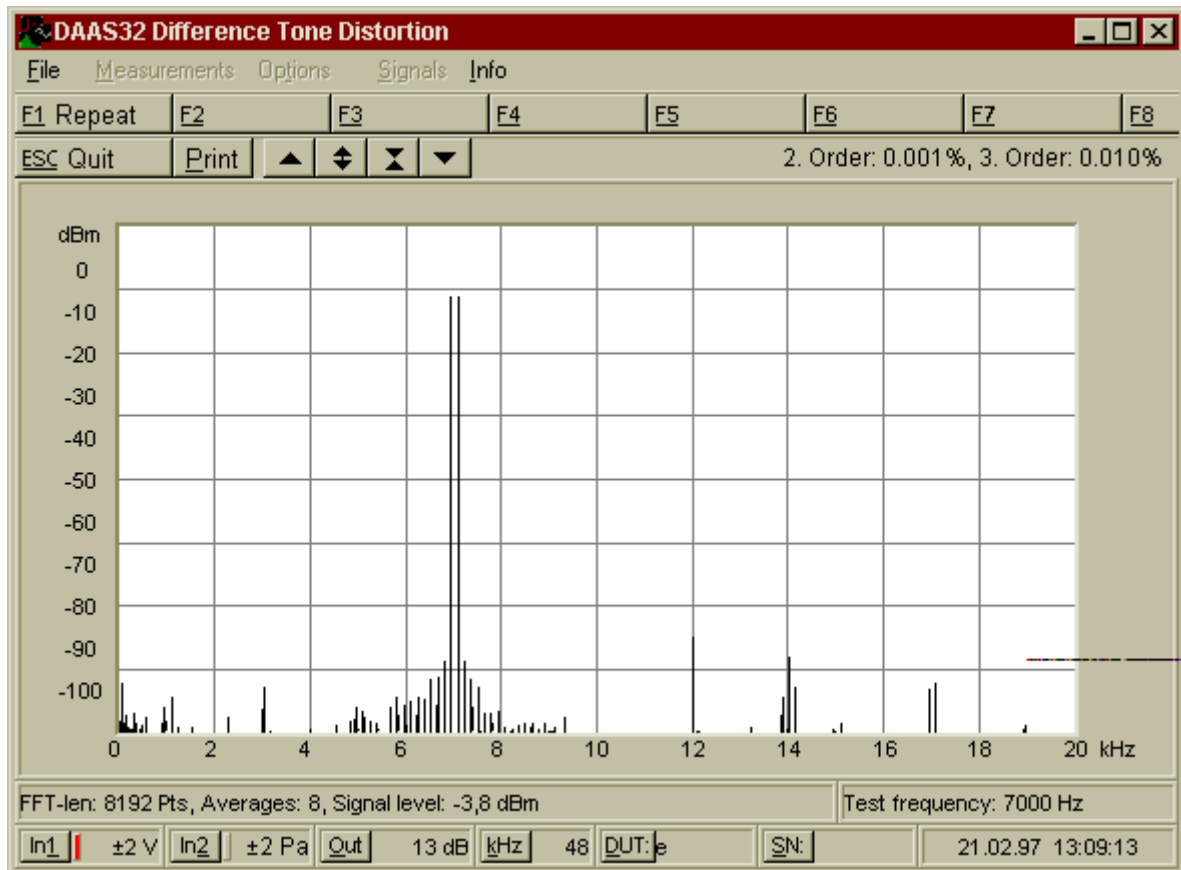
Like the Intermodulation factor measurement, the use of external test signals for the difference tone factor measurement is not possible.

3.6.5 Transient intermodulation distortion TIM

Transient intermodulation distortion originates from time delay in the inverse feedback of amplifiers and from different band width of the various amplifier stages.

TIM measurements are difficult to made. Different methods are suggested until now, and one of them is implemented in the DAAS®32.

This method uses a pink noise signal, which one octave/3 band removed:
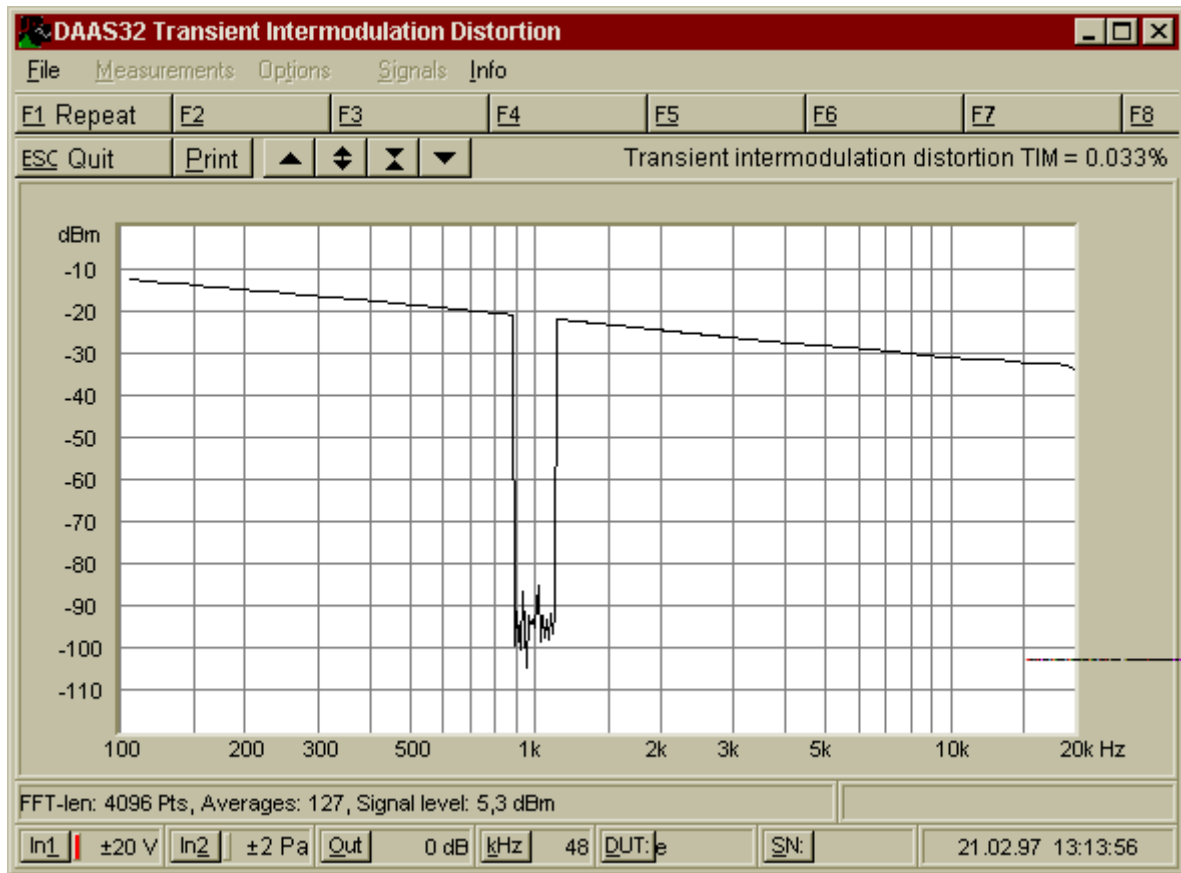
The level in this removed octave/3 band will be measured and related to the level in this band without removing.

