Audio Analyzer
ATB 805

HANDBOOK

H A N D B OOK

ATB 805

Project Manager

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Program development

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1. AN INTRODUCTION TO THE SYSTEM

1.1 GENERAL DESCRIPTION

The ATB 805 PC Measuring program combined with the ATB 805 USB box forms an exact and extensive Audio Analyser. The ATB 508 is a 2-channel analyser. The 2-channel is for the new coming ATB program and for other 2-channel measurement programs.

The following measurements are available:

Electric and acoustic amplitude frequency response, SPL
Electric and acoustic phase frequency response
Impedance measurement
Step response
Distortion THD and H2, H3, H4, H5
IMD and nonlinear distortion
Thiele-Small Parameter

The measuring system itself consists of the ATB 805 USB box and the ATB 805 PC measurement program to carry out measurements. Using the most modern measurement technology allows handling as simple as with an analyser.
The Windows ATB 805 PC measuring program contains all the properties for frequency characteristic measurement featured in the professional ATB precision program.
The ATB PC measurement program also allows a reliable car sound system check. As a DVD player is not usually installed in cars we have included the Auto-Test CD to ensure that all requirements are met for satisfying measurements in car.

The Windows measuring program corresponds to a frequency characteristic measurement reduced ATB precision program offering all the functions of the Windows operating system. For the PC we have developed a new measurement technology with optimised instrumentation.

For CAD or calculation program or data base the ATB PC data can be exported with the ATB Converter programme in many formats. The export is for example to MLSSA, LAUD, IMP and CLIO file.

1.2 INTENDED USE

The Audio Analyser ATB 805 is for the measurement of audio signals in home- or professional application.
The unite is connected via USB to the PC. The USB connected the ground from the PC to the ground of the ATB 805. A USB isolator is to use if there is a problem.
The product is only approved for connection to 220 – 240 V/50 – 60 Hz alternating voltage. It is intended for indoor use only. Do not use it outdoors. Contact with moisture, e.g. in bathrooms, must be avoided under all circumstances. Any use other than that described above will damage this product and involves the risk of short circuit, fire, electric shock, etc. The entire product must not be modified or converted, and the casing must not be opened.

⚠️   Observe all safety and assembly notes in these operating instructions!

1.3   SAFETY INFORMATION

⚠️ ⚡ The guarantee/warranty will expire if damage is incurred resulting from non-compliance with these operating instructions. We do not assume liability for any consequential damage. Nor do we assume any liability for damage to property or personal injury caused by improper use or failure to observe the safety information. In such cases the warranty/guarantee will expire.

Dear Customer:

The following safety information is intended not only for the protection of the device but also for the protection of your health. Please read the following items carefully:

• For safety reasons, any unauthorized conversions and/or modifications to the product are not permitted.

• All persons who operate this product, install it, assemble it, put it into operation or service it must be trained and qualified accordingly and must observe these operating instructions.

• Only a proper mains socket (220 – 240 V/50 – 60 Hz) of the public supply net must be used as voltage source.

• Set up the power analyser properly before operating it. Observe the operating instructions for it.

• When setting up the product, make sure that the mains cable is not pinched or damaged by sharp edges.

• Do not touch the mains cable if it is damaged. First, power down the respective mains socket (e.g. via the respective circuit breaker) and then carefully pull the mains plug from the mains socket. Do not operate the product anymore afterwards, but take it to a specialist workshop.

• This product is not a toy and not suitable for children. Children cannot judge the dangers involved when handling electrical devices.

• Never place containers containing liquids, e.g. buckets, vases or plants, on the device or in its vicinity. Liquids may get into the housing and impair electrical safety. This also poses great danger of fire or potentially fatal electric shock! If this is the case, first power down the respective mains socket (e.g. switch off circuit breaker) and then pull the mains cable from the socket. Disconnect all cables from the device. Do not operate the product anymore afterwards, but take it to a specialist workshop.

• Never touch the mains plug with wet or damp hands. There is the risk of a potentially fatal electric shock!

• Make sure the analyser is sufficiently ventilated when in operation. Never cover the device's ventilation openings with any magazines, blankets, curtains, etc. Keep a minimum distance of 15 cm from any other objects.

• Do not expose the product to any high temperatures, dripping or splashing water, strong vibrations or heavy mechanical stress.

• Never place any sources of open fire, such as lit candles, on or right next to the device.

• Consult an expert if you are unsure as to the function, safety or connection of the analyser.

• Never operate the product unatten

• Only use the device in a moderate climate and avoid any tropical environment.

• Do not leave any packing material unattended. It may become a dangerous toy for children.

• Also observe the additional safety information in the individual chapters of these operating instructions.

• Observe the safety and operating instructions of any other devices that are connected to the analyser.

• In commercial or public facilities, many laws and regulations of the employers' liability insurance associations must be observed for electrical systems and equipment, etc., e.g. regarding the prevention of accidents. As the operator, you are obliged to collect information on this from the appropriate offices and authorities and you are responsible for compliance with the applicable laws and regulations.

• If you are not sure about the correct connection or if any questions arise which are not covered by the operating instructions, do not hesitate to contact our technical support or another specialist.

2.   THE ATB 805 UNITE

Speaker out   function selector   output volume
                      calibration switch   channel 2 input selector

For the function Calibration and impedance the volume is disabled. For the impedance measurement it is set automatically corresponding to to the calibration measurement.

dBV	Spannung [V]
0	1,0
3	1,4
6	2,0
9	2,8
12	4,0
15	5,6

Table for output level for 9dB in the program

Microphone input, MC1 and MC2   audio out 1 = left, 2 = right
USB scket   digital out electrical, optitical

3. GETTING STARTED FOR WINDOWS WINDOWS 7, 8.1, 10

3.1 INSTALLATION OF THE USB DONGLE

The ATB PC Pro 64 program is protected with an USB dongle.
Insert the ATB program CD into your CD-ROM drive. In the workplace menu the driver can be found.
For Windows 8.1, 10 64bit the dongle driver do need a signature. Therefore the ATB dongle driver do not have it the process "Disable Driver Signature Enforcement" is required.

1. Start the operating system.
2. Windows 8.1: move the mouse in the upper right corner and click on Settings.
   Windows 10: Click on the Windows icon (start) at the bottom left
3. On the symbol "ON / OFF" button.
4. Shift Capitalize, click "Restart".
5. Only Windows 8.1: In the "Options" menu → "reboot"
6. In "Select Options"   →"Troubleshooting"
7. In "Troubleshooting"   → "Advanced Options"
8. In "Advanced Options"   → "Start settings"
9. In "Start settings"   → "Restart"
10. After the restart with the number "7" key "7 Driver Signature Enforcement
                                                 disable "call.

In the following time the operating system no signature is required when installing the dongle driver.

Installing the dongle driver

Connect the dongle to the USB connector, place the installation CD-ROM into the DVD drive 1.
Windows 7:   Start menu   → "Control Panel"
1. Windows 8.1:   Calling up menu "This PC" with Windows key + 1
                   In "This PC"   → "Control Panel"
1. Windows 10:   Click on the Windows icon (start) at the bottom left
                   In the menu   → "Settings"
2. Windows 7, 8.1: In the "Control Panel"   →"Hardware and Sound"
                   In "Hardware and Sound" → "Devices and Printers" →"Device Manager"
2. Windows 10:   In the "Settings"   → "Device"
                   In "equipment"   → "Connected bone"
                   In "Connected devices"   → "Device Manager"
3. Windows 7, 8.1: In "Device Manager"   → Other devices "ATB-LOCK USB" on
                                                 yellow triangle attention.
3. Windows 10:   In "Device Manager"   → "USB", "ATB-LOCK USB" on yellow
                                                 Triangle eighth.
4. Windows 7, 8.1, 10: Click "ATB-LOCK USB"   → "Properties ATB-LOCK USB"
5. Windows 7, 8.1, 10: In "Properties ATB-LOCK USB" → "Drivers"
6. Windows 7, 8.1, 10: In "Drivers" → "Update Driver"
7. Windows 7, 8.1, 10: In "Update Driver Software - ATB-LOCK USB" → "The computer
                                                 Search for driver software "
8. Windows 7, 8.1, 10: In the following menu   → "Search" and the DVD

Select drive: "Next"
9. Windows 7, 8.1, 10:   With "Browse" select the DVD drive → "OK" to complete
9. Windows 7, 8.1, 10:   In "Windows Security"   → "install driver software anyway"
10. Windows 7, 8.1, 10: The driver has been installed → "Close"

The dongle installation is shown in the folloing

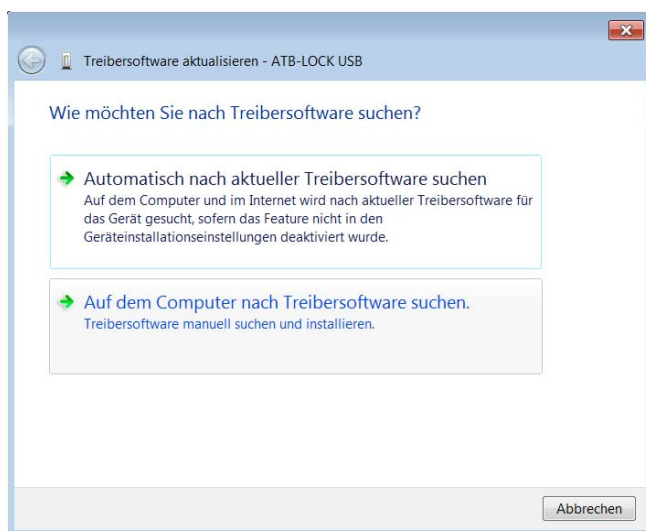
The ATB PC Pro 64 program is protected with an USB dongle.
Insert the ATB program CD into your CD-ROM drive. In the workplace menu the driver can be found.

By connecting the USB dongle to the port there will be an error. Windows can not install the driver automatically. The installation is to done manual.
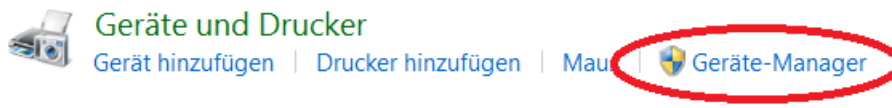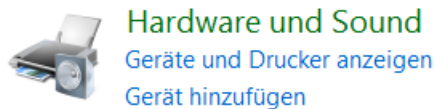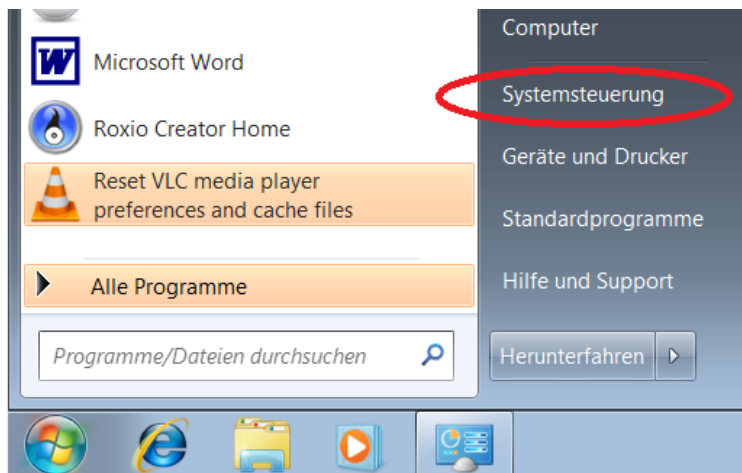
Driver software update
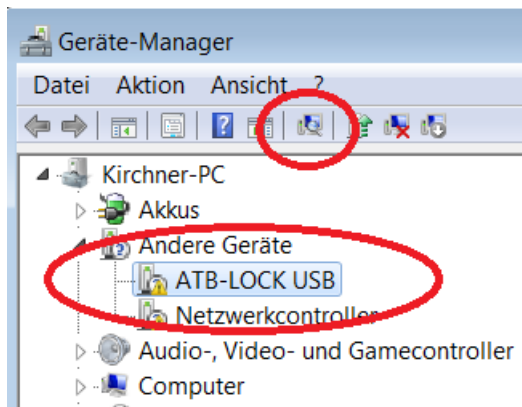
In the menu set the drive with the installation CD ROM
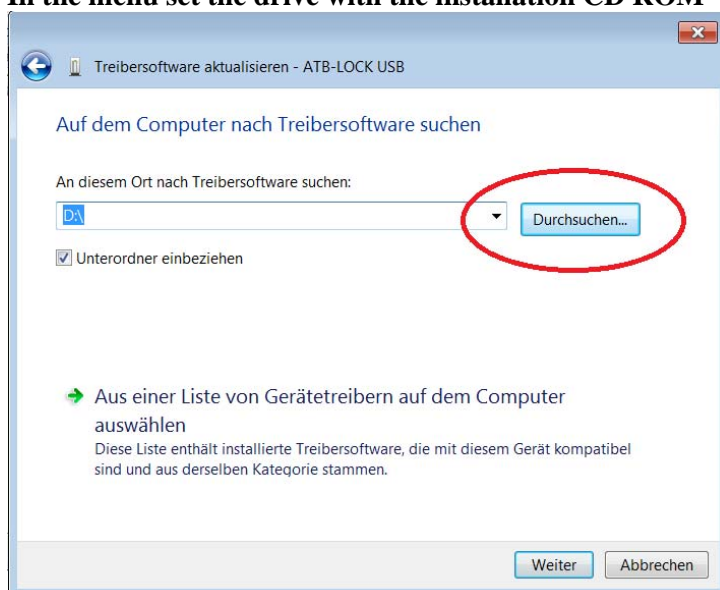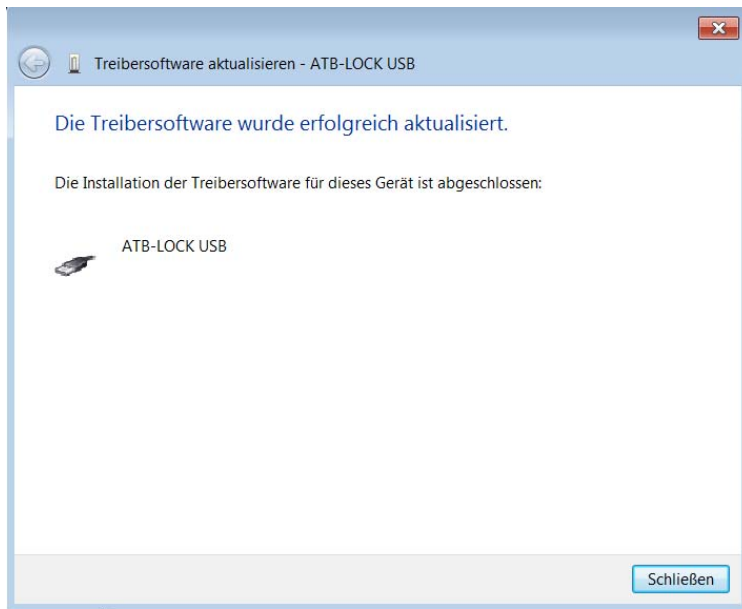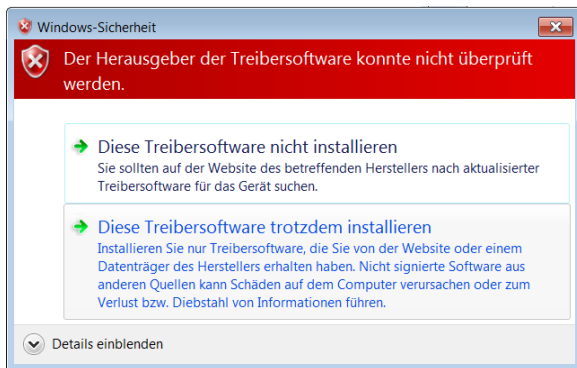
Ready!

If you want to check the correct installation of the dongle then open CONTROL PANEL in the START menu then open SETTINGS choose SYSTEM from the list then HARDWARE and then DEVICE MANAGER where you will find a list containing various devices including the BUS controllers :
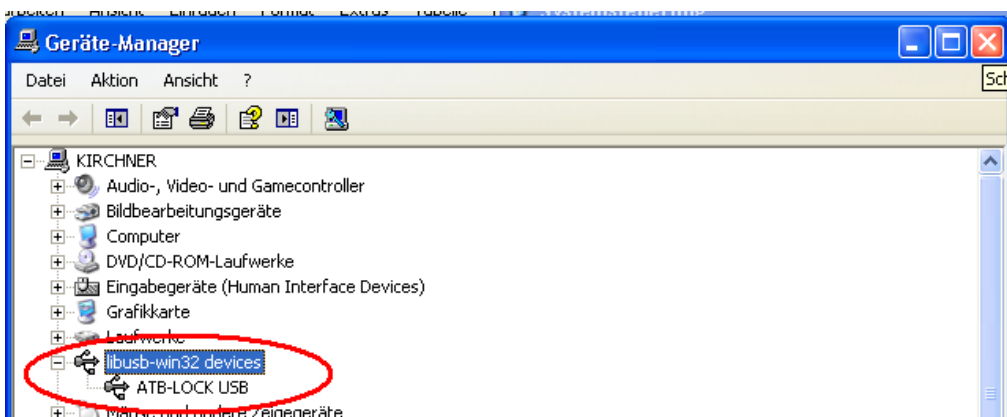
3.2   INSTALATION OF THE ATB 805 BOX

After connecting the ATB 805 via USB there will start an automatic installation.

Installation of the device driver

The device is ready to use.

3.3   INSTALLATION OF THE SOFTWARE

Place the DC „ATB-PC Pro Software" in the CD ROM drive of your computer.
Open the menu COMPUTER and select the CD.
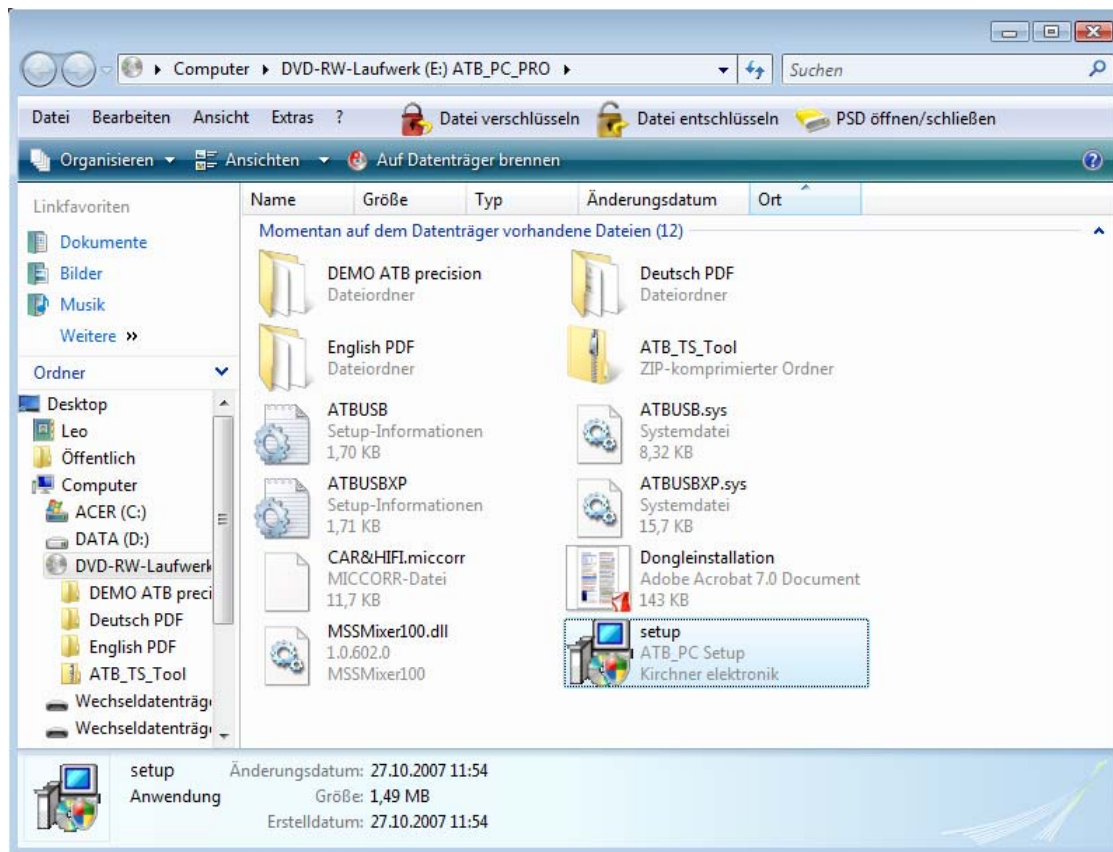
With double click "SETUP.EXE" the installation will start.
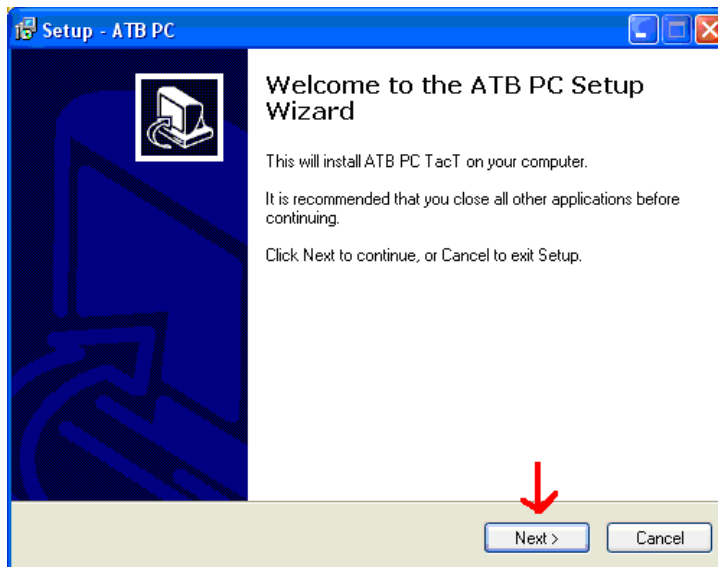
Follow the instruction.

3.4 SETTING UP THE INPUT AND OUTPUT Windows 7, 8.1, 10

Open the program with a double click at the icon.

1. INPUT
First you have to select LINE (USB AUDIO) or (2 channel USB AUDIO) in the INPUT menu.

There is no win-mixer in the Vista, 8.1, 10 program.

In the main menu is shown the correct setting

The menu can be found at
    SYSTEMCONTROL – HARDWARE AND SOUND – SOUND
In the SOUND menu Microphone will be selected.

2. OUTPUT

In the OUTPUT menu you have to select SPEAKER (USB AUDIO) or (2 channel USB AUDIO)
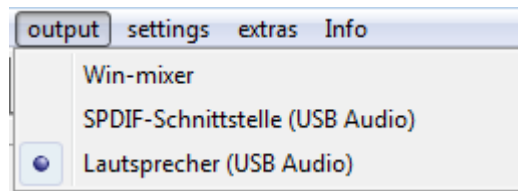
In the main menu the correct setting is shown

The menu Win-mixer shows the setting of the output.
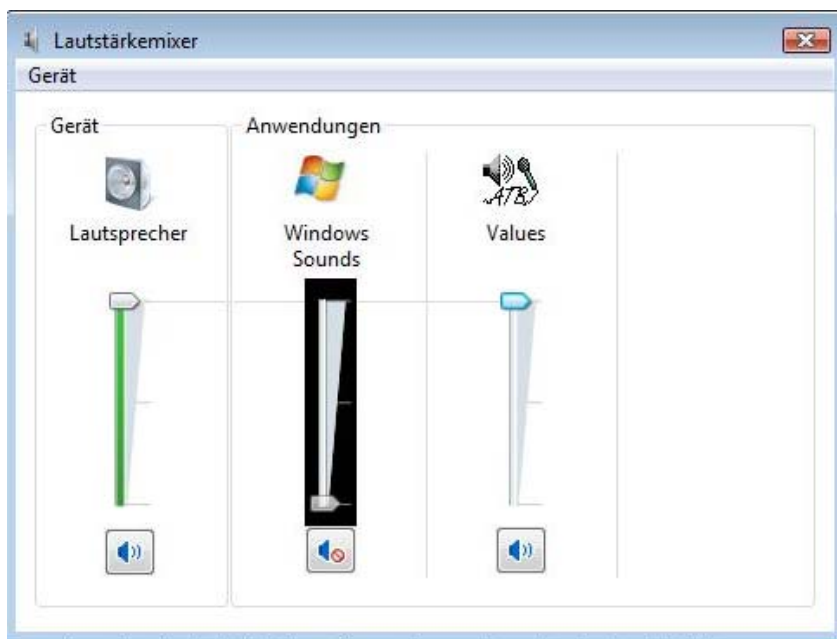
The menu can be found at
  SYSTEMCONTROL – HARDWARE AND SOUND – SOUND
In the SOUND menu Headphone will be selected.
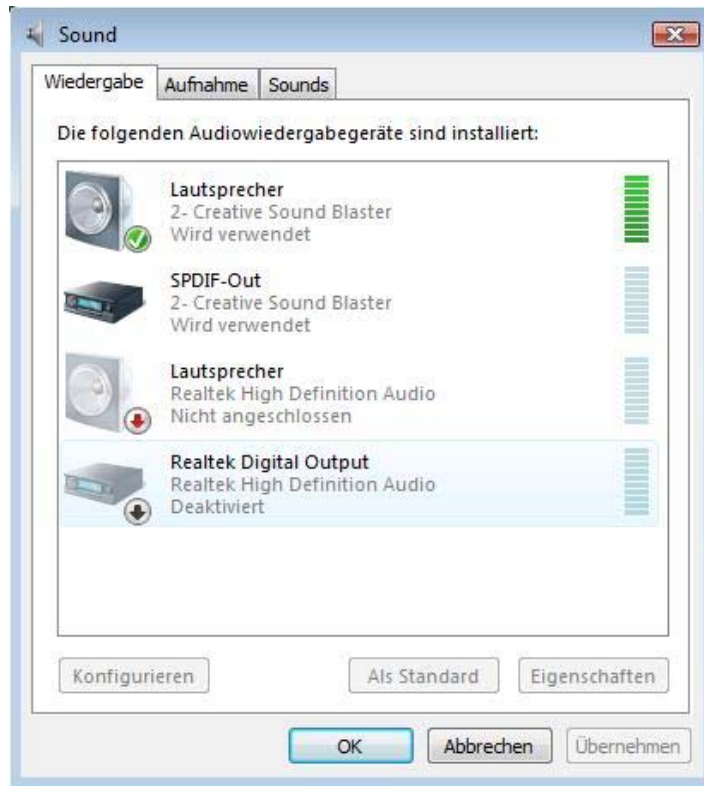
In the SOUND menu SPEAKER will be selected.

