Digital Audio Analysis System

Version Q-DAAS, DPMI 32 bit

User handbook

adm engineering
Steinmaate 24
D - 48529 Nordhorn
Germany
Tel. ++ 49 59 21 72 10 00
Fax ++ 49 59 21 72 10 01
eMail: info@admnet.de

Preface

The Quality Control Digital Audio Analysis System Q-DAAS was developed and produced by adm engineering, Germany.

The legal owner is allowed unlimited use of the enclosed software. The legal owner is allowed to make one security (back-up) copy of the software for his/her personal use. Further copying of the software, handbook or hardware is strictly forbidden and will result in legal action being taken. Reprinting the handbook or the information contained is only possible with written permission adm engineering.

adm engineering gives no guarantee that the contents of this handbook or the enclosed software are totally free of errors. Customer feedback in this respect is always welcome and we will attempt to correct any errors as soon as possible.

adm engineering gives no guarantee that Q-DAAS comprising the handbook, hard and software can always be successfully employed in any special situations. No liability can be accepted for any damage or claim resulting from the use of this system.

Various trade names and titles, protected by international law, are used in this handbook. The names remain protected even if they are not specially marked as such in the text.

Please read chapter 11 for your safety.

MS-DOS is a trademark of Microsoft Corporation
WINDOWS is a trademark of Microsoft Corporation
DR-DOS is a trademark of Digital Research
IBM PC, XT, AT, PC-DOS are trademarks of IBM
DAAS is a trademark of adm engineering, Nordhorn
internet: www.admnet.de

EMC-Act/CE marking

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

> Declaration of Conformity
> 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 should be applied with a protection-plugs. Use only shielded high quality cables with high quality cinch plugs (metal sleeve)
> .

Contents:

General

Preface

The thinking behind the development of the Digital Audio Analysis System was to produce a reasonably priced, PC supported measuring system for the quality control. From the original concept of a sampling system, analogue to digital converter and computerized evaluation techniques the system has developed to it's current form with wide ranging functions and features; the Q-DAAS.
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 need have no 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.

Q-DAAS includes many useful and important functions:

- Signal generation and measurement using a 16 Bit Digital/Analog- and Analog/Digital-converter,
- Sample Rates up to 64 kHz
- Tolerance masks for QC measurements
- Measurement of frequency and impedance with variable frequency resolution.
- Calculation of phase from frequency and impedance curves with automatic delay correction for acoustic measurement.
- Calculation of step response and waterfall spectrum
- Free field correction of acoustic frequency response curves
- Distortion measurement by virtually any frequency
- Evaluation of distortion curves vs. frequency
- Measurement of intermodulation distortion with variable frequencies.
- Calculation of Thiele Small parameters including equivalent volumes.
- Maximum frequency resolution approx. 0,5 Hz,
- Interface to speaker CAD program AudioCad and CALSOD or LEAP

Working with Q-DAAS 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.

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 oldest 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.
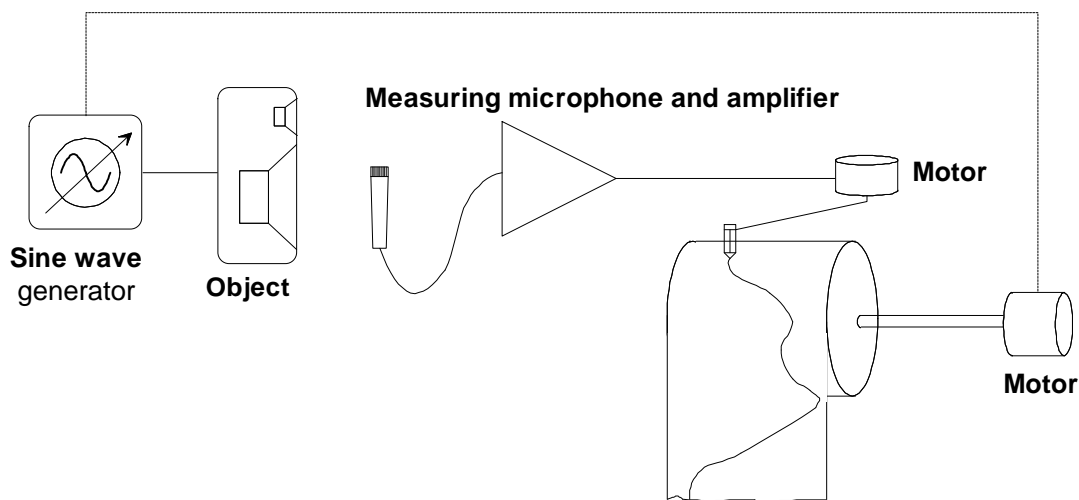
By the use of a plotter as shown above a continuous response curve can be obtained. If a bandpass filter, the middle frequency of which can be changed in synchronization with the generators frequency, is inserted between microphone and plotter, it is possible, within certain limits to remove room reflections, i.e. it is possible to achieve free field measurements within a sealed room. This principal 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.
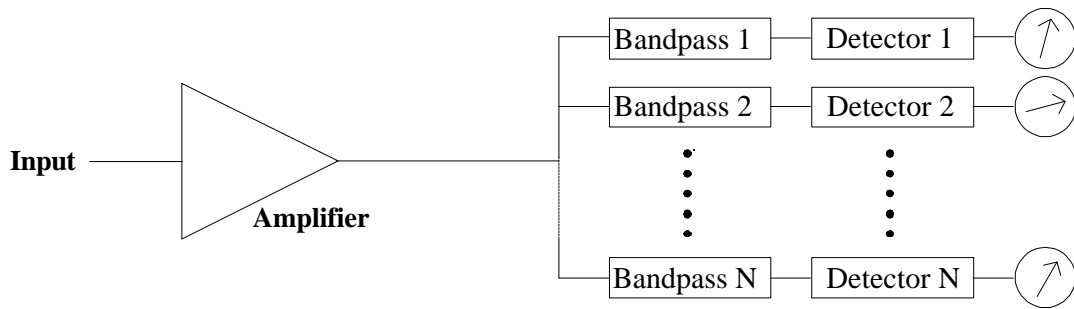
When the middle frequencies are so chosen that the filter bands border exactly with each other a complete spectrum can be achieved. With this system the entire frequency response curve can be obtained with only two individual measurements.
The measured object (e.g. a loudspeaker) is fed with a signal 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) are 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 whilst simulating the filters. The hardware requirements are therefore drastically reduced.
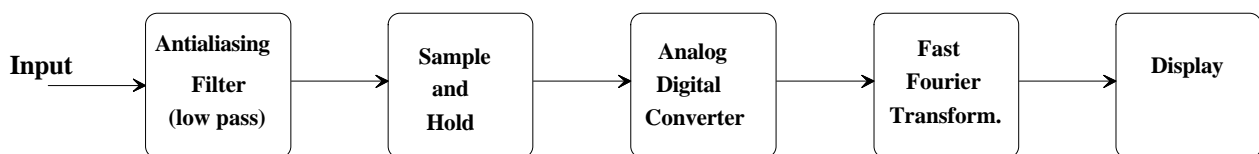
The measurement of frequency response is similar to that of the parallel filter Analyzer namely through evaluation of the signal at the input and output of the object under test. Theoretically if we used Periodic Noise it should be possible to obtain a frequency response by measuring the signal at the 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 for the various measurements:

◆ Pink noise, white noise and a Maximum-Length-Sequence
   (MLS) for frequency response and impedance measurements.
◆ 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.

Further signals can be requested or generated by the user.

Using the digitized signal, numerous analysis are possible:

◆ Waveform analysis,
◆ Settling time (i.e. Thiele Small Parameter)
◆ True RMS measurement of non sinusoidal signals,
◆ Analysis of transients and many more.

Through calculating the spectrum using the FFT algorithm, frequency evaluations can be undertaken.

◆ frequency and phase response/transfer function
◆ Step response / impulse response
◆ Waterfall spectrum
◆ Correlation analysis
◆ Impedance, magnitude and phase.
◆ Harmonic Distortion, Intermodulation, Difference-tone-factor,
◆ Analysis AM/FM (Modulation)
◆ Noise and sound pressure level measurements with differing weighting filter
◆ general spectral analysis with differing evaluation windows and resolutions.

Once the signal is stored in the computer in digital form, with suitable software, there is virtually no limit to the evaluation possibilities.

Installation and Start

Installing the ADC/DAC-card.

Switch off your computer and isolate from the mains 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 ADC/DAC card from its protective sleeve touch an earth point (e.g. a central heating radiator) to discharge any static which 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 you 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 an **AT-bus** 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 ADC/DAC-card is tested before dispatch. It is possible that an address conflict has occurred with another extension card. The ADC/DAC-card is supplied set to IO-address 358 Hex to 35F Hex. The address can be changed.

Installing the software

The software is installed using "INSTALL.EXE" on the program disk. This program copies the QDAAS files and sets the system up on your hard drive.

If you like to install the software under Windows 95, please read the file WIN95.DOC on the installation disk for help.

Change to the drive containing the installation disk with :-

A: [Enter]   or   B: [Enter].

Start the installation with

Install [Enter].

If your computer has a monochrome screen (e.g. Laptop), the installation can be started with "Install /b".
Press [F2] to select the hard drive to which DAAS is to be installed to and start the installation with [F1].

Start

Change directory to where the Digital Audio Analysis Systems has been installed to with

CD \Q_DAAS [Enter]

and start the program with

QDAAS [Enter].

If you have a SVGA-card supports higher screen resolution you can start the Digital Audio Analysis Systems with

QDAAS /800 [Enter]

or

QDAAS /1024 [Enter]

for display resolution of 800x600 Points or 1024x768 Points.

For proper operation with higher resolutions a VESA driver for your super -vga card is required !

Using the system

The graphic user interface of the Digital Audio Analysis System DAAS is totally
function-key or mouse controlled.
The following diagram shows the various screen areas.

```
┌─────────────────────────────────────────────────────────────────────────────┐
│ ┌──────────┐ ┌───────────────────────────────────────┐ ┌───────────────────┐ │
│ │ ESC End  │ │   D A A S 3L    Frequency response    │ │  05.12.92  15:47  │ │
│ └──────────┘ └───────────────────────────────────────┘ └───────────────────┘ │
│ ┌──────────────────────────────────────────────────┐   ┌───────────────────┐ │
│ │ Distance Loudspeaker - Mic : 36 cm, dBSPL normalised to 1 meter │   │ MO: Sample  Ser.No. 3 │ │
│ └──────────────────────────────────────────────────┘   └───────────────────┘ │
```

Info-box   Title   Measurement objekt   Clock

Message box / input box

Please enter the file name:
(without extension)

Data window

| F1 Repeat | F2 Free field | F3 Phase | F4 Print | F5 Export |
| F6 Waterfall | F7 Step response | F8 Freq. range | F9 Smooth | F10 load/save |
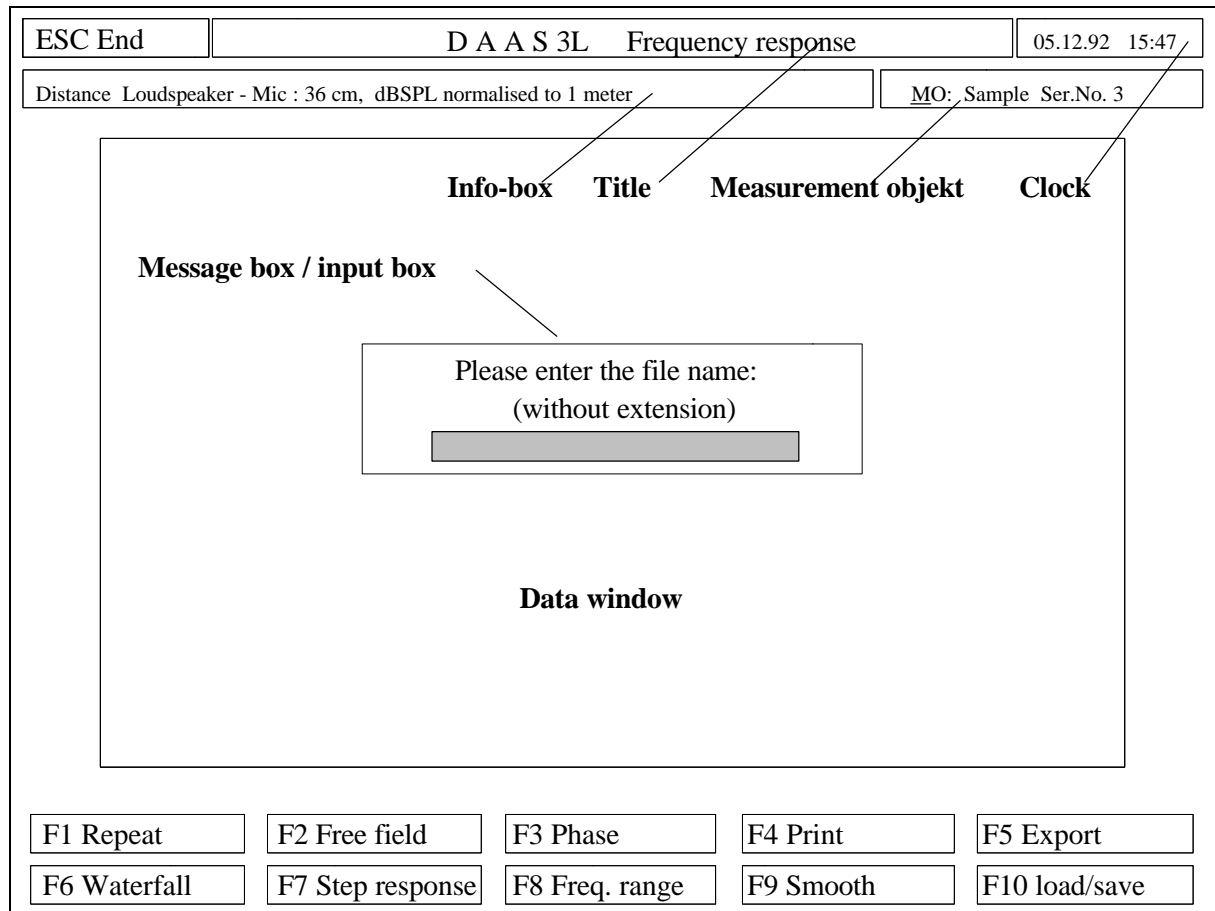
The first row contains the fields with the ESC key, the clock and the title. The ESC key
is labeled "Cancel" to cancel an action or with "End" to finish a measurement or the
program. The title field shows the name of the current measurement and in the clock
field the current date and time are shown. The field "measurement-object" shows the
first 11 characters of the objects manufacturers name and the first 11 letters of the
objects description. To enter details of the measurement object use the key combination
[Alt]+[M] or click on the field "Measurement object" with the left mouse button. In the
"info-box" additional information and results are shown.

In the data window all measured or calculated signals are shown. Click with the left
mouse button in the data window to insert a lettering. Press the right mouse button to

read-out time data (scope mode), frequency response values or impedance curve values at mouse cursor position.

The lower section of screen contains the function keys F1 to F10 through which complete program control is achieved.
All key strokes can of course be controlled by the mouse simply by clicking the left mouse button over the required function.

In the center of the screen there will, from time to time, appear a message or input box. This box gives details of the progress of the measurement or evaluation, general information, messages, warnings etc. This area is also used for entering data for example object or file names.
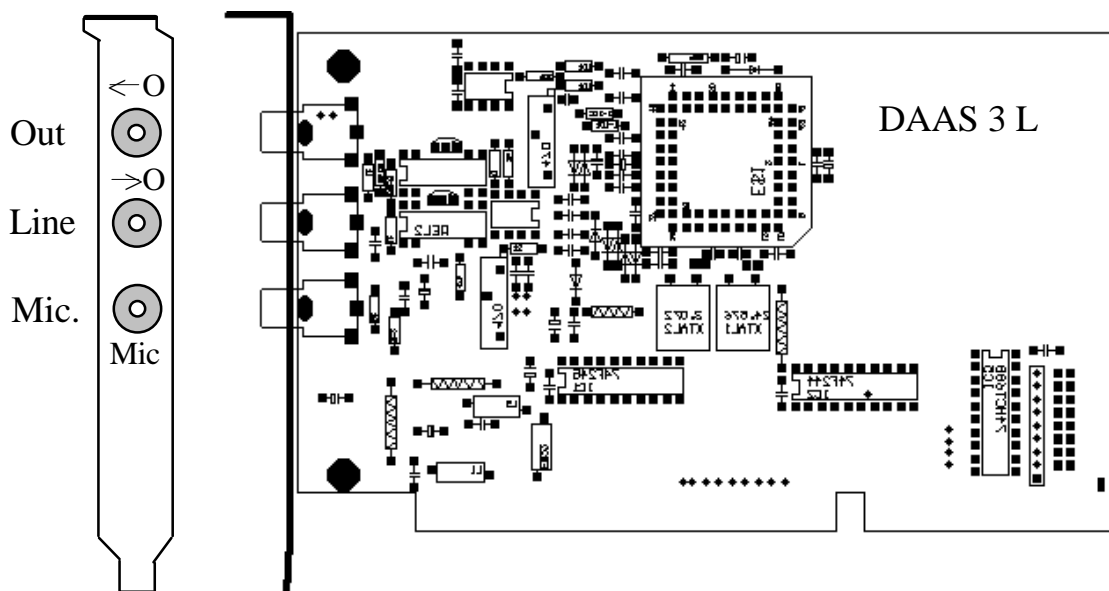
The DAAS card has 2 inputs and one signal output. The microphone is connected to the "Mic" socket. Using the microphone supplied the maximum sound pressure is ±2 Pa and ±8 Pa.
The input "Line" (Symbol →O) is the input for electrical measurements. Three ranges are available ±0.2 V, ±2 V and ±20 V.

All signals generated by DAAS 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 amp will be suitable. (NOT a bridged amp!!!)

Measurements

The most important and most commonly used measurements are pre-programmed into your Digital Audio Analysis System. 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. **Note**: please notice that the shown wiring scheme differ from the use of an additional Featurebox. Shown wiring scheme explain the basic structures of the system and should bring a higher degree of comprehension.

Frequency response measurements

Using DAAS 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. As in professional systems it is also possible to plot the response curves logarithmically or 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-analyse, the delay is calculated and corrected, so that the delay corrected phase curve can be plotted. From an object's (say loudspeaker's) frequency response curve, using DAAS it is possible to obtain two further important properties: the step response and the cumulative waterfall spectrum.

Choose in the main menu:

F1 Freq. resp..	F2 Impedance	F3 Thiele Small	F4Harm. Distortion	F5 RT 60
F6 Level	F7 Scope	F8 Spectrum	F9 Play Signal	F10 Options

......the frequency response measurement by pressing F1 or pressing the left mouse button with the mouse pointer over the respective field.
Firstly you must carry out a reference measurement on the input signal of the object to be measured.
There are 4 possibilities:
• electric - acoustic (e.g. Loudspeakers),

- electric - electric (e.g. amplifiers, crossovers),
- acoustic - electric (e.g. Microphone),
- acoustic - acoustic (Loudspeaker comparisons).

In the second two cases an acoustical reference measurement is necessary. Connect the signal output "Out" with the amplifier input and connect the speaker to the amplifier output. Position the microphone on the required reference location and start the reference measurement. If the sound level is too high or too low this can be adjusted using the amplifiers volume control or by choosing F3 "Output". The reference measurement should then be repeated.
Note: Measuring microphones (acoustic - electric), place the microphone close to the speaker, because the delay correction does not work in this direction.