Note that this measurement result is only valid for comparisons because the measurement method is not standardized.

The lower measurement limit is about 0.01 % corresponding to 80 dB. To improve the lower measurement limit increase the averaging time. This is done by increasing the signal memory, because the whole signal memory will be used for TIM measurements (see section 9.7).

3.7 Reverberation time RT 60

The determination of the reverberation time is of interest for several acoustical applications. An example of these is the calculation of the sound power from an acoustic power source (e.g. a speaker). The simplest way to calculate the reverberation time is to measure the acoustic level slope after shooting a pistol or switch off a loudspeaker supplied with noise. The measured acoustic level slope will be extended to 60 dB level difference, and the slope time for this 60 dB level difference is the reverberation time RT 60 of the room.
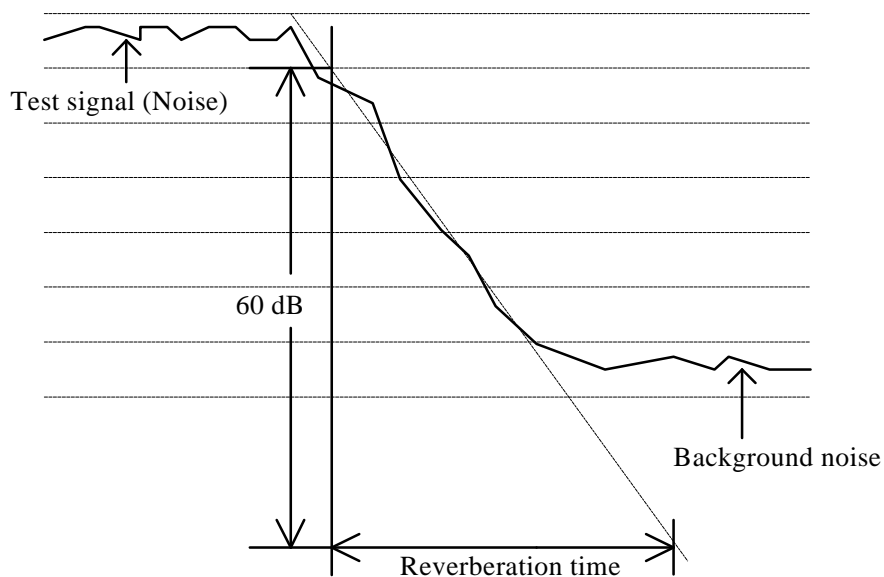
But the disadvantage of this method is that the reverberation time usually dependst on the frequency. Harmonic signals for supplying the speaker are not usable, because stationary acoustic waves will falsify the results. The DAAS®32 uses narrow band noise with variable center frequency to avoid this disadvantages. This method is very easy to use and don't need complex parameters to set.
Open the "Measurements" menu and choose "Reverberation time RT 60". It is also possible to start the RT 60 measurement by pressing Ctrl + F.

With "In 2" the input sensitivity can be set with ±2 Pa and ±8 Pa. The output level can also be set using "Out" and of course using the volume knob of the amplifier. Set up as follows and start the measurement by choosing the measuring frequency.
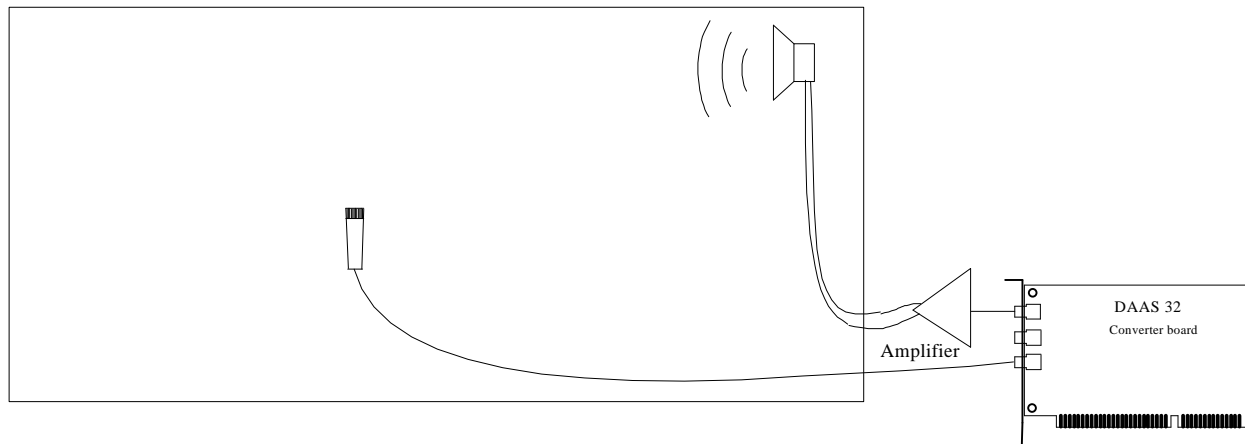
After the measurement the measured acoustic level curve is shown. A line to interpolate the slope will be inserted.
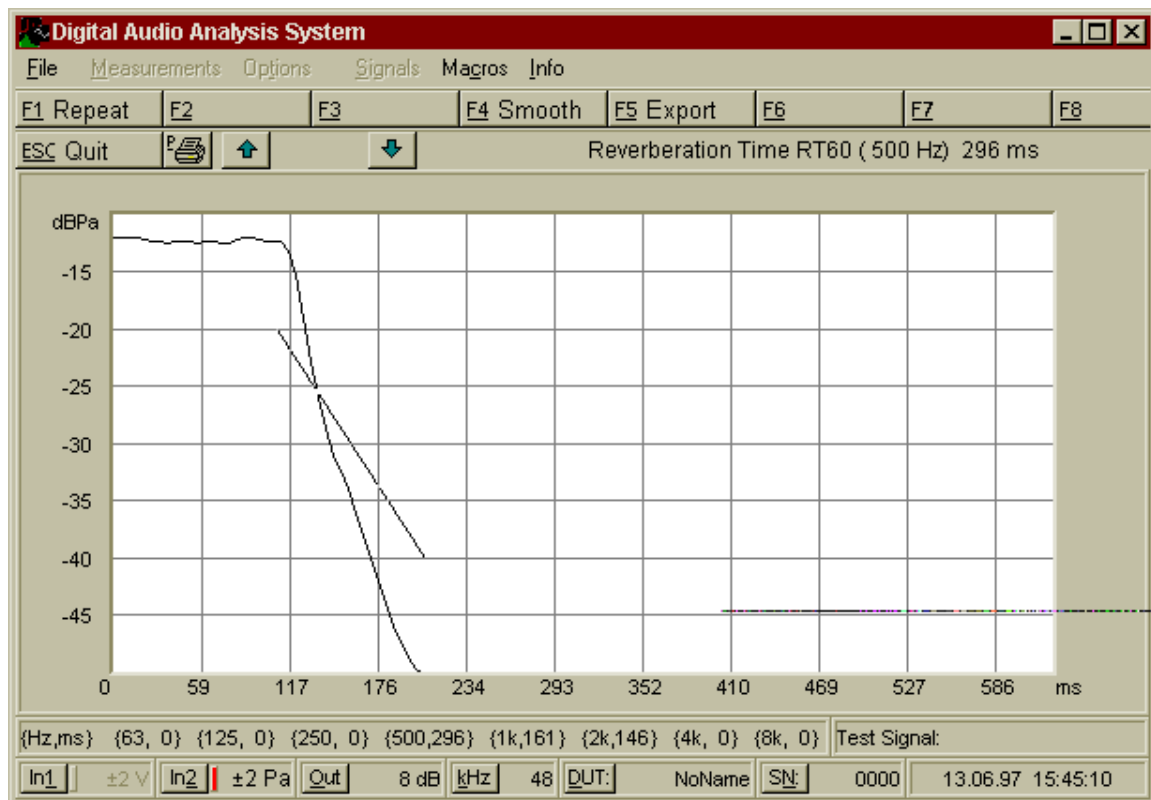
Shift this line until the slope of this line is identical with the slope of the acoustic level curve. To shift this line move the mouse to the upper end or to the lower end of the line, then press the left mouse button and move the chosen line end.