4.   CALIBRATION

4.1   SYSTEM-CORRECTION

Before you can do any measurement, the USB box will be tested and a correction file will be generated. In the System-correction, a correction curve will be generated for the input and output of the ATB 805 box.
The box is set to CALIBRATION. For the function Calibration and impedance the volume is disabled.
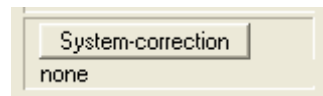
The System-correction menu is called up with a click on the [System-correction / none] button in the main menu.
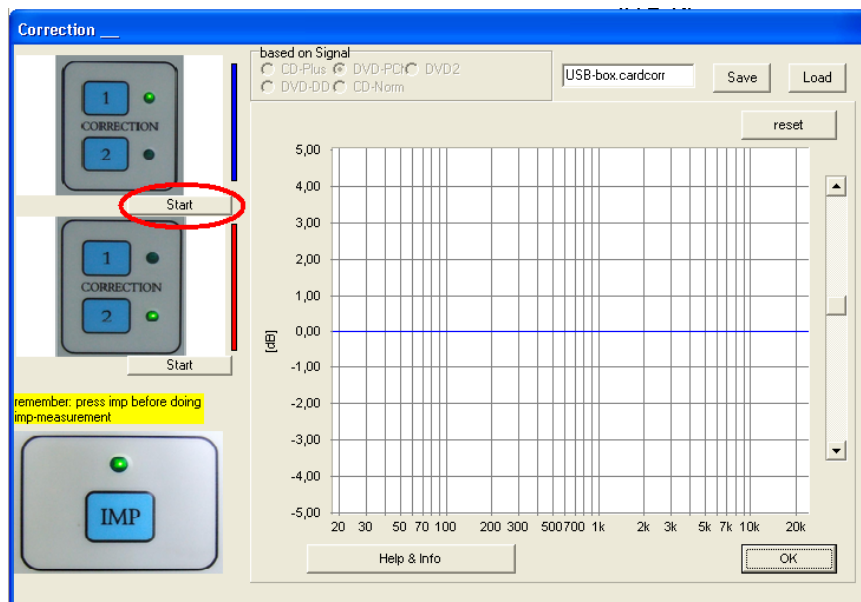
Switch the ATB 801 device to CALIBRATION and the switch in the back of the box to 1. Start the first measurement with START.
Attention: There is the message "signal too strong". This comes from a other setup and had no meaning for the ATB 805. For this and the next klick at OK.
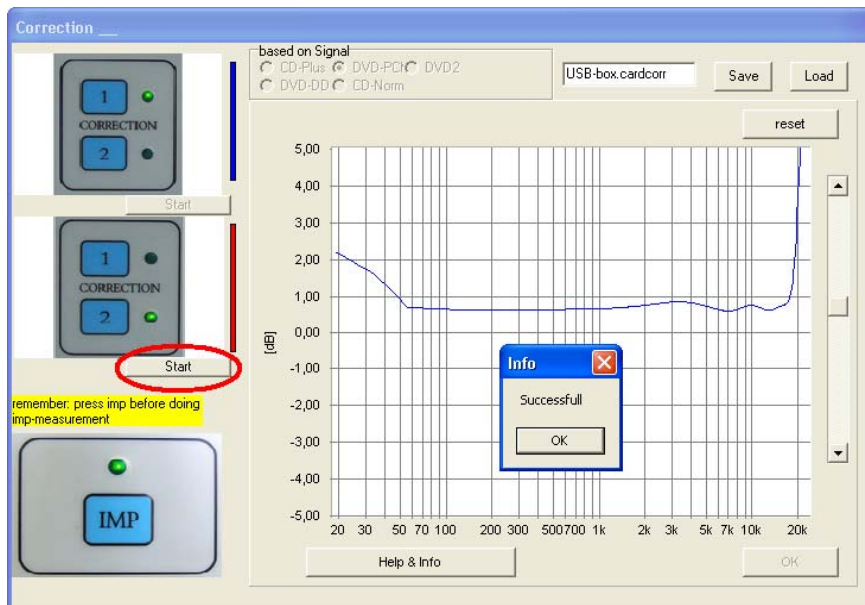
After the first measurement switch the ATB 801 Device to Correction 2 and start the second measurement. You will see the message" successful".
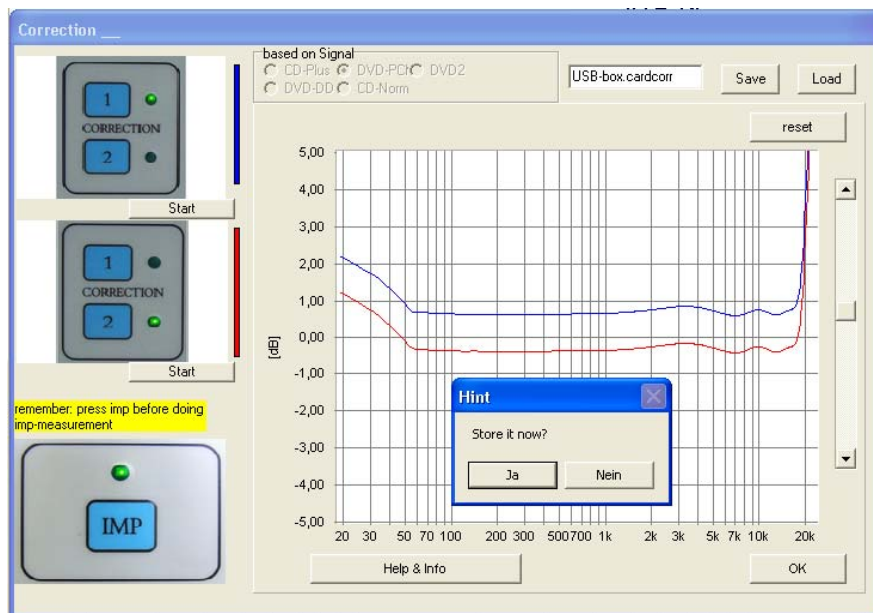
In the HINT menu the saving of the System-correction file is started with "Ja". The name could be "USB-Audio".
After saving, the correction curve is available for all following measurements and is automatically loaded at the start of the program.

4.2   SYSTEM-CORRECTION FOR THE SIGNAL FROM CD / DVD PLAYER

There are measurements which can be done with an external signal of a CD or a DVD player. For the test of the car sound and the HiFi sound system there is the Auto-Test CD. For the Surround equipment there is the Surround Test DVD.
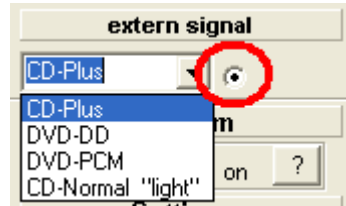
Choosing the signal

At this point you have the choice between selecting measurements with the Surround-Test DVD or the Auto-Test CD. For the Surround-Test DVD select the option DVD-DD (as shown above). CD-Plus you should select for the CD.
For the Measurement with DVD or CD signal there is a special system-correction.

For measure ring a new correction has to be done. Connect the CD or DVD player with the ATB 701 box. Switch the box to SPL.
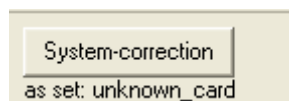
After pressing the button "system correction"

Start the two system correction measurements without switching the box to correction 1 or 2. Do not care about the message
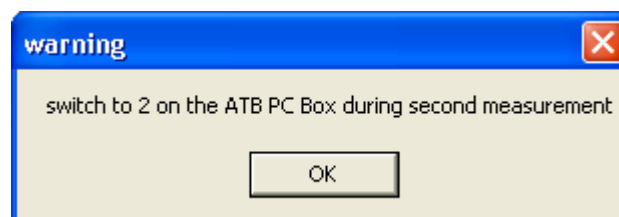
After the second measurement save the correction file with the name of the signal.

4.3 ERROR MESSAGE FOR THE SYSTEM-CORRECTION

Whilst CORRECTION is taking place, the following error indication messages may appear:

"Missing/weak signal" or "no signal"

The ATB 805 is broken.

"signal too strong"

The message was programmed for another unit. Please klick to OK

"insufficient record level"

The ATB 805 was broken.

"same level"

For the second measurement turn the calibration switch to position 2

5.   MICROPHONE

5.1   MICROPHONE CORRECTION

The ATB PC microphone MC 1 corresponds in precision of frequency response to the professional electret measuring microphones and does not need to be adjusted.

The ATB PC measurement program makes the adjustment of as many microphones as desired possible. This function allows many other adjustments:
1. The adjustment of bass elevation.
   First choose the position where loudspeakers and microphone are to be installed. A loudspeaker that has been already tested in an open space, large hall or outside should be chosen. After the test has been carried out the frequency characteristics are compared and the difference with MIC-CORRECTION curve negatively adjusted.
2. Acoustic adjustment of car interiors.
   A linear frequency characteristic is not recommended for car interiors. Frequency characteristics that comply with CAR&HiFi are audibly better.
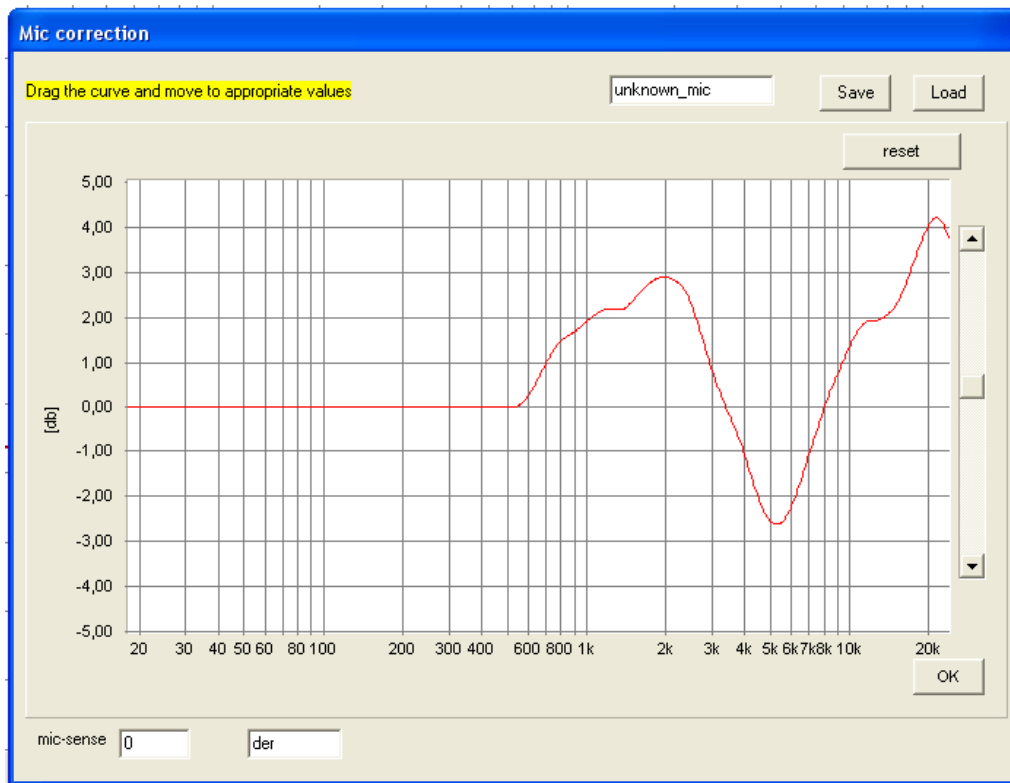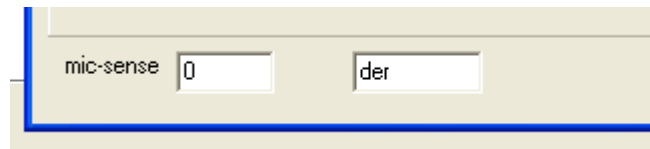
These frequency characteristics can also be adjusted with MIC-CORRECTION. Thus only the straight frequency characteristic must be taken into consideration. Audible adjustment is the appropriate adoption. Open MIC-CORRECTION in the SETTINGS menu.

Using the pointer the curve representing the recorded frequency characteristic of the microphone applied is adjusted. To carryout this negative curve is applied, a bank in the recorded frequency characteristic becomes a fall (break).

5.2   MICROPHONE SENSITIVITY

MICROPHONE SENSITIVITY is necessary for SPL measurements. Because of the differing sensitivity of microphones a correction factor is used. The correction factor is to set in MIC.-CORRECTION and is a part of the microphone adjustment. A frequency characteristic correction is not necessary when using the delivered microphone MC1. Nevertheless after ascertainment of the microphone correction factor is the unchanged curve with the name MC1 to be saved.

mic-sense	0	der

The correction factor is also to set in the parameter menu.

The ATB microphone is calibrated. The microphone has its own correction factor.

You have to write the factor into the mic-sense field.
Nevertheless after ascertainment of the mic-sense factor the curve is unchanged. Save it with the name MC1.

6.   PROGRAM

6.1   INPUT AND OUTPUT CONTROLLER

The OUTPUT level

The OUTPUT level can be set with the OUTPUT controller.
The displayed dB level is for the sin-Signals in dBV. 0db = 1V

The INPUT level
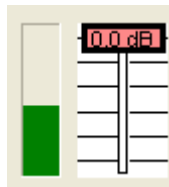
The INPUT level can be set with the INPUT controller. You can turn it up, but the bar should not turn red. The steps of the controller are 1.5dB. The value of the controller can not be seen in the plot. The plot shows independent from the INPUT controller and the input level.

6.2   DISPLAYS

The level meter

The level meter shows the RMS sound pressure level in dB.

The distortion meter

In the Mag measure with the sin signal is shown the total harmonic distortion.

The intermodulation meter

In the Mag measure with the RB signal is shown the Q-Index.

6.3 THE MEASUREMENTS

Frequency response of amplidude

Amplitude   **Sound pressure**   **Impedance**

| Mag | SPL | Imp |
| Phase |

Frequency response of phase

| Mag | SPL | Imp |
| Phase | ac: 1 kHz rect   ▼   Step

Step response

| Mag | SPL | Imp |
| Phase | ac: 1 kHz rect   ▼   Step

Measurement with analysis

Thiele-Small   **Distortion**   **Quality**

| Main | THD |
| TS | QC |
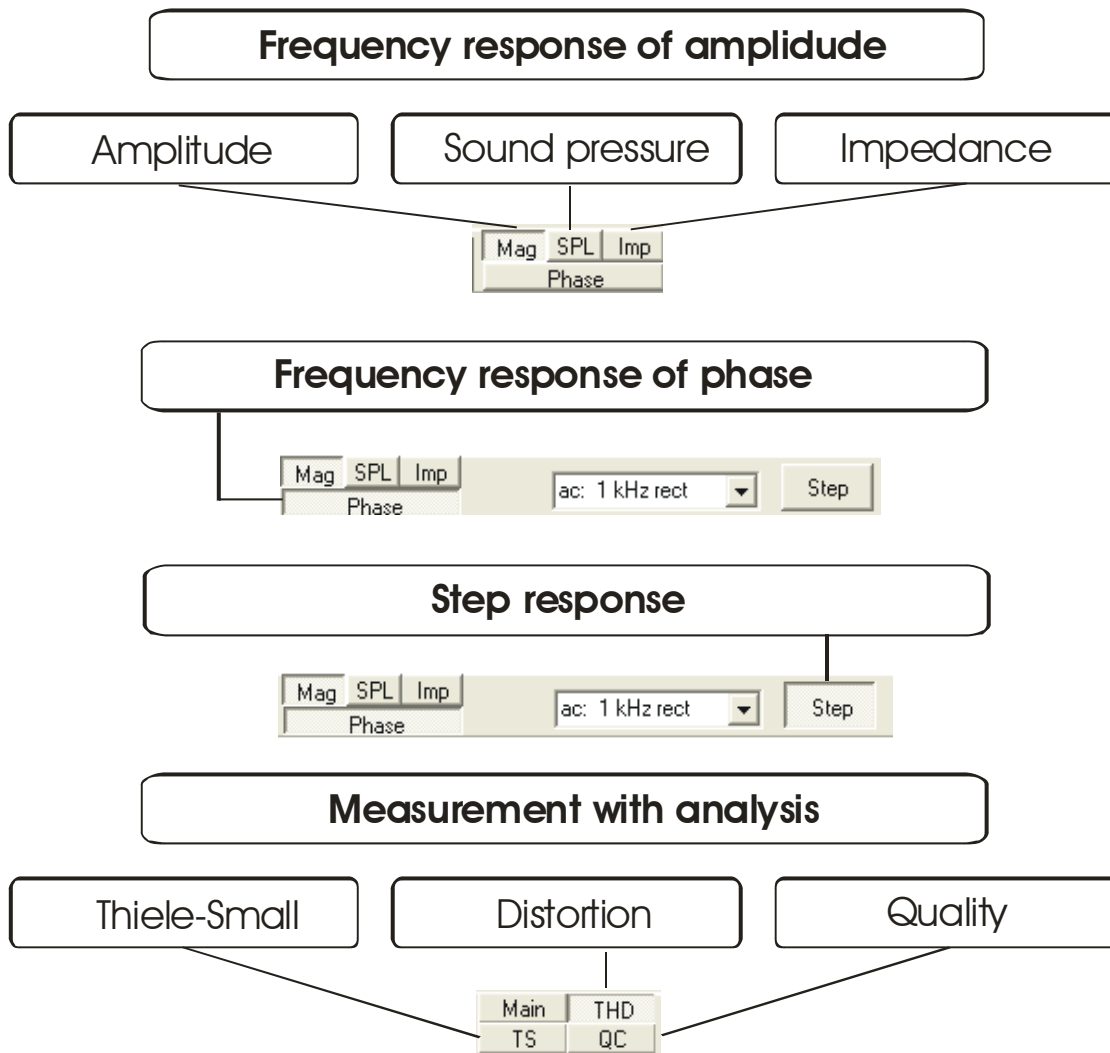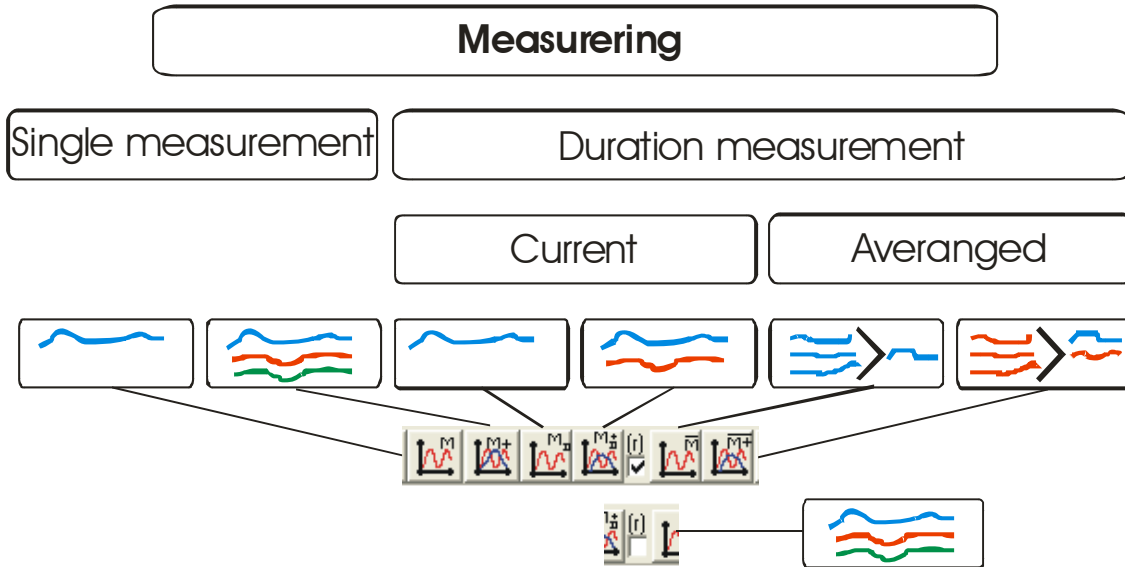
6.4 THE MEASURERING

6.5 DISPLAY ANG HANDLING
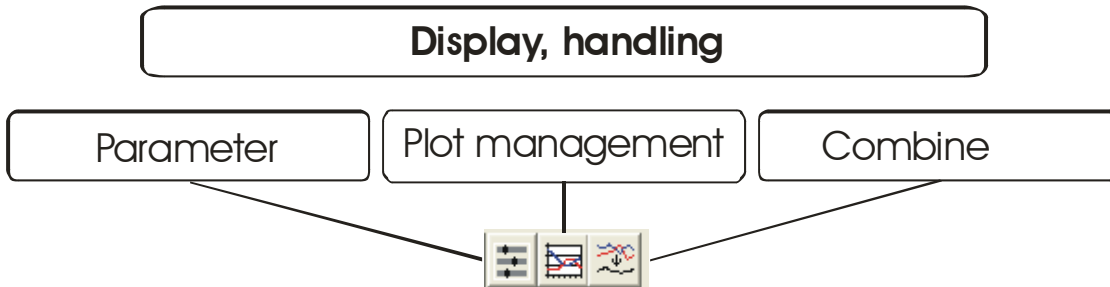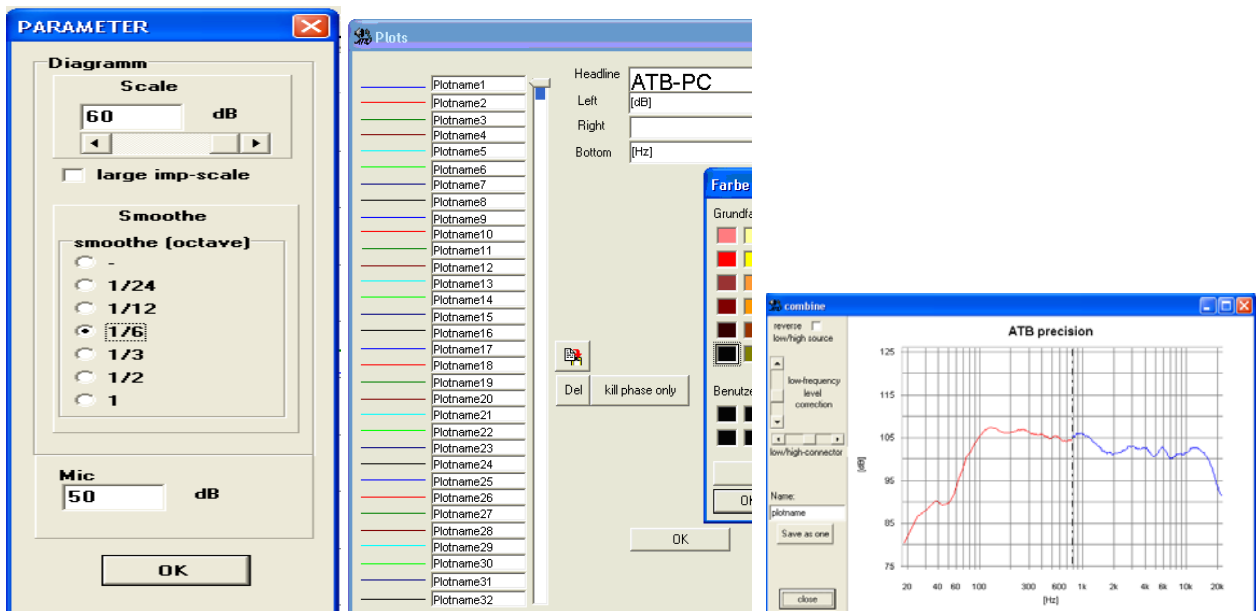
6.6   CARRING OUT THE MEASUREMENT

THE MEASUREMENT

For the measurement the speaker and the microphone has to be connected to the USB box.