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" .
The connections, using a speaker box as the object is as follows:
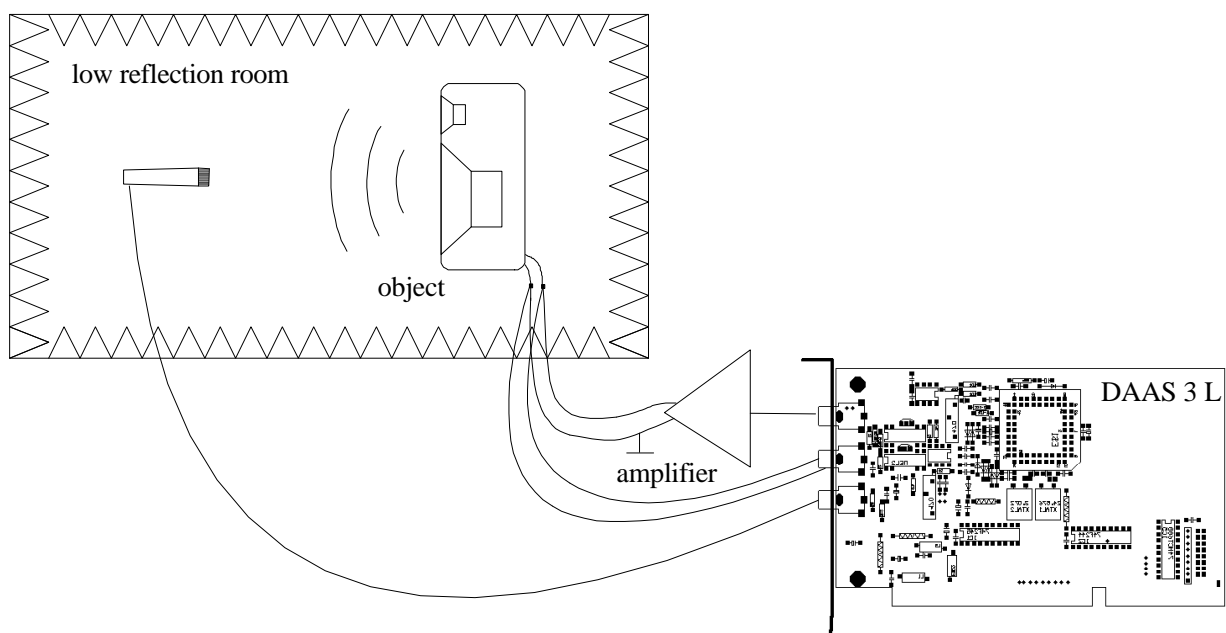
The electrical reference measurement is started by choosing the voltage range :-

F1 0,2 V	F2 2 V	F3 20 V	F4 Use Last	F5 load
F6	F7	F8	F9 Out Att.	F10 Sample Rate

The best voltage range depends on the object under test.

If you require to test an amplifier or some other piece of hi-fi equipment set the measurement level using [F9] "Out Att." to the range 20 to 40 dB (the equipment must not be over driven.....clipping!!) and choose the range "2 V" or "0,2 V".
When measuring loudspeakers the signal level is increased by the amplifier so generally the 20 V range will 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 [F9] for "Out Att.", then press [F2] for "Test". 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 use [Up] and [Down] to set the attenuation for the required level.
The volume level should now of course not be changed for the duration of the measurement.

[F5] is the "load" key. With this key 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 had not been changed in the meantime. A direct key to this function is [F4] „Use Last". This loads the last used reference signal.
Further details are contained in the section on "Loading and saving signals".

On completion of the measurement the following screen should appear:
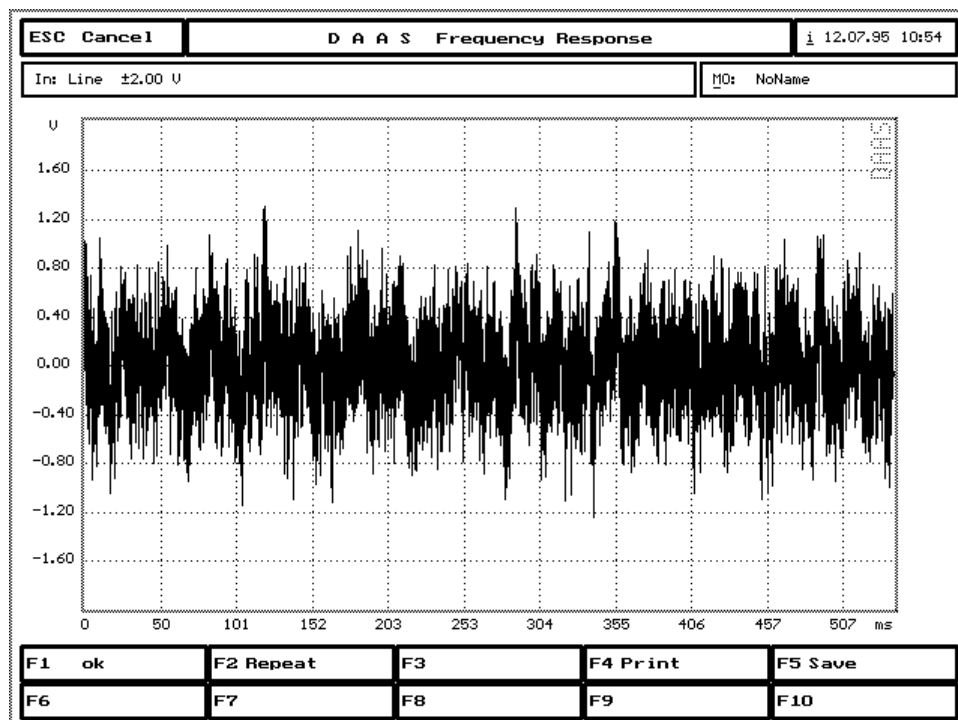
If the signal is too large a warning will appear on the screen, if it is too small the error will be higher.

In both cases the amplifier volume control should be adjusted and the operation repeated.

If required the signal can now be saved using F5.

Pressing [F4] will produce a screen hard copy on a 9 or 24 pin matrix printer or an HP-Laserjet II or create a PCX-File. (see section 6 on screen prints).

Press [F1], if the signal is OK and refrain from changing the amplifier settings. 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 measuring object with "Line" (measuring amplifiers etc.) or position the microphone (measuring Loudspeakers etc.). When everything is set start the measurement by choosing the range (± 0.2 V, ± 2 V or ± 20 V i.e. ± 2 Pa or ± 8 Pa). Choose a voltage range so that the range is fully used but over driving (clipping) does not occur. Alternatively load a signal from disk with F5.

If the measurement is OK it can be saved using F5 or displayed on the screen with F1.

If in the properties 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 when the distance is greater than 10 cm or when 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 'info box' at the top of the screen. Check that this distance is reasonable before continuing.

CAUTION: particularly with bass speakers it is normal for the automatic method to show a few centimeters 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 36 cms.

```
┌─────────────────────────────────────────────────────────────────┐
│ ┌──────────┐ ┌───────────────────────────────┐ ┌──────────────┐ │
│ │ ESC  End │ │  D A A S   Frequency Response  │ │ i 12.07.95 10:57│ │
│ └──────────┘ └───────────────────────────────┘ └──────────────┘ │
│ ┌──────────────────────────────────────────────┐ ┌────────────┐ │
│ │ Distance Loudsp.- Mic.: 36.5cm dBSPL normalised on 1 Meter │ │ MO: AUTO LAUTSPR.│ │
│ └──────────────────────────────────────────────┘ └────────────┘ │
```

(Frequency response graph: dBSPL/W/m on vertical axis from 75 to 105; horizontal axis frequency from 30 Hz to 20 kHz)

```
┌────────────┐ ┌────────────┐ ┌────────────┐ ┌────────────┐ ┌────────────┐
│ F1 Repeat  │ │ F2 Freefield│ │ F3 Phase   │ │ F4 Print   │ │ F5 Load/save│
└────────────┘ └────────────┘ └────────────┘ └────────────┘ └────────────┘
┌────────────┐ ┌────────────┐ ┌────────────┐ ┌────────────┐ ┌────────────┐
│ F6 Waterfall│ │ F7 Step resp.│ │ F8 Displ.range│ │ F9 Quick Repeat│ │ F10 Functions│
└────────────┘ └────────────┘ └────────────┘ └────────────┘ └────────────┘
```
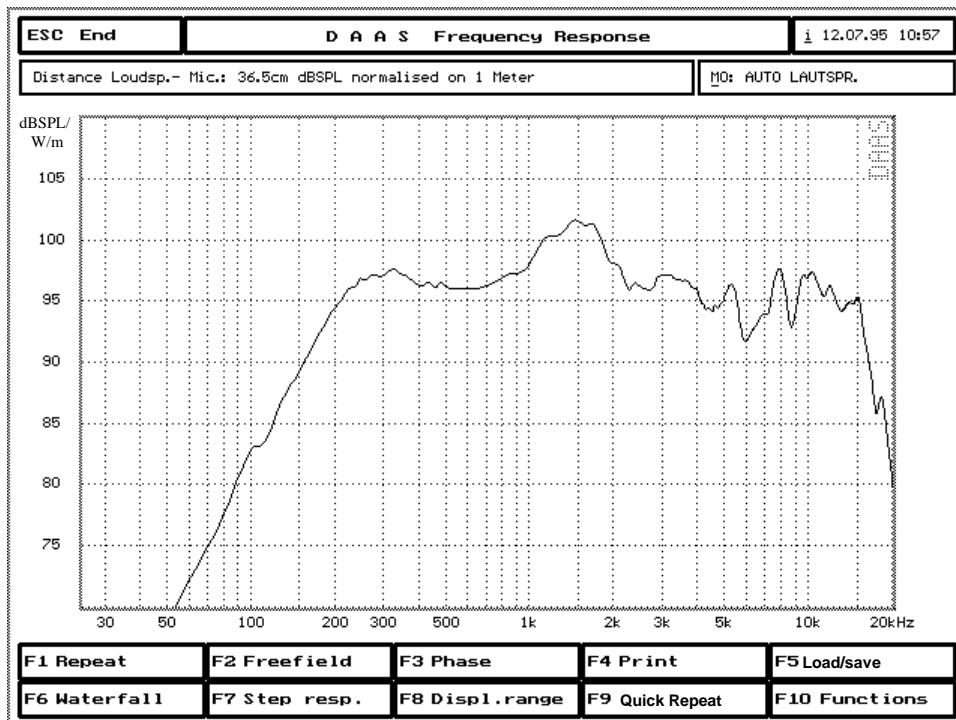
The scale can be altered under the "Options" menu. In this example dBSPL/W/m normalized to 1 meter has been chosen. This means that the curve is shown corrected as if the measuring distance was 1 meter and with a signal level of 1 Watt (the load resistor for the power calculation can also be set in the "Options" menu). 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 [F10] - [F2] 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]. If you are testing speakers which does not have the full range from 20 Hz to 20 kHz this feature should be used.

With [F10] - [F4] the curve can be smoothed. The smoothing is approx. one Terz (1/3 octave).
Frequency curves can also be saved and re-loaded via the [F5] key. The curve will be loaded ontop 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.

Using [F1] you can repeat the measurement like described before. Using [F9] you can repeat the measurement very quickly without choosing the input range etc. The via [F9] measured frequency curve will be inserted into the existing diagram using a new color. Note that if you use [F9] 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.

Using [F4] a screen copy can be printed on a 9- or 24- pin matrix printer, on a laser printer or a PCX file created.

Phase response curve

From the frequency response curve other properties of the object can be calculated. By pressing [F3] or using the mouse the phase response will be calculated and displayed.
If the automatic delay correction is successful or the distance from loudspeaker - Mic is entered, the following is shown :
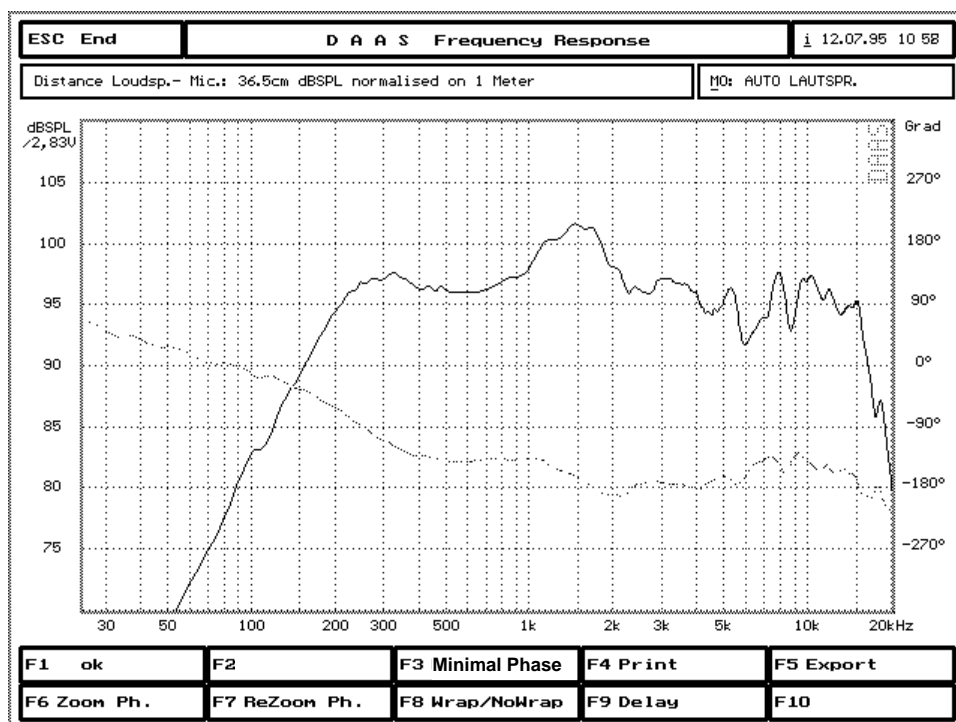
If the option 'Wrap/NoWrap' is activated (key [F8]), the phase response curve will be wrapped at ±180°.
The scaling of the phase response curve can be changed via the keys [F6] and [F7].

In order to differentiate between the lines, particularly for monochrome screens and printouts, the phase curve line style can be changed under the "Options" menu.

Minimum phase

Using the [F3] 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, 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").

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.

Group delay

From the phase response or the minimum phase curve we can determine the group delay curve. The shown group delay curve

```
┌─────────────────────────────────────────────────────────────────────┐
│ ESC End          │    D A A S  Frequency Response    │ i 12.07.95 10 58 │
├──────────────────┴───────────────────────────────────┴──────────────────┤
│ Distance Loudsp.- Mic.: 36.5cm dBSPL normalised on 1 Meter │ MO: AUTO LAUTSPR. │
└──────────────────────────────────────────────────────────────────────────┘
```

dBSPL /2,83V   Delay: 1.1ms + Kurve
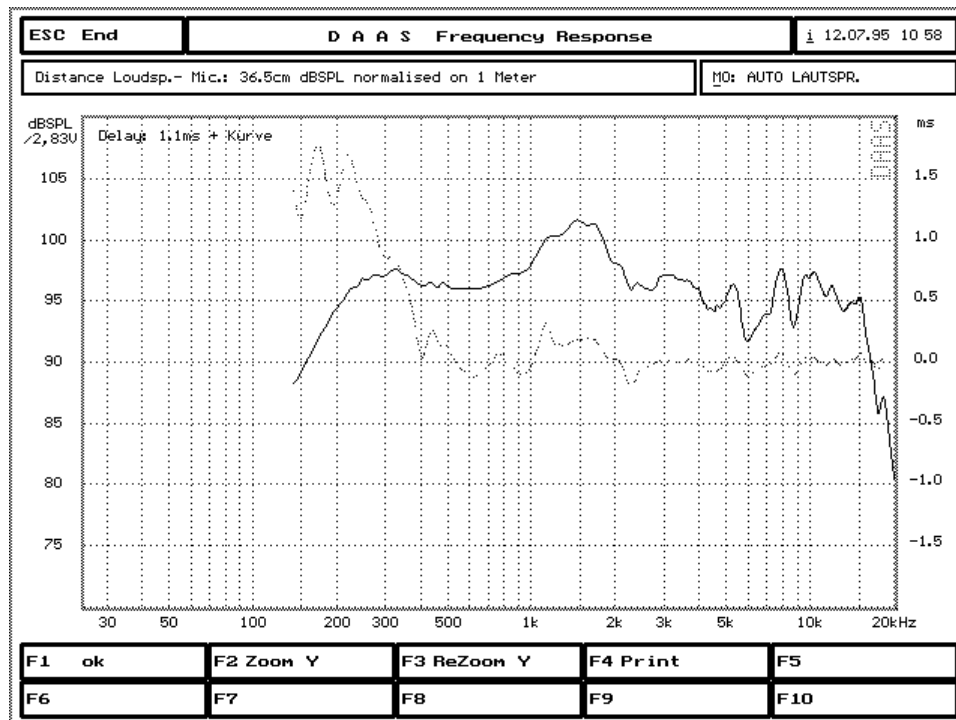
is the deviation from the via automatic delay compensation determined group delay.

Step response

From the measured complex frequency curve it is possible, by using an inverse Fourier transformation and integration, to calculate the step response. The step response is the sound pressure level response a loudspeaker produces when driven by an on-switched direct-current-level.
For clarity the time axis can be zoomed (Zoom In t and Zoom Out t).
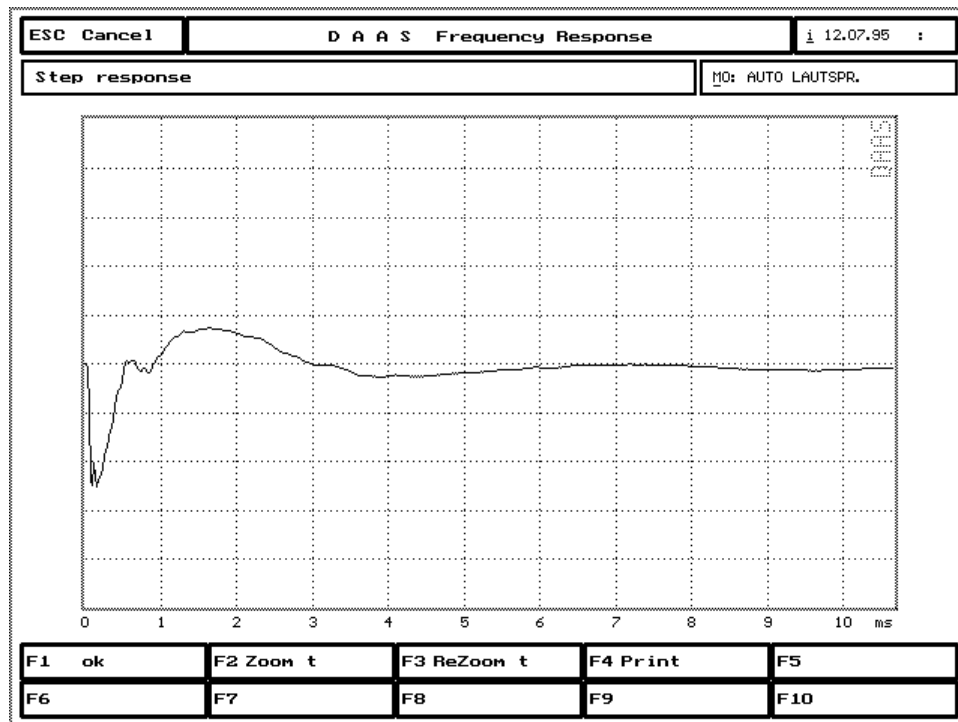
Free field correction

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

The "real" frequency response can only be obtained in a 'free field' measurement or in an appropriately sized 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 acoustic chamber is a function of it's size:

low reflection room

larger than the wavelength of the lowest frequency

microphone

object
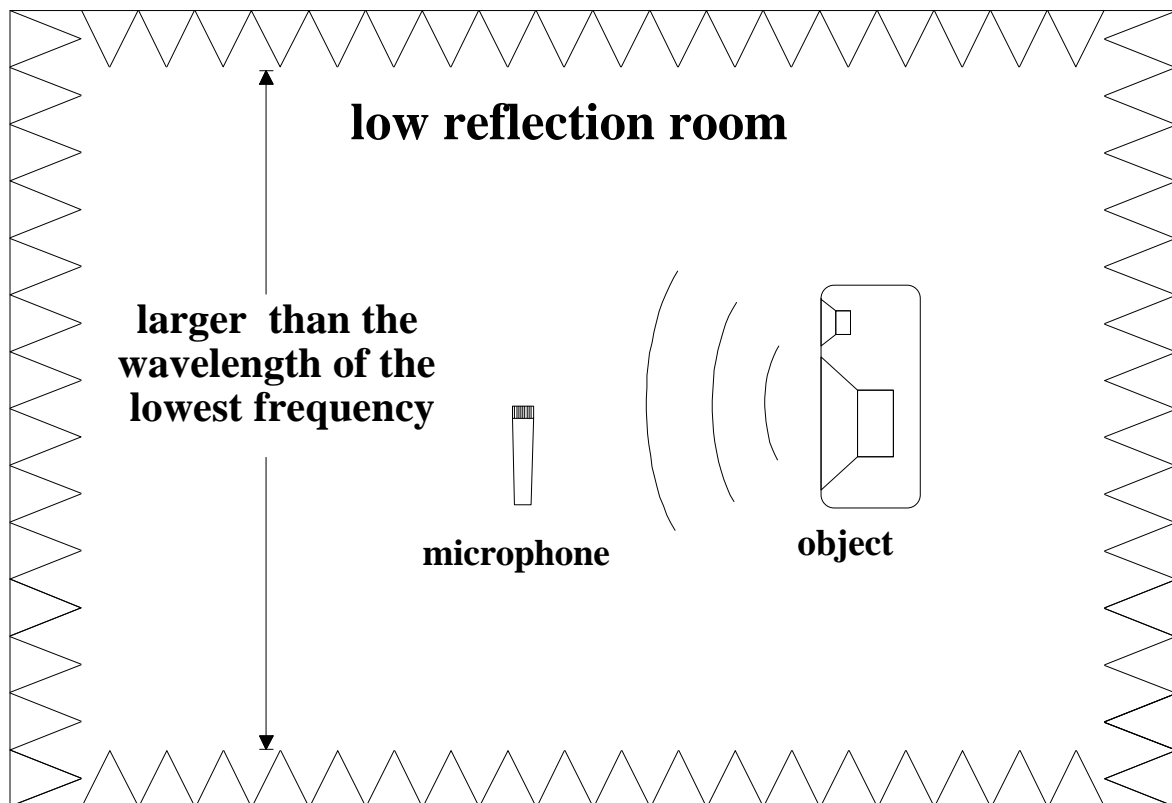
For a measurement down to 50 Hz the minimum size is 6,8 Meter (!) nett. The absorbent wedges must be 0,25 * wavelength i.e. 1.7m. This gives a total of around 10m !!!