Smooth the acoustic level curve using F4 if necessary.
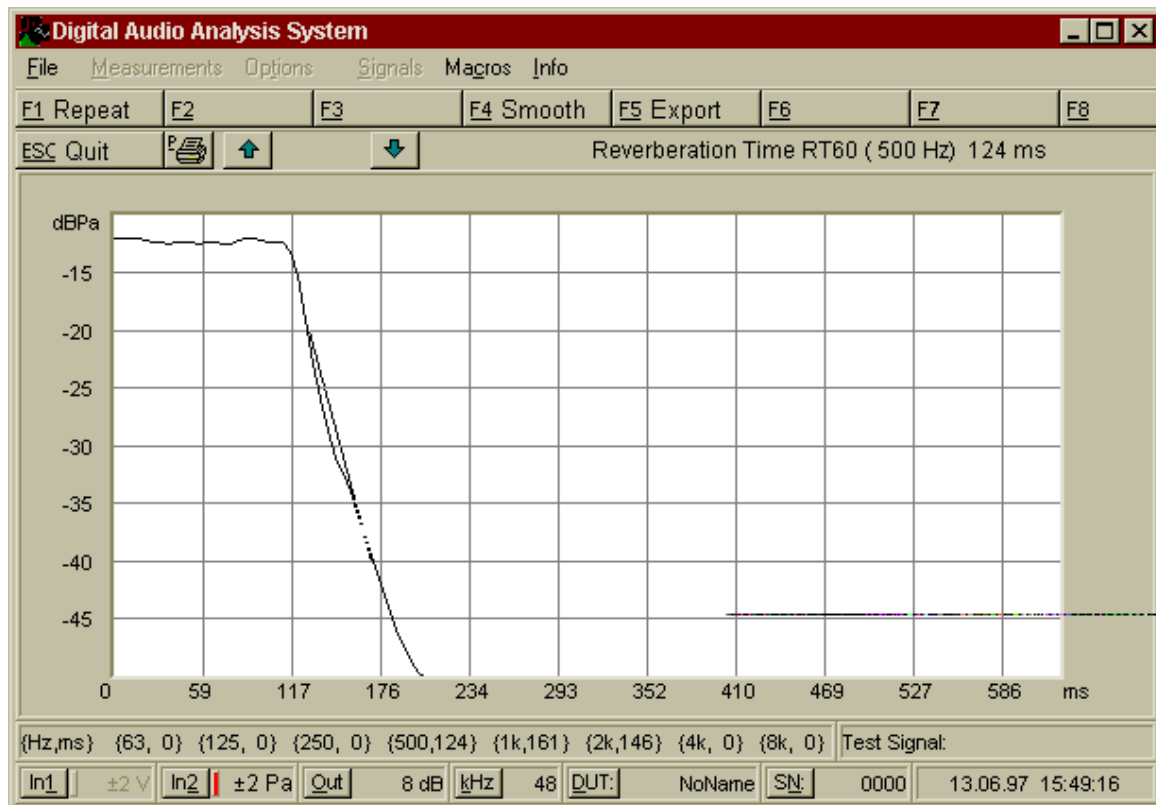
On the top of the screen in the info line now the reverberation time is shown. Additional all measured values since start of RT 60 measurement are shown in the status line. You can write this values to a text file by using the "Export" function.

3.8 Level and L_{EQ} Measurement

The Level and L_{EQ} Measurement implies a method to measure voltage levels and sound pressure levels. The signal level is shown as continuos undated signal level (Mode="Level"), as maximal level (Mode="Max Hold"), as L_m resp. L_{EQ} or as takt maximal leve $L_{Tm,3}$ resp. $L_{Tm,5}$.
The different averaging times "Fast", Slow" and "Impulse" and the various measuring modes are made in accordance with DIN IEC 651.
On F5 you can visit values on the previous measurement cycle.
On F4, various percentile levels such as L_{99}, L_{95}, L_{90}, L_m, are calculated on the previous measurement cycle.
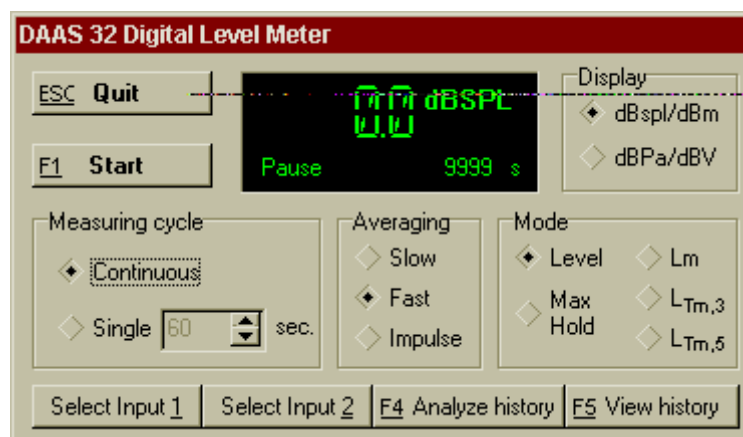
The measuring time can be set by using the "single" switch. On "Continuos", the measurement cycle will go on until the ESC key is pressed.

4 Signal Player

The "Signal Player" allows to play test signals or to generate and play some wave forms independent on any measurement.

To start the Signal Player open the menu "Signals" and choose "Signal Player". Adjust the output attenuation using F5.

Press F1 to play a signal file. Choose the signal file to play and select the number of repetitions: one time ('single') or repetitiously ('cont').

Or press F2...F4 to generate and play a sinus signal, a square signal or a triangle signal.

Start playing the signal by choosing the wave form frequency or by selecting the signal file. Now the wave form is played until the ESC-key is pressed.

The frequency of a sinus signal can be between 10 Hz and 27 kHz, the harmonic distortion is below 0.01 %.

The frequency of the other wave forms can be between 100 Hz and 10 kHz. Note that especially the high frequency square waves and triangle waves can be slightly different from the chosen frequency. The exact frequency is displayed in the message box in the middle of the screen. As well the high frequency square waves and triangle waves are slightly smoothed, because precise square waves and triangle waves require higher sampling rates.

5 Signal Recorder

The "Signal Recorder" allows to record signals independent on any measurement. Every recorded signal will be saved into a "WAV"-file and can be re-played every time.