The picture do not show the right positions of switches. It is SPL.

Switch the selctor to SPL and the volume to 9dB (SPL) for the 1w/1m SPL

Put the microphone 1m away and in front of the left front loudspeaker and connect the microphone to the microphone socket of the ATB 805 box. The microphone should be positioned at "ear height". Usually it is about 80-90 cm above the floor, as music is usually listened from people who are sitting (unless you are dancing of course!)

Select the SPL measure
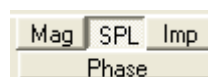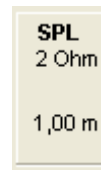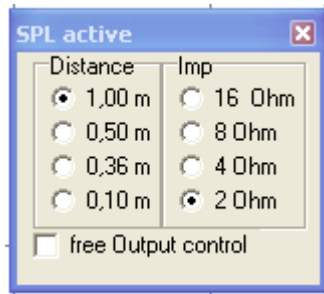
Following menu is shown

In the SPL active menu you can set:
   The impedance of the speaker. The program set the output power to 1W.
    The distance between speaker and microphone.
For a distance smaller than 1m the program shows the SPL for 1W/1m.
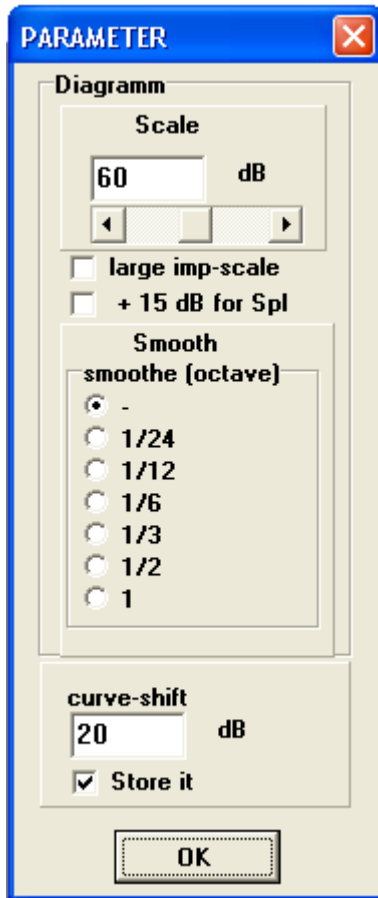
Choose "parameter" from the menu and position the smooth value to1/6:

Activate the button "duration measure"
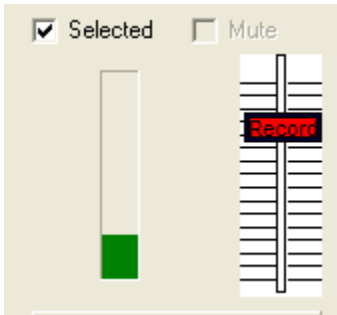
The modulation display indicates microphone-level.
With the slide controller (record) you can adjust the microphone sensitivity.
The green bar should fill 1/3 of the display window.
The bar turns red if the microphone input is overload (set to a too sensitive position).
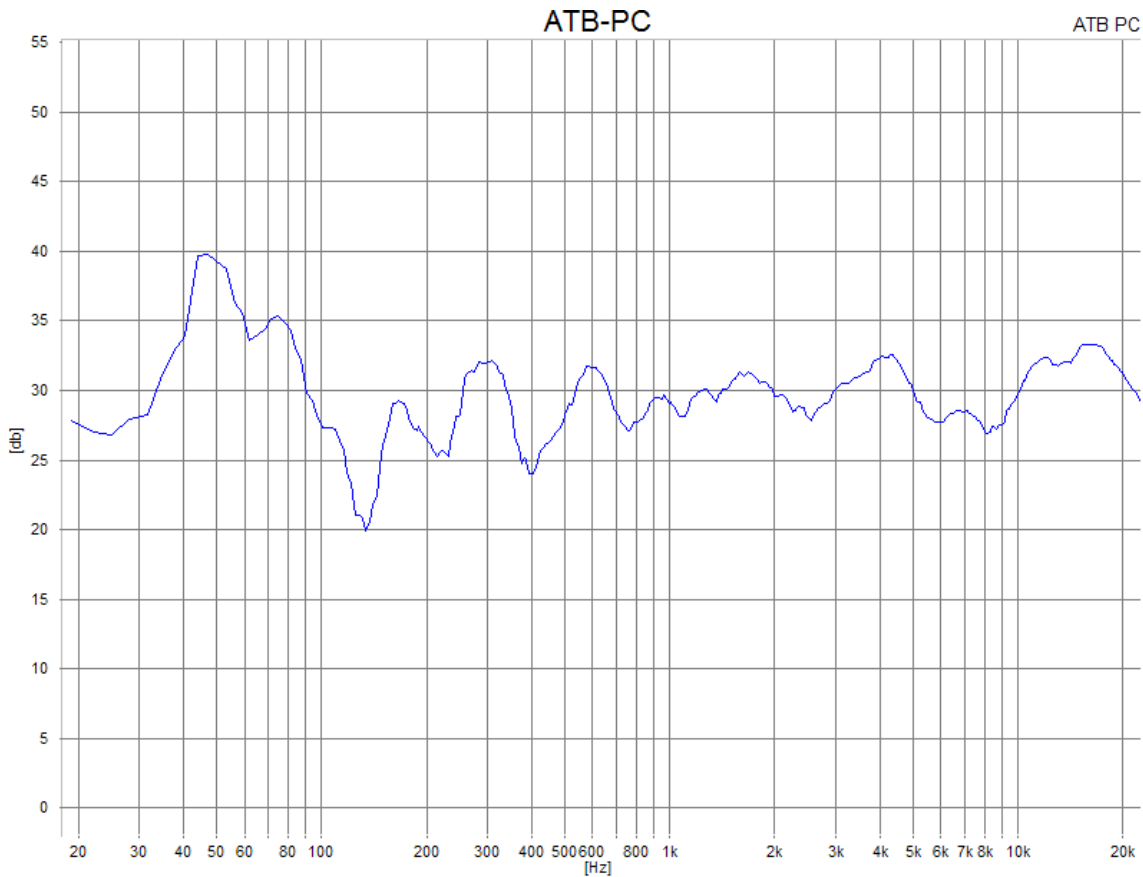
The frequency response curvature of the left hand loudspeaker should appear on the screen.

7.   MEASUREMENT SIGNAL

7.1   SETTING

Before carrying out a test the measurement signal must be adjusted.

Signal:
    BMH   BassMidHigh, full range noise for SPL and Impedance measurement
               It is the M-PN signal. It is equal to the PCM signal
    B   Bass, bass noise for woofer testing
    MH   MidHigh, noise for testing tweeter
    RB low   Rub&Buss signal for testing woofer, frequencies 180Hz, 234Hz
    RB mid   Rub&Buss signal for testing mid-speaker, frequencies 500Hz, 650Hz
    RB high   Rub&Buss signal for testing tweeter, frequencies 1,5kHz, 1,95kHz
    Sin   Sine wave for THD measurement.
    1/2Sin   positive impulse for testing the polarity in the QC measurement

For the measurement with extern signal there are following signals:
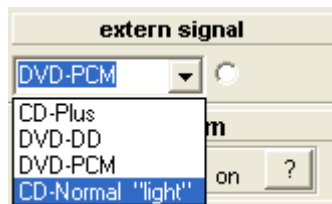
There are two signals on the CD and two signals on the DVD available for measurement purposes.

Signal:
    CD-Plus   full range noise for SPL and impedance measurement.
    CD-Normal   full range noise for fast SPL measurement
    DVD-PCM   full range noise from the DVD. It is like the BMH signal
    DVD-DD   Dolby Digital signal from the DVD

The signals DC-Plus and CD-Normal can not be found on the DVD.

CD-Plus is the usual test signal used in most measurement processes. It corresponds to the DVD signal in relation to its frequency distribution. The signal is a x 4 over sampling and it is almost analog. This signal is also the internal generator signal.

The signal CD-Normal is a measurement period of 1/3 s allows extremely fast results to be acquired. This makes the equalizing of filter switches very easy to carry out.

The quality of measurement reproduction is constant even when operating with high frequencies.

Dolby Digital (DD) signals and PCM signals are located on the DVD.
The surround test uses the surround test DVD and the DD signal as source.
The DD and the PCM signal differ in their higher threshold frequencies. The DD signal has a threshold of 20kHz and the PCM signal 22kHz.

The PCM signal is used for loudspeaker testing. It has a 24-bit resolution and a sample rate of 48kHz. Up to 22kHz can be measured with this signal.
The PCM signal is also used for digital measurements.

IMPORTANT!
The adjustments and system correction must comply with the test signal.

7.2   THE BMH, B AND MH SIGNAL

For measuring the BMH signal is used. It is a deterministic noise, a calculated signal, with the name M-PN (Metzner-Periodic-Noise). Through the function of a x 4 over sampling it is an analog signal. One of the considerations during the development of the signal was the adoption of the measurement technology to the later function of the instruments tested. A loudspeaker is for the transmission of music and speech. In correspondence with this the capacity of the loudspeaker is defined for the frequency range. In accordance with DIN is the performance at 100% total capacity as follows:

Low frequency   from 40Hz – 600Hz   62%
Midrange   from 600HZ – 4kHz   30%
Tweeter   from 4kHz – 20kHz   8%

Because of its steady energy distribution the Pink-Noise signal is adapted to the frequency characteristic of the music. The MLS (Maximum Length Sequence) signal is the exact opposite.
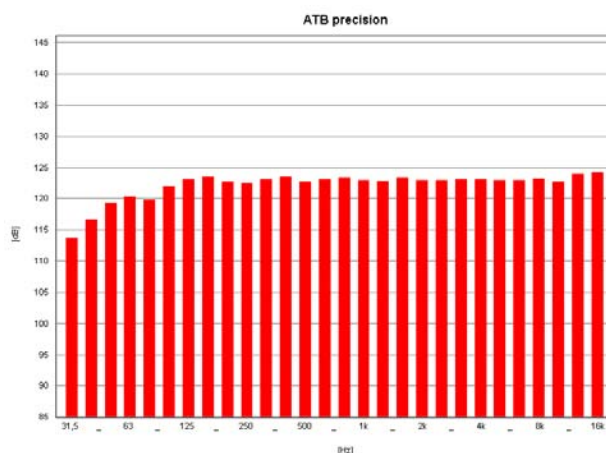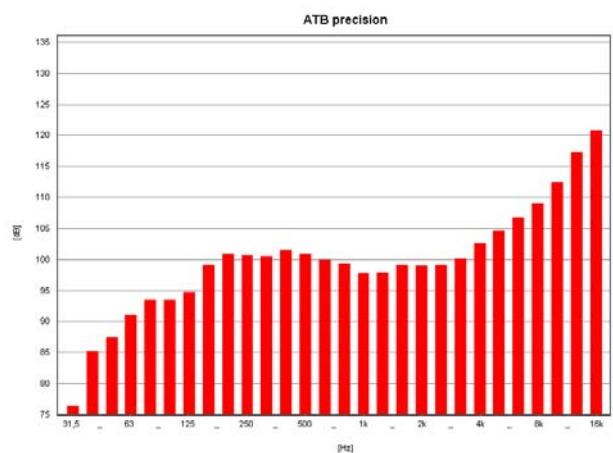
M-PN   MLS

The pictures show the analyser measurements of the M-PN and MLS signals. Both signals are a deterministic noise (a frequency mix). The M-PN signal shows a pink noise. Practically all 1/3 octave bars have the same amplitude. The drop in the low frequency range is not to be taken into consideration for measurements. The MLS signal tends to comply with white noise. This is easily audible when testing. The high frequencies have extremely high amplitude and the low frequencies are contained in the signal with –45dB. In opposition to 1/3 octave analysis the MLS signal appears after testing through the correlation with a straight frequency characteristic. A correlation has the following functions:

The loudspeaker is steered with the MLS signal shown above. If the loudspeaker has for example a linear frequency characteristic then the microphone used will record the same signal. It is changed into a digital signal within the measuring instrument. This signal as well as the MLS signal is withheld in the computer in digital form. During the correlation both signals are compared. The result (frequency characteristic) consists of a picture of the differences. If both signals are the same (in the case of the linear loudspeaker) the result is a straight line. If the loudspeaker does not transmit the low frequency range very well, then the difference between the signal measured and the output signal are pictured as a drop in the frequency characteristic.

That the MLS signal is not suitable for testing is proved by the following:

The loudspeaker is steered with the MLS signal. The energy distribution of the MLS the exact opposite of the music signal; in the high is a lot and in the low range nearly no energy.

The results of this are:

1. The extremely high energy of the high frequencies overload the high tones, so that a part of the frequency characteristic measured consists of dynamic compression and distortion. The result is that one of the music transmission frequency characteristics cannot be measured.
2. Because of the very small signals in the low frequency range an exact and reproducible measurement is nearly impossible. This leads to the need for at least 10 measurements with the MLS signal before an exact result (with adaptive windows) can be expected.

The M-PN Signal owns a steady energy distribution. It does not overload the high tones and has enough signal amplitude in the low frequency range. Each measurement is reliable (even without an average mean).

7.3   THE RBLOW, RBMID AND RBHIGH SIGNAL

The RB Signal is for the IMD, intermodulation distortion measurement. We give it the name Signal 13. The Signal 13 is for nonlinear distortion measurement.

Wolfgang Klippel, Germany, describes the distortion from nonlinear systems.

The basic for this measurement is a generator signal with consist of two sine waves. The signal is modulated at the nonlinear transfer response of the system under test. The result is the THD and the IMD spectrum.

The Signal 13 is for the quality control in the speaker industry. It is used for the Rub&buss measurement to test the mechanical quality of the speaker. In production line the speakers tend to have defects. These defects are cone displacement, rubbing voice coil and loose particles. The defect speaker produces an unwanted noise. An analysis of the noise shows short time and magnitude distortion. In former time the noise was tested with a THD measurement with analysing the energy of a high-order harmonics group. These are K13, K14,……..K20. For a real test there is the need of a non-static signal. Therefore, the newer systems used a sine-sweep or a combination of step and sine. The measurement with these signals has a problem with the surrounding noise. Therefore, the Signal 13 was developed from the Kirchner elektronik company. The signal consisting of the heterodyne of two sine waves with the ratio of 1:1,3. This

gives a non-static signal. The signal 13 is for testing distortion like THD, IMD und nonlinear distortion.
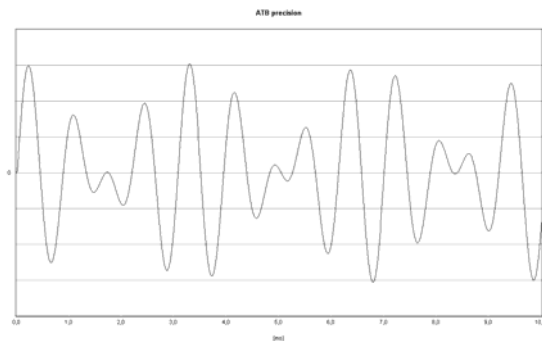
The time domain of the Signal 13
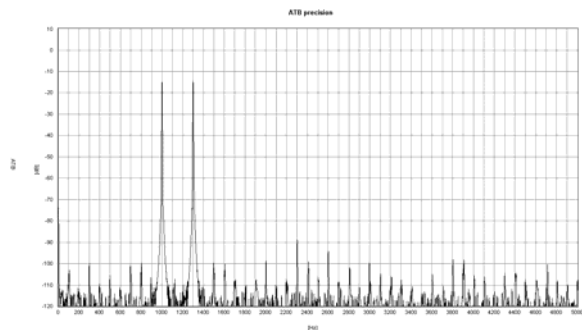
The frequency spectrum of the Signal 13

The time domain of the signal shows an always-varying wave. The signal excited all resonances of the unit under test. The resonances and the nonlinear transfer function show a spectrum with a great number of distortion frequencies.

For the calculation of the Q-index the amplitude of distortion frequencies are added and set into proportion to the amplitudes of the Signal 13. This is a very good test for the quality of a device.

For acoustic test of the speaker a quiet room is necessary. With the Signal 13 the Rub&Buss measurement is not so disturbed from the surrounding noise as other measurements. Because the frequencies below the Signal 13 are not considered, a HPF (High pass filter) suppresses low frequencies.

7.4   THE SIN SIGNAL

The sin signal is for the THD, harmonic distortion measurement.
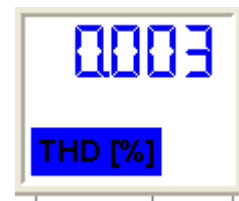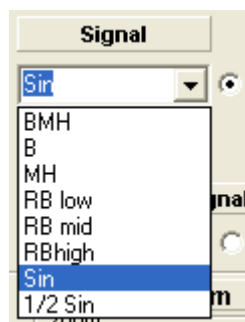
8.   PLOT MANAGEMENT

8.1   HANDLING SINGLE MEASUREMENTS

Up to 32 single measurements can be handled with the window in the program or the menu PLOTS.
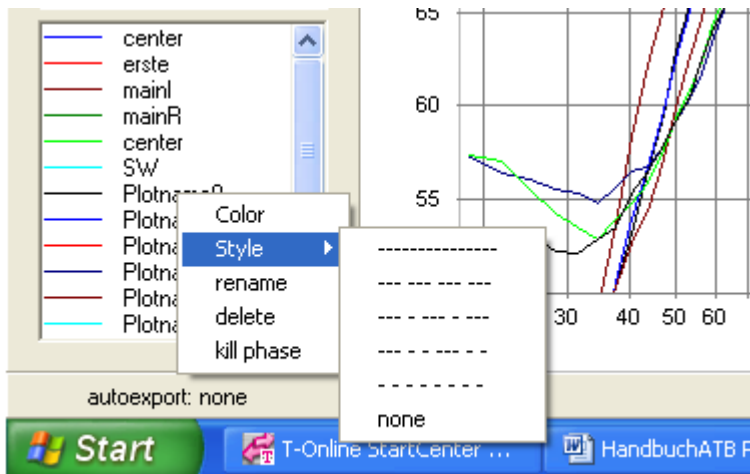
With a right mouse click at the plot name there is to activate the menu.
The menu performs following tasks:

1. To select a colour for the plot:
   Click the colour in the colour window.

2. Selecting PLOT line type with style:
   In front of the field for plot name the colour and line type are displayed.
   After selecting the plot to be changed with the cursor the colour/style menu is opened with a right mouse click.
   After selecting style the menu for the different types of line is displayed, choose the one desired.
   This function allows the different lines to be identified when printing in black and white.

Important! This only functions with a Print-line width of 1.

3. To name the plot:
   After actuation of the RENAME button enter the name of the graph in the NAME window.

4. To ERASE the graph:
   Actuation of the DELET button erases the graph.

5. To ERASE only the phase of a plot:
   Actuation of the KILL PHASE button only erases the phase of a graph.

The menu is opened with
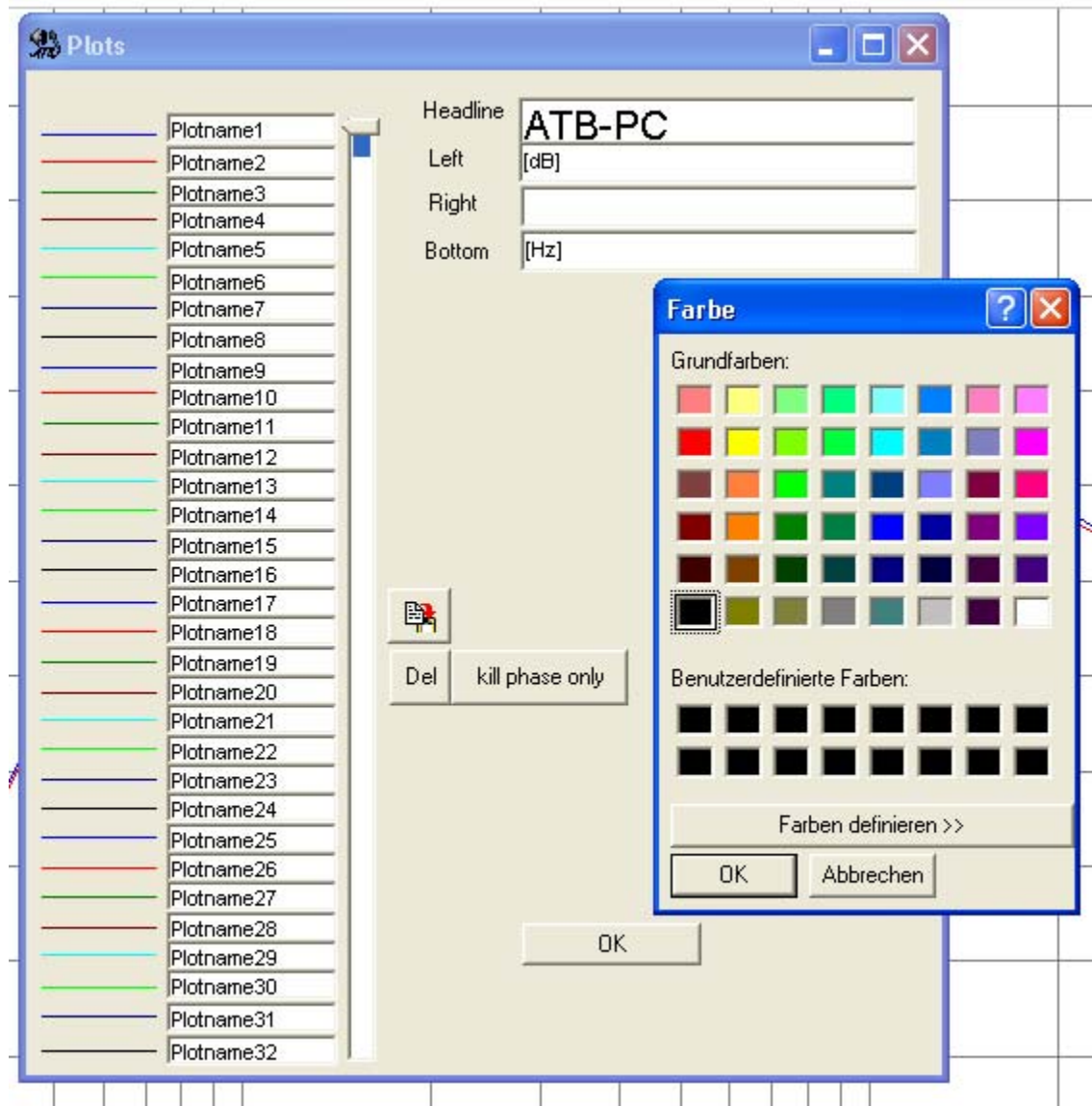
The PLOT menu performs following tasks:

1. To select a colour for the plot:
   Click the colour in the colour window.

2. Selecting PLOT line type:
   In front of the field for plot name the colour and line type are displayed.
   After selecting the plot to be changed with the cursor the colour/style menu is opened with a right mouse click.
   After selecting style the menu for he different types of line is displayed, choose the one desired.

This function allows the different lines to be identified when printing in black and white.

Important! This only functions with a Print-line width of 1.

3. To name the plot:
   Enter the name of the graph in the NAME window.

4. To copy the graph into the clipboard:
   Click on the COPY field. The measured values are copied into the clipboard in the ATB format.
   Retrieval can proceed with any Windows program.