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 - 20 cms. The room should not exhibit any 'hall' effects (echo).

microphone   **object**
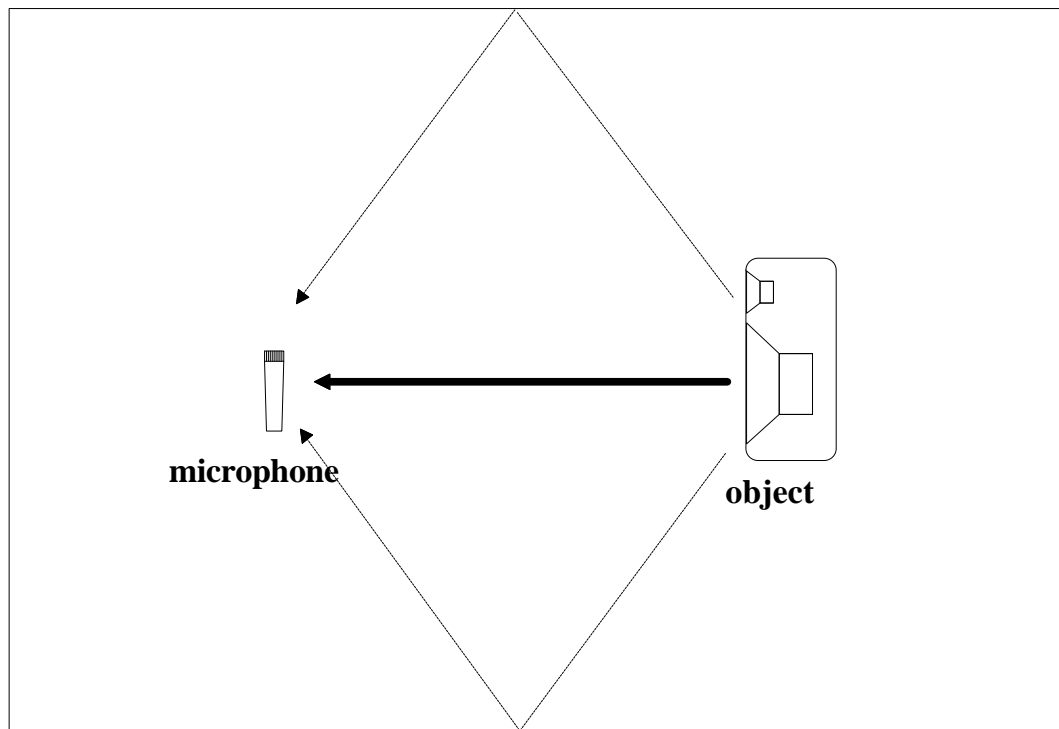
The undesirable reflections can be removed with the function [F2] "free field frequency curve calculation"

This function can be calculated from the speakers impulse response to a small needle impulse. In the wafeform of the impulse response it is possible to see any echoes and remove them. Search in the area of 1 to 10 milliseconds for the portion of signal which stops at zero milliseconds. Move the marker (small vertical line, shown in the diagram below near 4 milli-seconds) with the cursor keys([←], [→], [Ctrl]+[←] and [Ctrl]+[→]) infront of this portion of the signal. All echoes to the right of the marker will be removed.

```
ESC                    D A A S   Frequency Response        i 12.07.95 10 59

Calculated Impulse response                              MO: AUTO LAUTSPR.
```

Please move the marking point to the
left side of the first room echo
and press RETURN or accept the default.

```
F1 Fertig      F2    ◄       F3    ►        F4 Zoom Y      F5 ReZoom Y
F6 ETC         F7    ◄◄      F8    ►►       F9             F10
```
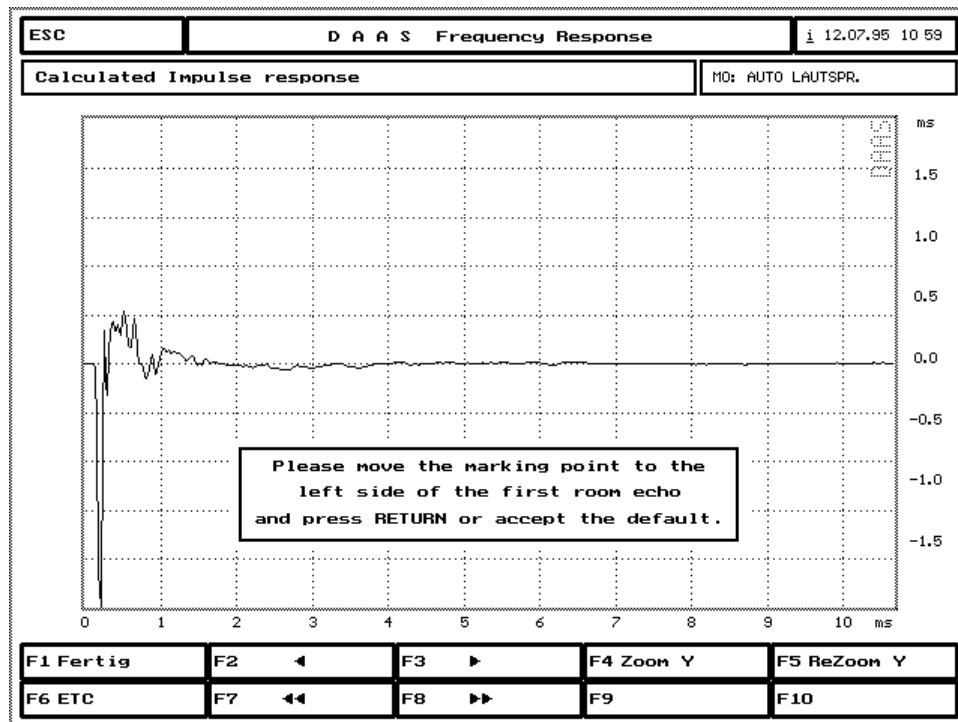
Note that cutting out the echoes reduces the frequency resolution. The resolution is a result of 1/measuring time. If the marker is position at 5 milliseconds the resolution will be 200 Hz.

If you are unsure accept the suggested setting and by pressing [F1]. Now the frequency response will be calculated from the impulse answer and cleared of errors due to reflections.

The frequency range displayed always begins at 100 Hz despite the reduced frequency resolution. The area which is likely to be of little use due to the reduced resolution is marked by a line on the frequency axis scale.

```
┌─────────────────────────────────────────────────────────────────────┐
│ ESC End          D A A S  Frequency Response     i 12.07.95 10 59    │
├─────────────────────────────────────────────────────────────────────┤
│ Anechoic frequency response            MO: AUTO LAUTSPR.             │
└─────────────────────────────────────────────────────────────────────┘
```



```
┌──────────┬──────────────┬──────────┬──────────┬──────────────┐
│ F1    ok │ F2 Waterfall │ F3 Phase │ F4 Print │ F5 Export    │
├──────────┼──────────────┼──────────┼──────────┼──────────────┤
│ F6       │ F7           │ F8       │ F9       │ F10 load/save│
└──────────┴──────────────┴──────────┴──────────┴──────────────┘
```
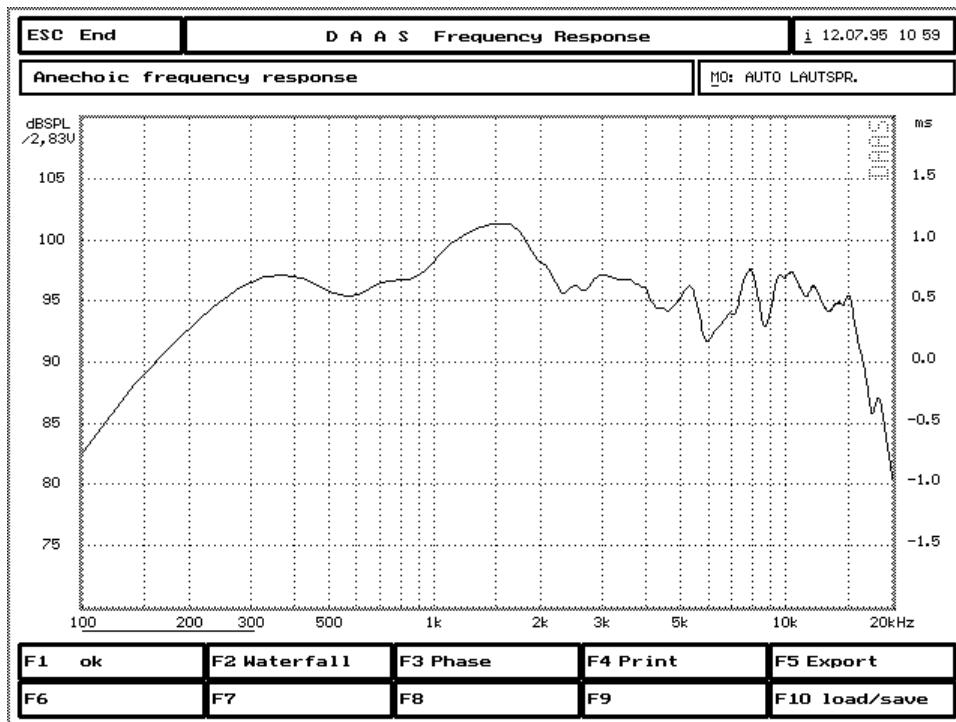
The curve can be saved using [F10] = "load/save" and re-loaded at a later date.
A corrected phase response is also available under [F3].

Waterfall diagram

From the above corrected frequency curve it is possible to calculate the waterfall diagram. This shows, for all frequencies, the speakers cumulative spectral decay. It gives indications of undesirable resonance's in enclosures and drive units.

Altogether between 48 and 72 curves are calculated. The first curve is the frequency response with the test signal at the input of the speaker, all others are without. An ideal loudspeaker would have the second curve under the frequency axis i.e. it would radiate no sound as no signal is present on the input. On a real loudspeaker this is however not the case.
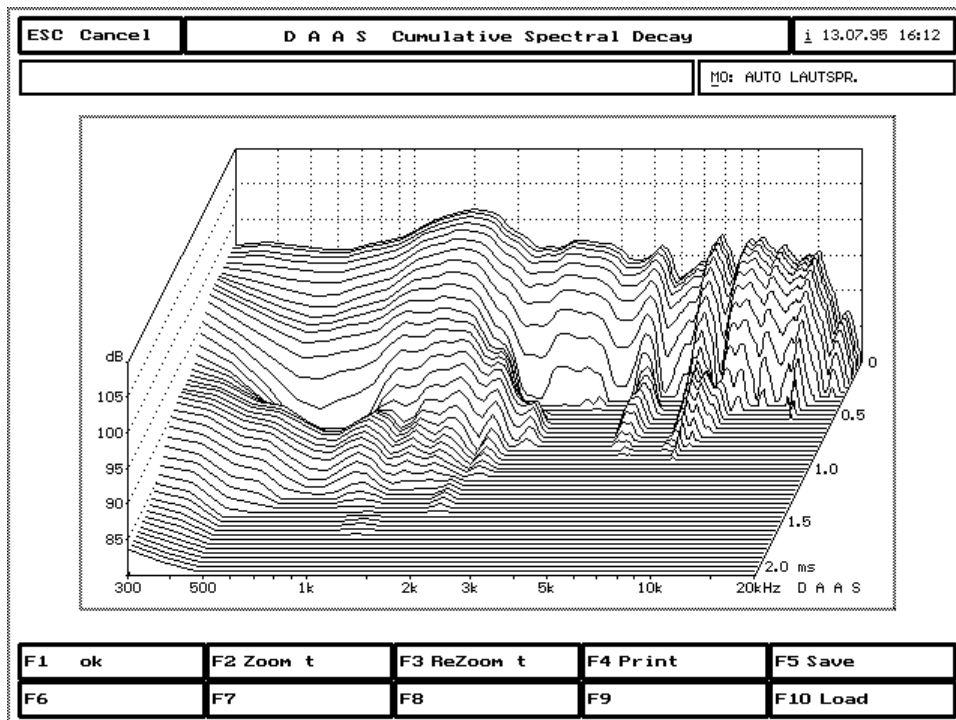
Here too the time axis can altered with Zoom In t and Zoom Out t. However by using the zoom function all the curves are recalculated. The default setting for the time axis can be changed.

The lowest frequency shown (in this example 300 Hz) can, within the 200 to 2000 Hz range, be changed, when, for example, measuring tweeters.

Using [F5] and [F10] the waterfall diagram can be saved and loaded. When a waterfall diagram is loaded the current screen is deleted.

If you require to use your own software to create waterfall diagrams for use with DAAS3L, information as to the data format necessary can be supplied on request.

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.

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 the your results are obviously wrong, check the signal to noise ratio using the level measurement. Measure the level without changing the set-up again then disconnect the speaker and measure again. Check the Terz 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 10 -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.

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 on the same amp) the input signal, in this case on the amp output, need only be measured and saved once. Do not change the amp settings after carrying out the reference measurement

Export

Frequency and phase curves can be saved for future use by other user programs, speaker CAD programs etc. Available formats are AudioCad, CALSOD, Akabak® and the hifisound database. The files can be read by any text editor (except the hifisound database).
Press [F10] - [F1] to activate the export function.

The name of the export file is made up of the first three letters of the object manufacturer's name and the first five letters of the objects name. Press [Alt]+[M] or click the mouse on MO field and enter the name.

The export of electrical-acoustical frequency response curves always has a scaling of dBSPL/W/m normalized to 1 Meter.

Combine with near field measurement

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 combination of a frequency response curve and a near field curve. So you can get a complete frequency response curve without the problems mentioned in section "Free field correction".

To combine curves do as follows:

1. Carry out a frequency response measurement under usual conditions,
2. Place the microphone close to the speaker (near field measurement),
3. Press [F10]-[F6] to start the function "Combine with near field measurement",
4. Enter the frequency you wish to combine the two curves.

Now measurement is repeated and the frequency response curve values below the chosen combine point are replaced by the new measured values.

The combined curve can be saved and loaded like each other curve.

Measure and insert impedance curves

Especially for QC measurements it may necessary to measure and display frequency response curves and the impedance curves together. You can do this with the Q-DAAS system like described below.

Preparation: Carry out a complete impedance curve measurement and set the display range (Ohm/Div) to appropriate values. This settings are used later.
Quit the impedance measurement, short or remove the serial resistor, and start the frequency response measurement.

After complete the frequency response measurement, insert the serial resistor again and press [F10] - [F5] to start the impedance curve measurement. The Y-Axis adjustment (Ohm/Div) used by the impedance measurement before is used now.

It is highly recommended to use a switch box to insert respectively to short the serial resistor.

Comparison of Microphones

For comparison of microphones (measurement acoustical-acoustical) it is possible to correct the absolute level of the resulting curve.
To use this level correction do as follows: Create an ASCII-File "MIC-CORR.TXT" in the Q-DAAS folder. This file must consist on two lines. Each line contain one correction factor. The two correction factors are added to every measurement acoustical-acoustical.
Example for "MIC-CORR.TXT":
-5
3

If this file does not exists, no value will be added. The correction values are displayed in the upper part of the screen.

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:

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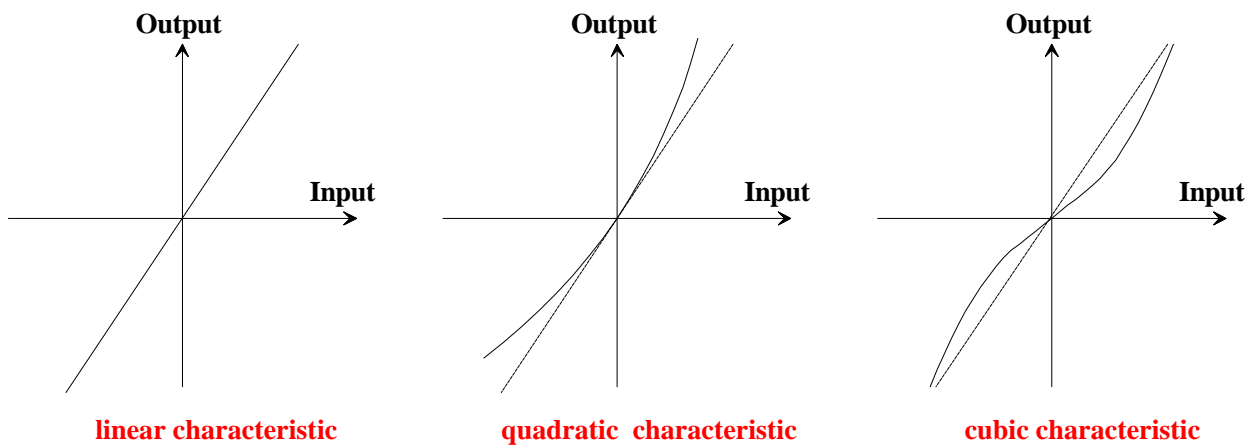
Generally gain characteristics are made up of elements of linear, quadratic and cubic characteristics.
When a sine wave of frequency f is distorted, overtones, or harmonics, are created which were not present in the original signal. Quadratic characteristics create only even (2*f, 4*f etc.) and cubic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

Non linear distortions are usually measured by one of two methods: using the harmonic distortion or the intermodulation factor.

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. 6 kHz. Above 6 kHz the third harmonic (and the most particularly damaging harmonic) lies outside the audio frequency range.

The Digital Audio Analysis System has a sine wave of variable frequency in the range 20 Hz to 6 kHz for the measurement of harmonic distortion, (under 100 Hz the sampling rate must be changed).