To record a signal file

1. Open the menu "Signals" and choose "Signal Recorder".
2. Enter the signal length.
3. Enter the file name.
4. Choose the input and set the input sensitivity using "In", "In 2" or F3.
5. Press F1 to start recording.

Note that the whole signal must fit into the signal memory. Change the size of signal memory if necessary (see section 9.7).

6 Measurements with various frequency resolution

The system can be driven with three different sampling rates. The sampling rates 6 kHz, 48 kHz and 64 kHz are available and can be adjusted by using "kHz" button. The default sampling rate is 48 kHz.
In the frequency range the frequency resolution is conversely proportional to the sampling rate. The sampling rate in conjunction with FFT length the will designate the frequency resolution:

$$\text{frequency resolution} = \text{sampling rate} \, / \, \text{FFT length}.$$

Use 6 kHz if you need high frequency resolutions in lower frequency ranges. Except for the two tone distortion measurements and for the TIM measurement all analyses can be carried out at the higher resolution resp. the lower sampling rate. Frequency and impedance response can achieve a maximum resolution of about 0.1 Hz! Note that the frequencies of all test signal files are also reduced by a factor of 8. The test signal "1KHZ.WAV" has, with reduced sampling rate, 125 Hz. The useable frequency range is thereby extended down to 3-10 Hz (depending on the measurement). However the upper limit is reduced to 2500 Hz.

Use 64 kHz sampling rate to get knowledge about the behavior of impedance or frequency response curves above 20 kHz.

Further changes to the parameters are not necessary. The 'Anti-Aliasing-Filter', necessary with reduced frequency limits, is built into DAAS®32 as standard.

7 Saving and loading signals

Before each measurement and after each screen plot it is possible using the F5 key to save resp. to re-load signals.
On pressing F5 the selection box appears for the file name. Only files shown in "Load/save" directory (see section 9.1) which belong to the current measurement are shown. When saving, numerous additional pieces of relevant data necessary for future evaluations are stored with the signal.

Every reference signal for frequency response measurements, for impedance curve measurements and for Thiele Small Parameter measurements is saved with the special name "LASTREF". On leaving and later returning to the measurement it is not necessary to carry out a new reference measurement, providing of course the set-up has not been changed. Simply use the "F3 Use last..." button to re-load the last reference signal.

To load earlier measured signals you must consider the following points:

- Loading a frequency response or impedance curve signal you should also load the reference signal to be sure that all conditions like signal level and so on are recovered.
- Be sure that you've not changed the analysis parameter like test signal, FFT length, weighting window or number of averagings in the meantime.
- Loading impedance curve signals be sure that the serial resistor used by the signal to load is identical with the corresponding entry in the status line. If not, the results will be false.

8 Macro recorder

The macro recorder allows to record a sequence of keystrokes and mouse actions and play them back at a later time. Up to 9 macros are possible. Every macro can be named.

To record a sequence:

1. Open the "Macros" menu, select "Record Macro" and choose the macro number.
2. Enter the keystrokes and mouse actions that you want to record,
3. Press S or click the button Stop Record" to finish recording,
4. Now enter the designation for this macro.

To play back a macro open the "Macros" menu, select "Play Macro" and then choose the macro. Or simply press Alt + macro-number (e.g. Alt + 3).

By using the macro recorder you must consider the following aspects:

- You can play back a macro only at the same place where you have recorded it. For example, a macro for repeat the frequency response measurement, played in the options-menu will cause everything but not repeat the measurement.
- The macro recorder records also the time period between every action. This means if you make a pause between two actions while recording this pause will also appear by playing the macro.
- All mouse positions and moves are recorded in relation to the screen. If you resize or move the DAAS$^®$32 window in the mean time the mouse clicks go wrong.
- The running oscilloscope and the running spectrum analyzer cannot be stopped via macro.
- The signal player in repetitiously mode ('cont') or by playing wave forms cannot be stopped via macro.

After quit the Digital Audio Analysis System all macros will be kept.

9 Options

To change some options open the "Options" menu and select the area where you want to make changes.
Click the appropriate buttons or boxes to make your changes. Or use the keyboard like follows:

- To move to the next box press the TAB key.
- To move to the previous box press Shift + TAB.
- To toggle a check box press the SPACE bar.
- To change a value enter the value or press the up or down arrow.
- To change radio button positions press the up or down arrow.

Using the "Default"-button you can always recall the default settings for all adjustments in the actual area.

9.1 General options

Use this option	To do this
Actual Configuration	The options and various parameters of the DAAS$^{®}$32 are saved to the file 'DAAS32.CFG' by leaving the system. Starting DAAS$^{®}$32 this file containing the options is loaded. By using the "load" and "save" keys it is possible to generate various option files. By loading an option file the parameters will be active immediately.

Directories	Here you can set a separate path for export files, for temporary files and for signal files to save and load.
Show electrical Spectra as	Display scaling in dBV or dBm (electrical spectrum only).The scaling dBm means dB relative 1 milliwatt on 600 ohms = dB relative 775 mV on 600 ohms. It is also possible to set the load resistor to other values than 600 ohms.
Print options	Select the printer and the print options here. Check the "Black&White" box if you want to printout on Black&White printers. All curves with the different colors are then converted to black. To increase the letter size on printouts reduce the DAAS®32 main window before printing. To decrease the letter size on printouts increase the DAAS®32 main window before printing. Use F6 to change the printer and the printer adjustments like resolution, darkness and so on.
Export Format	Data format and number of data points for export frequency response curves and impedance curves. Four formats are available: 1. AudioCad: Frequency curves have the extension '.AMP', Frequency phase curves have the extension '.PHA', Impedance curves have the extension '.IMP', Impedance phase curves have the extension '.PHE'. 2. CALSOD: Frequency curves have the extension '.SPL', Impedance curves have the extension '.IMP'. 3. AkAbak®: Frequency curves have the extension '.AKD', Impedance curves have the extension '.IMP'. 4. Harmonic Design: For special use only.
Confirm Quit Measurements	Checking this box causes that DAAS®32 queries "Quit? Yes or No" by leaving a measurement.
Message Beeps	Check this box to activate the signal- and warning tones.

9.2 Setting the parameters for frequency response measurements

Here the parameters for the frequency response measurements and the associated functions can be set.

Use this option	To do this
Speaker freq. response in	The speakers frequency response curve can be displayed with 3 different scalings: • dBPa/V (dB Pascal/Volt, 1 Pa = 10µbar) • dBSPL/V (dBSPL = dB relative 20 µPa, 1 Pa = 94 dBSPL). • dBSPL/W/m, xx Ω, corrected to 1 meter at 1 Watt at a load resistor of xx Ohms. The setting dBSPL/W/m, xx Ohms is coupled with the automatic distance measuring. If the distance is greater than 5 cms an automatic conversion into dBSPL / Watt / Meter will be carried out. This is good for near field measurements in the range 15 to 50 cms.

The option dBSPL/W/m, xx Ohm is necessary when the frequency curve is to be exported to loudspeaker CAD programs which require data in dBSPL / W /m format.
By normal electrical measurements the frequency response curve is shown in dB.