5. To ERASE the graph:
   Actuation of the DEL button erases the graph.

7. To ERASE only the phase of a graph:
   Actuation of the KILL ONLY PHASE button

8.2   CURSOR

The cursor enables plot value evaluation. Activate the function with a double click on the plot you wish to evaluate (the mouse cursor appears as a cross when led over a curvature). The window VALUES will appear.
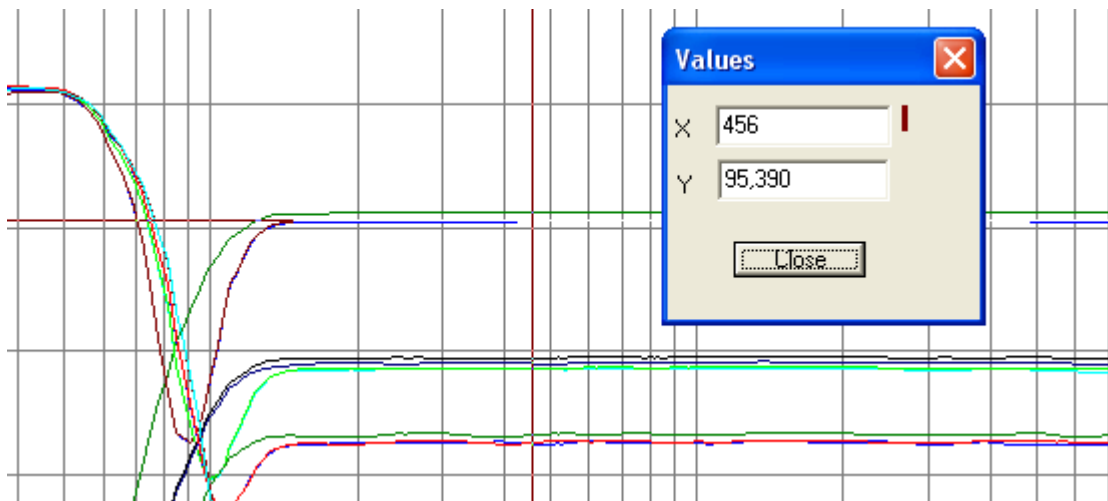
The VALUES displays following values:
        X = Frequency
        Y = Amplitude
When the values are read out the frequency is predefined. The X value is selected with the cursor.

8.3   THE ZOOM FUNCTION

A zoom function is contained in the main menu. After clicking ZOOM with the left mouse button
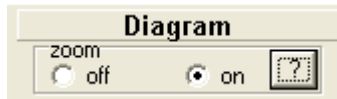
it is possible to (the left mouse button is held pressed) mark the area intended for closer inspection. The area selected will then automatically be enlarged. The frequency range remains unchanged for further measuring. To return from the zoom function NORMAL is then selected.

The frequency range can also be adjusted with the zoom function.

9.   GENERAL PARAMETERS

9.1   RESOLUTION

The resolution gives the scale of the Y-axis. Its upper limit is fixed at 0dB, which automatically gives the lower limit a negative value. Reduction of the shown can thus be compared to a zooming in to an existing diagram. 'Zooming' presupposes, however, adequate transformer level control.

9.2   SMOOTHING

The parameter SMOOTH is there to smooth out a line described by a plot curve.

```
          Smooth
smoothe (octave)
   ⦿ -
   ○ 1/24
   ○ 1/12
   ○ 1/6
   ○ 1/3
   ○ 1/2
   ○ 1
```
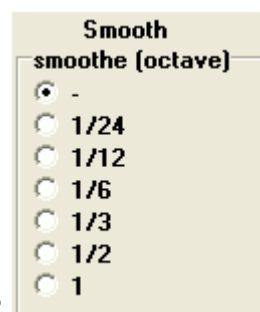
The smoothing of the plot is set in the PARAMERTER   menu.

It is in the nature in particular of acoustical measurements to produce rather coarse graphs. Such irregularities in the graph may blur its major characteristics, hence the smoothing option.
The SMOOTHING parameter determines the smoothing radius, which is the number of previous and subsequent measured values used in calculating the value currently displayed. The setting range is displayed in octaves (1/24, 1/12, 1/6, 1/3, 1/2, 1).

9.3   THE SMOOTHING FUNCTION

Smoothing implies that the different plots are redrawn, while the measured values remain intact. To return to the original (unsmoothed) plot, change the smoothing factor back to 0. Except for waterfall diagrams where for optical reasons some smoothing is indispensable, a smoothing parameter 0 implies that there is actually no smoothing. Apart from such intended smoothing there is no hidden measured value manipulation.
The resultant of a smoothed plot is similar to that of a wobble measurement, or, in other words, the smoothing radius corresponds to the wobble range of the sinusoidal wave.

10.   WINDOWS - PROGRAM

10.1   STARTING ATB PC Pro

If the program group ATB for Windows is shown as a symbol, it must be opened with a double

click on the symbol.

After this the ATB is opened.

10.2   THE CONFIG FILE

The complete settings of the program are written in the config file C:\WINDOWS\ATB_Prec.ini.

11.   OPENING DATA FILES

With the command FILE • OPEN you can open a file saved on the disk drive.

The following pop up window will appear:
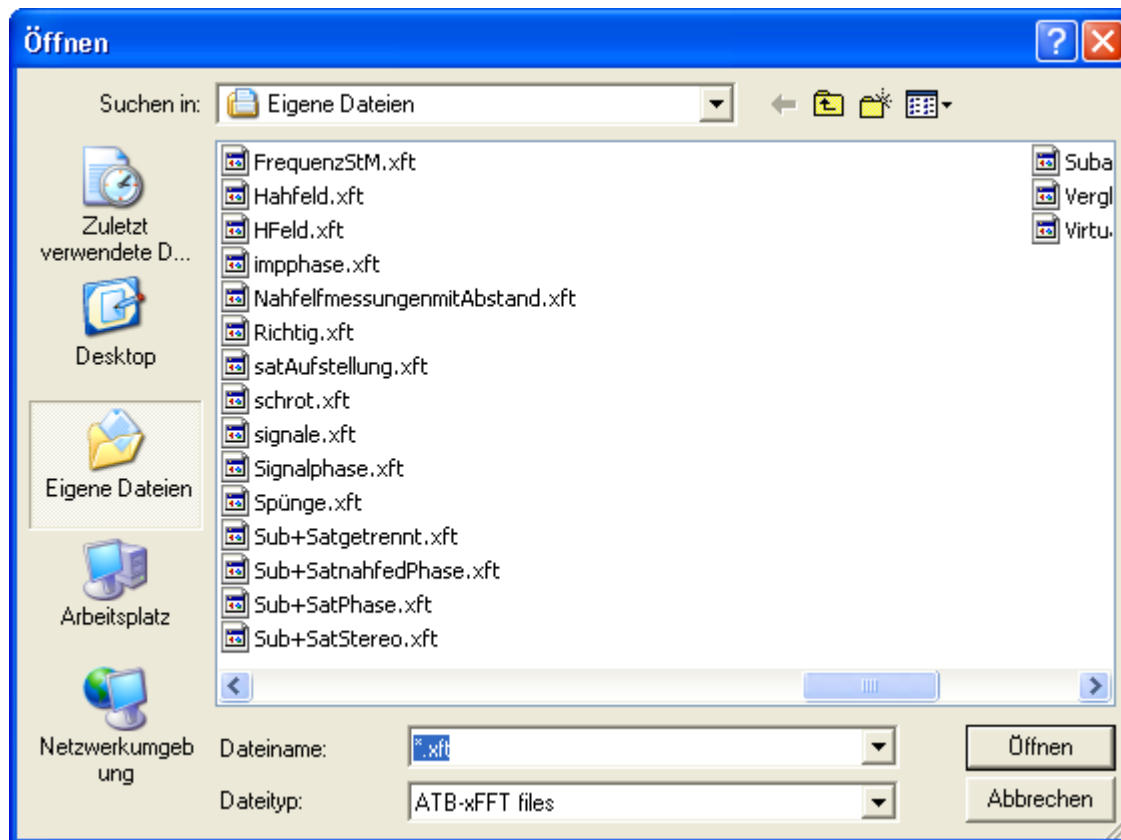
The pop up window shows the data in a file.
Disk drive and files can be changed with SEARCH IN:
Click on the desired data, the name will appear in the box FILE NAME. Clicking on open will open the following menu:
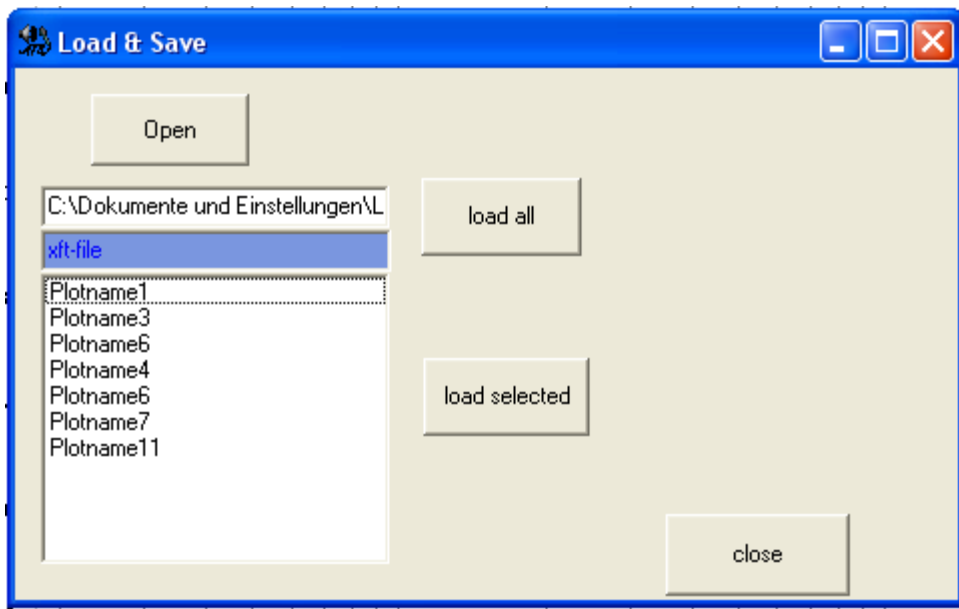
The buttons in the menu have the following functions:

OPEN   Clicking on OPEN jumps back to the menu OPEN to enable a new data
                    selection.

LOAD ALL   loads all plots in the file list (previous plots will be deleted).

LOAD SELECTED   loads a selected plot into the diagram.

CLOSE   closes the function LOAD SELECTED

The name of the opened file is shown in the program.

12. SAVING DATA

After creating or changing a file, the file should be saved. This enables you to retrieve and work with the file later.

When saving for the first time the following pop up window will appear:
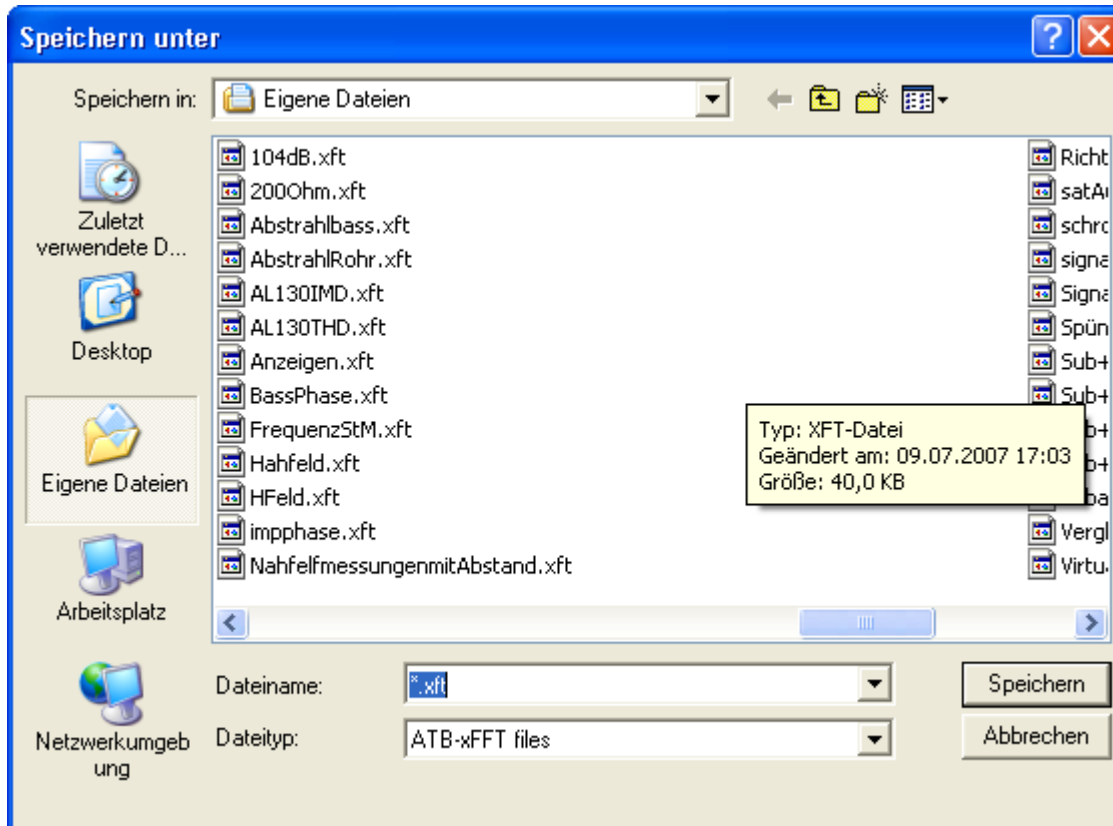
The name of the data to be saved should be written in the box data name. It is suggested that the file extension .xft be used. When you click SAVE, the file will be saved under the name you have chosen, in the dir you have selected. The pop up window is closed by clicking SAVE. Also you can break OFF the saving by clicking CANCEL.

.

13. EXPORTING DATA

The ATB program has many export formats so that data can be exported to other programs. It is an ASCII text format. This allows the data to be read by many programs.
A lot of CAD programs need a special format to read loudspeaker data for example. To be compatible with nearly all known programs the ATB PC Pro has the function to export data in several formats.
The EXPORTMENU is to open in the menu FILE

File	Measure	Info
Open		
Save		
Autoexport (ATB)		
Export		
Exportmenu		
PrintConfig		
Print		
End		
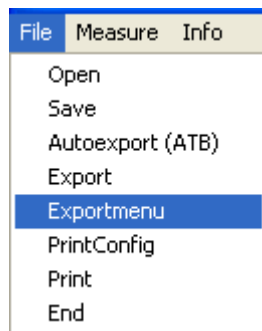
In this menu you have to select the format.

Exportmenu

format
- ⦿ ATB normal (*.txt)
- ○ ATB pure data (*.txt)
- ○ LAUD(*.frd)
- ○ LAUD(*.zma)
- ○ MLSSA(*.fmp)
- ○ MLSSA(*.fm)
- ○ IMP(*.frd)
- ○ IMP(*.zma)
- ○ IMP(*.zfr)
- ○ Clio-SPL(*.txt)
- ○ Clio-IMP(*.txt)

Delimiter
- ○ ;
- ⦿ |

Dec-seperator
- ○ ;
- ○ ,

☐ define as standard   [Export now]
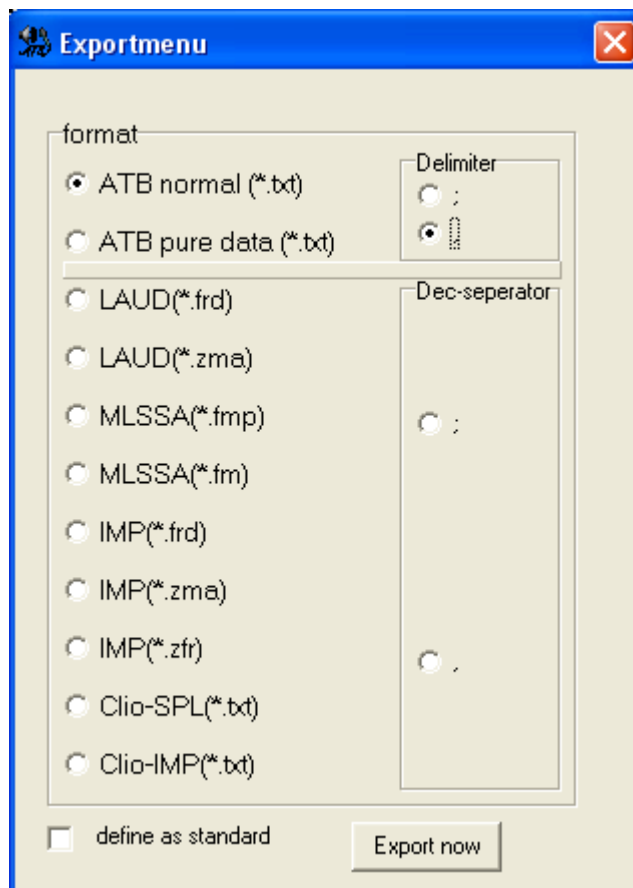
When saving data in the ATB export format it is important to differentiate between saving measurements with multiple curves and singular measurements.

1. Saving measurements.
   Open EXPORT in the menu FILE.
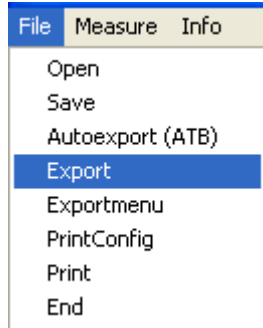
   The pop up window SAVE AS will appear and the measurements saved.

2. Saving plots with the clipboard.
   After measuring the menu PLOTS is opened. Pick out the plots to be saved with the pointer. After that press the clipboard button:

   The plot data will be stowed in ASCII Format in the clipboard.

3. Auto export
   The auto export function saves the latest frequency response curvatures automatically in a predefined file. Thus the results are open for evaluation from other programs.
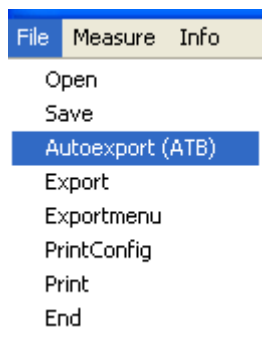
In the context menu the target path can be chosen. The separator/slash can be selected using the EXPORT MENU.

14.   PRINTING

Of course you wish to see your documents in black and white, to do this they have to be printed.
A printer must be connected to your PC to enable the printing of a diagram. The printer must be
installed properly, power on and switched to »On Line«. Your printer can only receive data from
your PC when the printer is switched On Line.

14.1   ADJUSTMENT

By opening printconfig in the menu FILE you can choose the printratio (length/height) and the
print line width
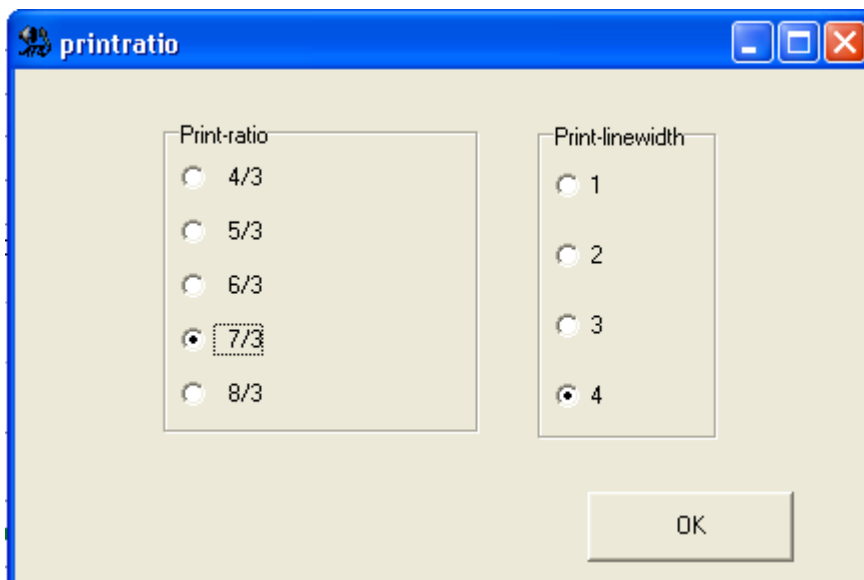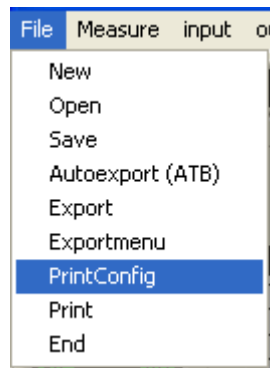
14.2   PRINT START

The button prints the whole document.

If PRINT is activated in the menu FILE a similar print menu will appear
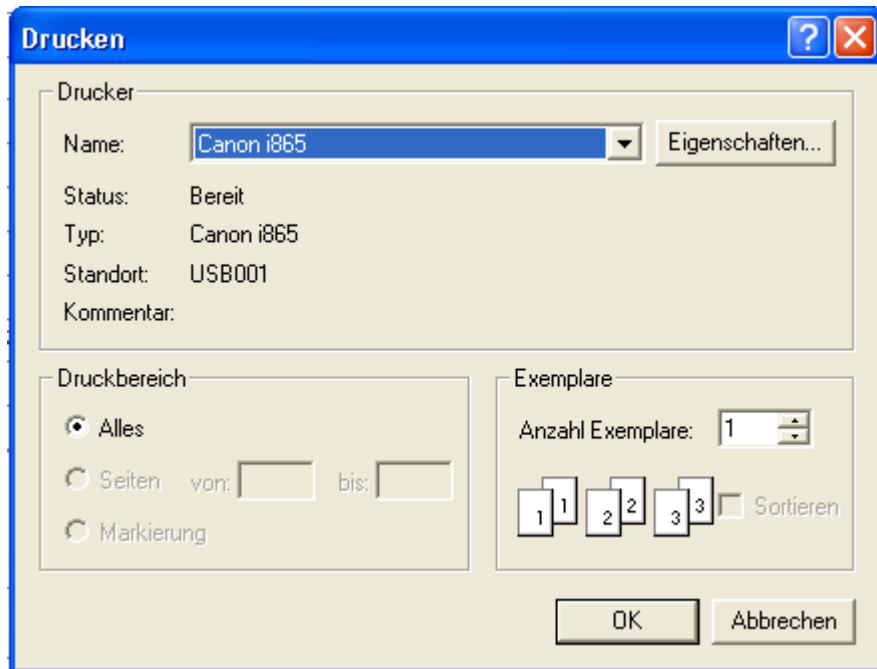
In COPIES you can choose the amount of copies to be printed.

15. WORKING WITH THE WINDOWS CLIPBOARD

The Windows clipboard is available to any application. It is an outstanding tool when exchanging data between different applications.

15.1 COPYING

To copy the diagram produced from all the measurements made into the clipboard press:

15.2 COPYING SINGLE MEASUREMENTS

Single measurements can also be copied into the clipboard. The format for the files saved is the ATB-format. These data are ASCII-Data. The copying procedure for single measurements is described in the chapter EXPORTING DATA.