The set-up for measuring a speaker is as follows:

object

amplifier

DAAS 3 L
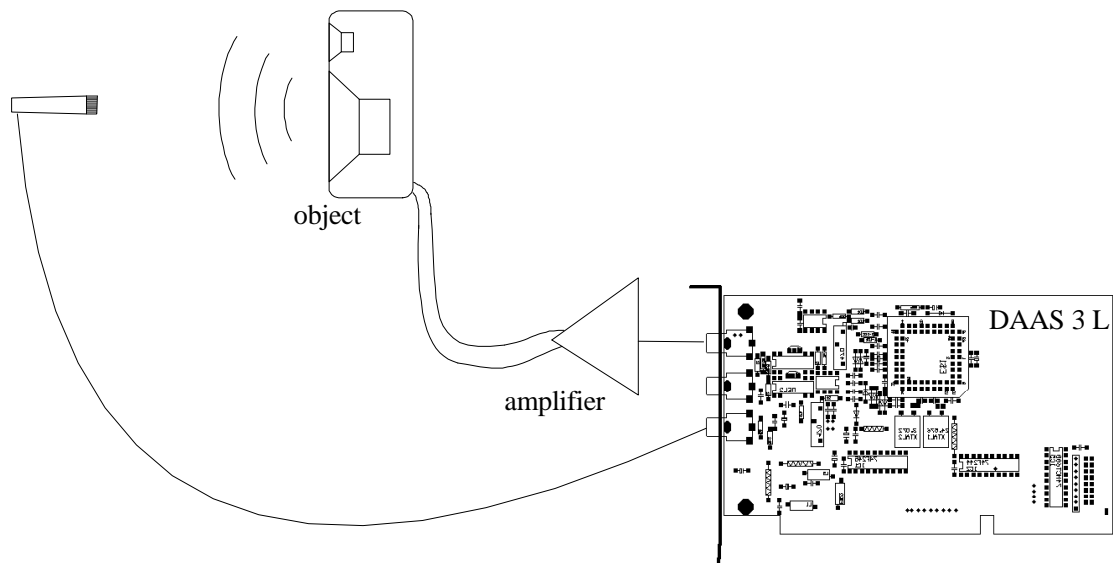
Choose harmonic distortion measurement in the main menu using [F4] - [F1] and then using [F2] the input (Line = electric, Mic = acoustic, (see display in top left of info box) and the measurement range (±0.2 V, ±2 V or ±20 V i.e. ±2 Pa or ±8 Pa), with [F3] the output level attenuation and with [F4] the measuring frequency. Then [F1] = Start. After the measurement (short sine tone) the measured signal is displayed. If necessary change the output level and re-measure until the input level range occupies approx. 1/2 of the display. An ideal setting is as follows:-
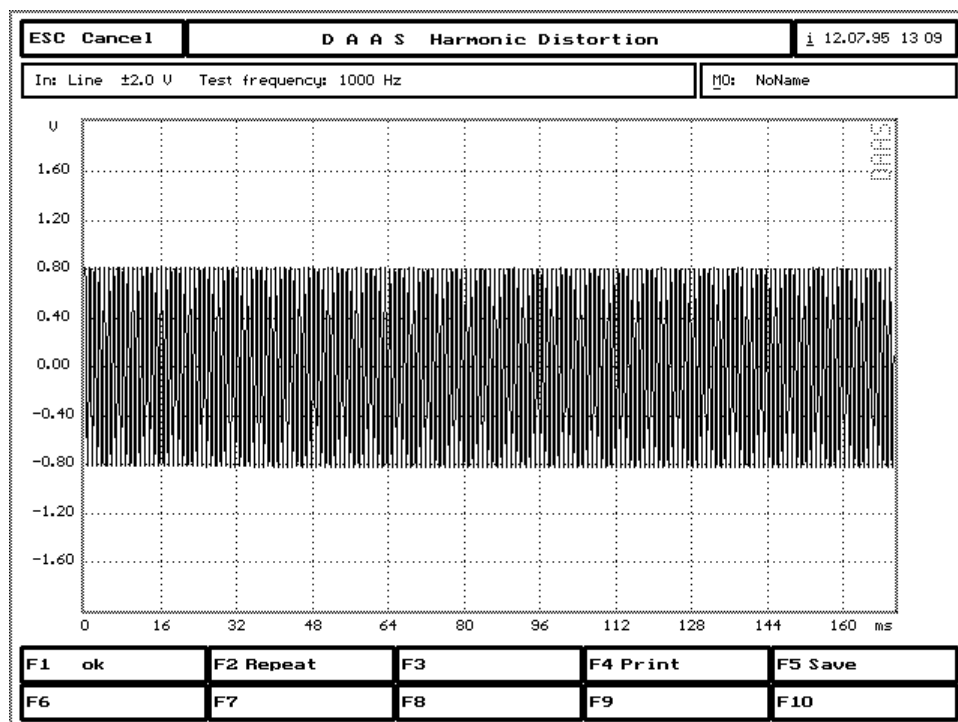
If the signal is too large, a warning appears if too low, accuracy in the lower range will be lost.

Use [F5] to save or load the signal at this point.

Press [F1] for "measurement OK", and the harmonic distortion will be calculated.

```
┌─────────────────────────────────────────────────────────────────────┐
│ ESC  End              D A A S   Harmonic Distortion      i 12.07.95 13:09 │
├─────────────────────────────────────────────────────────────────────┤
│ k2: 0.005%,  k3: 0.004%,  k4: 0.007%,  k5: 0.002%    │MO:  NoName       │
├─────────────────────────────────────────────────────────────────────┤
│ dBV                                                                   │
│                                                                       │
│  -5                                                                   │
│                                                                       │
│ -20                                                                   │
│                                                                       │
│ -35                                                                   │
│                                                                       │
│ -50                                                                   │
│                                                                       │
│ -65                                                                   │
│                                                                       │
│ -80                                                                   │
│                                                                       │
│ -95                                                                   │
│                                                                       │
│   100   200  300   500     1k      2k    3k    5k     10k   20kHz      │
├─────────────────────────────────────────────────────────────────────┤
│ F1 Repeat    │ F2 Zoom Y  │ F3 ReZoom Y │ F4 Print  │ F5 more...     │
├─────────────────────────────────────────────────────────────────────┤
│ F6           │ F7    ▲    │ F8    ▼     │ F9        │ F10            │
└─────────────────────────────────────────────────────────────────────┘
```
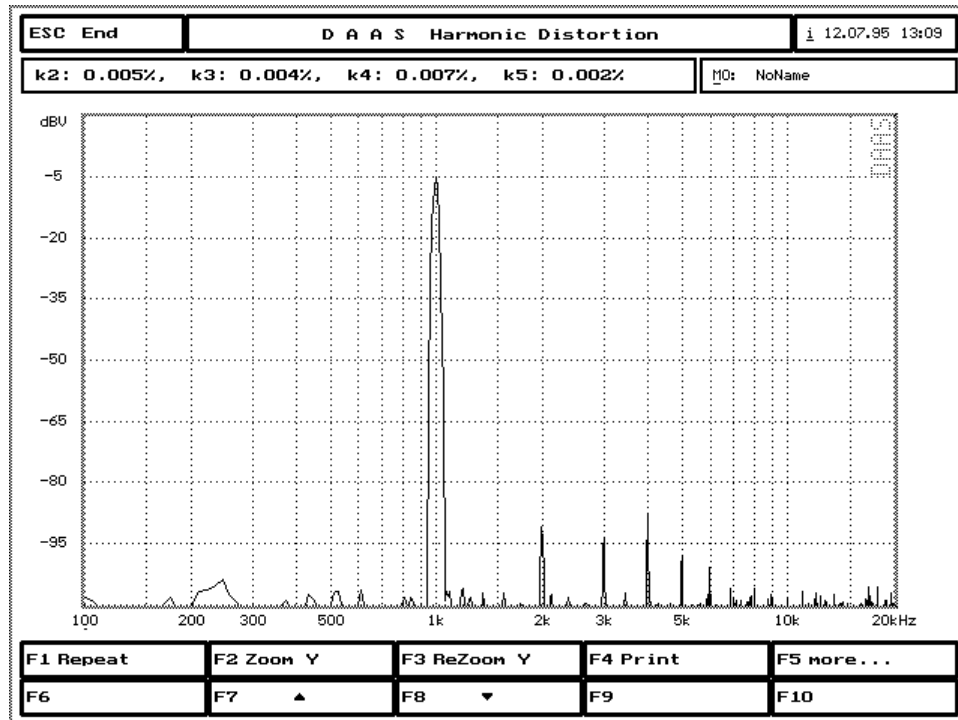
The following important factors will be displayed:-

q k2: 2nd Harmonic distortion.

q k3: 3rd Harmonic distortion

q k4: 4th Harmonic distortion

q 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. Total harmonic distortion is calculated as follows:-

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

6th harmonics and higher do not affect the measurement in practical terms. Therefore everything above the fifth is discarded.

Use [F2] / [F3] to change the display size and [F7] / [F8] to move.

CAUTION: for the harmonic distortion measurement it is not possible to use external signals e.g. from a sine wave generator.

Harmonic distortion curve

Often, just using one frequency for determining harmonic distortion is not enough. DAAS enables up to 20 points to be used in the range 100 Hz to 8 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] = "Measuring frequency" to set the frequency limits as opposed to the measurement frequency. The number of points is then determined by the specified frequency range.
On starting the measurement the following screen is shown:
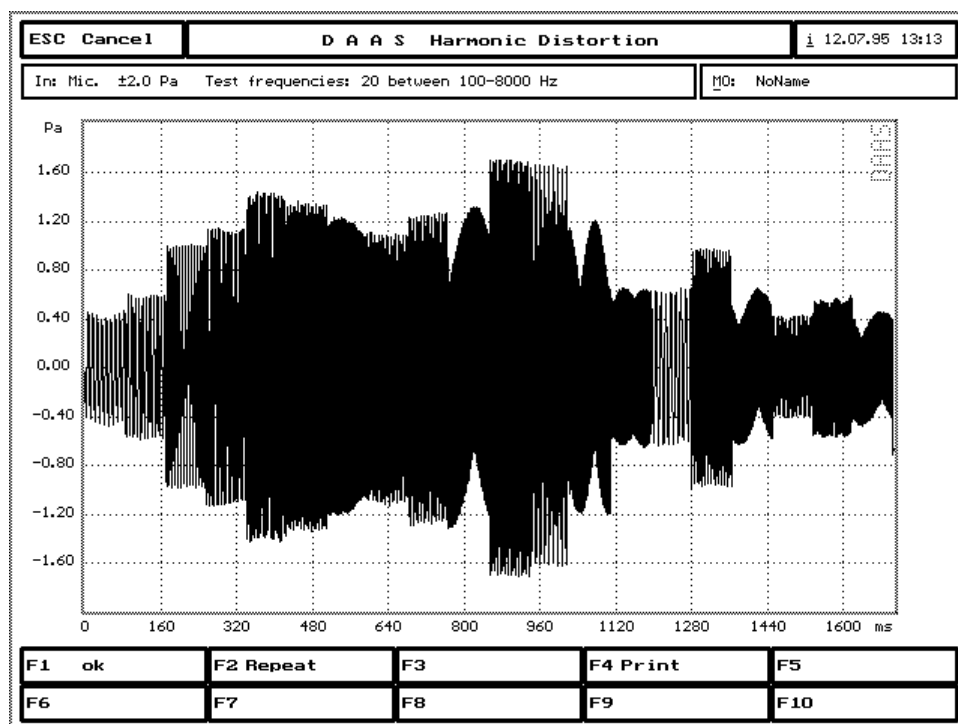
The signal tones are combined to form a single picture. Left are the low frequencies and right the highest. Press [F1] to calculate the harmonic distortion curve:-

```
┌─────────────────────────────────────────────────────────────────────┐
│ ESC Ende │          D A A S  Harmonic Distortion          │ i 12.07.95 13 14 │
├─────────────────────────────────────────────────────────────────────┤
│ In: Mic.  ±2.0 Pa    Test frequencies: 20 between 100-8000 Hz │ MO:  NoName │
├─────────────────────────────────────────────────────────────────────┤
│  %                                                                    │
│      ┌─ -- K2 (2. Harmonische)                                        │
│      └── K3 (3. Harmonische)                                          │
│                                                                       │
│ 3.16                                                                  │
│                                                                       │
│                                                                       │
│ 1.00                                                                  │
│                                                                       │
│                                                                       │
│ 0.32                                                                  │
│                                                                       │
│                                                                       │
│ 0.10                                                                  │
│                                                                       │
│                                                                       │
│ 0.03                                                                  │
│                                                                       │
│                                                                       │
│ 0.01                                                                  │
│       100        200   300      500        1k        2k   3k    5k  8kHz │
├─────────────────────────────────────────────────────────────────────┤
│ F1 Repeat │ F2      │ F3      │ F4 Print │ F5       │
├─────────────────────────────────────────────────────────────────────┤
│ F6        │ F7      │ F8      │ F9       │ F10      │
└─────────────────────────────────────────────────────────────────────┘
```
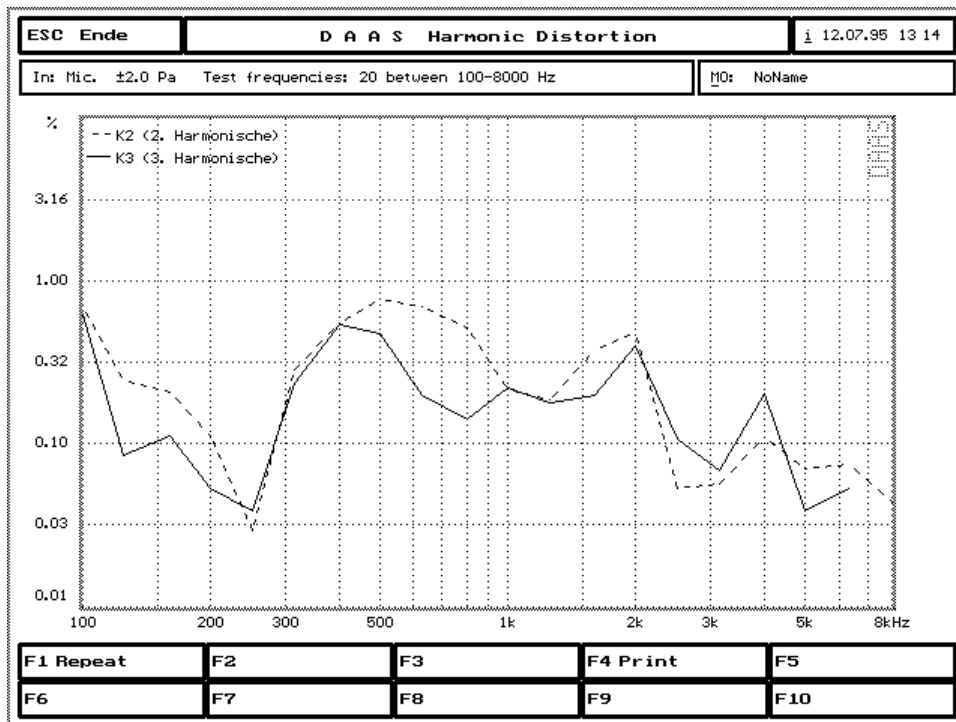
Second and third order curves are shown.

The number of points can be increased, with a resulting increase in memory, hard drive and calculation time requirements. Details can be supplied on request

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 intermodulations factor measurement is identical to distortion measurement.
Q-DAAS provides a composite signal for the intermodulation factor measurement.
The signal comprises two tones with an amplitude relation of 4:1. The tone with the largest amplitude can be selected within the range 100 Hz to 1000 Hz. The second tone has a fixed frequency of 4 or 5 kHz:
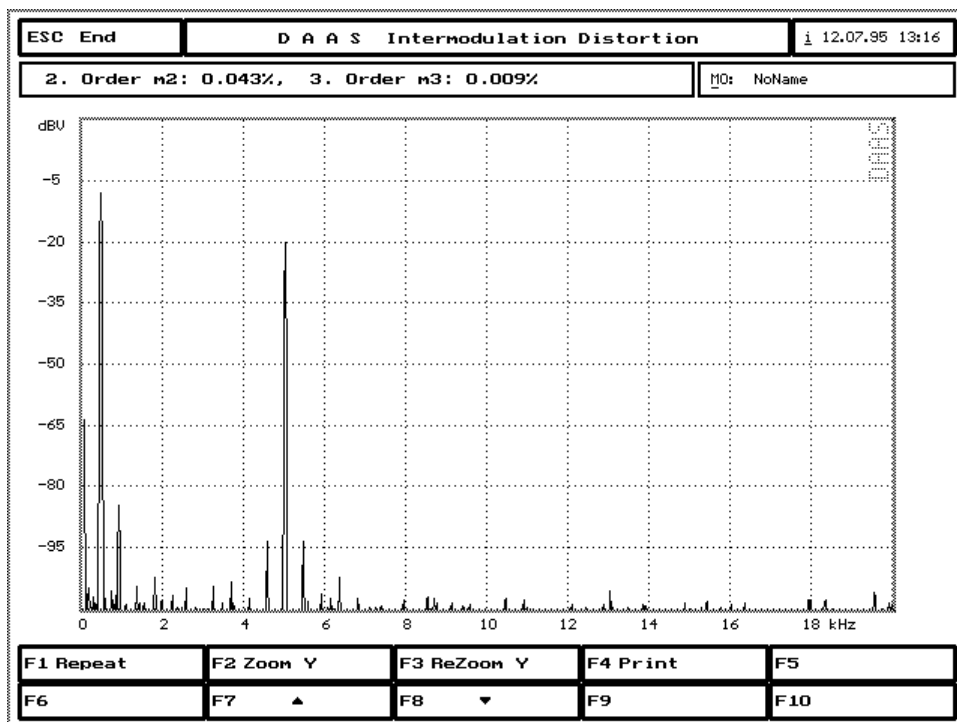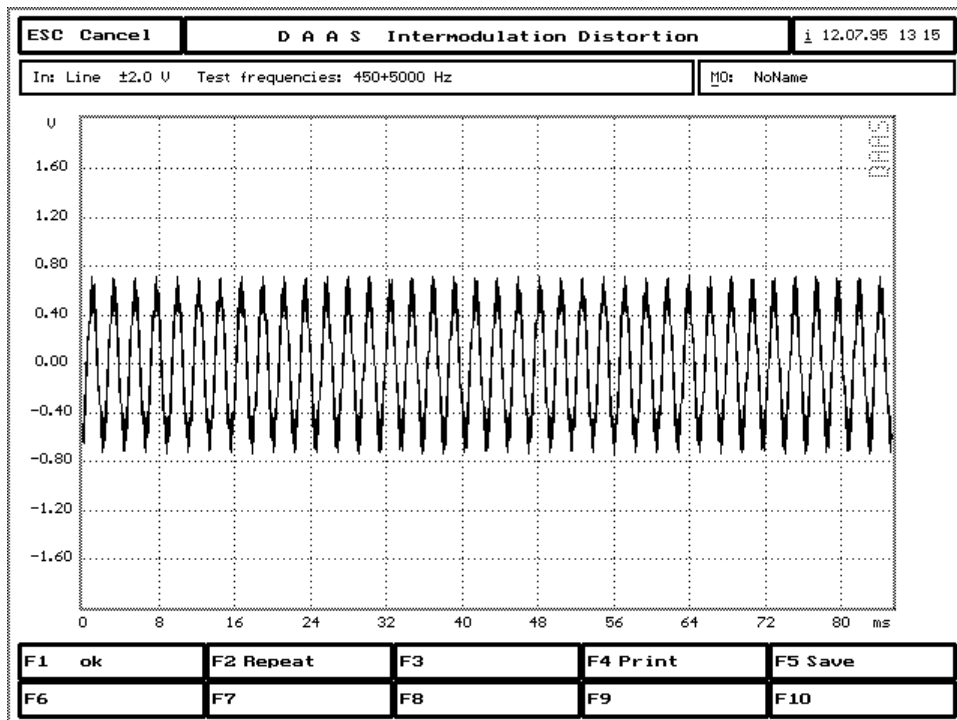
Only the second and third order harmonics are used in the evaluation.
The evaluation is carried out as recommended in DIN 45403.4.
The lower measuring limit in comparison with distortion measurement is slightly higher.
This is because the DIN recommends an amplitude ratio U1:U2 of 4:1, which would
otherwise result in U2 being 12 dB under the required level.