Delay Correction

Delay compensation during acoustic measurements automatic or manual. The normal setting for this option is: 'automatic', i.e. no distance is requested as it is measured automatically. The 'manual' setting is only for use when the automatic setting is producing obviously false results or for other special applications.
CAUTION The calculated distance between speaker and microphone can sometimes be a little bit greater than the true distance, because the delay of the speaker, the microphone an the other equipment influences the result.

Smooth

The frequency response curves, the phase curves and the group delay curves are smoothed with a special low pass filter, so that the relative frequency resolution is constant. In conjunction with a change of the FFT-length for the frequency response measurement the following frequency resolutions respectively smoothing factors are available:

FFT length	Smooth 1	Smooth 2	Smooth 3
65536 Pts.	Octave /96	Octave /48	Octave /24
32768 Pts.	Octave /48	Octave /24	Octave /12
16384 Pts.	Octave /24	Octave /12	Octave /6
8192 Pts.	Octave /12	Octave /6	Octave /3
4096 Pts.	Octave /6	Octave /3	Octave /1.5
2048 Pts.	Octave /3	Octave /1.5	Octave /0.75

For acoustic measurements the 'Octave/3 bar' -option is available.

Automatically go to
Impulse Response

Checking this box causes the system shows first the impulse response instead of the frequency response curve.

Speed of Sound | As the speed of sound varies with air pressure, humidity and temperature the measurement distance (object - microphone) will only be exact if the actual speed of sound at the time of measurement is entered here. The default is 343 m/s which is a good average at normal room temperature and average humidity.

9.2.1 Options Waterfall Spectrum

Use this option | To do this

Number of spectra | Number of lines in waterfall spectrum (number of spectrums), either 48, 60 or 72.

Frequency limit | Lower and upper frequency limit of the waterfall spectrum.

Time smooth | The time slices of the waterfall spectrum can be smoothed. This is useful because the FFT analysis in conjunction with the 'Rectangular' weighting window causes leakage effects. These leakage effects can be reduced by smoothing the rising flank of the weighting window. But the disadvantage is the reduction of time resolution. This means a steep drop off appears smoothed.
The smoothing range is 1 point (no smoothing) to 40 points (max. smoothing). Use small values for tweeters and middling values for all other speakers. Default is 20 points.

9.2.2 Frequency response System Setup

WARNING The System options for the frequency response measurements form a vital part of the evaluation algorithms. Changes may well have a negative effect on system performance (increase in evaluation time, false results and large errors)! All settings are optimized by the manufacturer and should only be changed for special applications and only by persons who are fully familiar with the techniques of digital signal evaluation.

The optimum test signals for frequency response measurements are signals which were created with help from the DAAS®32 FFT and therefore contain all the properties to fit exactly in the analysis. The signals are 'WNxxxx.WAV' and 'PNxxxx.WAV'. 'WN' means White Noise, 'PN' means Pink Noise, and 'xxxx' means the sequence length of 2048 to 16383 points. **Using these test signals the FFT length must be adapted to the sequence length! Also the WINDOW must be 'Rectangular'!** Select an overlap of 0 %.
The default test signal is 'PN8192.WAV' in conjunction with a FFT length of 8192 pints, the 'Rectangular' window and an overlap of 0 %.
Alternative test signals are 'PINKNOIS', 'WHITE_NS.WAV' and 'MLS.WAV'. Using these or other test signals, the 'Hann' window in conjunction with an overlap of 20%-50% is recommended.
The number of RMS-averages can be between 2 and 64. More RMS-averages involves more stable results, but increases the calculation time and memory requirement. Using the test signal 'PNxxxx.WAV' or 'WNxxxx.WAV' the number of RMS-averages should be between 2 and 6, other test signals possibly requires more averages.

Frequency response measurements can also be made with a special sweep signal. This sweep signal (PNSWEEP.WAV) contains a short periodic noise sequence at the begin of the signal for determining the distance by acoustic measurements. To use this sweep signal for frequency response measurement simply choose the test signal 'CHIRP.WAV'. The analysis parameters are set automatically to the following values:

- FFT length: 8192,
- Averagings: 42,
- Overlap: 60 %,
- Window: Hann,
- Start Analysis: 2048 Points.

Don't change this adjustments by using "SWEEP.WAV"!

The FFT length in conjunction with the sampling rate will designate the frequency resolution:

$$\text{frequency resolution} = \text{sampling rate} / \text{FFT length}.$$

"Start analysis transfer function at" designates the waiting time between test signal start (t=0 in the time signal window) and start of the analysis to prevent errors due

to acoustic delay and transient behavior. Default: 2048 Points = 43 ms (using 48000 Hz sampling rate).

The options "Begin de-wrap phase at" and "De-wrap if phase step greater than" influence the de-wrapping phase function. De-wrap the phase curve means if the phase curve achieves e.g. 180° it will go on to greater values e.g. 200° instead of jump to -180°.
The option "Begin de-wrap phase at" designates the lower frequency to start with de-wrapping the phase curve.
The option "De-wrap if phase step greater than" designates the minimum phase step size for assuming this is a 360° wrap. If a phase step is greater than this value the system adds or subs 360°.

If you are using the "PNxxxx.WAV" or. "WNxxxx.WAV" test signals, and the error message 'Signal file not found, destroyed, or too short for the chosen measuring or for the chosen test signal length!' appears, the signal file might be too short. In this case you can extend the signal by using a sound editor. Take care to copy the whole signal in a buffer, and put it several times into the new file without a gap!

9.3 Options impedance curve and TSP measurements
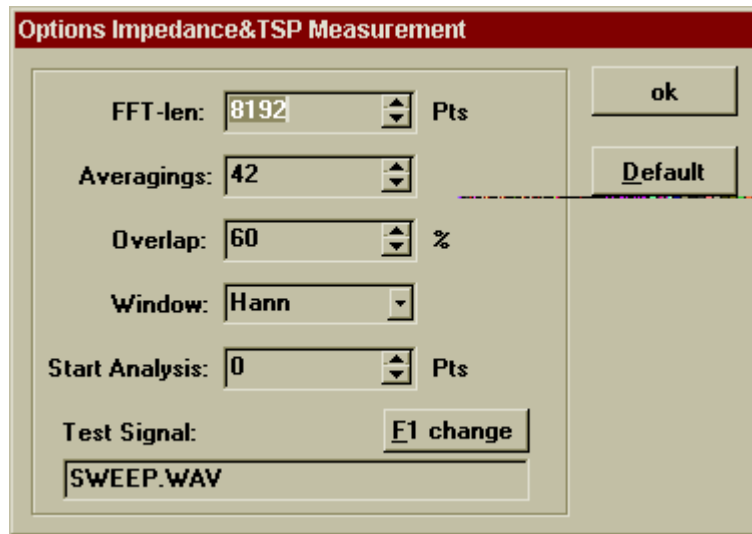
WARNING The options for impedance curve and Thiele Small parameter measurements form a vital part of the evaluation algorithms. Changes may well have a negative effect on system performance (increase in evaluation time, false results and large errors)! All settings are optimized by the manufacturer and should only be changed for special applications and only by persons who are fully familiar with the techniques of digital signal evaluation.

Measuring impedance curves of speakers or other non linear objects it is strongly recommended to use the test signal 'SWEEP.WAV', because the impedance of these objects depends on the amplitude of the test signal (the objects are non-linear). Using other test signals like Pink Noise or White Noise the impedance curve may come distorted or contain noise components.