16.   DIAGRAM

16.1   DIAGRAM INSCRIPTION

The diagram can be marked in the window PLOTS. PLOTS is opened with

Measurement diagrams can be marked in the PLOT window. The diagram elements, HEADLINE, LEFT, RIGHT and OPTION, are those above each diagram. Options are also available for marking the diagram axes and the required units. This window is adjusted to the M-TYPE (measurement type). The HEADLINE is shown to be presenting ATB precision. It can, however, be overwritten by the user for each relevant measurement.
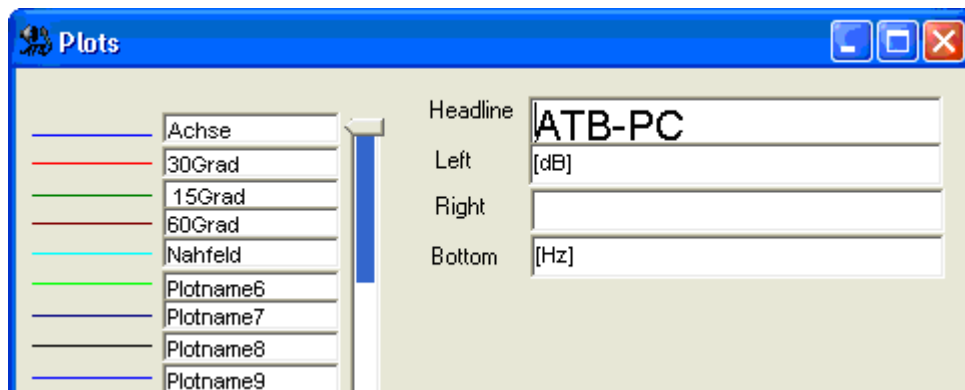
Set the mouse cursor in the field to be written in and activate the menu "script".
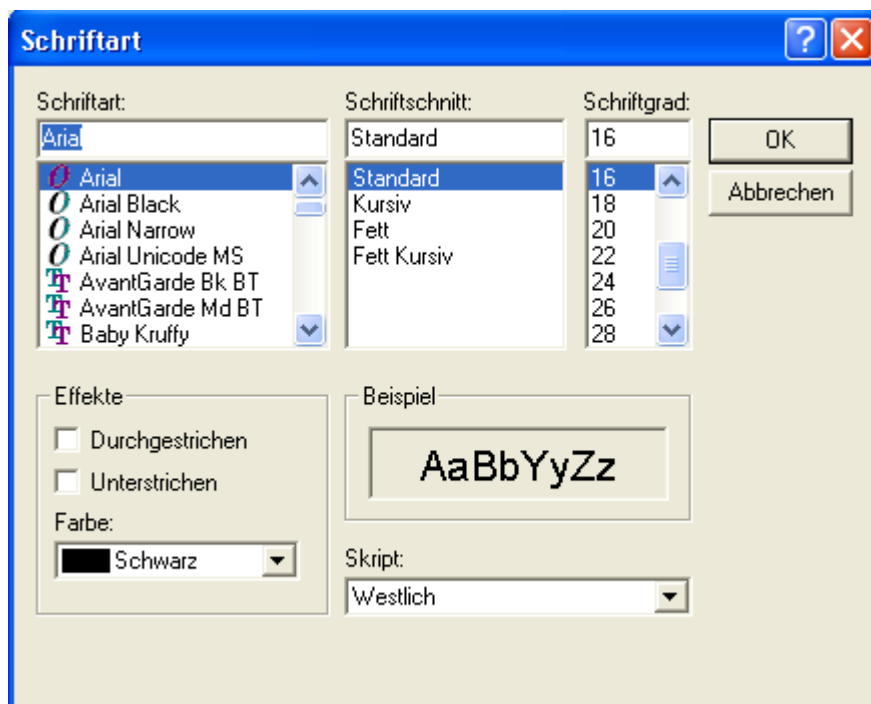
17. MEASUREMENT PROGRAM

17.1 PARAMETERS
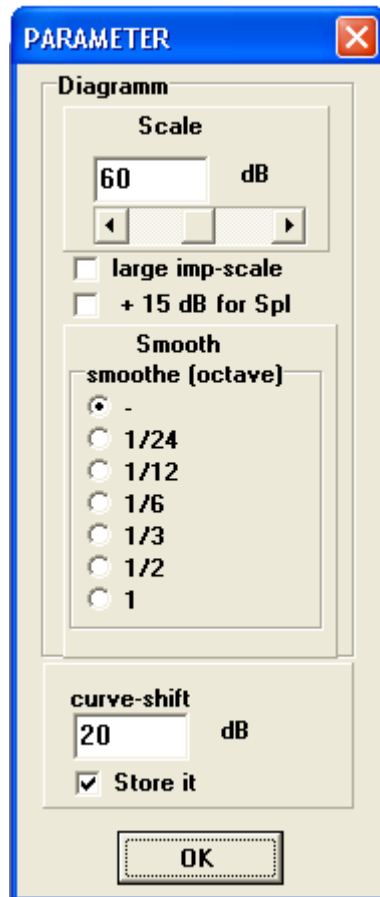
SMOOTHING
  Smoothing radius in octaves, 1/24, 1/12, 1/6, 1/3 = Terz, ½, 1

SCALE (RESOLUTION)
  Field of scale from 5 to 60dB

MIC
 In this window there is to set the lowest dB level for the dB scale.

LARGE IMP-SCALE
 This Function is for the Thiele-Small parameter testing. If the impedance curve is higher than
 50Ω. The function is set to get an impedance scale up to 250Ω

CURVE.SHIFT
 This parameter is for the scale of the Mag measurement. It set the scale to right input level.
 The parameter depends on the input adaptor.

17.2   SIGNAL

The program has the capability to task various assignments.
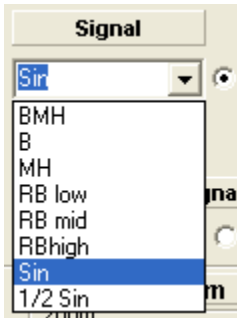
Before the measurement the signal _____ has to be selected.

For the measurement with the CD or DVD select the EXTERNAL SIGNAL
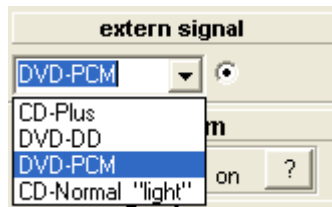
It is possible to use either the CD or the DVD as a signal source for measuring. The CD is used for car interiors (most cars lacking a DVD Player). The surround sound system can only be tested in conjunction with the DVD. The signal source and signal type are to be chosen in the signal menu.

The signals Normal und Plus are to be found on the CD.
The signals Dolby Digital (DD) und PCM are to be found on the DVD.
The signal DVD-PCM (intern) is for the internal generator

17.3   MEASURING WITH CD, DVD

The signal for the Optitest can come either from CD or DVD player. The signals for the program are on the Auto-Test CD or the Surround-Test DVD. It is especially important for car sound systems to use the CD. The advantages are:
1. There are no complicated cable systems from ATB output to the (hard to reach) sound system input necessary. This is especially important when serial testing.
2. The whole system is measured. The qualities of CD player, equaliser, amplifier and loudspeakers are tested at once. This is especially important when testing the surround system. The qualities and adjustments of the Dolby Digital Decoder are included in the test.

17.4   THE MEASUREMENTS

An especially important function of the Optitest measurement is the mean of the measurements. With this function the mean average of the measurement is in real-time, so that results can be read whilst measuring is in progress.

The Measurements mean average M and M+ are activated with the buttons shown above. The number of mean averages, for the test duration, is unlimited.

Another function of the Optitest is the M *D* und M+*D* continuous measurement. This test is a constant measurement where the actual measurement is shown. This allows the comparison of instruments, new adjustments will be shown immediately.

With the buttons

testing is started and stopped. After a test new starts it is possible keep the first plot (to allow comparison) or to allow it to be deleted. This is done by marking the box next to the M+D button.

To compare instruments a target plot is measured or loaded. After this the test is carried out with M+D on the instrument to be compared. For the duration of the test only the last measurement is shown. If the new plot complies with the referred plot then the instrument has been equalized and the test is stopped.

16.5   COMBINE

The Optitest has the special function COMBINE
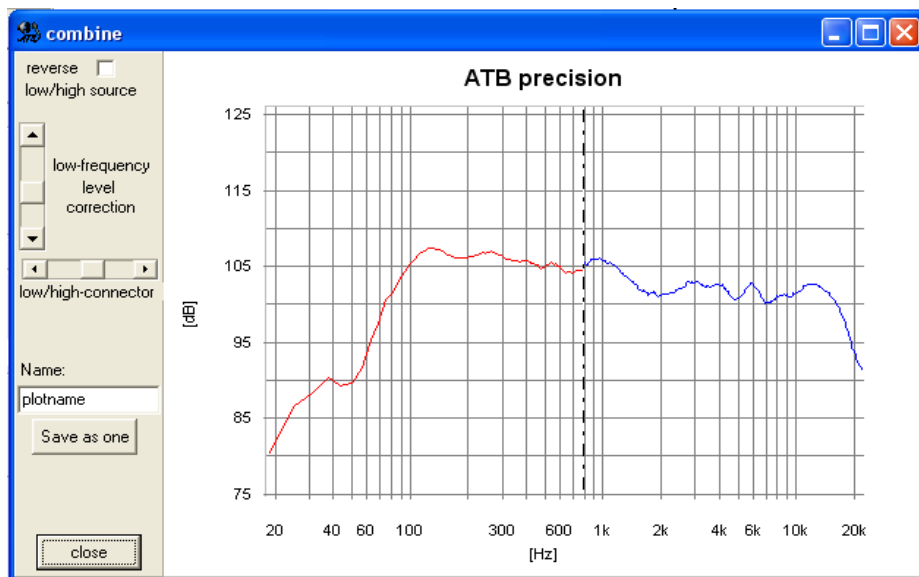
The menu allows the simple combination of near field and distance (SPL) measurement. The

combine menu is opened with the button .
 In this menu the last two measurements will be shown.
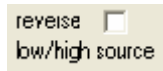
With the field **low/high source** (reverse ☐) can the distribution of near field and distance (SPL) plots are changed.

The next step is to set the cut frequency between distance (SPL) and near field plots with the **low/high-connector** low/high connector.

Low/high-connector can be adjusted to make room resonance vanish. To make both plots into a single line the near field plot is to be set with the **low-frequency level correction** low-frequency level connection.
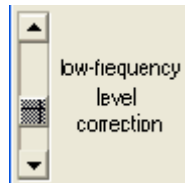
For proper results the following is to be considered: The remote field measurements are to be carried out as an SPL measurement. The microphone distance is 1 metre. The amplifier output-voltage is set to 2,8V for 8Ω loudspeakers and 2V for 4Ω loudspeakers. This measurement provides the reference level for the plot and will not be changed. Through adoption of the near field measurement the plot remains realistic.

17.6   MEASUREMENT DESCRIPTION

Carrying out a measurement is very simple, thanks to the ATB 701 box. For the external signal measurement there is a new correlation type. The test signal can come from either a CD or DVD.

The strong correlation used by ATB proves itself with insensitivity against delays. This means no distance need be adjusted when measuring loudspeakers. Digital switches and equalizers can also be measured without problems.

Whilst measuring, the output voltage is to be chosen so that the instrument or loudspeaker is at listening volume.

The input voltage range is chosen so that the curve is in upper range of the diagram. It should not surpass the top edge though.

One of the four test signals (according to which operation) is used for measuring.

18. MAGNITUDE MEASUREMENT
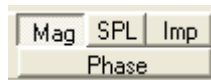
18.1 SET-UP FOR THE MEASUREMENT

For the measurement of amplifier, mixer or equaliser the Mag measurement is used. The input of device is connected the magnitude output of the ATB 805.

For the Mag measurement the ATB 805 selctor is switched to Magnitude. The Audio Out level is set in the programme. The volume is disabled for the Audio Out.

 The picture do not show the right selector position. It is Magnitude.

The Input is the magnitude in. Mag low for audio level and Mag high for power amplifier output.

18.2 THE MAG MEASUREMENT

The measurement is to open with [Mag].
The device under test is connected with the adapters. The input level for the device is set with the OUTPUT controller.

The measurement is stated with the [button image] button. The Input controller is to set that the level indicator does not get read.
The parameter CURVE-SHIFT is set according to the input adapter.
For the above shown adapter CURVE-SHIFT = -22.2 dB for the dBV scale.

19.   AMPLITUDE-FREQUENCY RESPONSE, SPL
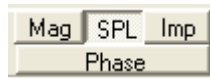
19.1   SET-UP FOR THE MEASUREMENT

SPL frequency response measurements are carried out in conjunction with the MICROPHONE IN socket. The speaker is connected to the speaker out.

The selector is switched to SPL and the volume to 9dB (SPL).

  The picture do not show the right
selector position. It is SPL.

19.2   THE SPL MEASUREMENT

To see the sensitivity for 1W/1m in the plot there is the SPL measurement .
It is important that the Mic-sense be set in the MIC-CORRECTION menu.
After opening the SPL active menu
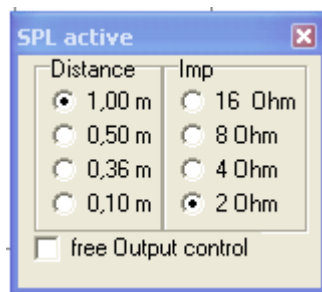
There is to set the impedance of the speaker. With this information the box set the output power of 1W.
The next is to set the distance from the speaker to the microphone. If the distance is not 1m, the program calculates the right value.

The function FREE OUTPUT CONTROLL sets the OUTPUT controller to set the SPL level for the THD or IMD measurement.

19.3 NEARFIELD MEASUREMENT

How to carry out acoustic measurements with loudspeakers and rooms are described in the following chapters.

By the near field measurement, the physical property of sound getting weaker over the distance is made of use. A measurement microphone is placed 10cm away from the woofer. This way the direct sound is many times greater than reflected sound, so that only the direct sound is measured. The measurement is only carried out up to 300Hz and extended by another measurement for the frequency range above 300Hz. The second measurement is made at a distance of 1m to measure both mid and high tone speakers at the same time.
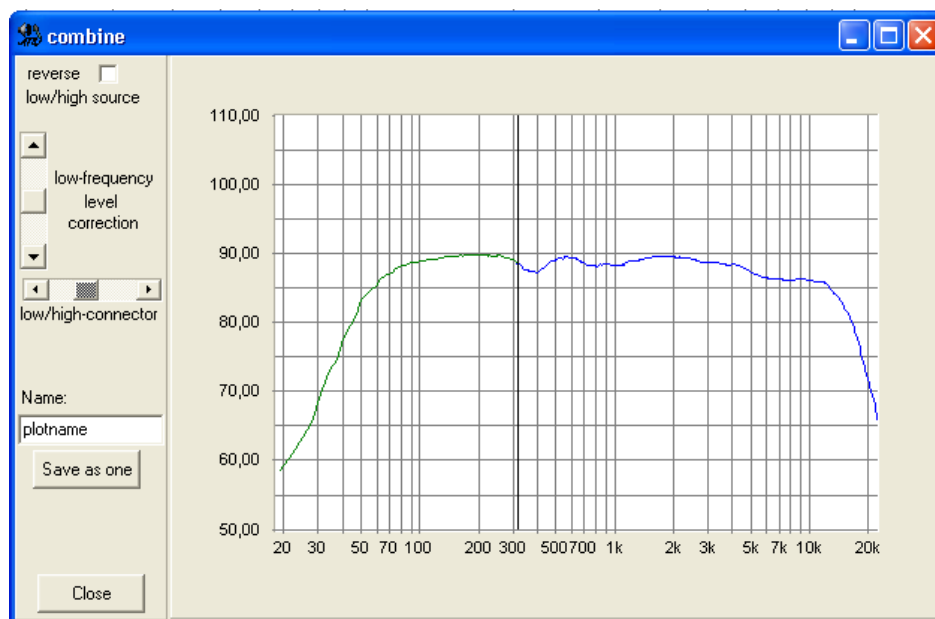
Fig. 19.1 the menu Combine for the addition of low and high tone measurements

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

Pressure ~ Power x Area

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

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

blended out by a time window. Because of the small time delay difference it lies within the time window.
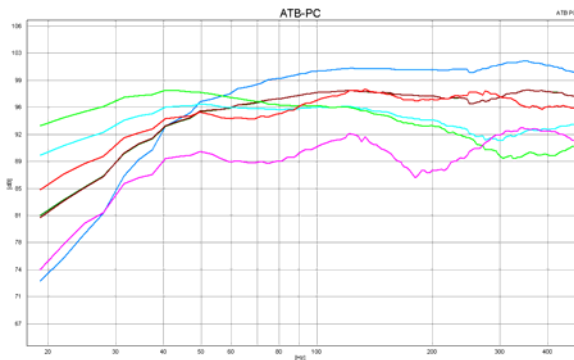
Fig.19.2
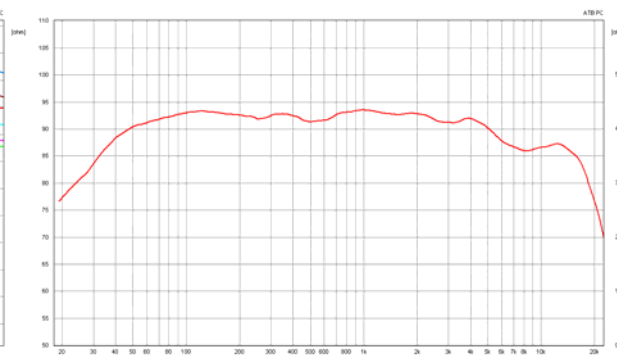
Fig. 19.3

Fig. 19.2 shows the measurement of the bass reflex opening at different distances. The follow up of distance to the rim of the bassreflex opening is green, light blue brown and blue. The measurement was carried out with near field correction. The red curve is the boarder area measurement and violet the measurement at 1m. Both measurements were carried out without correction.

Fig. 19.3 shows the resulting frequency curve of the loudspeaker. It is made up of the brown curve, which equals the boarder area measurement and the violet curve. The curves were connected together in the Combine menu.
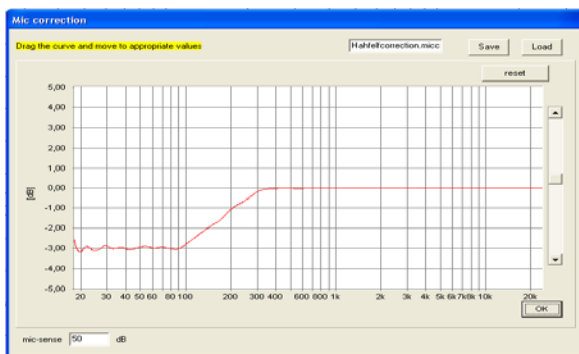
Fig.19.4
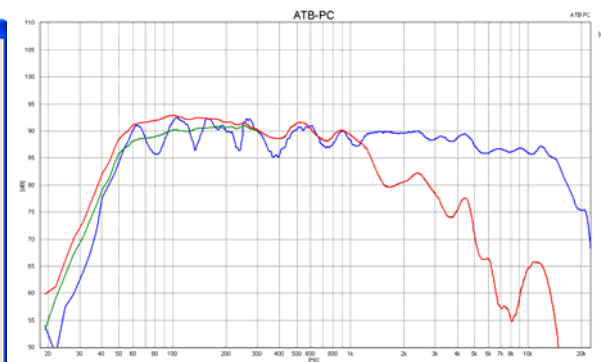
Fig. 19.5

A problem with the near field measurement is the measure of bundling, a gain of sound pressure in the near field area. The gain can be up to 3dB in the low frequency range. The ATB PC measurement system has a Mic-correction function. Here a correction curve for near field measurement is read in. The Fig. 19.4 shows the correction curve that is read into the Mic-correction. Once setup the Mic-correction curve is called up before near field Measurements. The frequency range measurement, Fig.19.5, shows the near field measurement without correction, red and with correction, green. The green curve equals the blue curve of the measurement at 1m distance. With the described measurement process, the low frequency range can also be correctly measured in normal rooms.

19.4   THE BOARDER AREA MEASUREMENT

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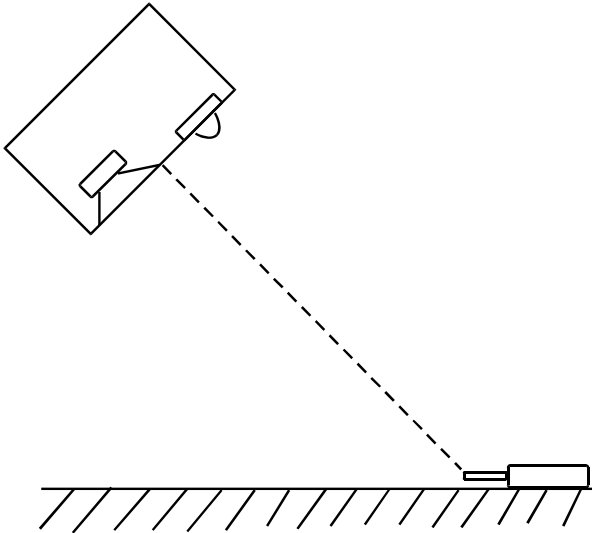
The picture shows the positioning of loudspeaker and microphone.

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

As far as the room resonances go, the rule is the bigger the measurement room the better.

20.   IMPEDANCE MEASUREMENT
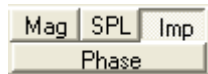
20.1   THEORY

Impedance is the frequency dependent resistance of alternative currents. It consists of the value Z in Ω and the phase $\varphi = °$. The measurement displays both values above the frequency range.
The phase φ displays the values of reactive and active resistance which together result in the value Z. $\varphi = 0°$ consists of resistance only and $\varphi = 90°$ consists of reactive resistance (reactance) only.
The impedance measurement is needed when developing and testing loudspeakers. It is a value for the demand on amplifiers from loudspeakers. Smaller Z values demand powerful amplifiers.
The phase φ reveals possible instability in amplifiers, which can create oscillation and damage loudspeakers. The phase angle should not be $> + 60°$.
The impedance measurement enables the testing of electronic devices, resistors, capacitors and spools.

20.2   ARRANGEMENT

The speaker is to connect to the speaker out socket.

The selector is switched to IMP. The volume is disabled.

The picture do not show the right function positions. It is IMP

20.3   MEASURING

The impedance measurement uses the same graphics as used when amplitude frequency response is measured.
The measurement range reaches from 20Hz -20kHz. The right-hand axis displays the value Z. The range is from $0 – 50 \ \Omega$.
The change over between amplitude and phase measurements is initiated with the button

after changing over the usual buttons are available. The constant measurement using averages is not necessary for impedance measurements and is deactivated.

The measurement is started with:

The result is displayed in Ω on the right-hand axis of the diagram created.

The two buttons at the left are for a single measurement.
The two buttons in the middle are for a duration measurement with shows always the last curve.
The two buttons at right are for a duration mean measurement. This measurement is important for the impedance because errors will be out of the curve.

There are some professional speakers with impedance higher than 50 Ω.
For them there is the switch in the PARAMETER menu to get the 250 Ω scale

☐ large imp-scale

20.4   IMPEDANCE MEASUREMENT FROM 12HZ

For the setting of bass-reflex boxes there is the need of a measurement in the range below 20Hz. This measurement is carry out in the Thiele-Small measurement.
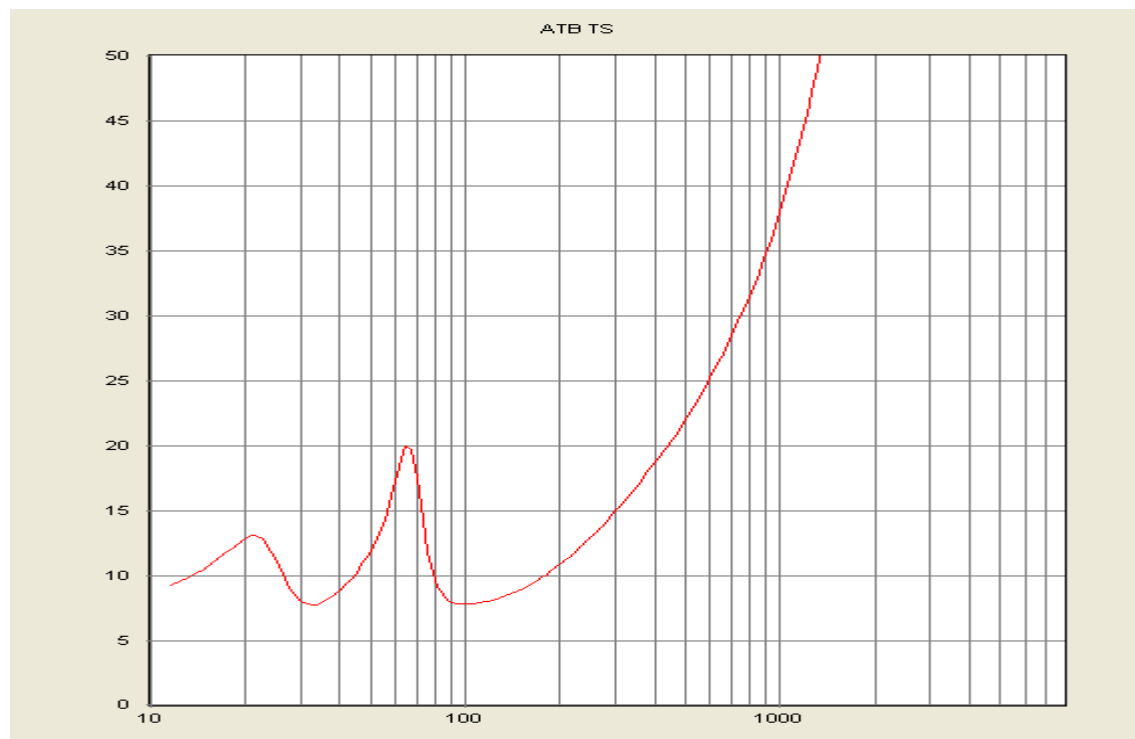
21.   PHASE MEASUREMENT

21.1   BASICS

The phase is used when describing sinus shaped processes such as mechanical oscillation, alternate currents, radio waves and sound waves.
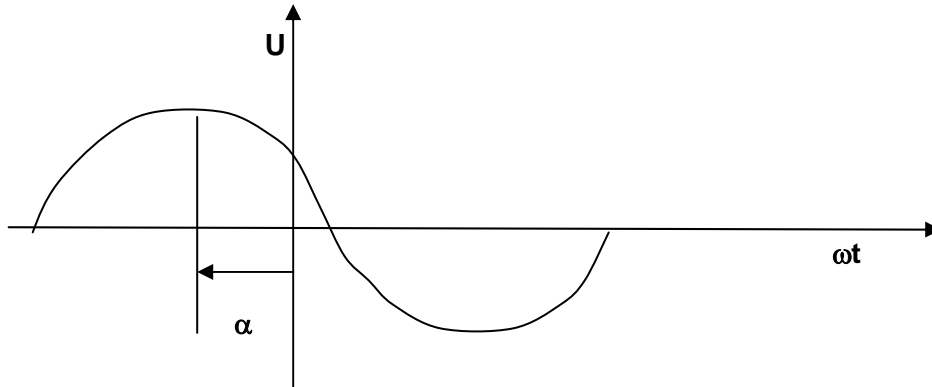
U(t) = Us cos (ωt+α)
t Time,   U(t) Instantaneous value of voltage, Us   peak value (max. Amplitude),
f   Frequency,   T = 1/f   Period duration, ω = 2π f = 2π/T   Angular frequency,
α Null phase angle

The phase angle α already appears in the in the basic equation. It is purposefully positioned with the point of reference TIME 0. With two or more oscillations (waves), as with e.g. acoustic signals, the phase angle is of absolute importance for measurement description.