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

Impedance measurement

Using the Digital Audio Analysis System Q-DAAS it is possible to measure the impedance response and the associated phase response of a speaker or crossover.
As with the frequency response measurement this is carried out in two stages.
For the reference measurement connect the output "Out" via a 600 Ω ($\pm$1%) resistor to the speaker. Connect "Line" input between "Out" and the resistor. Set the output level attenuation to 0 dB. Start with [F1].
After successful reference measurement press [F1] to start the spectrum analysis.
Now connect the input "Line" to the loudspeaker as follows:

object
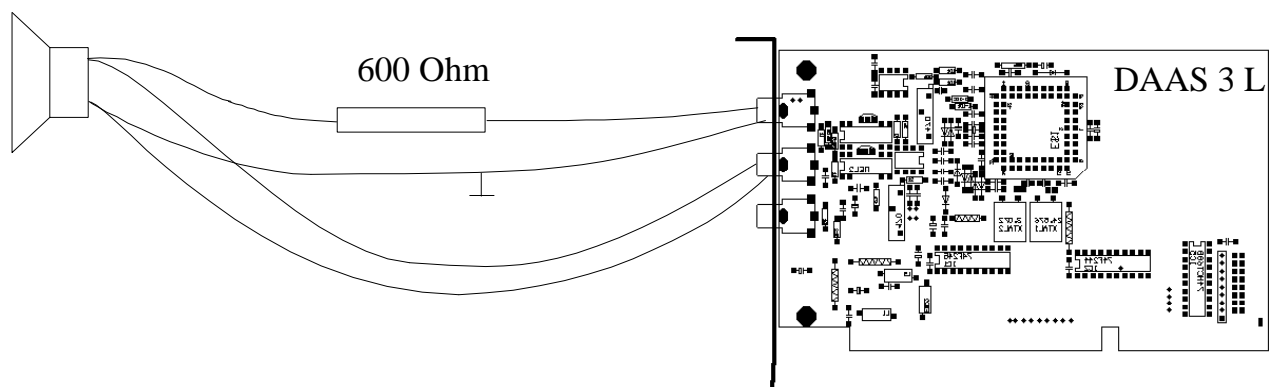
Connect the input "Line" directly to the object as shown to avoid measuring the cable!
Now start with the actual measurement.
Generally the 0,2 V range will be used. Only when the resistance is more than about 100 Ohm a warning will be given and then the 2V range will have to be selected..
The signal should be as follows:
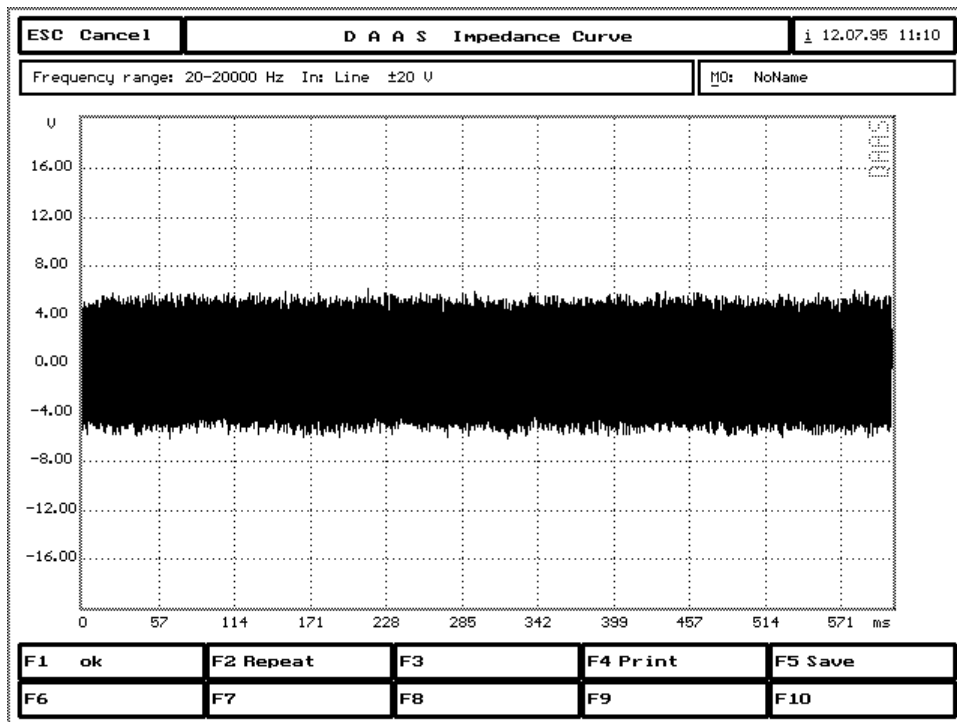
As before the signal can be saved or loaded with [F5].

If the signal is of the right magnitude press [F1] to start the calculation and display the impedance curve.
A test measurement on a 960 µH coil gave the following impedance curve.
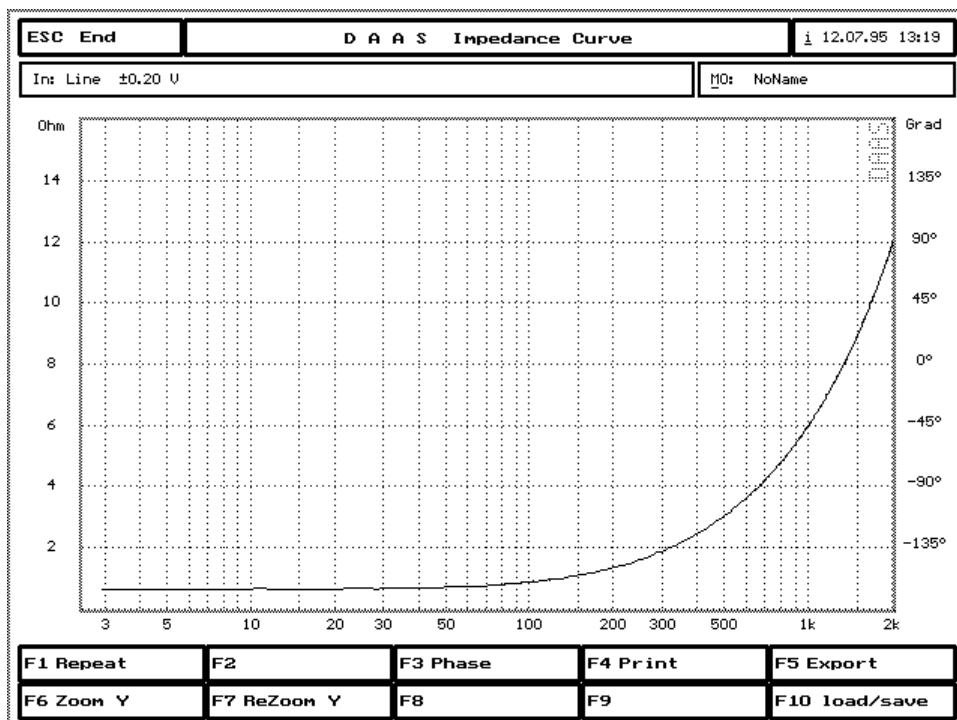
Using [F6] and [F7] the scaling can be changed from 1 Ohm/Div to 32 Ohm/Div.
Press [F3] to calculate and show the phase curve.
Key [F10] is labeled "load/save". Using these keys the impedance curve can be saved
and later loaded. A curve load at this point will be shown ontop of the current curve i.e.
the existing curve/s will not be deleted.

Using [F10] the resistor value can be changed (Default = 600 W).
CAUTION: This resistor setting is used in the impedance calculation. If signals are
loaded which have been measured with different resistors, inaccurate impedance values
will be shown!
When using resistors less than 600 W a measuring amplifier should be connected
between "Out" and the resistor. In this case set the output level attenuation to 20 dB.
Set the amplifier volume control so that for the reference measurement the signal peaks
are at least 5V.

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. The following diagram shows
the impedance and phase curve of a 960 µH coil and the calculated inductance.
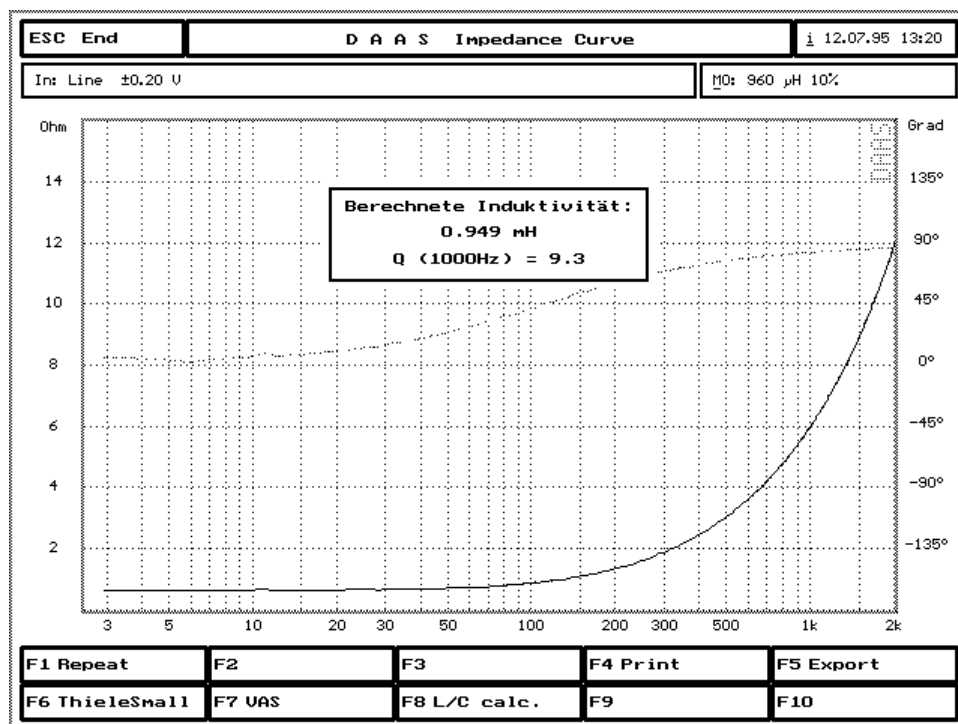
The range for inductance is approx. 6 μH to about 10 mH and for capacitance approx. 1 μF to 100 μF. For larger values the sample rate must be reduced.

Data export

The impedance and phase curves can be exported for use in other programs, speaker CAD programs etc. The files are exported in standard Audiocad or CALSOD format, Akabak® and the hifisound database. The files can be read with any text editor (except the hifisound database).
Press [F5] to activate the export function.
The name of the export file is made up of the first three letters of the object manufacturer's name and the first five letters of the objects name. Press [Alt]+[M] or click the mouse on MO field and enter the name.

Thiele Small Parameters

Press [F3] in the main menu to start the Thiele Small Parameter measurement. The handling of the Thiele Small Parameter measurement is identical to the impedance measurement.
After successful impedance measurement the DC resistance of the driver is requested. If the resistance in the lower frequency range is (very level?) the value can be obtained from the diagram, otherwise the resistance must be measured using an Ohm-Meter. Qms, Qes and Qts plus the resonant frequency are then calculated.

```
ESC End            D A A S  Thiele Small Parameter        i 15.03.96 11 04

In: Line  ±0.20 V                                    MO:  NoName

Fms:  198.7 Hz  Res:   0.0 Ω    Nref:   0.00 %   Re:   3.4 Ω   Le: 0.1 mH
Qms:    4.03    Rms:   0.0 kg/s  Bl :   0.0 N/A  Rp:   8.9 Ω
Qes:    1.54    Cms:  0.00 mm/N  SPL :   0.0 dB  Lp:   1.8 mH
Qts:    1.12    Mms:   0.0 gr   Vas :   0.0 l   Cp: 363.3 µF
```

F1 Repeat   F2 VAS   F3 Fit   F4 Print   F5 Export
F6 Zoom Y   F7 ReZoom Y F8   F9   F10
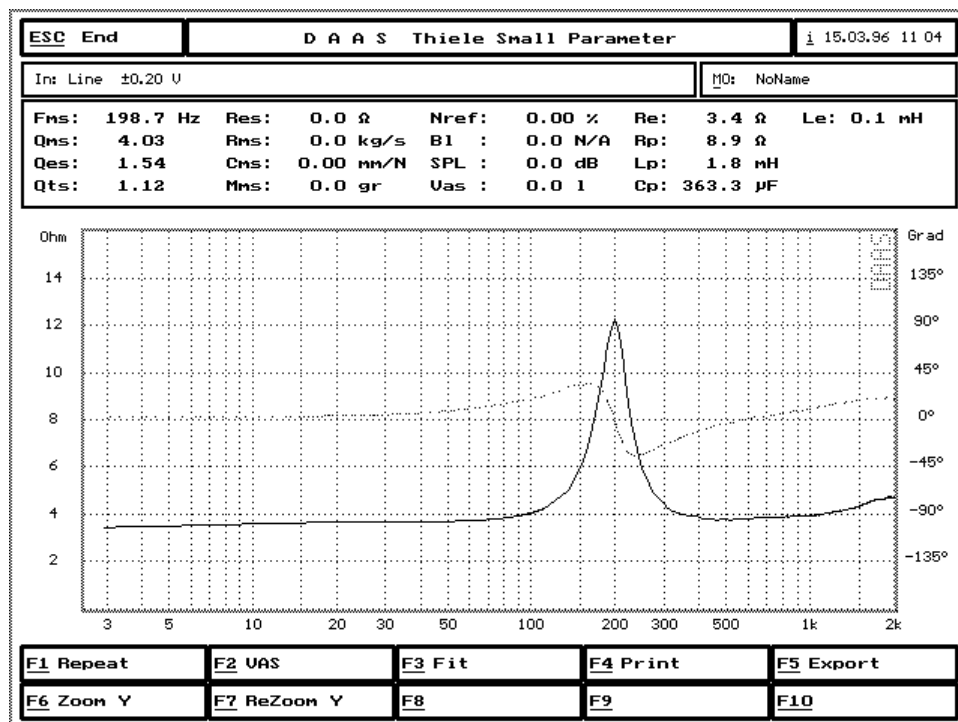
Hint: Except for tweeters it is recommended to use the sample rate 6 kHz (default).
Measuring a bass speaker it is recommended to use resistors between 100 and 470
ohms.

During Thiele Small Parameter measurements the following warnings may appear:

● "Phase zero cross point / 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. / sample rate * FFT-Length from the
Resonant freq. (Impedance maximum)). This appears mainly when a speaker shows
two resonance's very close together.

❑ "The frequency resolution is too low . . . ".
Reason: The impedance maximum for the set resolution is too low. The error is too
large, reduce the sample rate.

❑ "Resonant freq. is not equal to $\sqrt{f1 * f2}$ . . . ".
Reason: The resonant frequency is not in the geometrical center of the frequencies f1
and f2. Theoretical this should not appear, but it is not always the case.

VAS-measurement.

To calculate the equivalent air volume VAS and other characteristics such as power factor, Bl, SPL, Cms, Mms etc. the normal parameter set must first be obtained (see 0). Then press [F2] = 'VAS' and the measurement is repeated using a small weight on the speaker. Modeling clay is recommended. This should be formed in a ring and pressed around the dust cover. Use 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.

Level measurement

With this measurement you can measure the noise level and the signal to noise ratio on amplifiers and recorders (with or without A-weighting). Also dB(A) and dB 20µPa = dBSPL can be measured.
For the frequency curve measurement the measurement level can also be set. This is useful if you wish to feed a speaker with a predetermined signal-level or if you require a certain level at the measuring point (microphone position).
The measurement of noise level is very simple. Connect the object to be measured to the "line" socket and set up the microphone. Use [F2] to select the input and sensitivity. [F5] can be used to load a previously saved signal.
Press [F1] to start the measurement and the signal will be shown as follows:
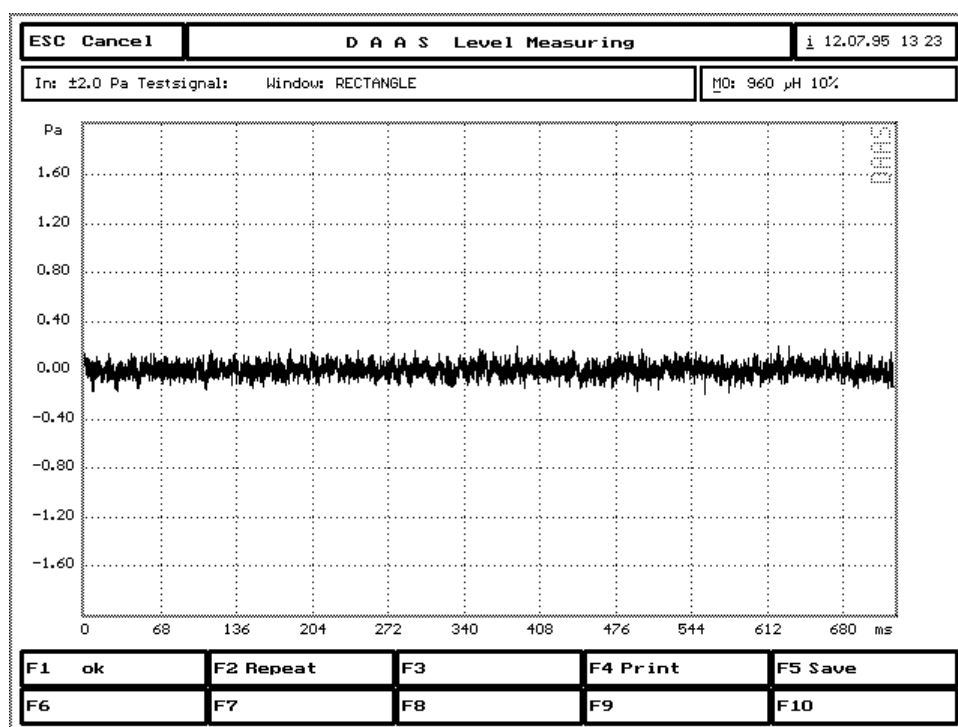
Accept with [F1] and the level will be calculated and shown.
The following diagram shows the Terz - Spectrum and measured level in a PC case.
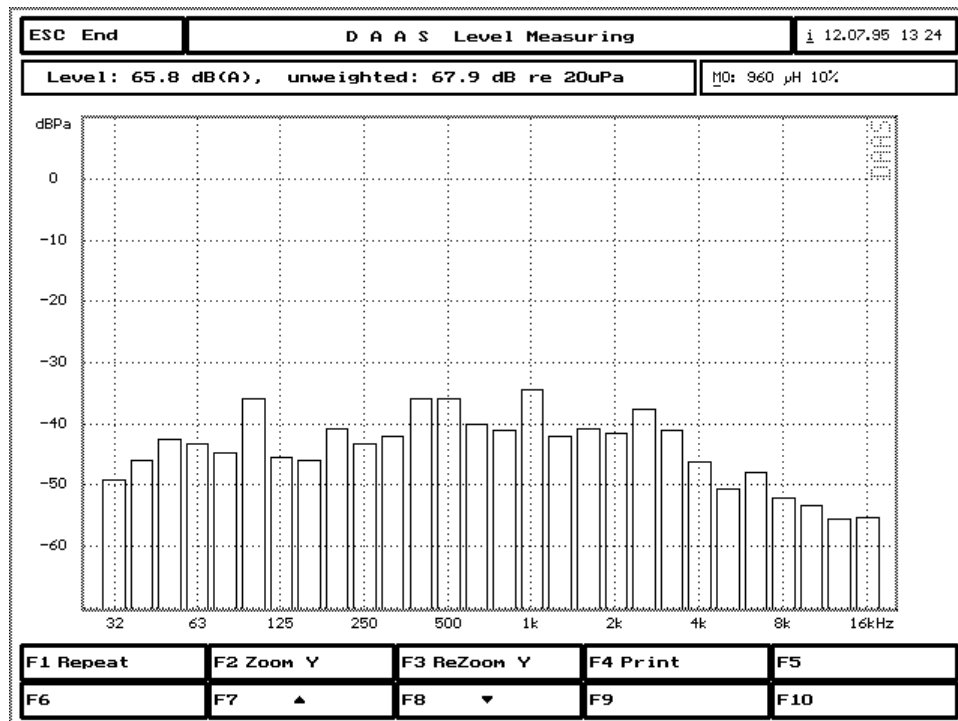
To measure the signal to noise ratio of an amplifier, feed the input with a sine wave
(say 1kHz) and measure the level on the output. (20 V range, load resistor on output!).
Increase the volume setting and repeat the measurement until the output voltage is just
about to be limited. Note the level. Switch the input signal off and measure the
amplifier noise in the 0,2 V range. The difference between the two levels is the signal
to noise ratio, A-weighted and without weighting.
Should you need other curves (B, C, D, P.53/ O.41 etc.) please contact your supplier.
The internal generator can be used as the signal generator. Set the required signal (see
under "Options") and this will appear on the "Out" socket. The level of the test signal
can be altered using [F3].
To set the measuring level for the frequency response measurement select under
"Options" the signal which is set for frequency response measurement. Connect the
amp output to "Line" (or "Out" if no amp is used) and connect the speaker and Mic.
Set the level with the amp volume control (or with F3) and repeat the measurement until
the required level is obtained. The frequency response measurement can now be carried
out with the pre-set level.

Oscilloscope

With the oscilloscope function, time signals up to 20 kHz and transient events up to a length of approx. 22 secs. can be displayed and printed.
The chosen test signal is shown synchronized with the oscilloscope.
Choose oscilloscope in the main menu with [F7]. In the following menu the measurement parameters can be set.

F1 Start	F2 Input	F3 Output	F4 Time +	F5 Test signal
F6	F7	F8	F9 Time -	F10 Sample rat

With [F2] the input sensitivity can be set, either the Mic input with ±2 Pa and ±8 Pa or the line input with ±0.2 V, ±2 V and ±20 V.
If you are using a DAAS test signal, the output can be set using [F3] or of course using the amp volume control.