The sine wave sweep signal 'SWEEP.WAV' ensures a constant magnitude during the whole measurement.

If the test signal 'SWEEP.WAV' is selected, all analysis parameters are set automatically to the following values:

- FFT length: 8192,
- Averagings: 42,
- Overlap: 60 %,
- Window: Hann,
- Start Analysis: 0 Points.

Don't change this adjustments by using "SWEEP.WAV"!

If you are sure that the test objects are really linear (e.g. measuring electrical networks) you can also use other test signals for measuring impedance curves. In this case the optimum test signals are signals which were created with help from the DAAS®32 FFT and therefore contain all the properties to fit exactly in the analysis. The signals are 'WNxxxx.WAV' and 'PNxxxx.WAV'. 'WN' means White Noise, 'PN' means Pink Noise, and 'xxxx' means the sequence length of 2048 to 16383 points. **Using this test signals the FFT length must be adapted to the sequence length! Also the WINDOW must be 'Rectangular'!** Select an overlap of 0 %.
Alternative test signals are 'PINKNOIS', 'WHITE_NS.WAV' and 'MLS.WAV'. Using these or other test signals, the 'Hann' window in conjunction with an overlap of 20%-50% is recommended.
The number of RMS-averages can be between 2 and 64. More RMS-averages involve more stable results, but increase the calculation time and memory requirement. Using the test signal 'PNxxxx.WAV' or 'WNxxxx.WAV' the number of RMS-averages should be between 2 and 6, other test signals possibly require more averages.

9.4 Options spectrum analyzer

These options are described in section 8.5, "Spectrum analyzer".

9.5 Desktop Colors

Here you can change the Desktop Colors. To get the default colors select "Default Colors".

9.6 Control external equipment

The DAAS®32 system allows to control external switching boxes like the adm featurebox with power amplifier, microphone amplifier, symmetrical inputs and so on. The control unit uses an interface module connected to the computers serial port. This interface module provides 8 control lines. In these options you can define which line is activated at the different measuring steps.
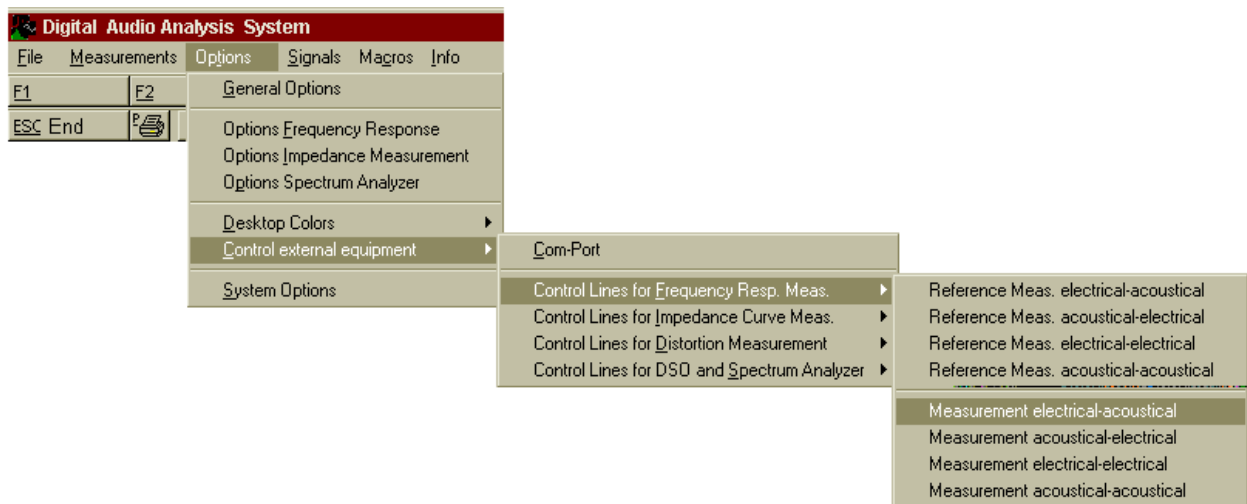
Select the serial port where you have connected the interface module, or select "none" if you don't have one.

Select the control lines like follows:
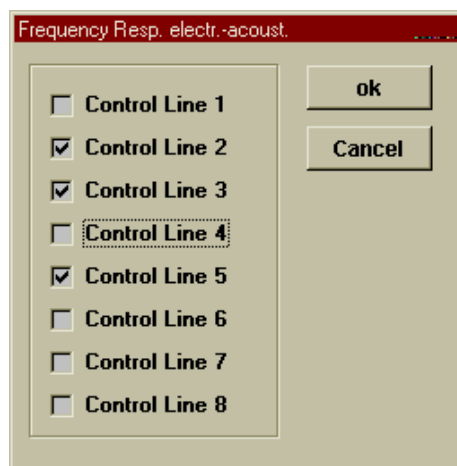
In this example the control lines 2,3 and 5 are activated if you are measuring an electrical - acoustical frequency response.

9.7 Setting the system parameters

CAUTION!

The system settings form a vital part of the evaluation algorithms. Changes may well have a negative effect on system performance (increase in evaluation time, false results, large errors, and system crash)!

All system settings are optimized by the manufacturer and should only be changed for special applications and only by persons who are fully familiar with the techniques of digital signal evaluation.

Use this option	To do this
Signal memory	Since the Digital Audio Analysis System is running under WINDOWS 9x it is possible to use up to 32 Mbytes for test signals. CAUTION Use not more than the half of the physical memory of your computer!
Max. Value for DAC	Maximum peak value of the sinus signals generated by DAAS®32. Range: 16 bit integer resp. 32767. Default: 30000.

Distortion measurement

FFT length   FFT length distortion measurements. Default: 8192 Pts.
                            Range: 4096 Pts. to 16384 Pts.

Averagings   Time averaging for all distortion measurements. Default 8.
                            Possibilities 1 to 64.
                            More averages involve a better noise suppression, but
                            increase the calculation time and memory requirement.

Start analysis   Waiting time between test signal start (t=0 in the time
                            signal window) and start of the analysis to prevent errors
                            due to delay and transient behavior. Default: 2048 Points =
                            43 ms (using 48000 Hz sampling rate).

Microphone   Here you can select one of four pre-defined microphones.
                            Each microphone has its own sensitivity adjust and its own
                            correction curve (MIC1.INI...MIC4.INI). See section 9.8
                            for details.

Address of Converter
Board   Selects the address range of the DAAS®32 converter
                             board. The normal setting is address 358h-35Fh. If this
                             setting leads to address conflicts with other boards the
                             range can be changed by setting jumpers on the converter
                             board. If the jumper setting does not correspond with the
                             setting under this menu heading, the error message "No
                             answer from converter board..."appears.

                             The following address ranges are possible:

                             • 1D8h - 1DFh, Jumper setting: J1 . . J8 = 0 0 0 1 1 0 1 0
                             • 2D8h - 2DFh, Jumper setting: J1 . . J8 = 0 0 1 1 0 0 1 0
                             • 320h - 327h, Jumper setting: J1 . . J8 = 1 1 0 0 0 1 1 1
                             • 350h - 357h, Jumper setting: J1 . . J8 = 1 0 0 1 0 0 1 1
                             • 358h - 35Fh, Jumper setting: J1 . . J8 = 1 0 0 1 0 0 1 0

                             "1" means jumper set, "0" jumper open.