U(t) = Us1 cos(ωt+α) + Us2 cos(ωt+α+φ)
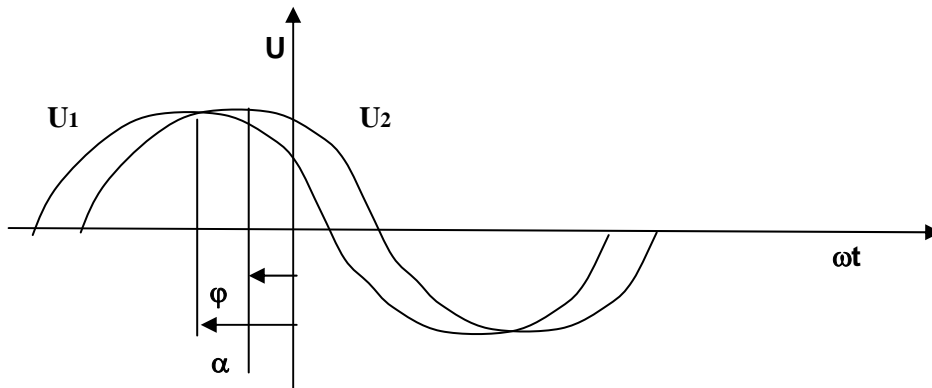
The angle φ defines how the oscillations (waves) overlap.

Examples of how the angle φ effects the waves :

φ = 180° => blanked out

U

U_1 U_2

ωt

φ
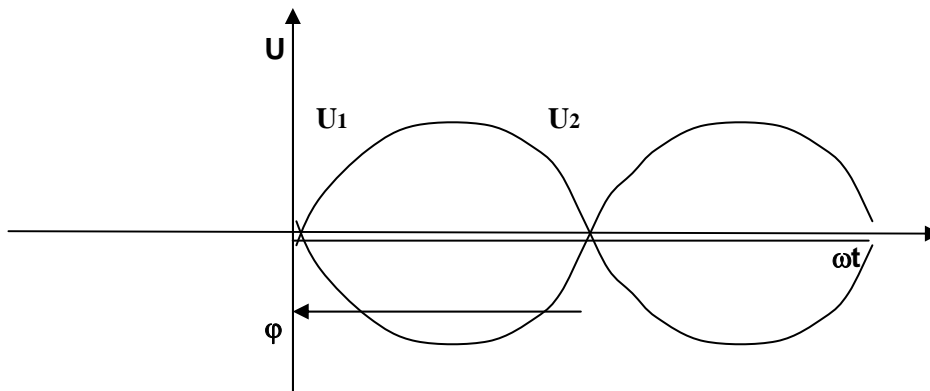
φ = 0°, 360° => maximum amplitude

U

U_2

U_1

ωt
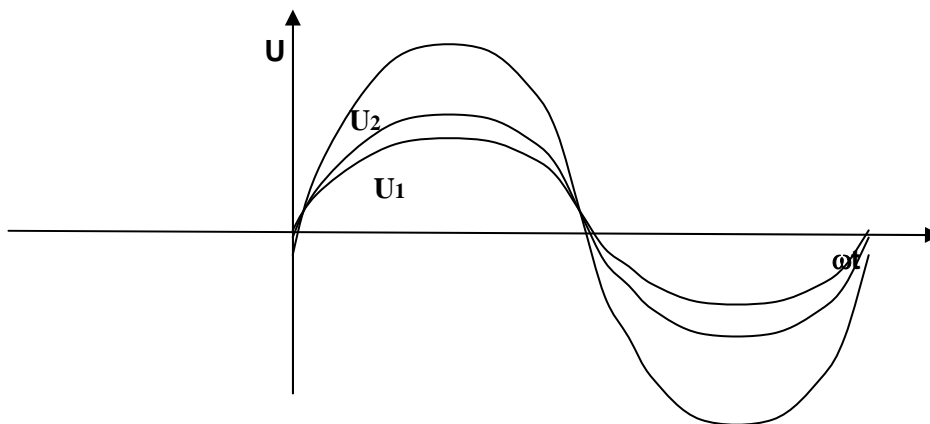
φ = 90°, 270° => partly amplified, or also partly blanked out

U

U_2

U_1

ωt
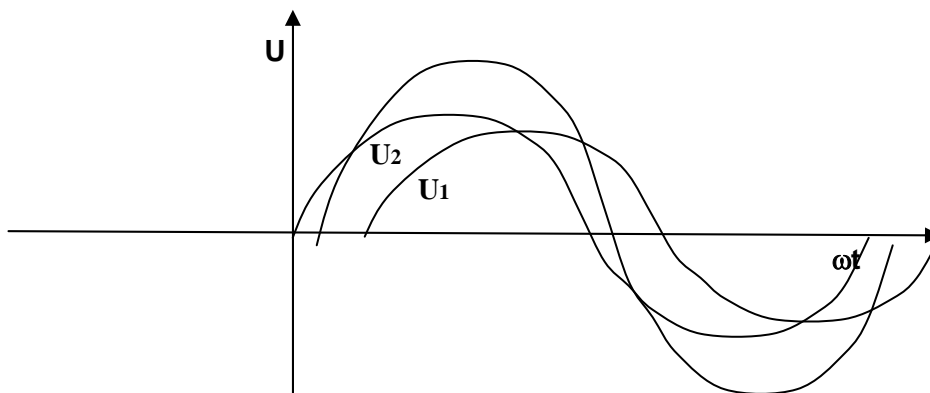
The phase and loudspeakers

Loudspeakers also have to comply with the basics of physics. Amplitude and phase are essential for a correct description of the sound pressure emitted. The common theory that the phase can be determined in the frequency response is <u>not</u> true. This only complies to simple electronic circuits. The complicated transfer function, including duration, inhibits phase recognition in the amplitude frequency response. A box with an absolutely smooth frequency response can produce extreme and therefore audible phase jumps.

A common method for detecting the phase relation in loudspeakers is as follows:

To determine the phase relation from the frequency response in the transition between e.g. the high and middle tones of two single loudspeakers, the crossover network is constructed to produce the maximum blanking out (extinction). When, after switching the poles of one of the loudspeakers, the frequency response is well balanced then the phase position is also correct. The phase values are usually about 180°, 360° or 540° apart and the transient behaviour is bad.

This method is not successful beyond the break frequency.
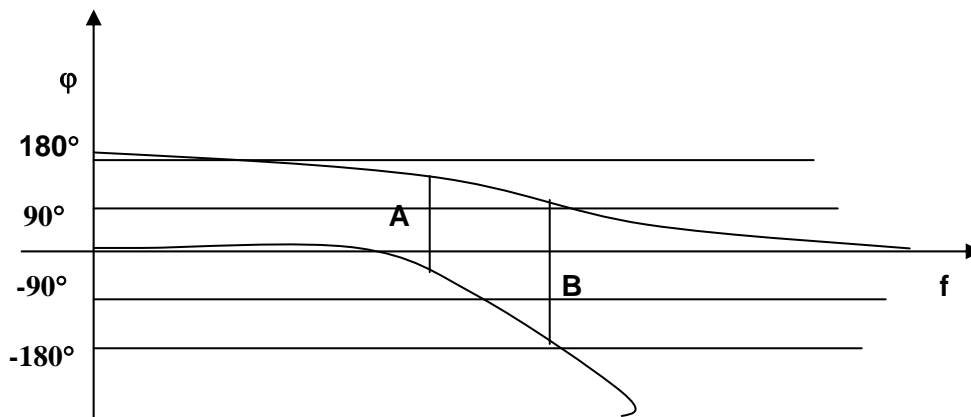
The figure shows the phase relation of the low and high pass

A = the determined phase of 180° using the method described above

B = phase angle of 270° . This angle partially erases the sound elements of the middle and high tones. A switching of poles on the loudspeaker shows the same frequency response, thus the fault cannot be detected with the usual method. With a phase angle of 90°, 270° the loudspeakers are too loud and produce an annoying sound quality. In this case the usual frequency response measurement has no meaning.

Many tests show that a balanced phase frequency response is absolutely necessary for a natural sound reproduction.

21.2   ELECTRICAL PHASE MEASUREMENT

The electrical phase measurement is carry out in the Mag measure.

21.2   IMPEDANCE PHASE MEASUREMENT

With the electrical phase measurement we can measure for example loudspeakers impedance. This measurement is essential for a fault free operation of loudspeakers. Large phase angles show a capacitive and inductive behaviour of the loudspeaker impedance. The result is an overloading of the amplifier. The amplifier can also become instable and cause oscillation. After a system correction has been made available the loudspeaker can be connected to the clamps of the Testbox. Using the button IMPEDANCE we can switch from frequency response measuring function to the impedance measuring function and activate phase measurement function. In phase the option ELECTRIC is selected.

With the phase measurement the quality of soundcard and CD/DVD player are shown. For frequencies > 5kHz the measurements can be faulty. This is shown by the toward negative angle descending curvature.
Whilst measuring it is important that there are no other noise sources in the vicinity. Noises can cause the measurement to be incorrect because the loudspeaker acts like a microphone during the impedance measurement.

21.3   ACOUSTICAL PHASE MEASUREMENT

The acoustical phase measurement of a loudspeaker can only be carried out in quiet rooms. This test is more sensitive to disturbances than the frequency response measurement. Strong reflection in the in which the test is being carried out can also influence results. In small rooms the distance should not be less than 0.5m.

The measurement should be carried out as a continuous measurement:
The automatically phase calculation is highly complex and faults can occur. This can be easily recognized as the curvature will appear to be broken up. Next to the phase button there is a drop down list with a series of adjustment possibilities to support the measurement.
Whilst the continuous measurement is running one of the three ranges are selected.
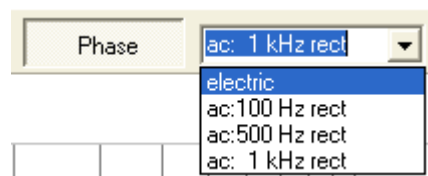
The correct range has been found when the phase curvature remains the same. When the curvature is intact in the high amplitude range then a correct result has been attained. When measuring loudspeakers separately it is only possible to compare the phase behaviour. Phase differences of 180° cannot always be correctly determined. After the simultaneous measurement of loudspeakers the results are once again correct. Because distance measurement is not necessary with ATB PC, every measurement technician gets the same results.

22.   STEP RESPONSE

22.1   THEORY

The step response shows the amplitude and the phase simultaneously. It is calculated using both measurements. The first thing done is to calculate the impulse of the sinus oscillation through a complicated addition, with value and phase. Because the impulse is a mathematical factor, ideally the amplitude is infinite; its image is hardly determinable. For instance the frequencies are not distinguishable. Therefore the integration of the step response is calculated with a mathematical function. The image of the step response entails all the important information about the qualities of the transfer distance.
The ATB PC program contains an analysis function for the step response. Usually all frequencies within the range of 20 Hz – 20 kHz of the step are displayed. This is especially difficult by loudspeakers e.g. when testing the cross over between subwoofer and middle/high tones, it is better to not even starting to show the high tones. Therefore in the ATB PC program the frequency range for the frequencies displayed by the step response can be adjusted.

Step response of a linear transfer distance

Step response of the loudspeaker
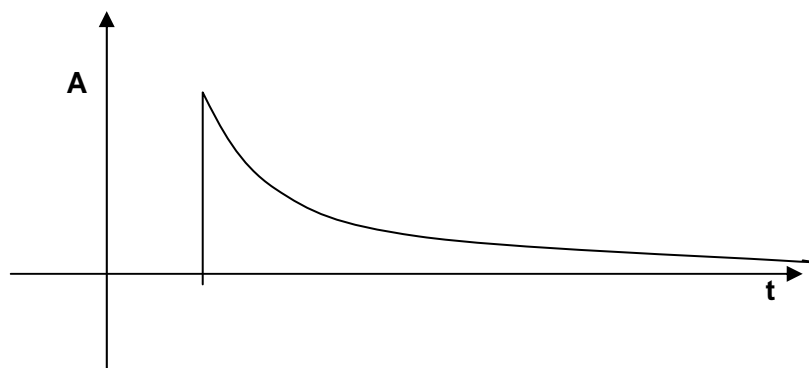
The signal ascends steeply and falls in correspondence with the e – function. The descent is caused by the differentiating function of the air.

The picture shows an ideal loudspeaker. The steep ascent of the step response displays a high top frequency border. The e – function's corresponding descent shows a linear frequency characteristic. The lower frequency border can be deducted from the steepness of descent. The deciding factor in this measurement though is the statement brought about by the phase relation.
Because the step response only shows ascent and descent this means that all loudspeakers are on the same acoustic level and are poled identically. The phase is also linear. Average loudspeakers display many peaks and falls depending on the number of single chassis.

22.2 THE MEASUREMENT

The step response is not a measurement on it's own but an evaluation of frequency characteristic and phase.
The window for the display is activated with the button:
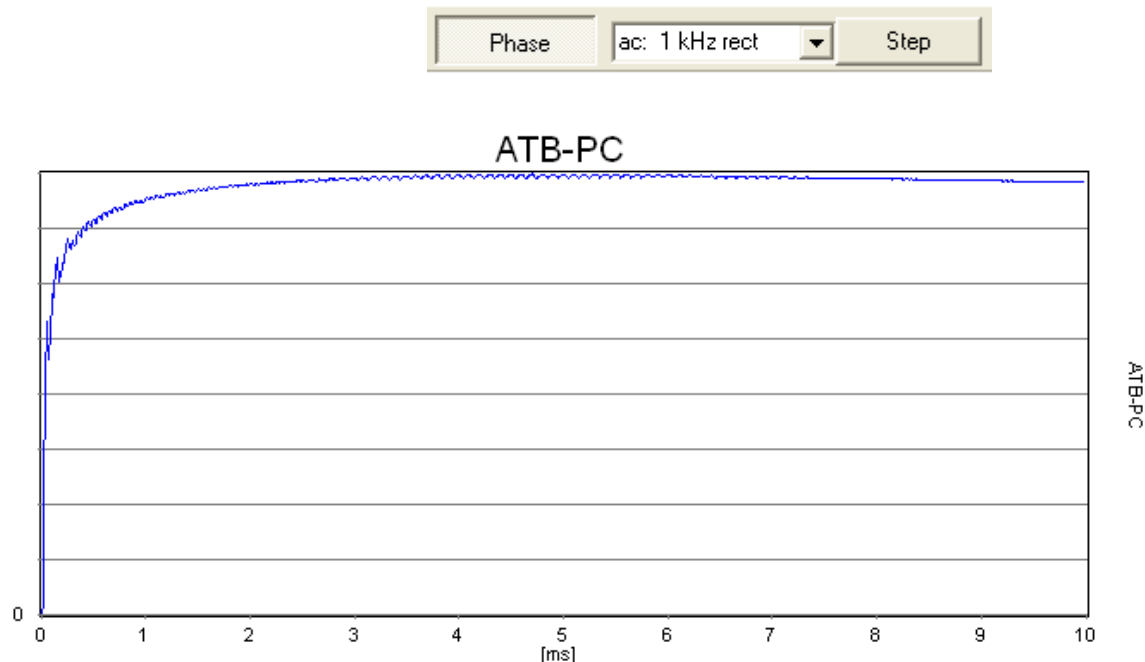
The plot shows the electrical measurement.
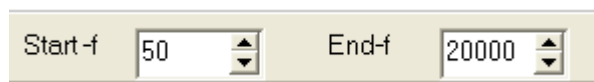
The display for the lower and top frequencies is adjusted under the curve. The scale for the x – axis is in msec. It can set with the START-F controller.

22.3   SETTING OF THE DELAY WITH THE STEP

The following plots show a speaker combination from midrange and tweeter.
Between the speakers is a delay, the sound of the speaker doing not reach the microphone at the same time.

ATB-PC
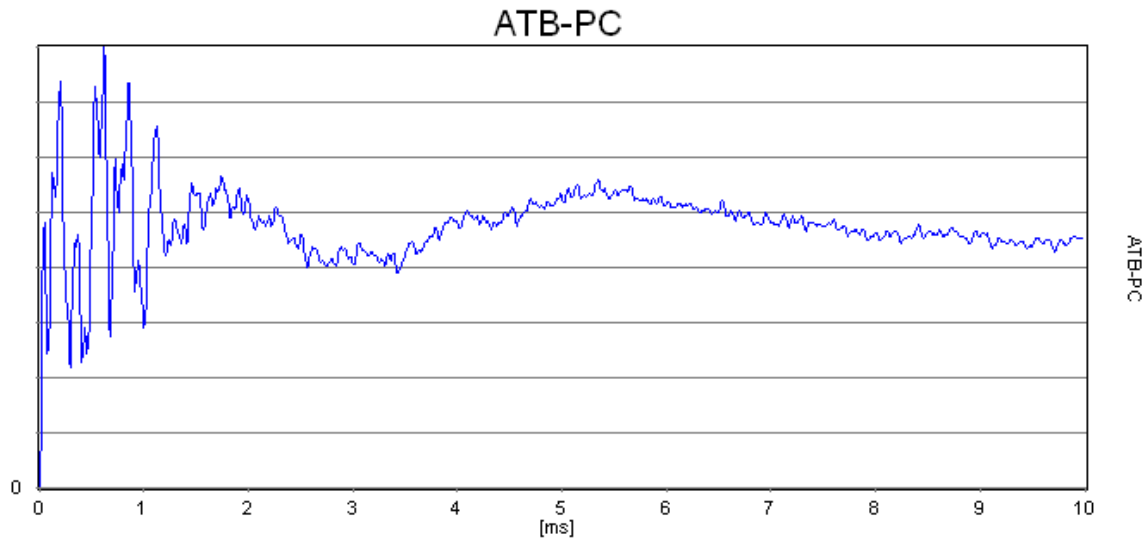
The plot shows the sound pressure of a combination with the tweeter behind the midrange.

ATB-PC
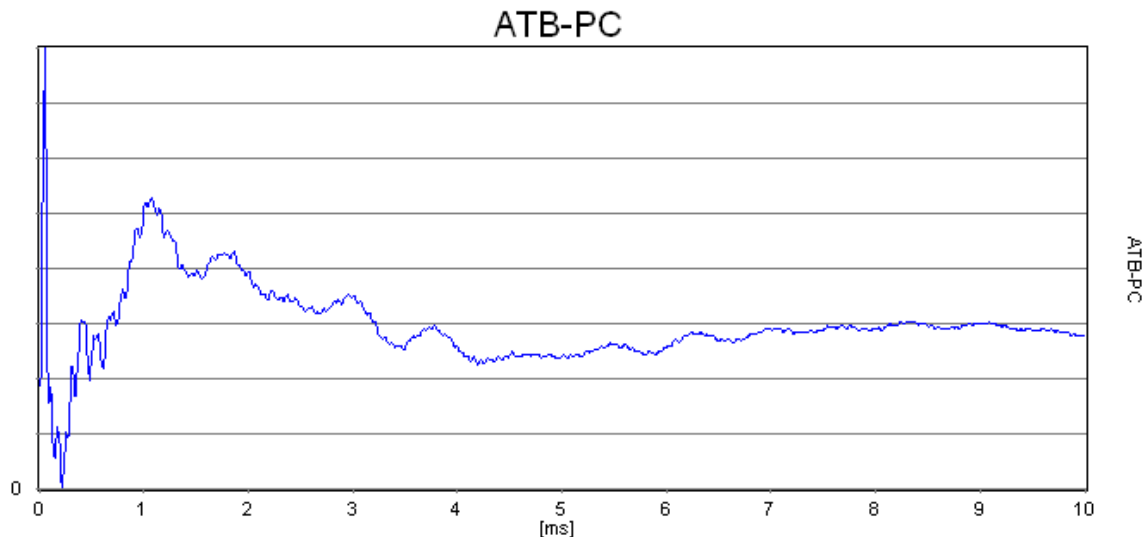
The plot shows the sound pressure of a combination with the tweeter before the midrange. The tweeter needs a delay to be in time.
To set the delay the distance of from the two speakers is measured. Then the difference is calculated. For the difference, S in meter, the delay time, T, is calculated.

$$C = 340m/sec \quad T = S \text{ m sec} / 340m$$

The next step is to vary T to get the plot shown below.

ATB-PC
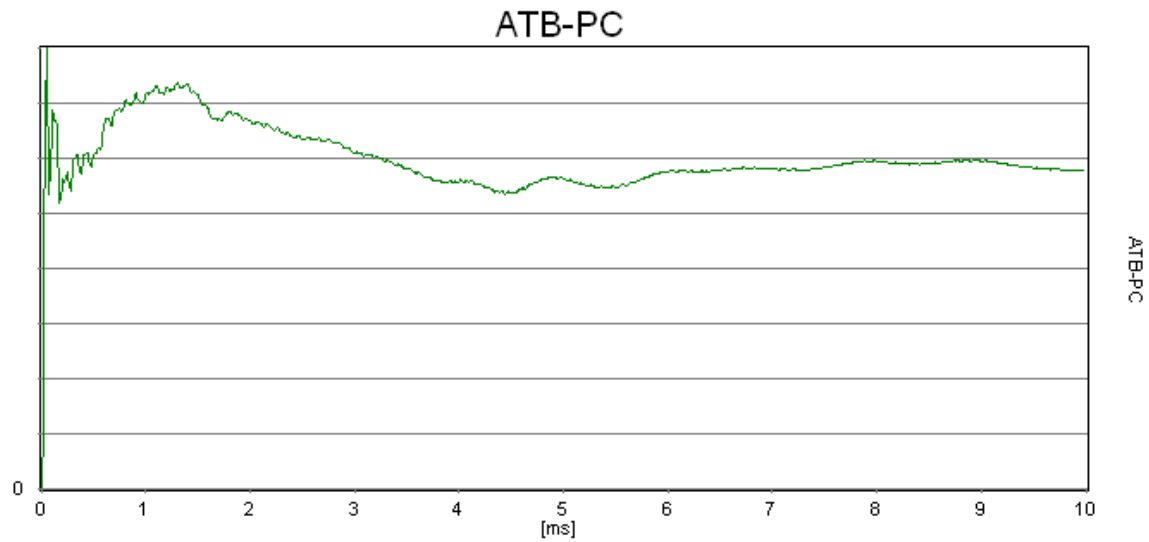
The plot shows at the start of the signal the sound pressure of the tweeter. After there is the signal of the midrange. Between the two signals there is a small drop. This comes from the crossover. The crossover is not right.

ATB-PC
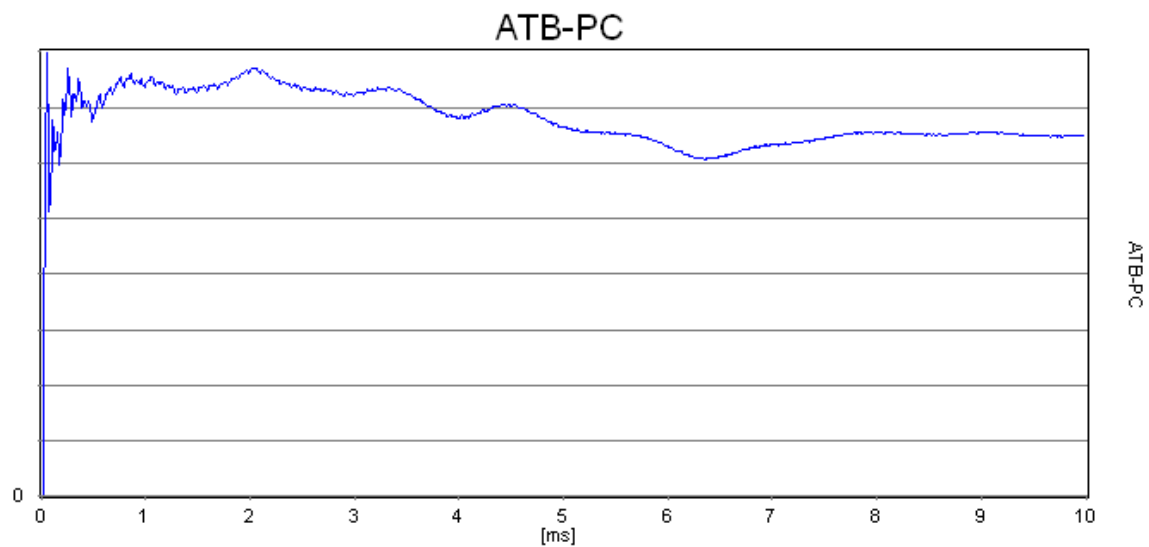
The plot shows the step function of a time right speaker. The signal is very near to the signal of the electrical measurement shown before.