The time range (number of points) can be changed with [F4] and [F9] (minimum 5 ms and maximum 2.7 secs. at 48 kHz sample rate, by reduced sample rate up to maximum 22 secs.).
The sample rate (number of points per second) can be changed from 48 kHz to 6 kHz using [F10]; this will have the effect of increasing the measurement time and frequency resolution by a factor of 8 in any future spectral analysis.
Start the measurement with [F1].
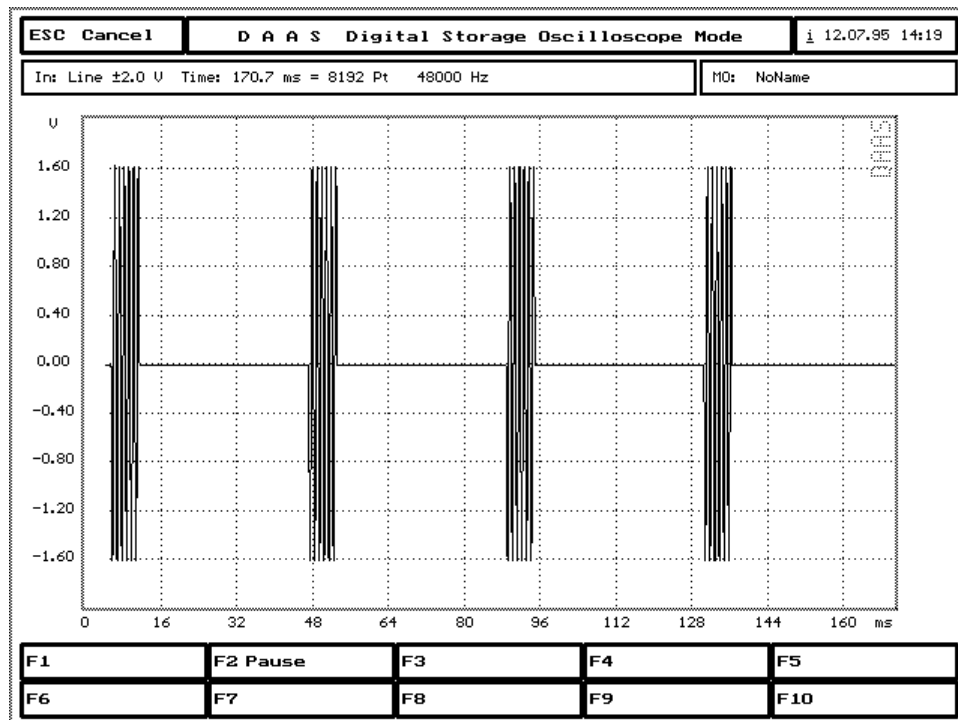
The measurement will be repeated until [Esc] is pressed or [F2] for pause.
When the system is paused, [F1] is used to continue.
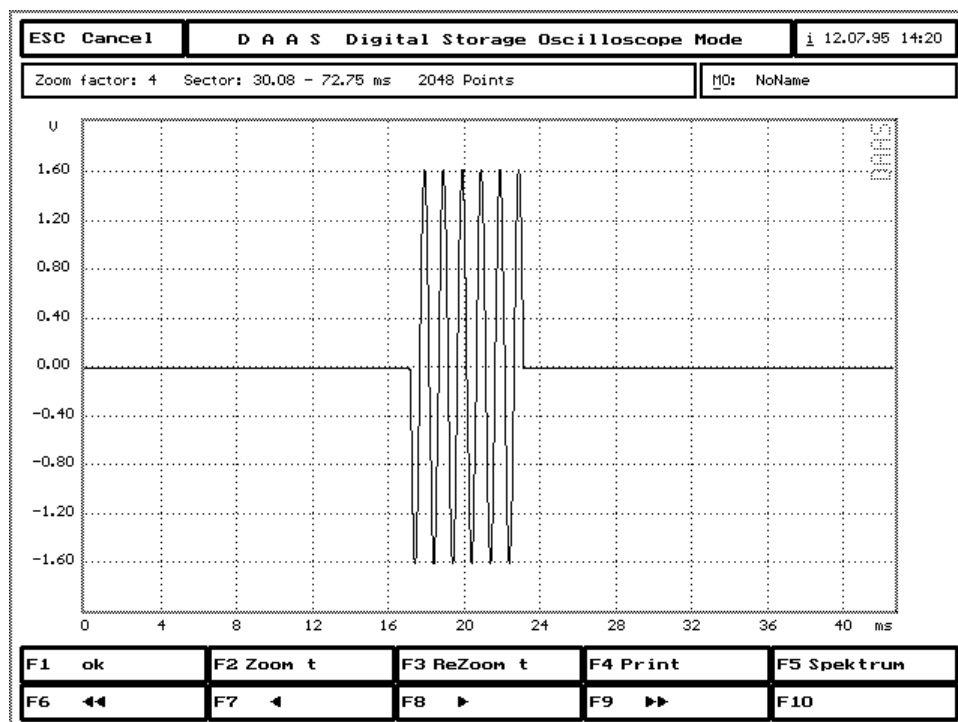
In "Pause", mode the signal can be printed using [F4]. Interesting signal sections can be
expanded and compressed using [F2] and [F3]. With [F6], [F7], [F8] and [F9] the signal

can be moved around, displayed and printed. [F7] and [F8] moves the signal one point, [F6] and [F9] half a point.

Alternatively the cursor keys ([←], [→], [Ctrl]+[←] and [Ctrl]+[→]) can be used.

The "info box" shows details as to the zoom factor and position of the signal.

Using [F5] a spectral analysis can be carried out on the chosen section of the signal.
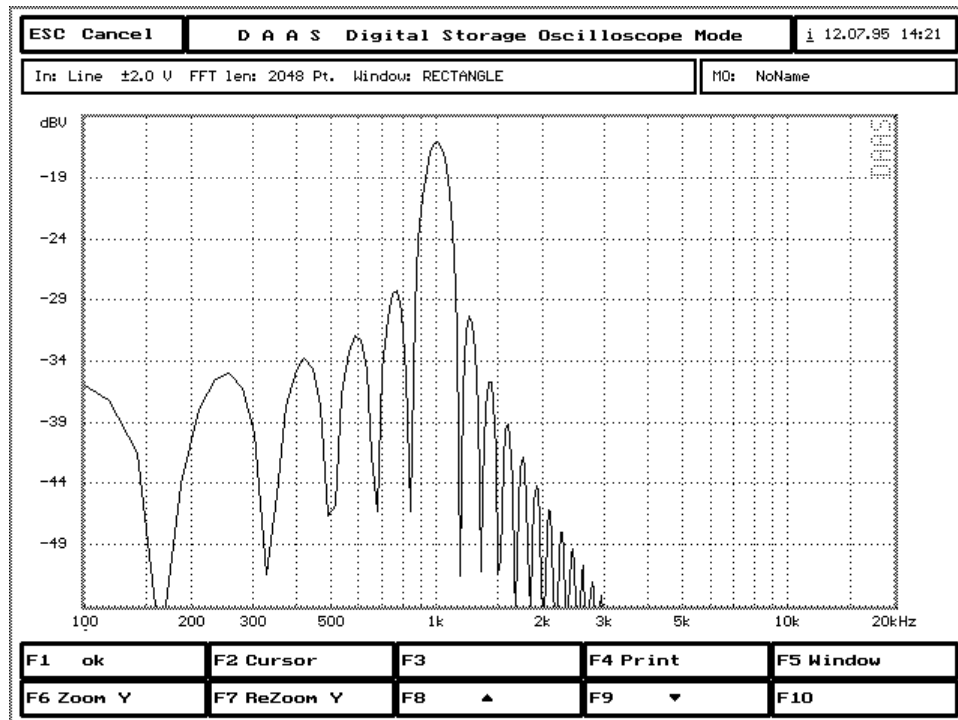
The right mouse key can be used to move the marker along the curve and read out the values for amplitude and frequency.

With [F5] the evaluation window is selected. The following windows are available.

◆ Square (for transients),
◆ Hann (for general use),
◆ Blackman,
◆ Kaiser (with B=10.056, Side-Lobe-attenuation, approx. 100 dB),
◆ Flat Top (for exact amplitude setting).

The display range can be changed with [F6], [F7], [F8] and [F9]. These functions are identical to the "Spectrum Analyzer".

Spectrum Analyzer

With the Spectrum Analyzer, time signal spectrums in the audio range to 20 kHz and transient events (up to a length of 4096 points) can be displayed and printed.
The chosen test signal is shown synchronized with the oscilloscope.
By using a special test signal and special set-up, frequency response measurements can be carried out without time delay problems and without a reference measurement.

[F8] selects the Spectrum Analyzer, and the following menu appears.

F1 Start	F2 Input	F3 Out Att.	F4 FFT-Params	F5 Test signal
F6 Zoom Y	F7 ReZoom Y	F8 ↑	F9 ↓	F10 Sample Rate

[F2] sets the input sensitivity. The Mic input has two ranges ±2 Pa and ± 8 Pa. If you are using a DAAS test signal the output can be set using [F3] or by using the amp volume control. The test signal must be selected under [F5], after which it then is synchronized with the measurement cycle on the "Out" output socket.
Using [F4] the analysis length (length of the DFT/FFT), the window function, the smooth factor and the number of averagings can be adjusted.
The analysis length can be set between 512 Points and 4096 Points.
The following windows are available.

- ◆ Rectangle (for transients),
- ◆ Hann (for general use),
- ◆ Blackman,
- ◆ Kaiser (with B=10.056, Side-Lobe-attenuation, approx. 100 dB),
- ◆ Flat Top (for exact amplitude setting).

Default is the Rectangle-window.
The smooth function should only be used in conjunction with noise signals. Note that the "Terz"-smoothing is a smoothing to Octave/3 bandwidth. The result is not equal to a Octave/3-spectrum.
The averaging can be disabled or set to exponential or linear. If exponential or linear is selected, the number of averagings should be set to an appropriate value.

[F6], [F7], [F8], and [F9] can be used to set the visible data-window as required.
[F1] starts the measurement.

```
┌─────────────────────────────────────────────────────────────────────────┐
│ ┌─────────────┐ ┌────────────────────────────────┐ ┌───────────────────┐ │
│ │ ESC Cancel  │ │ D A A S  Spectrum Analyser Mode│ │ i 12.07.95 14:22  │ │
│ └─────────────┘ └────────────────────────────────┘ └───────────────────┘ │
│ ┌───────────────────────────────────────────────┐ ┌───────────────────┐ │
│ │ In: Line  ±2.0 V  FFT len: 4096 Pt.  Window: KAISER│ │ MO:  NoName      │ │
│ └───────────────────────────────────────────────┘ └───────────────────┘ │
│                                                                           │
│  dBV                                                                      │
│                                                                           │
│  -15                                                                      │
│                                                                           │
│  -30                                                                      │
│                                                                           │
│  -45                                                                      │
│                                                                           │
│  -60                                                                      │
│                                                                           │
│  -75                                                                      │
│                                                                           │
│  -90                                                                      │
│                                                                           │
│ -105                                                                      │
│                                                                           │
│      100   200  300   500      1k     2k   3k   5k    10k   20kHz         │
│ ┌────────────┐┌────────────┐┌──────────┐┌──────────┐┌──────────┐         │
│ │F1          ││F2 Pause    ││F3        ││F4        ││F5        │         │
│ ├────────────┤├────────────┤├──────────┤├──────────┤├──────────┤         │
│ │F6          ││F7          ││F8        ││F9        ││F10       │         │
│ └────────────┘└────────────┘└──────────┘└──────────┘└──────────┘         │
└─────────────────────────────────────────────────────────────────────────┘
```
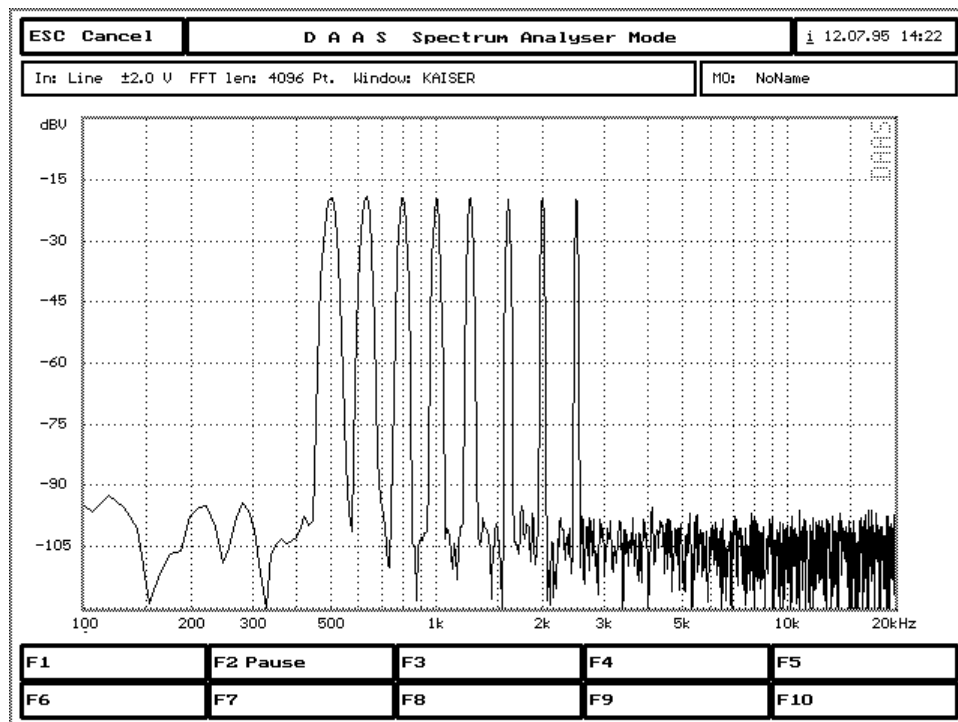
The measurement will be repeated until [Esc] or [F2] 'pause' is pressed. [ESC] or [F2] can be clicked with the left mouse button, however if DAAS is busy calculating, a slight delay may be experienced.

On pressing the "Pause" key, the measurement will be halted and a marker placed on the maximum level. At the same time the frequency and effective value at this point are shown in the info box.

```
ESC Cancel          D A A S  Spectrum Analyser Mode       i 12.07.95 14:22

RMS value: -18.9 dB V  Frequency: 632.8 Hz               MO:  NoName
```
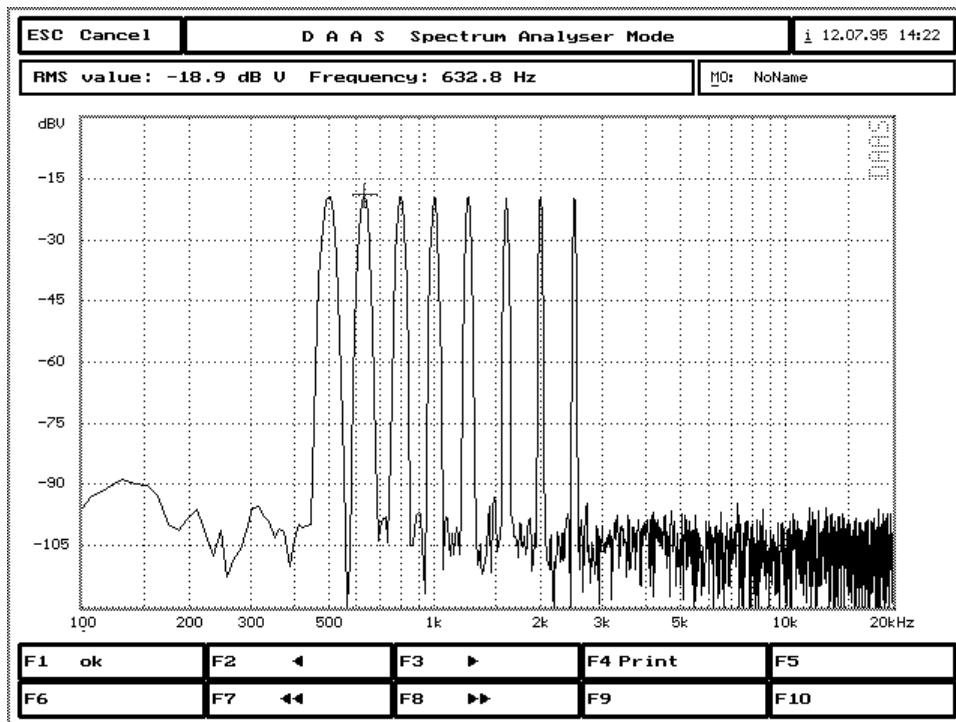
The marker can be moved now using [F2], [F3], [F7] and [F8] or by moving the mouse in the appropriate direction. The display of the effective value and frequency is then updated.
Alternatively the cursor keys can be used [←], [→], [Ctrl]+[←] and [Ctrl]+[→].

The system waits until [F1] is pressed to continue. [F4] can be used to make a printout. [Esc] is used to go back a menu where the parameters can be changed and the measurement repeated.

Frequency measurement 'virtually' in real time

The DAAS software package contains a number of special test signals which were created with help from the DAAS - FFT and therefore contain all the properties to fit exactly in the analysis. Two of these signals "WN2048.DAT" and "WN4096.DAT", are particularly suitable for the measurement of frequency response curves in the Spectrum Analyzer.
The spectrum comprises (when used with a 2048 or 4096 point FFT and the square window) a line without the statistical variations which normally occur when measuring a noise signal.

```
ESC Cancel          D A A S   Spectrum Analyser Mode      i 12.07.95 14:23

                                                          MO:  NoName

dBV

-15

-30

-45

-60

-75

-90

-105

    100      200   300     500      1k       2k    3k    5k       10k      20kHz

F1                F2 Pause        F3            F4            F5
F6                F7              F8            F9            F10
```
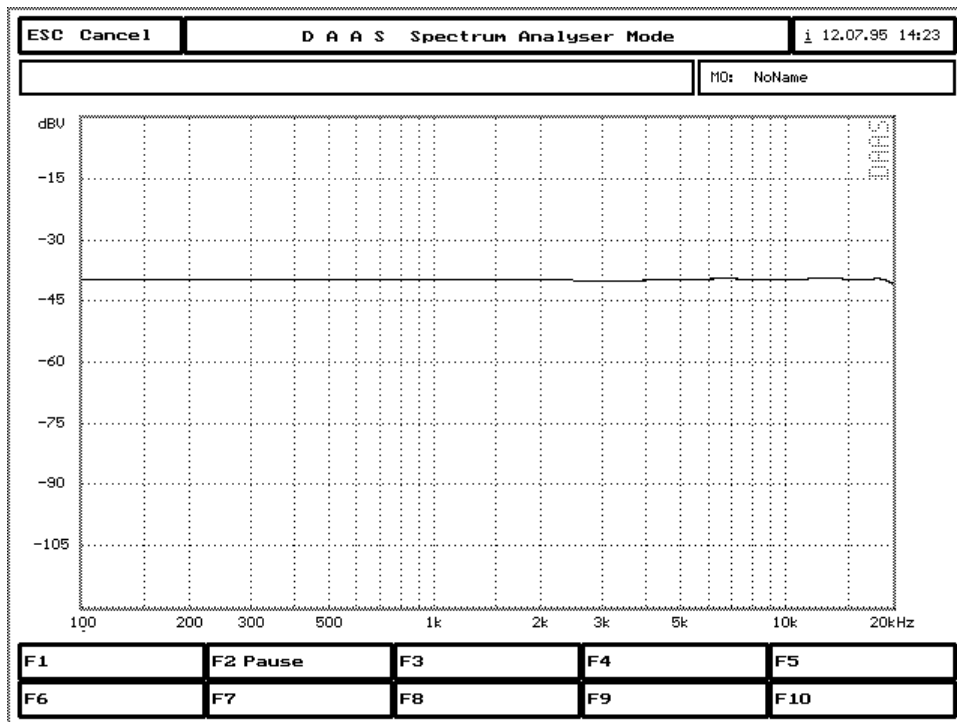
Due to the measurements being continuous, this method is particularly suitable for, for example, fine tuning of crossovers and speaker enclosures. The following sketch shows a typical set-up.