                             The following picture shows the position of the address
                             jumpers.
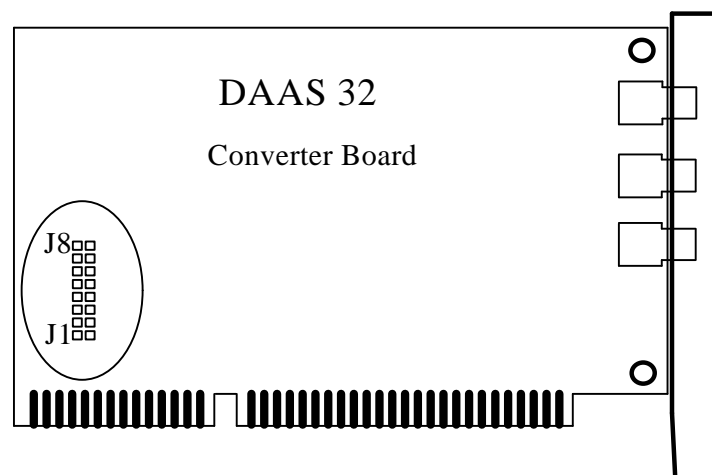
9.8 Calibration and adjustment

The "Line" input and the "Mic." input should be regally calibrated. Particularly if you want to use a new microphone, a calibration of the microphone and the "Mic." socked is highly recommended.
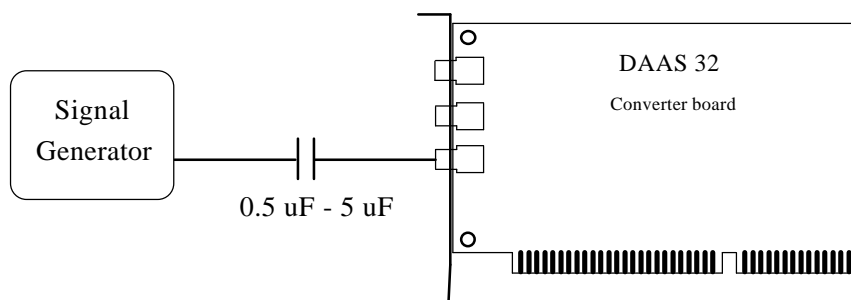
9.8.1 Microphone socked

Equipment required:

- Precision signal generator
- Reference digital voltmeter
- A capacitor in the range 0.5 uF to 5 uF

Procedures:

1. Connect the precision signal generator to the "Mic." socked,
2. Use the capacitor to remove the DC from the Phantom supply,

3. Select a signal frequency between 500 Hz and 2 kHz,
4. Open the System Options menu,
5. Set the generators output level to about half the entry in the "Mic. socked sensitivity" field (e.g. Mic. socked sensitivity: 120 mVrms, appropriate output level: 60 - 100 mVrms),
6. Press F4 to carry out a level measurement,
7. Compare the measured level with the generators output level.

If there is a difference, the Mic. trimmer will have to be adjusted. The adjustment point for the "Mic." socked is shown below:
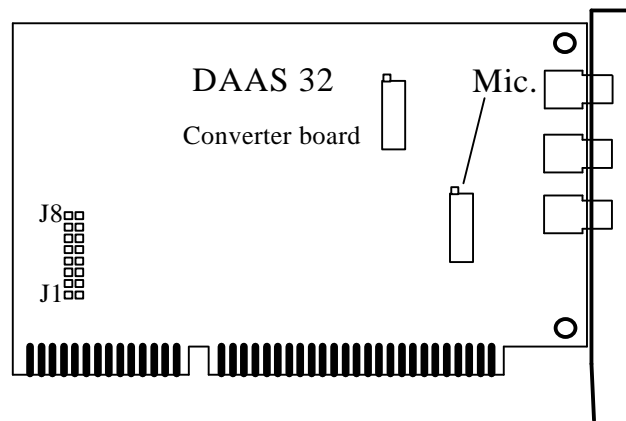
9.8.2 Microphone

Equipment required:

- A sound calibrator with a well defined signal level (i.e. 94 dBSPL resp. 0 dBPa at 1 kHz).

Procedures:

1. Place the Microphone into the sound calibrators output as described in the calibrators manual,
2. Activate the calibrator,
3. Open the System Options menu,
4. Press F3 ("Calibrate selected Mic."),
5. Enter the calibrators sound level.

After calibration process, the microphones sensitivity is calculated.
The calculated sensitivity appears in the appropriate field and should match the manufacturers specification.

Instead of this calibration process, it is also possible to set the microphone sensitivity entry manual to the specified value (see data sheet from the microphone manufacturer). In this case the accuracy of any acoustical measurements is reduced.

9.8.3 "Line" socked

Equipment required:

- Precision signal generator
- Reference digital voltmeter

Procedures:

1. Connect the precision signal generator to the "Line" socked,
2. Select a signal frequency between 500 Hz and 2 kHz,
3. Carry out a level measurement (see section 3.8),
4. Compare the measured level with the generators output level.

If there is a difference, the Line trimmer will have to be adjusted. The adjustment point for the "Line" socked is shown below:
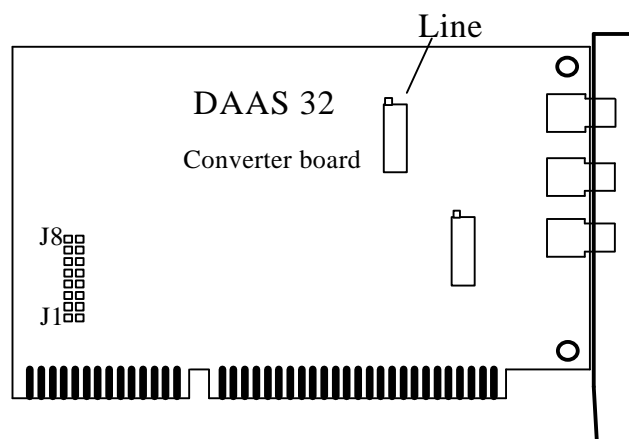
9.8.4 Adaptation of a new microphone

This section describes the process to adapt a new microphone to the system.
At first, take a look into the data sheet of the new microphone. Look for the sensitivity of the microphone. For example, this value is about 30 mV/Pa.
Choose the maximum sound level input range you wish to use (e.g. 12 dBPa resp. 106 dBSPL). Convert this sound level into the corresponding Pascal value. For this example, the result is 4 Pa (RMS).
Now calculate the required "Mic." socked sensitivity like follows:

$$\text{"Mic." socked sensitivity} = 4\ \text{Pa} * 30\ \text{mV/Pa} = 120\ \text{mV}$$

Enter this calculated "Mic." socked sensitivity into the appropriate field. Now you must calibrate and adjust the "Mic." socked as described in section 9.8.1. Finally calibrate the microphone as described in section 9.8.2.

9.8.5 Microphone correction curves

The DAAS®32 system allows to correct the frequency response of a non ideal measuring microphone. This feature works in conjunction with a data file containing the frequency response data in ASCII text file format. The file name is "MIC?.INI" ("MIC1.INI" for microphone 1, "MIC2.INI" for microphone 2, and so on). If this file exists in the DAAS®32 folder, it will be considered, if not, no corrections will be made.