23.   HIELE-SMALL PARAMERTER
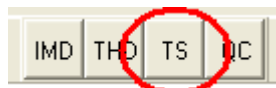
23.1   THE ADJUST FUNCTION

The Thiele-Small measurement used a special frequency range with the start frequency of 12Hz. Therefore the measurement needs an own calibration. After the installing of the program there is to run once the adjustment function.

By the first start of the Thiele-Small measurement there is following message

For the adjustment the ATB 805 device selctor is switched to calibration switch to 1. The volume is disabled.

After the adjustment is to start with a click at the ADJUST button
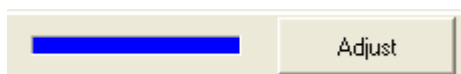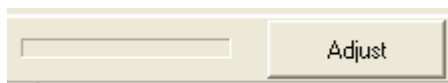
The bar get blue and there is to seen the plot below for a short time. The adjustment file is saved by the program automatically.
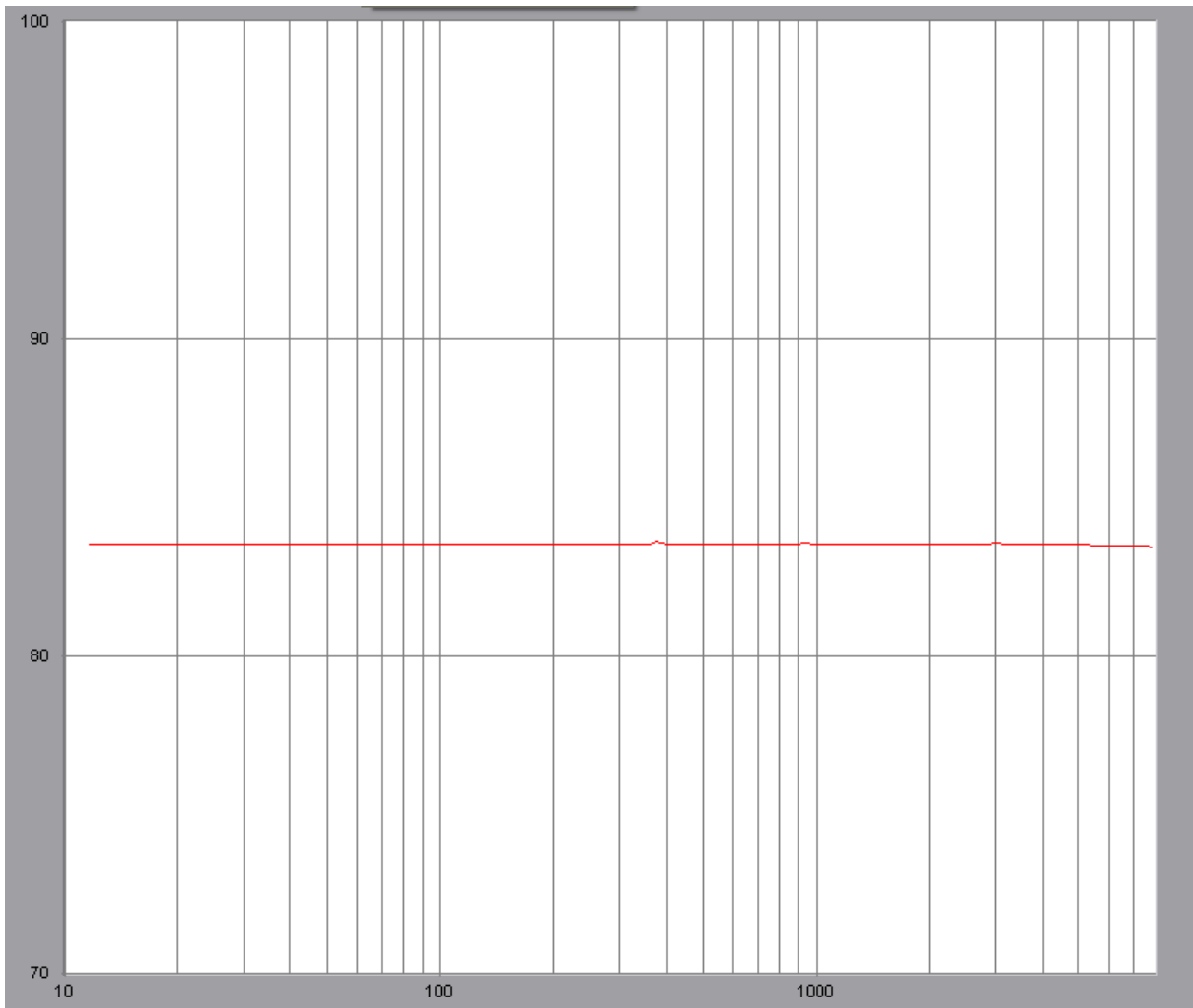
THE MEASUREMENT

The speaker free air measurement is implemented through FREE AIR MEASUREMENT. During this measurement, the driver must be in a vertical position, which would be the normal as-installed position, and the nearest large surfaces must be at least 1 m away.

The entire set of parameters, i.e. the free air values including VAS, can be measured with the weight method.

The weight method allows drivers to be measured that have a permeable bead and/or dust cap. The mass should be selected such that the change in the resonance frequency is a minimum 20%. For conical speakers having membranes of up to 13 mm in diameter, the weight can be approx. 10 grams; big woofers may require 100 grams and more.

23.2   THE METHOD

The Thiele-Small method determines the parameters in compliance with the conventional three-point method.
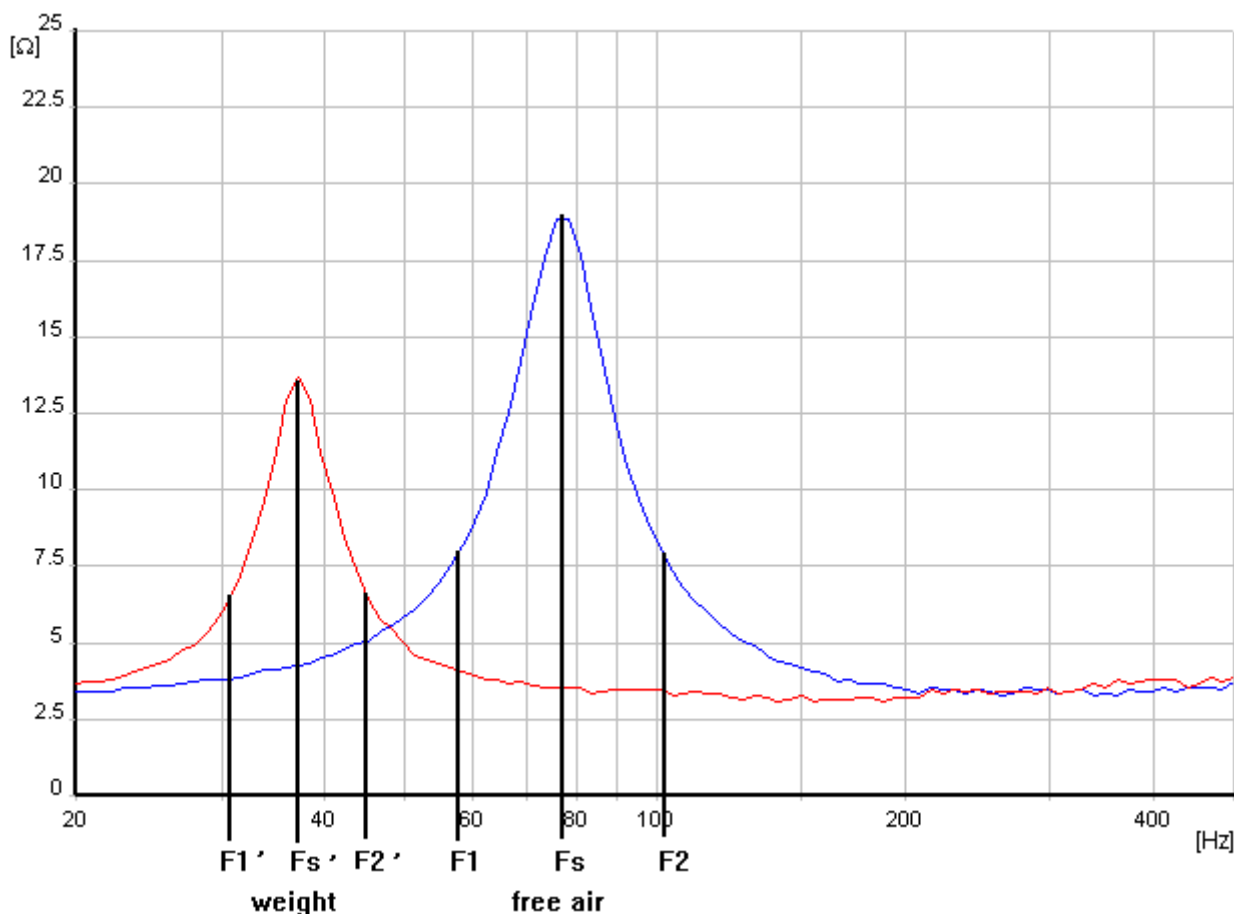
The diagram shows the impedance curve for the free air and the weight options (low resonance frequency). The values the program uses for calculation are also plotted. The program searches these automatically; they are not shown during measurement.

As regards determination of values, the following should be noted:
The theory of Thiele-Small parameter determination starts from a highly simplified speaker diagram, which inevitably implies that not all the speakers behave in compliance with the theory. The impedance curves of these speakers do not follow the theoretical lines. For this reason, the ATB program provides an algorithm adjusting the impedance curve measured to the theoretical curve. The parameters thus established comply with the Thiele-Small theory and allow speaker enclosures to be calculated. The fact that these TS parameters are numerical quantities is evident from a comparison of the resonance frequency measured and the resonance frequency displayed. The peak of the resonance curve measured is found to deviate from that of the Fs calculated with the parameters. The algorithm used for approximation of the measured impedance curve and the theoretical curve complies with the known curve fitting method, is however much more exact than measuring instruments employing the FFT method. FFT measurements only furnish a limited number of values (points) within the low-frequency range.

The impedance curve, which always is an approximation, is thus calculated from only a few values.

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For the Thiele-Small method, the ATB uses a sinus sweep. Reference can be made to up to 500 measured values at a frequency spacing of 0.01Hz. Since the frequency deviation remains below 1%, the ATB is one of the most precise Thiele-Small measuring units.

22.3   THE VOICE COIL INDUCTANCE

For the development of a crossover there is to know the voice coil inductance.
The inductance is measured by the TS program. The inductance is complex impedance. Therefore the normal L meters are wrong.
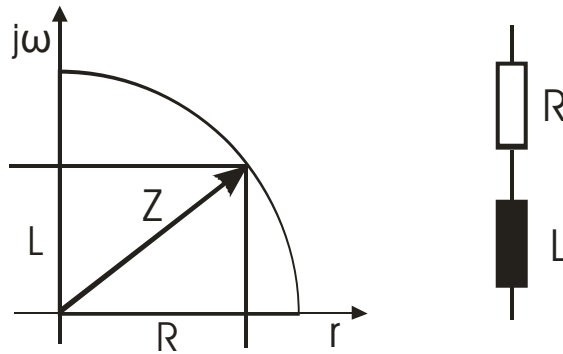
The calculation of the voice coil inductance L.
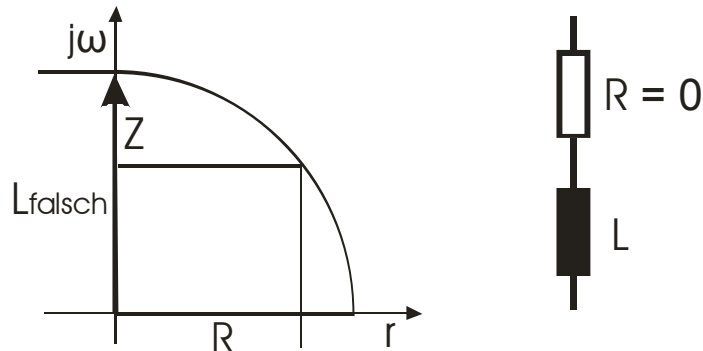
The picture shows the calculation of a normal L meter. For this calculation R is very small, and this is right for a normal inductance. But for speaker it is wrong.

23.4   PROCEDURE

All parameter measurements should be done with a broken-in speaker, because new speakers change their resonance frequency after a run-in period of five to ten hours by up to 15%! Woofers may be broken in by running them for about one hour with a sinusoidal signal of around 25 Hz. The membrane should move visibly, make sure however that the admissible linear travel is not exceeded.

The following settings have to be made:

1. DC RESISTANCE
   The DC resistance determined with the aid of a multimeter

2. ADD MASS
   When applying the weight method, enter the mass into the MASS field.

3. SURFACE
   When applying the weight method, additionally enter the speaker membrane surface as a parameter into the SURFACE field. Should the membrane surface not be specified, it can be calculated with the formula
   r x r x 3.14 = F; r = membrane radius + 0.5 x bead

4. Check Range

   The range for the calculation.

The ATB 805 selector is switched to IMP. The volume is disabled.

The picture does not show the right selector position. It is IMP.

Measuring always commences with the free air option .

Measurement having been completed, the parameters Fs, Qe, QM and Qt are shown below the diagram.

RDC	6,80	Ohm	Qe	1,11	
SD	30,00	cm²	Qt	0,78	
Qm	2,70		fs	123,94	Hz

Before proceeding with the measurement, the values should be checked for plausibility. Any value of 1.00, for instance, implies that no evaluation was possible.

The shape of the impedance curve has to be such that the amplitude values for F1, Fs and F2 appear in the diagram (cf. Fig. Thiele-Small Theory).

Should this not be the case, change the parameters INPUT, LOWER LIMIT and UPPER LIMIT.

To determine the VAS parameter, start a second measurement.

Now, the second impedance curve will be plotted.

VAS	1,13	l
No	0,19	%
Mms	1,85	g
Rms	0,53	kg/s
Cms	0,89	mm/N
B*L	2,97	Tm
SPL	84,70	db 1W/1m
SPL	85,41	dB 2.83V/m
L	0,24	mH

For a measurement to be correct, weight or enclosure have to be selected to comply with the conditions below. This is revealed by the resonance of the impedance curves (cf. Fig. Thiele Small Theory).

Fs = resonance of the free-air option
Fs' = resonance of the weight or box option.

In the weight measuring mode, the mass is too low if
        Fs' > Fs x 0.8

To leave the menu, to come back to the main menu, there is the
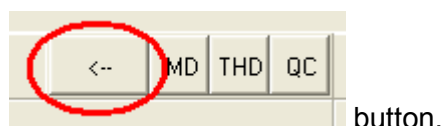
 button.

24. DISTORTION MEASUREMENT
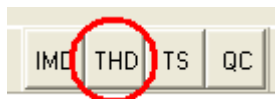
24.1 THE SETTING IN THE MAIN MENU

The setting for the Distortion measurement is carry out in the main menu. In the SPL measure is to make a test measurement. The microphone is to place in a distance of 1 m to the speaker.
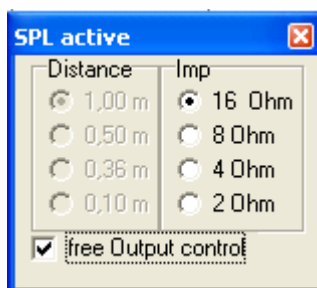
In the SPL active menu is to select   free Output control.

The measurement is to start with the duration measure.
With the OUTPUT controller is to set the SPL level fort he measurement. The INPUT level controller is to set that the level meter bar does not get read. This overload of the input will produce distortion.

For the following distortion measurement a distance of the microphone is to reduce to 0.5m or 0.36m. The setting of the OUTPUT controller, the power for speaker, is not to change.

After the THD measurement is started.
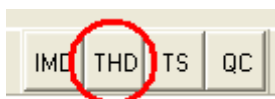
24.2 THE THD MEASUREMENT
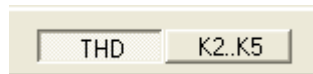
The Total Harmonic Distortion measurement used the sine-sweep as test signal. The frequency range is 20Hz – 7kHz. The harmonics are measured up to K8.
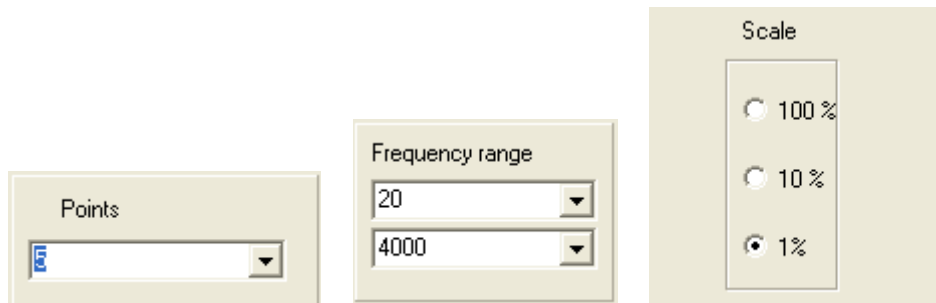
It is to set under point the number of the test frequencies and the Frequency range.
Scale is fort the resolution of the plot.

The level meter may not get read

For the interrupt of the measurement is the ESC key.

To leave the menu, to come back to the main menu, there is the
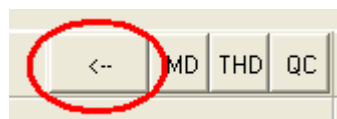
button.

24.3   THEK2, K3, K4 UND K5 DISPLAY

For the analysis of the THD there could be shown the harmonics K2, K3, K4 and K5.

To get the plot click at .
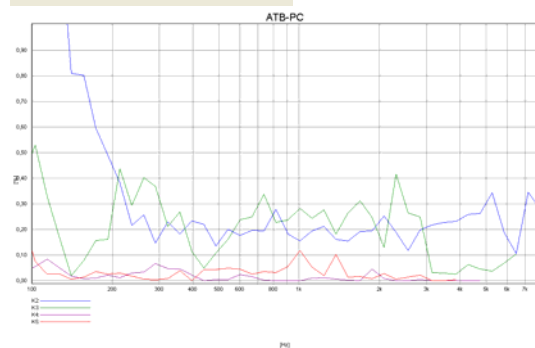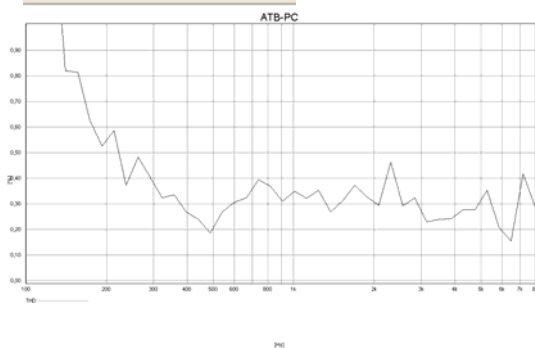
25. DISTORTION SPECTRUM

25.1 SETTINGS

The setting for the Distortion measurement is carry out in the main menu. In the SPL measure is to make a test measurement. It is described in the THD measurement.
For the max. precision the OUTPUT controller is in the +15dB range.

The measurement is called up with the   button
To leave the menu, to come back to the main menu, there is the
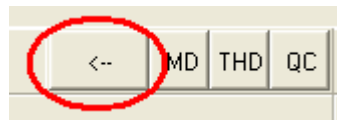
button.

25.2 THE IMD MEASUREMENT

For the Intermodulation Distortion measurement is used the RB signal. The measurement is to test unlinear characteristic curves.
For woofer there is the RBlow, for midrange the RBmid and for tweeters the RBhigh signal.
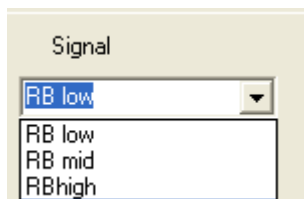
             The display shows the Q-Index.

The measurement is a frequency analyser and shows the intermodulation and harmonics.
(Plot from the Mag measure)
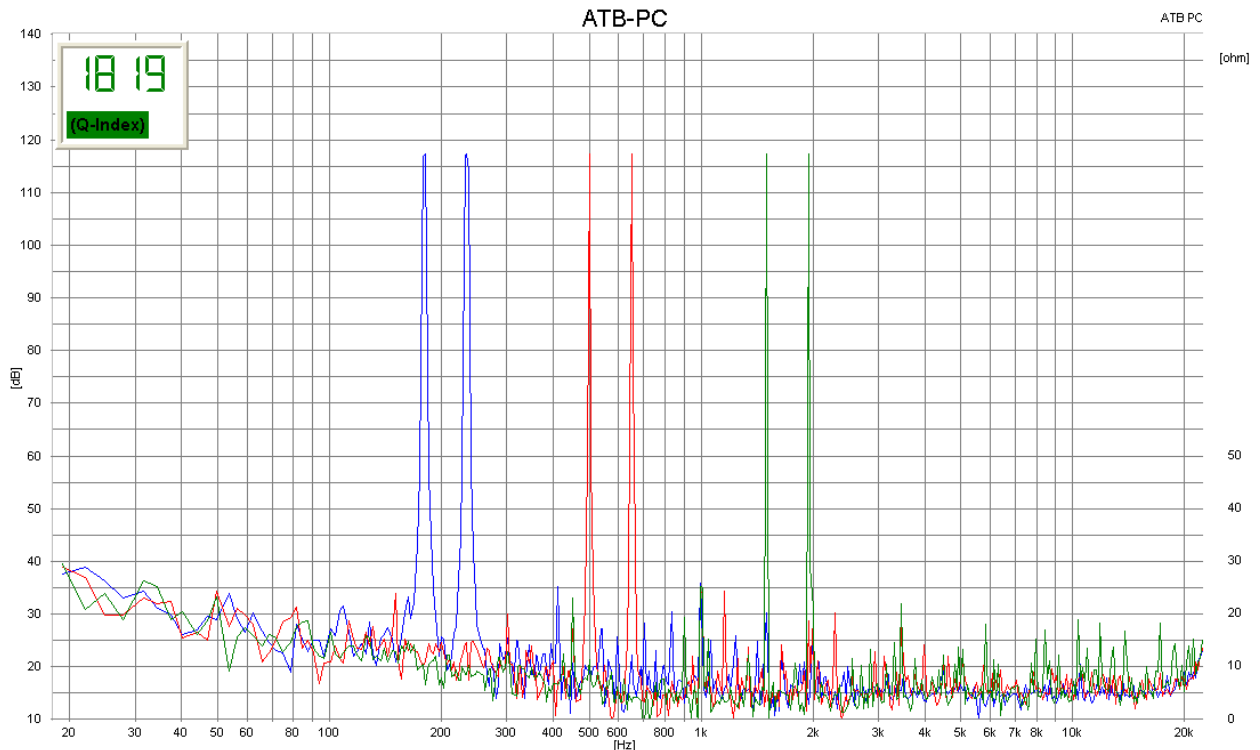
To get the distortion there is to calculate the amplitude for following frequencies.

Signal	Fundamental f1 f2	K2	K3	K4	IMD2 f2-f1 f2+f1	IMD3 f2+2f1	IMD4 f2+3f1
RB low	180	360	540	720	54		
	234	468	702	936	414	594	794
RB mid	500	1,0k	1,5k	2,0k	150		
	650	1,3k	1,95k	2,6k	1.15k	1,65k	2,15k
RB high	1,5k	3,0k	4.5k	6,0k	450		
	1,95	3,9k	5,85k	7,9k	3,45	4,95	7,95k

The first frequencies f1, f2 are the fundamentals, K2, K3 and K4 are the harmonics. To get the distortion in dB there is the formula
Amplitude in dB of the fundamental – Amplitude in dB of the harmonic = distortion in –dB.
To get the distortion in dB there is the formula

$$10^{\frac{dbWert}{20}} = \text{Distortion in \%}$$
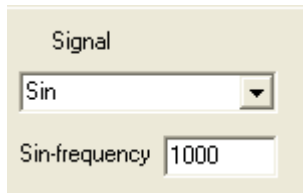
-20dB =   10%
-40dB =   1%
-60dB = 0.1%

25.3   THE DISTORTION MEASUREMENT

The distortion measurement uses the sin signal.
The harmonics are measured up to K8.

Signal

Sin ▼

Sin-frequency 1000

After selected sin the Sin-frequency is set.

4920

THD [%]

In the display the THD is showing in %.