Set the amp volume control or the 'Output' using [F3] a level well above the surrounding noise level. In the Spectrum Analyzer menu press [F2] and 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.

objekt

amplifier

DAAS 3 L
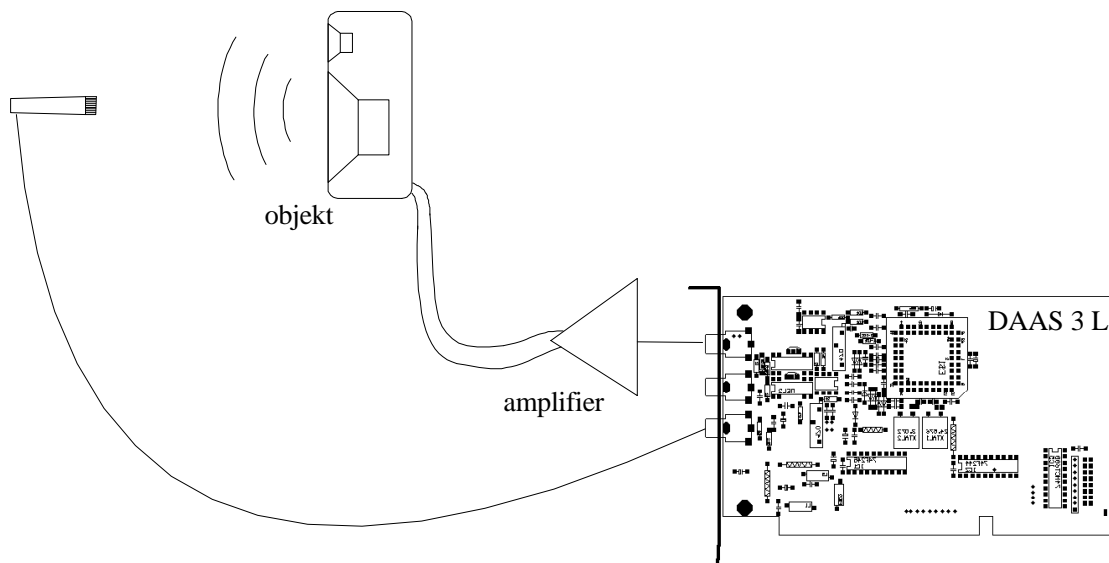
The spectrum now shown is directly the frequency response curve of the object (speaker). Note that the curve shown is a relative curve, the absolute values (e.g. 2 Pa/V) cannot be read directly from here.
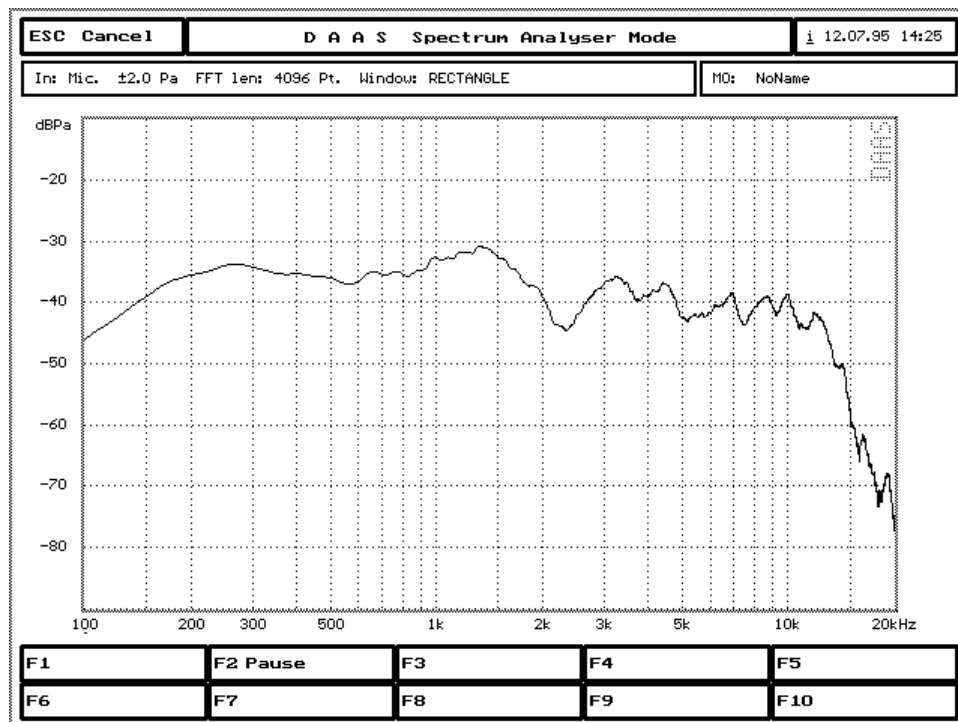
For this frequency response measurement the smoothing option mentioned above is very useful. Set the smoothing option to 'yes' to get a soft curve, or set the smoothing option to 'Terz' to get octave/3 bars.

This frequency response measurement with the Spectrum Analyzer can also be carried out with reduced sample rate so the higher resolution (measurement range 0-2kHz) can also be used.

Pressing [F2] = 'Pause' and then [F4] will produce a screen hard copy on a 9 or 24 pin matrix printer or an HP-Laserjet II or create a PCX-File.

Reverberation time

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.
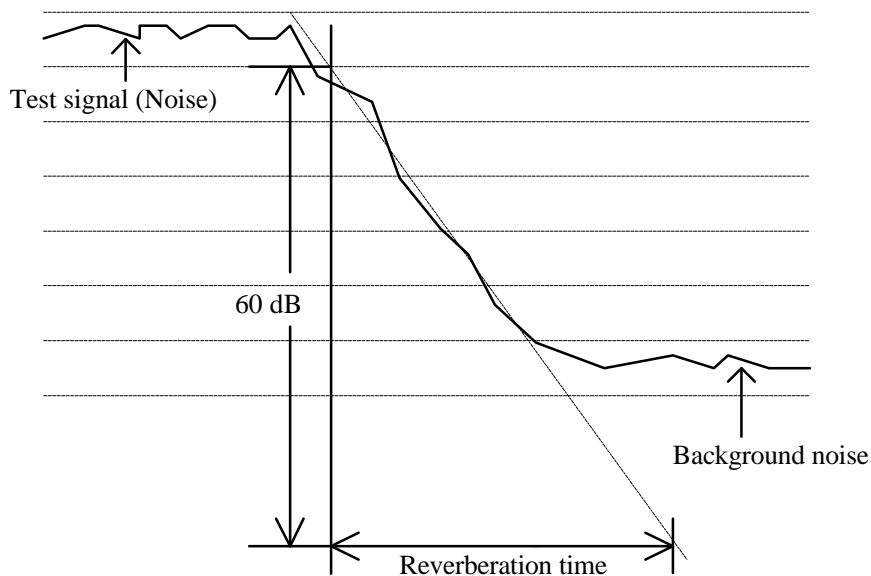
But the disadvantage of this method is that the reverberation time usually depends on the frequency. Harmonic signals for supplying the speaker are not usable, because stationary acoustic waves will falsify the results. The Digital Audio Analysis System uses narrow band noise with variable center frequency to avoid this disadvantages. This method is very easy to use and don't need complex parameters to set.
Choose the reverberation time measurement by pressing [F5] in the main menu. With [F5] the input sensitivity can be set with ±2 Pa and ±8 Pa. The output level can also be set using [F10] and of course using the volume knob of the amplifier.
Set up as follows and start the measurement by choosing the measuring frequency.
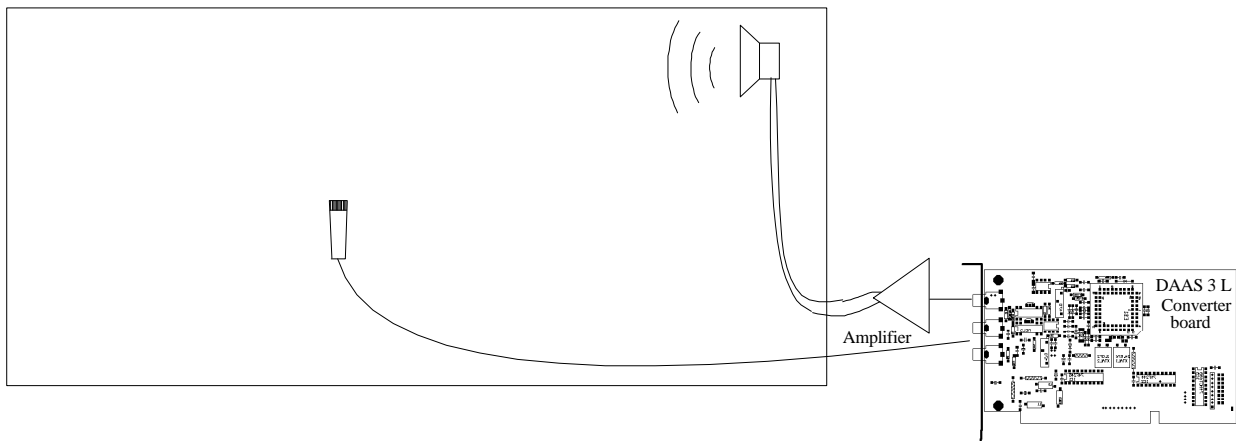
After the measurement the measured acoustic level curve is shown. A line to interpolate the slope will be inserted.
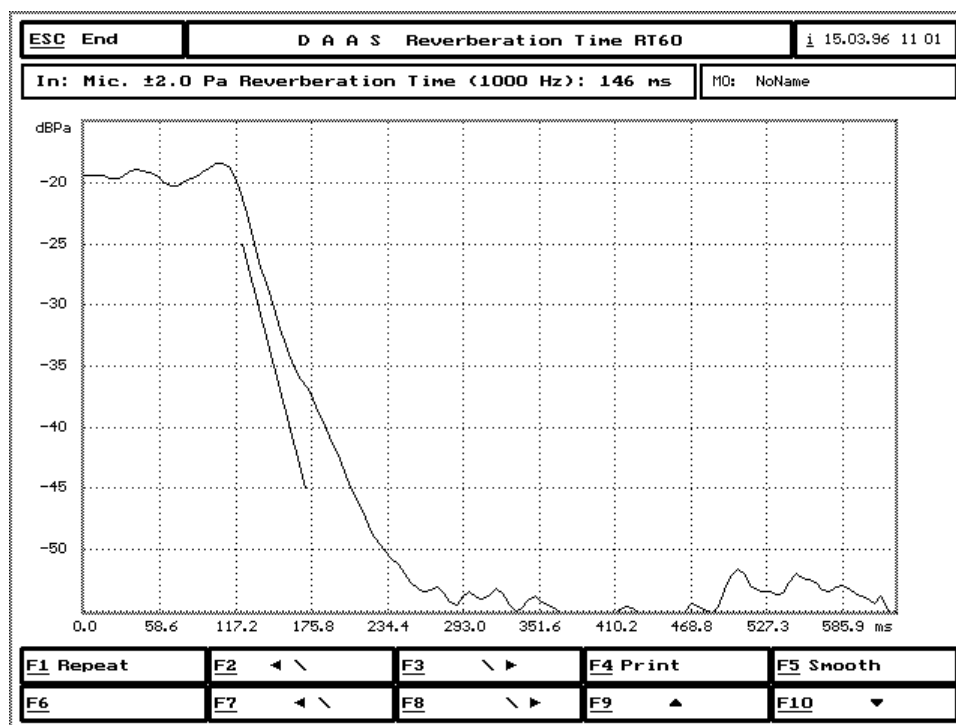
Shift this line using [F2] / [F3] and [F7] / [F8] until it is identical with the slope of the acoustic level curve. Smooth the acoustic level curve using [F5] if necessary.

Note: To get the correct slope the most standards recommends to consider the decay curve between -5 and -35 dB, but recently the upper part of the decay curve seems to be important. This "Early Decay Time" is especially in the lower frequency range a little bit lesser.
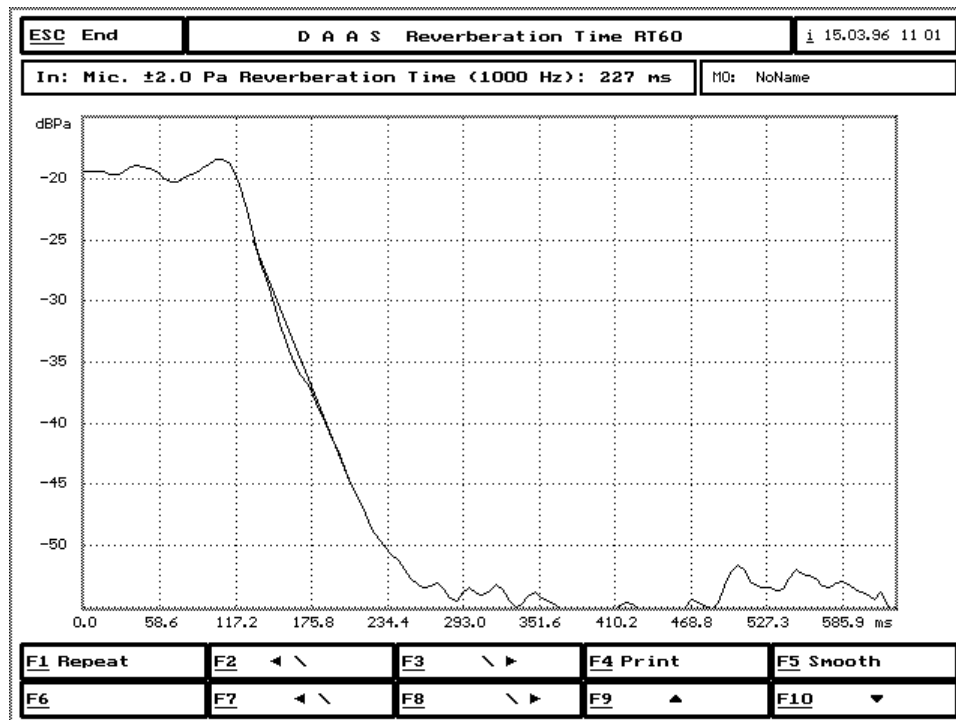
In the info line on the top of the screen now the reverberation time is shown. Note that smoothing the acoustic level curve will increase the lowest measurable reverberation time.

Pressing [F4] will produce a screen hard copy on a 9 or 24 pin matrix printer or an HP-Laserjet II or create a PCX-File.

Signal player

The 'Signal player' allows to play test signals and generate/play some wave forms independent on any measurement.
Choose the 'Signal player' by pressing [F9] in the main menu.
Set the output attenuation using [F4].
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 the 'Signal player' by choosing the wave form frequency or by selecting the signal file.
The wave form will be played until the [ESC]-key is pressed. The exact frequency is displayed in a message box in the middle of the screen.

Measurements with various sampling rates

The system can be driven with the three sampling rates

- 48 kHz,
- 6 kHz,
- 64 kHz.

The default setting is 48 kHz which gives a resolution in the frequency range of max. 4.6 Hz (frequency response and impedance curve measurements). The screen viewing area is hereby from 20 Hz to 20000 Hz. To obtain a higher resolution in the lower frequency range choose the sampling rate of 6 kHz.
Except for the intermodulation factor measurement all analyses can be carried out at the higher resolution. Frequency and impedance response can achieve a maximum resolution of about 0,59 Hz! The screen viewing area is now reduced to approx. 10 Hz to 2000 Hz.
The frequencies of all test signals are also reduced by a factor of 8. For example the test signal "1KHZ.DAT" has, with reduced sample rate, 125 Hz.

To get results above 20 kHz select the sampling rate of 64 kHz.

The sampling rate can be set in the options menu.
Further changes to the parameters are not necessary. The 'Anti-Aliasing-Filter', necessary with the various frequency limits, is built into DAAS as standard.

Saving and loading signals

Before each measurement and after each screen plot it is possible using the [F5] key to save and load signals.
On pressing [F5] the selection box appears for the file name. The file is selected using the cursor keys of by pressing or clicking [F2] or [F3] with the left mouse button. Only files shown in current directory which belong to the current measurement are shown. Having selected the file press [Enter] or click [F1] with the mouse to load it. You can also enter a file's name directly. Enter the file's name (without extension).
Only files which belong to the current measurement can be loaded. When saving, numerous additional pieces of relevant data are stored with the signal which are necessary for future evaluations.

Every reference signal, for frequency response and impedance curve measurements, is saved with the name "LASTREF". On leaving and later returning to frequency response or impedance curve measurements it is not necessary to carry out a new reference measurement, providing of course the set-up has not been changed.

Tolerance mask, Quality control functions

Preface

For the afterwards described measurements it is necessary to install the Featurebox to the Q-DAAS converterboard. **Don`t forget to activate the COM-Port for the Featurebox in the Q-DAAS options ([F10],   [F1]), default is COM2** (see page 64 and Featurebox manual)

It is a good advise to work at first with the standard functions of the QDAAS (similar to DAAS3L+) for frequency - and impedance measurement response, to become familiar with the measurement system (that's why the chapter „Tolerance mask, quality control functions" is not the first one in this handbook).

For the frequency response measurement and the impedance curve measurement there are tolerance masks available. Press [F10] - [F3] *after* complete a frequency response measurement respectively [F2] after complete an impedance curve measurement to get into the mask menue. Use [F1] or [F2] to load or save masks. Press [F3] to draw a new mask on the screen. If you loaded a mask, or drawn a new one, the mask will be inserted in the diagram after each measurement. [F4] allows to import a new mask, generated from a former done frequency response.

There are two different masks available, one for impedance curves, and one for frequency curves. Each mask consists on an upper and a lower limit line. Each limit line can consist up to max. 99 single frequency points.

The tolerance mask files can be read and edit by any text editor (ASCII). The file extension is "MH" for frequency curve masks, and "MZ" for impedance curve masks.
Do not change any key word!
Do not change the position of the numbers and key words!

To convert a measured frequency response curve to a mask line, do as follows:

After a complete frequency response press [F10] and [F1] to export the frequency response curve. Afterwards press [F10]-[F3] and [F4] to import and convert a frequency response curve to a mask.. Select the file name and decide, if the file should be an upper limit line or a lower limit line. Now you have the opportunity to shift the imported limit line. Enter the dB value and do the same with the remaining mask line. In many cases it is not necessary to use such a precise mask in nearly all cases a mouse-drawn line will be sufficient. In a noisy ambient it is always meaningful to use a mouse-drawn mask.

Automatic adaption of tolerance mask

For QC measurements the Q-DAAS system can shift the frequency response mask lines automatically to every measured curve (automatic adaption of tolerance mask). This is very helpful if the devices under test have different transmission factors (sensivity). To use this feature do as follows

1. Activate the feature "Adaptation mask lines to Curve" in the (main) options menu
   ([F10],   [F2] and then   [F1]   (means the „shift key")
2. Carry out a frequency response measurement,
3. Enter a new set of mask lines. Now the Q-DAAS system detects the difference between the upper mask line and the measured curve at e.g. 1 kHz (the frequency which builds the referencepoint for the adaption, may be modified in the file DAAS3L.SYS).
At all following measurements the mask lines are automatically shifted to the same distance from the curve at 1 kHz. The detected difference between the upper mask line and the curve will be stored by the Q-DAAS configuration file and used until the above described procedure is repeated.

Checking the mask

For QC measurements the Q-DAAS system can check if the measured curve is inside or outside of the mask lines. The result of this check is indicated in the upper left corner of the data screen with a "good"/"no good" message. The two key words for the statement "good" or "not good" can be changed in the options [F10], [F12]. Also the letter size is variable. On mask „failed" an additional warning tone (beep) can be activated. Default settings are:
mask passed: „good"   (statement displayed always in green color)
mask failed: „bad"   (statement displayed always in red color)
letter size: 6
beep: off

Additonally, an electrical signal for good/bad is available at an output connector on the rear side of the Featurebox (only optional available). An automatic start of the Macro No.1 is possible as well. This makes it possible to implement the QDAAS system into an automatic production line. For more informations concerning the control lines please refer to the Featurebox manual.

Note:
If no mask is selected, no check will be done.

Combined quick repeat measurement (only in combination with the *Featurebox*)

Connect the featurebox with the Q-DAAS converterboard and the DUT (device under test). At first a (standard) impedance measurement must be completed. Select the desired display scale for the impedance curve and draw the necessary mask lines. After that, leave the impedance measurement and proceed a frequency response measurement. Then, a combined (frequency response - and impedance response) measurement can be made with [F12].

Screen print

A screen print is always available using the [F4] key. Printouts are available on a 9 or 24 pin dot matrix printer and additional drivers are also available for HP Laserjet or as a .PCX file. Under the "Options' menu press [F8] to select the format required. The .PCX files can be imported directly into other applications e.g. Windows paintbrush, Winword and various DTP programs.

A screen print can also be obtained using the DOS program "GRAPHICS.COM".
The method is quite simple:
From the DOS prompt run the program "GRAPHICS.COM".
To obtain the print out hold down the [Shift] key and press [PrtScr].
See your DOS handbook for further details.

To print in color on a color printer the Windows "Clipboard" can be used. Start DAAS from within Windows and when the required screen is shown press[Alt]+[PrtScr]. The screen will now be stored on the "Clipboard". The image can now be down loaded from the clipboard by another Windows application e.g. Paint etc.
Refer your Windows handbook for more details on this procedure.

Macro recorder

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

To record a sequence:

1. Start the recorder by pressing [Alt]+[R]
2. Then choose the Macro 1-9 by pressing [Alt]+[F1] .. [Alt]+[F9]. The start of record mode is indicated by two short tones
3. Enter the keystrokes and mouse actions that you want to record (a small dot above the data window indicates the record mode),
4. When you want to stop recording, press [Alt]+[R] again. This will be shown by the quit-DAAS-tone sequence.