The file should contain the frequency response curve of the measuring microphone as couples of frequency- and dB-values, equal to the AkAbak® export format. Example:

```
   100        -3.2
   110        -3.1
   150        -1.1
   350         1.0

             .
             .
             .
  1000         2.5
  1200         0.9

             .
             .
             .
 12900        -1.2
 15000        -2.2
```

To inspect the actual correction curve, start the DAAS®32 with the command line option "/DEMO" and carry out a speakers frequency response measurement. If "MIC?.INI" does not exists, the frequency response appears a flat line. If "MIC?.INI" exists, the correction curve appears.

You can measure you own correction curves by using the frequency measurement in conjunction with a high precision reference microphone. To do this, measure the frequency response of the measuring microphone you want to use, and export the result by using the AkAbak® export format. Edit the export file with a simple text editor. Remove the values you don't want to correct (usually the lower frequencies

below 200 Hz and the upper frequencies above 15 kHz). Then rename the file to "MIC?.INI".

Working with microphone correction curves you must consider the following points:

- The frequency points must not be equidistant.
- The frequency curve must not lie near to 0 dB. All values are normalized to 0 dB at 1 kHz.
- Any "MIC?.INI" file can contain up to 499 frequency points.
- This feature is not made for using poor microphones! The compensation range should not exceed ± 5 dB.

10 Technical data DAAS®32

10.1 DAAS®32 converter board

Function: PC-Bus-card with quartz controlled 16 Bit-2 channel-
                   Analogue/Digital-converter- and Digital/Analogue-
                   converter plus input and output amplifier

Inputs: electric (Line).: ±0.20V, ±2.0V and ±20V,
                   software switchable,
                   input impedance 100 kOhm,
                   Microphone input with Phantom supply 8 V,
                   Resistance 8,2 kOhm. Input sensitivity from DAAS
                   microphone: ±2 Pa ad ±8 Pa, software switchable.
                   Max. voltage: 2Vss

Output: Output resistance <10 Ohm, minimum load: 600 Ohm,
                   output level to 6 Vs, software controllable in 1.0 dB-steps
                   up to 90 dB attenuation.

Sampling rate: 64 kHz, 48 kHz and 6 kHz

Resolution: A/D-converter: 16 Bit,
                   D/A-converter: 16 Bit

Address range: 358 Hex to 35F Hex (default, adjustable),

Connections :   Phono (Cinch) for 'Line', 'Out' and 'Mic.'

Temperatures: In use: +17° to + 35° C, no condensation, humidity: 10% to
                   75%. Storage temp.: 0 to + 50° C, no condensation

Warm up time: typ. 15 minutes to specified accuracy.

10.2 Software DAAS®32

Function type:	FFT-Analyzer
FFT-Length:	512 - 65536 points
Frequency resolution:	up to 0.1 Hz (6 kHz sampling rate)
Weighting windows:	Rectangular (UNIFORM),
	Hann (or Hanning),
	Blackman,
	Kaiser (B=10.056, Side-Lobe-attenuation approx. 100 dB)
	Flat Top.
Overlap:	0 % to 50 %

Analysis   frequency response via cross spectrum, magnitude and phase,
delay measurement via cross spectrum and Hilbert Transform,
group delay via phase,
impedance curves via cross spectrum, magnitude and phase,
Minimal phase and excess phase,
Step response, impulse response and ETC,
Waterfall spectrum (cumulative spectral decay),
Free field correction of acoustic frequency response curves,
Distortion measurement via harmonic analysis,
Thiele Small parameters including equivalent volumes
Digital storage oscilloscope function with trigger, time selective frequency analysis and waterfall diagram,
Spectrum analyzer function with linear or exponential averaging and smoothing,
Reverberation time analysis (RT60).

Test signal format   Standard WAV format with the following parameters:
- one channel (mono),
- 16 Bit signed (linear coded, ± 32768, INTEL format),
- 48 kHz sampling rate.

11 System requirements

- Intel® Pentium® processor or compatible, 100 MHz or better,
- Operating system WINDOWS 95® or later,
- VGA, min. 640x480 points, 256 colors, 800x600 points recommended
- one free 16 bit slot,
- Microsoft® - Mouse or compatible,
- one hard drive, one 3,5" floppy drive or one CD-ROM drive,
- min. 16 Mbytes memory,
- 20 Mbytes free hard drive space

12 Important points concerning safety and use

- Use DAAS®32 only for the given applications. If you are not sure DAAS®32 is meant for, inquire a specialist.

- If you are using a hi-fi amplifier as measuring amplifier, you must never shortcut the input ground and the output ground of the amplifier!
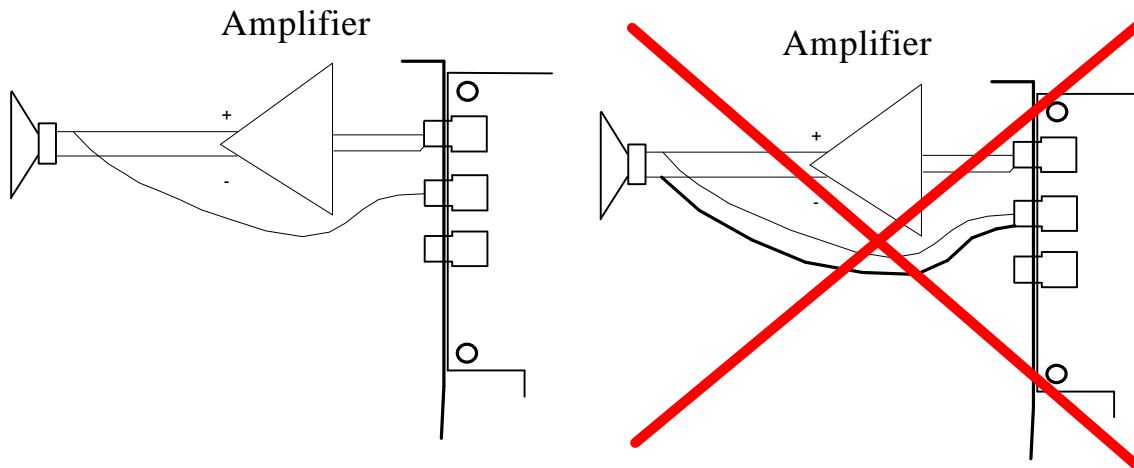
A connection between the input ground and the output ground of a hi-fi amplifier may result a high frequency oscillation with a high amplitude. This may damage your amplifier!

- You must not use amplifiers with bridged power stages!

- No voltages of any sort are to be applied to the output socket.

- Before connect or disconnect a device switch off all concerned devices.

- Be careful with static charges as these can easily lead to damage.

- Keep the DAAS®32 converter board sockets clean and apply a little contact spray occasionally. Don't use any sort of aggressive cleaning substance.

- After long use the DAAS®32 converter board may become dusty which can lead to errors. Clean the board occasionally with a soft brush and lightly blow any deposits away. Don't use a rag due to the build up of static charges which can damage the components.

- Problems with SCSI 2 - adapters or other hardware: The card is factory set at address 358 Hex to 35F Hex. Interrupts and DMA channels are not used. The card address can be changed see section "Setting the system parameters"

All information and specification are subject to change without notice.