To play back a macro simply press [Alt]+Function key (e.g. [Alt]+ [F3]).

CAUTION Only sequences of keystrokes and mouse actions will be recorded. You can play back a macro only at the same place where you have recorded it. For example playing a macro for repeat the frequency response measurement in the options-menu will cause everything but not repeat the measurement.

The following actions cannot be recorded:

1. Lettering in the data window
2. Stopping the scope in repetition mode
3. Stopping the signal player in repetition mode

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

Options

The "option" key [F10] is used to set various measuring system parameters.
The following menu is then shown

```
┌──────────────────────────────────────────────────────────────────────────┐
│ ESC End          D A A S  Change Options          i 22.04.99 16 13         │
├──────────────────────────────────────────────────────────────────────────┤
│ Frequency range: 20 Hz - 20.0 kHz                  MO:  NoName             │
│                                                                            │
│   ┌────┐                                                                   │
│   │ F1 │ Signal out attenuation                       : 36.0 dB            │
│   │ F2 │ Signal out attenuation for Impedance meas. : 30.0 dB              │
│   │ F3 │ Export format                                : Audiocad           │
│   │ F4 │ Sounds and warning tones                     : no                 │
│   │ F5 │ Store temporary data to                      : RAM                │
│   │ F6 │ Sample rate                                  : 48000 Hz           │
│   │ F7 │ Input range overflow warning optical         : no                 │
│   │ F8 │ Printer                                      : none               │
│   │ F9 │                                                                   │
│   │F10 │ Drive:\Path\ for all Macros                  :                    │
│   │F11 │ Drive:\Path\ for export files                :                    │
│   │F12 │                                                                   │
│   │↑F1 │ RS232-control port ...                                            │
│   │↑F2 │ Options frequency response+impedance ...                          │
│   │↑F3 │ Adjust desktop colors ...                                         │
│   │↑F4 │ System options ...                                                │
│                                                                            │
│ ┌──────────┬──────────┬──────────┬──────────┬──────────┐                  │
│ │ ☐        │ F2       │ F3       │ F4       │ F5       │                  │
│ ├──────────┼──────────┼──────────┼──────────┼──────────┤                  │
│ │ F6       │ F7       │ F8       │ F9       │ F10      │                  │
│ └──────────┴──────────┴──────────┴──────────┴──────────┘                  │
└──────────────────────────────────────────────────────────────────────────┘
```

The various settings are as follows:

[F1]
The level at the "Out' socket i.e. the attenuation can be set here in the range 0 to 90 dB.

[F2]
The level at the "Out' socket i.e. the attenuation for the impedance and for the Thiele-Small parameter measurements can be set here in the range 0 to 90 dB.

[F3]
Data format 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. Database: hifisound database has the extensions ".DBF" and ".FRQ".

[F4]
Activate and de-activate the signal- and warning tones.

[F5]
Use RAM memory for temporary storage. DAAS uses 512k when this setting is on
"RAM". If enough memory is not available switch this option off and the data is written
temporarily to the hard drive.

[F6]
Changes the system sampling rate. Default is 48 kHz which gives a resolution in the
frequency range of max. 4.6 Hz (frequency response and impedance curve
measurements). The screen viewing area is hereby from 20 Hz to 20000 Hz. To obtain
a higher resolution in the lower frequency range choose the sampling rate of 6 kHz. The
frequency resolution is then about 0,59 Hz. The screen viewing area is now reduced to
approx. 10 Hz to 2000 Hz. With a lower sample rate the frequency of all test signals is
reduced by a factor of 8.
To get results above 20 kHz select the sampling rate of 64 kHz.

[F7]
By an overflow in the input level range a warning signal sounds and a message is
shown in the info-box. With [F7] the screen can also be made to blink if this occurs.

[F8]
Is used to set the output device.

• all 9- and 24-dot matrix printers (Epson-compatible),
• Laser printers HP-II-Series (or compatible),
• Output as PCX-File (Black and white),
• Deskjet Color printers (or compatible).

 [F9]
reserved

[F10]
Drive and path for all macros.

[F11]
Drive and path for export files.

[F12]
reserved

   [F1]
RS232 control port. Select desired COM - port of your PC to communicate with the
F*eaturebox*.
Select the control lines of the Featurebox (refer to *Featurebox* handbook !) In doubt,
do not change the control line settings; most are correct preset for common
measurements.

   [F2]
Options for frequency response and impedance response. See menue setting parameters
for frequency and impedance measurement.

   [F3]
Adjust desktop colors

   [F4]
System options. See menue setting system options.

[ESC]
Press [ESC] to leave this menu, as with all others.

On closing DAAS all settings are saved.

Setting the parameters for frequency response measurements

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

```
┌──────────────────────────────────────────────────────────────────────┐
│ ESC End          D A A S  Change Options        i 22.04.99 16:17 │
├──────────────────────────────────────────────────────────────────────┤
│ Frequency range: 20 Hz - 20.0 kHz                   MO:  NoName   │
│                                                                    │
│   ┌────┐                                                           │
│   │ F1 │ Display speakers frequency response as : dBSPL/1.00W/1.000m/8ohm │
│   │ F2 │ Extra Smooth automatically             : yes            │
│   │ F3 │ Delay detection acoustical, electrical : automatic,none │
│   │ F4 │ Default display range waterfall plot   : 4.0 ms         │
│   │ F5 │ Lower display limit waterfall plot     : 300 Hz         │
│   │ F6 │ Number of Waterfall-Spectren           : 48             │
│   │ F7 │ Anechoic Frequency Resp.: cut echos at : 3.3 ms         │
│   │ F8 │ Speed of sound                         : 343 m/s        │
│   │ F9 │ Line style for the phase curve         : Dots           │
│   │F10 │ Detect Speaker Polarity at             : 400 Hz         │
│   │F11 │                                                          │
│   │F12 │ Message on Tolerance Mask passed/failed: Good/Bad,Size 6 │
│   │↑F1 │ Autom. adaption of Mask Lines to curve : ±6 dB          │
│   │↑F2 │                                                          │
│   │↑F3 │ Read Out Frequency Resp. Values...                      │
│   │↑F4 │ Message on Phase good (+) or bad (-)...                 │
│                                                                    │
├──────┬──────┬──────┬──────┬──────────────────────────────────────┤
│ ░1   │ F2   │ F3   │ F4   │ F5                                     │
├──────┼──────┼──────┼──────┼──────────────────────────────────────┤
│ F6   │ F7   │ F8   │ F9   │ F10                                    │
└──────┴──────┴──────┴──────┴──────────────────────────────────────┘
```

[F1]
The loudspeaker frequency curve can be displayed in 3 different scales.:

◆ dBPa/V(dB Pascal/Volt, 1 Pa = 10 μbar),
◆ dBSPL/V(dBSPL = dB relative 20 μPa, 1 Pa = 94 dBSPL)),
◆ dBSPL/W/m corrected to 1 meter.

The setting dBSPL/W/m is coupled with the automatic distance measuring. If the distance is greater than 10cms an automatic conversion into dBSPL / Watt / Meter will be carried out. This is good for near field measurements in the range 15 to 50 centimeters. **CAUTION:** do not alter the level settings after carrying out a reference measurement.
The option dBSPL/W/m is necessary when the frequency curve is to be exported to loudspeaker CAD programs which require data in dBSPL / W /m format.

[F2]
Automatically smooth the frequency response curve.

[F3]
Delay compensation during acoustic measurements automatic or manual.

[F4]
Display range for waterfall spectrum. (range 1,7 to 13,3 Milliseconds).

[F5]
Lower display frequency of waterfall spectrum in the range 200 Hz to 2000 Hz.

[F6]
Number of lines in waterfall spectrum, either 48, 60 or 72.

[F7]
Cut of point for echoes. This point is normally set during measurement to remove room reflections. This setting is useful when the same room is being constantly used and known reflections need to be removed. The default setting is approx. 3.5 milliseconds.

[F8]
Set the 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 if 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.

[F9]
Changes the line style to enable better differentiation, particularly useful on monochrome monitors.

[F10]
This option is made for QC measurements of loudspeakers and looks for the phase at the chosen point. If the phase is greater than zero, the polarity of the speaker is "+", otherwise the polarity is "-". The polarity is shown in the upper right part of the screen. The reference frequency for inspecting the polarity is selectable.

[F11]
reserved

[F12]
Message on tolerance mask passed or failed
For QC measurements the Q-DAAS system can check if the measured curve lies inside
or outside of the mask lines. The result of this check is indicated in the upper left
corner of the data screen with a "good"/"no good" message. Here it is possible to key
in the message text for the „good" or „no good" state. You also can activate an
additional "no good" - warning tone and change the message text size.

   [F1]
Also for QC measurements the Q-DAAS system can shift the frequency response
mask lines automatically to every measured curve. Here this feature can be enabled or
disabled. The search range for automatically adaption of the curve can be preset here.

   [F2]
reserved

   [F3]
Read out frequency response values.
To automatically readout and display a frequency response values at a fixed frequency
point enter here a frequency greater than zero. If this entry is greater then zero, the
corresponding frequency response value is shown on screen after each measurement. If
more than one frequency point is given, Q-DAAS will show an average value
calculated from the result of all frequency points.

   [F4]
Possiblity to key in the message text for the polarity check.
e.g. „good"(+) and „bad"(-)

Setting the screen colors and screen saver

This menu is used for setting the screen colors and is activated by pressing [F11] in the "options" menu. the changes are immediate on pressing the function keys or clicking with the mouse. [F11] can be used to activate the three default settings. (useful for laptop/LCD/Plasma screens). The delay before the screen saver activates is set with [F12].
63 colors are available. In monochrome mode these will be shown as gray scales so some scales may appear double.

```
┌──────────────────────────────────────────────────────────────────────────┐
│ ┌ ESC End ──┐ ┌──────────── D A A S  Change Options ──────────┐ ┌ i 12.07.95 11 15 ┐ │
│ └───────────┘ └───────────────────────────────────────────────┘ └──────────────────┘ │
│ ┌ Frequency range: 20 Hz  -  20.0 kHz ──────────────┐ ┌ MO:  NoName ──────┐ │
│ └───────────────────────────────────────────────────┘ └───────────────────┘ │
│   ┌──────────────────────────────────────────────────────────────────┐     │
│   │    ┌───┐                                                          │     │
│   │    │F1 │  Background color                                        │     │
│   │    ├───┤                                                          │     │
│   │    │F2 │  Text color                                              │     │
│   │    ├───┤                                                          │     │
│   │    │F3 │  Signal window                                           │     │
│   │    ├───┤                                                          │     │
│   │    │F4 │  Signal                  _____                        │     │
│   │    ├───┤                                                          │     │
│   │    │F5 │  Scale-lines             .......                         │     │
│   │    ├───┤                                                          │     │
│   │    │F6 │  Scale-text              1 2 3 ms                        │     │
│   │    ├───┤                                                          │     │
│   │    │F7 │  Screen limit lines                                      │     │
│   │    ├───┤                                                          │     │
│   │    │F8 │  Phase                   ...................             │     │
│   │    ├───┤                                                          │     │
│   │    │F9 │  Waterfall               ..............                  │     │
│   │    ├───┤                                                          │     │
│   │    │F10│  Background Waterfall                                    │     │
│   │    ├───┤                                                          │     │
│   │    │F11│  Preset 1, 2 or b&w                                      │     │
│   │    ├───┤                                                          │     │
│   │    │F12│  Sreensaver aktive after 16 min                          │     │
│   │    └───┘                                                          │     │
│   └──────────────────────────────────────────────────────────────────┘     │
│ ┌ F1 ──────┬ F2 ──────┬ F3 ──────┬ F4 ──────┬ F5 ──────┐                    │
│ ├ F6 ──────┼ F7 ──────┼ F8 ──────┼ F9 ──────┼ F10 ─────┤                    │
│ └──────────┴──────────┴──────────┴──────────┴──────────┘                    │
└──────────────────────────────────────────────────────────────────────────┘
```

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.

```
┌─────────────────────────────────────────────────────────────────────┐
│ ESC  End  │            D A A S  Change Options    │ i 12.07.95 11 16 │
├───────────────────────────────────────────────┬─────────────────────┤
│ Frequency range: 20 Hz - 20.0 kHz             │ MO:  NoName          │
├───────────────────────────────────────────────┴─────────────────────┤
│                                                                       │
│     ┌────┐                                                            │
│     │ F1 │  FFT options frequency response...                        │
│     ├────┤                                                            │
│     │ F2 │  FFT options impedance measurements                       │
│     ├────┤                                                            │
│     │ F3 │                                                            │
│     ├────┤                                                            │
│     │ F4 │                                                            │
│     ├────┤                                                            │
│     │ F5 │  Window harmonic distortion          :    KAISER          │
│     ├────┤                                                            │
│     │ F6 │  Window intermodulation distortion   :    KAISER          │
│     ├────┤                                                            │
│     │ F7 │  Acoustic input range (peak value)   :    2 / 8 Pa        │
│     ├────┤                                                            │
│     │ F8 │  Start analysis Imped./Dist./Spectrum :   42.7 ms         │
│     ├────┤                                                            │
│     │ F9 │  Address of the converter board      :    358h-35Fh       │
│     ├────┤                                                            │
│     │F10 │  Delay acoustic measurings           :    no              │
│     ├────┤                                                            │
│     │F11 │  Digital filter delay                :    52 Taps         │
│     ├────┤                                                            │
│     │F12 │                                                            │
│     └────┘                                                            │
├───────────┬───────────┬───────────┬───────────┬─────────────────────┤
│ F1        │ F2        │ F3        │ F4        │ F5                   │
├───────────┼───────────┼───────────┼───────────┼─────────────────────┤
│ F6        │ F7        │ F8        │ F9        │ F10                  │
└───────────┴───────────┴───────────┴───────────┴─────────────────────┘
```

The individual menu points are as follows:

[F1]
Window, RMS-averagings and overlap for frequency response measurement. Note: if you use the test signal "PNCHIRP.DAT" these options are fixed.

[F2]
Window, RMS-averagings and overlap for impedance measurement. Note: if you use the test signal "CHIRP.DAT" these options are fixed.

[F5]
Window for harmonic distortion measurement. Default: KAISER.
Alternatives: SQUARE, FLAT TOP.

[F6]
Window for intermodulation distortion measurement. Default: KAISER.
Alternatives: SQUARE, FLAT TOP.

[F7]
Setting the microphone sensitivity.
Default : two settings ±2 Pa and ±8 Pa.
Possibilities: ±1 Pa and ±4 Pa, ±2 Pa and ±8 Pa, ±4 Pa and ±16 Pa.
CAUTION: Should this setting be changed then the microphone must be re-calibrated and the microphone pre-amplifier reset.

[F8]
Waiting time between Test signal start (t=0 in the time signal window) and start of the analysis by all distortion measurements and spectral analysis. Default: 2048 Points = 43 ms

[F9]
Address range of the ADC/DAC board. The normal setting is address 358h-35Fh. If this setting leads to address conflicts with other boards the range can be changes by setting jumpers on the DAAS board. If the jumper setting does not correspond with the setting under this menu heading [F9], a system crash is a certainty !!
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

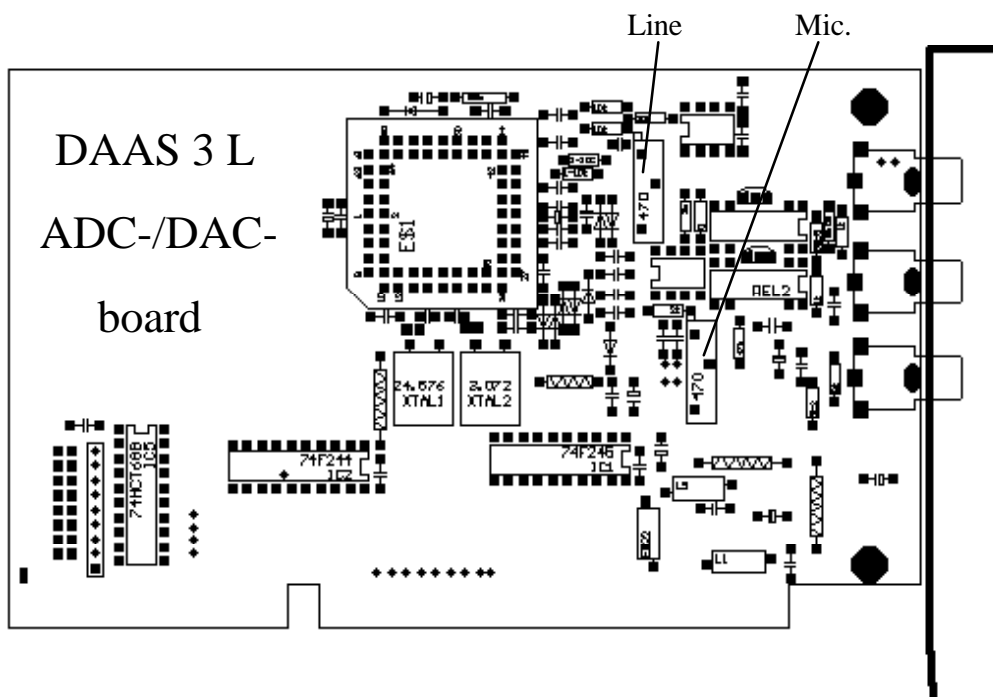"1" means jumper set,   "0" jumper open.

[F10]
Additional waiting time between Test signal start (t=0 in the time signal window) and start of the analysis by all acoustic measurements.
Default: No.
Changing this option is necessary only in conjunction with some special ADC/DAC boards.

[F11]
Compensation of the ADC/DAC converters digital filter delay. The "AD1848" needs 52 Taps, the "CS4248"/"CS4231" needs 18 Taps.

[F12]
Change type of converterboard. Do not change.

Calibration

The inputs should be regally calibrated, particularly the mic input.
In order to carry out the calibration a calibrator with a defined signal level (i.e. 94 dBSPL e or Pa_{eff}) should be used. Carry out a level measurement and compare this with calibrators level. If there is difference the microphone pre-amp will have to be adjusted.

The electrical input should also be tested. Use a defined voltage source with a frequency between 500 Hz and 2 kHz. Carry out a level measurement and compare the measured level with the actual level. If there is a difference the measurement amp will have to be adjusted.

If an adjustment is necessary it will be necessary to gain access to the DAAS card and adjust the trimmers as shown below.

Technical data Q-DAAS

ADC-/DAC-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: 100 mV, Phantom supply 12 V, Resistance 8,2 KOhm. Input sensitivity from DAAS microphone: ±2 Pa ad ±8 Pa, software switchable. Max. allowed input 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.
Sample 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
Connections :	RCA-Phonoplug 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.

Software Q-DAAS

Function type: FFT-Analyzer

FFT-Length: 512 - 4096 points

Calculation time: FFT 4096 points: app. 0,35 seconds,
                        FFT 1024 points: app. 0,08 seconds,
                        measured on a 386/33 with HANN-window.

Resolution: Frequency and impedance measurement to 0,59 Hz

Evaluation widows: SQUARE (UNIFORM),
                        HANN (or HANNING),
                        BLACKMAN,
                        KAISER (B=10,056),
                        FLAT TOP

Overlap: 0 % to 50 %

Analyses: see preface in this handbook.

Signals: special noise - and sine testsignals produced for a maximum
                        performance of Q-DAAS.

Signal format: 16 Bit linear coded: range ±32767, INTEL-Format
                        (Low Byte first)

Graphic: SVGA 1024X768 pixels, variable colors
                        SVGA 800x600 pixels, variable colors
                        VGA: 640x480 pixels, variable colors

System requirements

◆ IBM-AT 386 / 486 or compatible (<u>not</u> IBM PC/XT, IBM PC/AT 286 or compatible)
◆ Graphic card VGA / SVGA
◆ Windows95® (only in a fullscreen DOS-Session), PC/MS-DOS® Version 5 or later
◆ Microsoft® - Mouse or compatible (not essential but easier to work with)
◆ 1 Hard drive , 1 Floppy drive.
◆ 8 MByte RAM (using with WINDOWS95: 16 MB)
◆ min. 10 MByte free hard drive space
◆ math. Coprocessor (FPU) is not required.

Important points concerning safety and use

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

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

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

❑ Don't use amplifiers with bridged power stages!

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

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

❑ After long use the Q-DAAS 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 "options".

All information and specification are subject to change without notice.