In the plot the harmonics are shown.
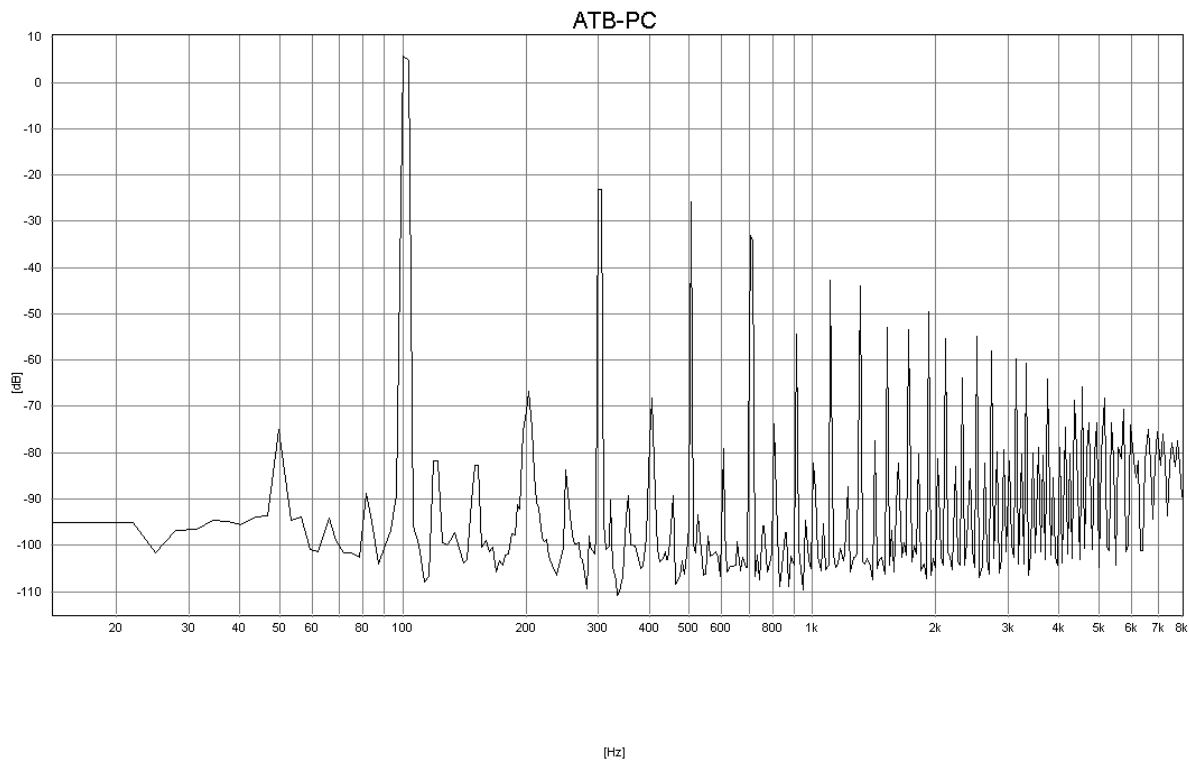
26. MEASURING CAR INTERIORS WITH THE CD

Before testing, the system-correction is to be measured once. The system-correction is for comparison of the USB box. Thus any CD/DVD Player can be used.

For comparison, the left audio output of the CD/DVD player is connected (through the measuring adapter) to the microphone input on the soundcard. The RESOLUTION is set to CD-Plus. The CD is played left with Signal Plus. First of all the modulation is set with the controller so that the bar takes up the first half of the scale. After this the system-correction menu called up and the measurement can begin. If the plot lies within the diagram the menu can be closed with OK.

 After that a test measurement is started with M. When the plot appears as a straight line (only dropping when above 20kHz) then the system-correction has been successful.

After opening the system-correction menu the plot can be given a name. The name should entail the name of the soundcard and CDnormal. The correction curve is saved (with SAVE) and used for the following measurements. Only when the soundcard or resolution is changed there is a need for a new system-correction.

26.1 ROOM CORRECTION
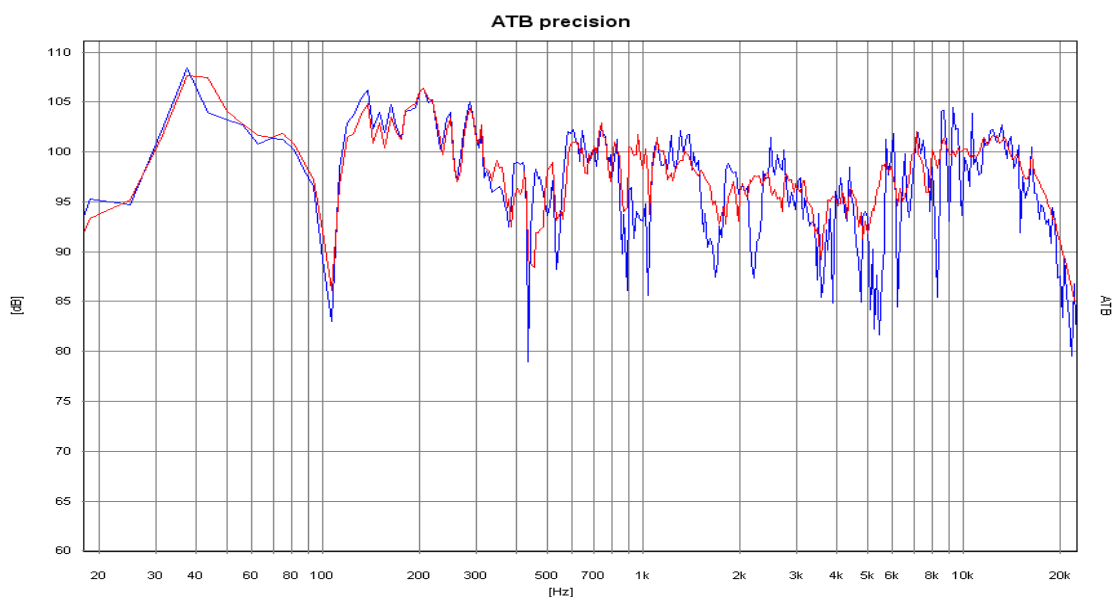
In car frequency character: blue = single measurement in Standard position,
red = room correction

Acoustically considered, because of the lack of room within automobiles there is no proper or steady sound distribution. The interference of the sound waves that result shows a different frequency character for each microphone position when measuring. A test with the standard position shows interferences but no certain frequency characteristics. Only when a measurement with multiple microphone positions in head height of the passengers is carried out can a satisfying frequency characteristic for sound be attained. The ATB Optitest operates by showing then mean average of the frequency characteristics of multiple microphone positions.

26.2 NEAR FIELD MEASUREMENTS
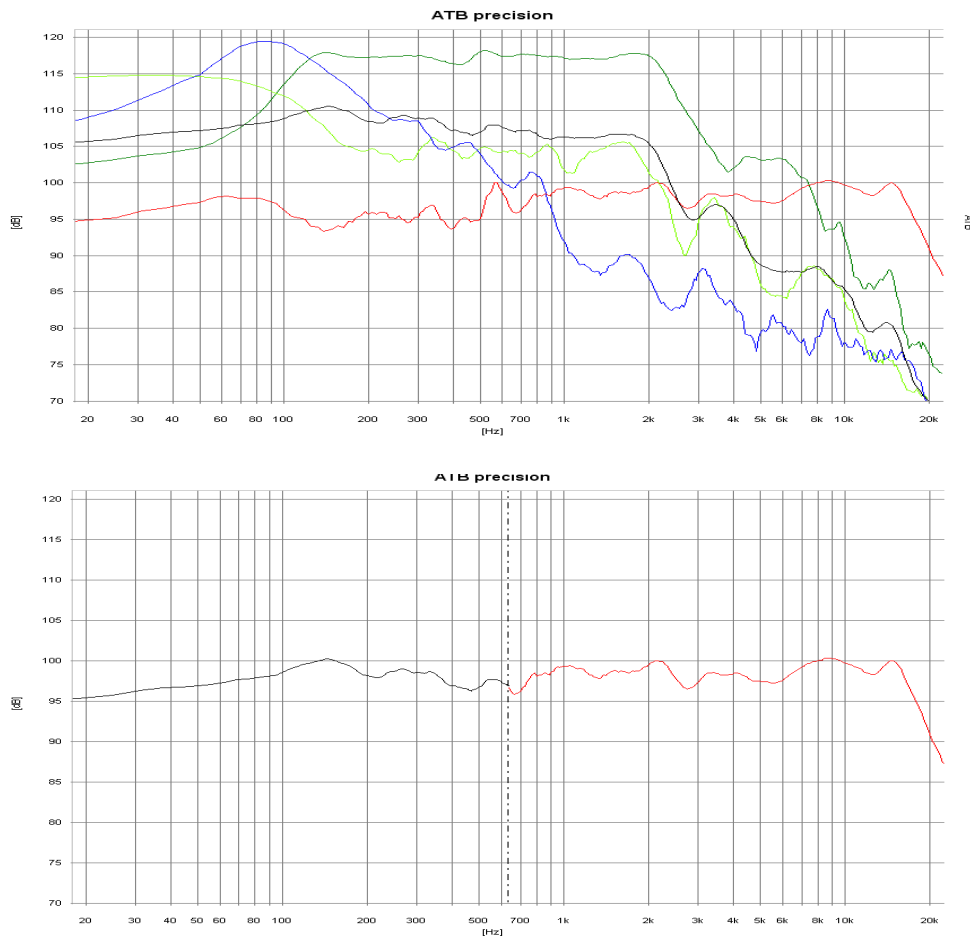
Near field measure: blue = Bass, green = Middle, red = room corrected Middle- High tone, light green = Bass reflex, black = Bass mean average, Middle and Bass reflex

The room correction for the low frequency range cannot be carried out as described. To suppress room resonance the low frequencies are measured in near field. Because of the near field effect during simple measurement the results are falsified, through a super elevation in the sound pressure curve and a loss of sound in the bass reflex opening. The ATB Optitest measurement avoids this effect by moving the microphone between woofer and Reflex opening whilst measuring. Start and end points of the movement are the edges of Bass loudspeaker and vented opening. This is how the influence of the near field effect is avoided. The mean average of the frequency characteristics measured whilst moving show the real low frequency given. In the "Combine" menu the bass and middle-high tone frequencies are put together. The transitional frequencies and amplitudes can be chosen at will.

27.   SOUND SYSTEM CALIBRATION USING THE DVD

For a real cinema feeling in your living room, as well as the proper picture, sufficient sound is necessary. For good sound first of all you need good loudspeakers. Sound differences between high quality DVD players and amplifiers are present but are not as important for attaining the cinema sound. More important is the proper adjustment of the amplifier and the balance between loudspeakers and room acoustics. This is only possible with measuring technology. Thanks to computers and the ATB-PC program it is possible even for the interested nonprofessional to measure and adjust his/her home system.

Before testing, the system-correction is to be measured first.
For comparison, the left audio output of the CD/DVD player is connected (through the measuring adapter) to the microphone input on the soundcard. The RESOLUTION is put to DVD DD. After starting the DVD player, with the surround single signal left, the modulation next to the controller is adjusted. The controller is to be adjusted so that the bar takes up the first half of the scale. After this the system-correction menu is opened and the measurement can begin. If the plot lies within the diagram the menu can be closed with OK. After that a test measurement is started with M. When the plot appears as a straight line (only dropping when above 20kHz) then the system-correction has been successful.
After opening the system-correction menu the plot can be given a name. The name should entail the name of the soundcard and DVD DD. The correction curve is saved (with SAVE) and used for the following measurements. Only when the soundcard or resolution is changed there is a need for a new system-correction.

27.1   AMPLIFIER

 To operate the amplifier (Receiver) it is connected to the television.

In the amplifier menu (Receivers) the setting SPEAKER SETTING is to be actuated. All SPEAKER are to be set to LARGE and the SUBWOOFER to ON.

Further adjustments are, in CHANNEL BALANCE the volume for all loudspeakers is to be set at 0dB.

In the CHANNEL DELAYS menu the different distances have to be adjusted. These adjustments are important because the sound waves spread through the air with the speed of sound. Because the sound from a surround system comes out of the speakers at the same time but is to reach the listener at the same time, CHANNEL DELAY is adjusted to compensate. With home systems the surround loudspeakers are nearer to the listener. This is why the tone is delayed, to make sure that all tones are heard at the same time as the front loudspeakers. To adjust the difference, the distance of the front loudspeakers – surround loudspeakers is ascertained and noted in the menu. In some menus the entries are to make in msec. The msec are calculated with the formula 1m = 2.94msec. The values do not have to be 100 % perfect. Important is that the tone is first heard from the front loudspeakers. Some manufacturers recommend for small rooms 10ms, middle 15ms and 20ms for large rooms. The centre is delayed with about 2-3ms if possible.

27.2   MICROPHONE

 The microphone is connected to the MIC INPUT.

Microphone positioning:
First of all the seating places for the audience are decided. The microphone is positioned on the back of the sofa or armchair, so that the head of the microphone is in the same position as the head of the spectator. If two different seating places are favoured then the microphone is placed between the two places.

27.3   DVD Player

Connecting the DVD player
The DVD player is, with its coaxial cinch or optical digital audio output, to be connected to the amplifier (receiver). The video signal is also connected to the amplifier.

Starting the Surround-Test DVD
The DVD is started and the single measurements are chosen in the menu.

27.4   SINGLE MEASUREMENTS

Front loudspeakers FL, FR
The measurements are carried out as single measurements and are started with the front loudspeaker FL. The position of the loudspeaker is to be tested with this measurement. The test is started with the measuring program M *D*. The volume for the M-PN is to enlarge until the plot is in the upper half of the diagram.
The following measurements were done in Analog.on Studio. The loudspeakers used are reliable, favourable in price and quality.
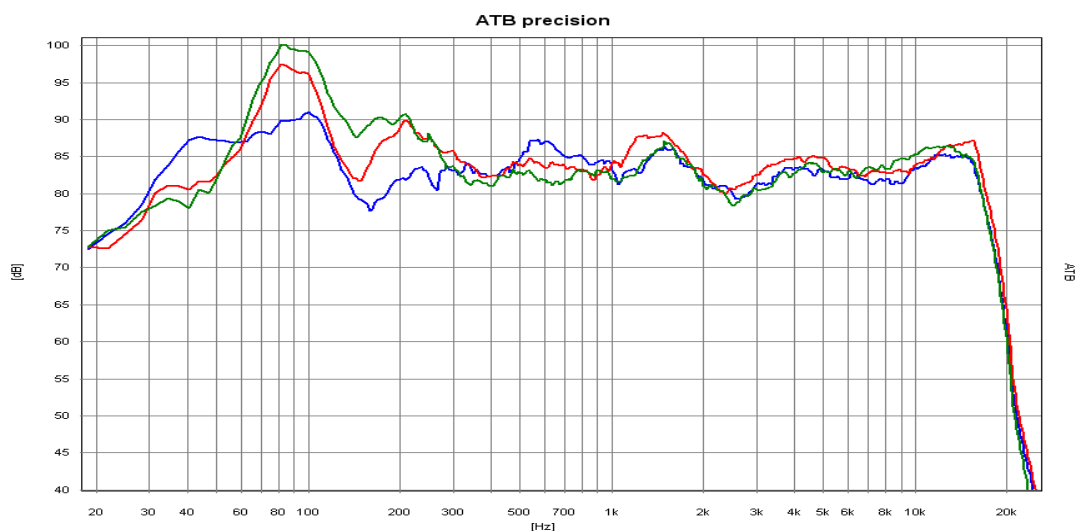
The blue curve is the FL loudspeaker, the red curve is the FR loudspeaker with large and the green curve with small distance from the corner of the room.
The position of the loudspeaker is now sought for where the frequency characteristics show no great elevations or extreme falls. A fall in the high tone range is compensated by

positioning the loudspeaker obliquely in the direction of the seating position. Elevation or drops in the Middle tone range can be compensated through a greater distance to the bordering walls. A super elevation in the low frequency range shows a corner position. Enlarging the distance to the back lowers the elevation. It is not yet possible to avoid an elevation in the low frequency range within a living room (using the room size as a function). Because the ear does not react critically to bass, but even finds it pleasant, a compromise between a minimal bass accent and room comfort is certainly possible. After measuring FL, FR is to be placed symmetrically to FL. The symmetrical positioning is more important than small irregularities in the frequency character. The measurements are started with M+D. Differences in low frequency playback between the left and right loudspeakers are not important because the audience hears the sum of both loudspeakers. Large drops or elevations in the frequency characteristics of the FR loudspeaker make it necessary to reposition the loudspeaker. FL is positioned symmetrically to the new position of FR and then measured. If FL displays large breaks in linearity then both loudspeakers should be positioned between both optimal positions.
Small loudspeakers should be positioned so that the sound pressure curve runs steadily above 100 Hz.

Centre
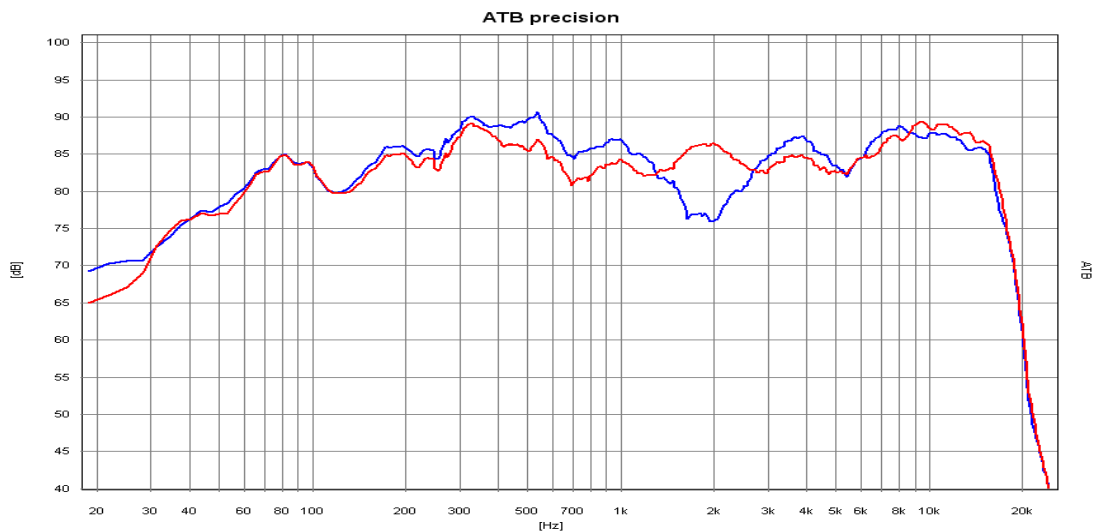The centre is for voice playback. Therefore the centre has to have a steady frequency characteristic. The high tone playback should not be emphasized, this making the voices sound too hard. The low frequency range may drop below 100 Hz, being transmitted from either the Subwoofer or the FL, FR loudspeakers.

The plots show the frequency characteristics of the centre with a small distance from the room floor (blue curve). The red curve shows an increased distance of 20 cm.
When positioning the centre it is important to consider the distance to the floor. With the M+D measurement the Centre is measured with different distances to the floor. The distance with the steadiest frequency characteristics should be chosen.

Surround Loudspeaker

Special rules are needed for the positioning of surround sound loudspeakers. The Home Cinema owner should be aware of the fact that films are not made for the living room but for the cinema. Just as the picture format is not suitable for a television screen so has the sound been mixed for the cinema. The surround sound in a cinema is transmitted with up to 24 small loudspeakers. The loudspeakers are positioned on the walls and the back of the hall. When positioning it is important that each spectator has a loudspeaker in the near so that a direct sound portion reaches him/her. The rest of the loudspeakers create a diffused sound. These conditions should be reproduced in the living room. Because the room does not allow more than two surround loudspeakers, the loudspeakers should create a direct and diffused sound. This is why the usual direct sound radiating loudspeakers are not suitable. Even the earlier used THX Dipoles with diffused sound radiation are not really ideal. The frequency characteristics of the surround loudspeakers should lie between 100Hz and 10k'Hz. The border frequency of 100Hz allows that the low frequencies (that radiate to the front) can be found at the back. The high frequency range should not be emphasized. In cinemas the high frequency range above 10kHz is limited to stop the singling out of each loud speaker. For measuring the microphone is turned in the direction of the loudspeaker. M+D is used to carry out the measurement.

ATB precision
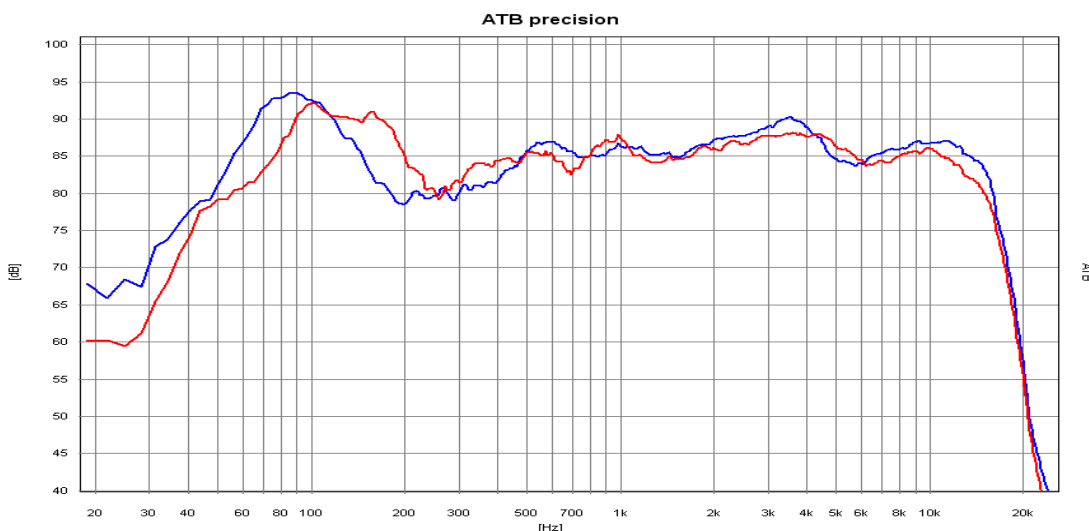
The picture shows both S loudspeakers. The blue one is in a corner.
The loudspeaker position is changed until the rough inconsistencies are gone. To measure the volume the microphone is put into its old position.

Subwoofer

The Subwoofer takes over the low frequency range. The steady transmission of the lower frequencies in living rooms proves to be a problem because the size of the rooms is similar to the wavelengths of the sounds. Therefore the sound field in the room is not steady and the sound pressure owns Maxima and Minima. This means that the lower frequencies are either louder or quieter in different positions in the room. The Subwoofer must be placed so that no minimum occurs at the seating position.
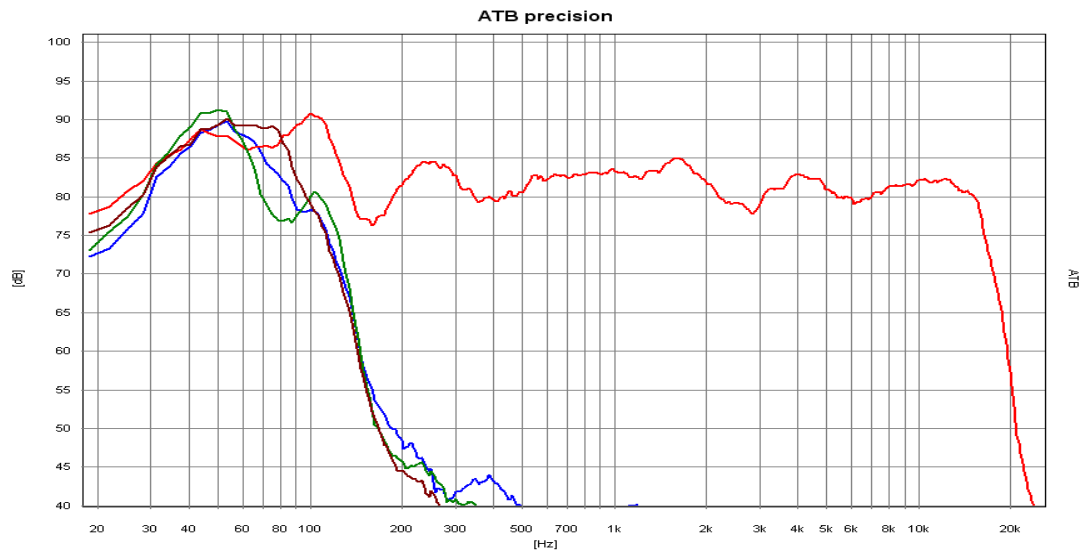
The picture shows plots for the SW in different positions. The difference to each position is 20cm. The red curve is the FL loudspeaker.

Simultaneously it is an advantage to place the subwoofer between the FL and FR loudspeakers. Through multiple tests with M+ and repositioning of the Subwoofer it is possible to find the ideal position quite quickly. This is the position where the measured plot is at its highest.

Volume adjustment

The loudspeakers of the surround system must have the same volume. The loudspeakers are measured with single signals and the adjustments are made in the amplifiers menu under CHANNEL BALANCE. For the loudspeaker FL 0dB is chosen and the sound pressure is measured with M.
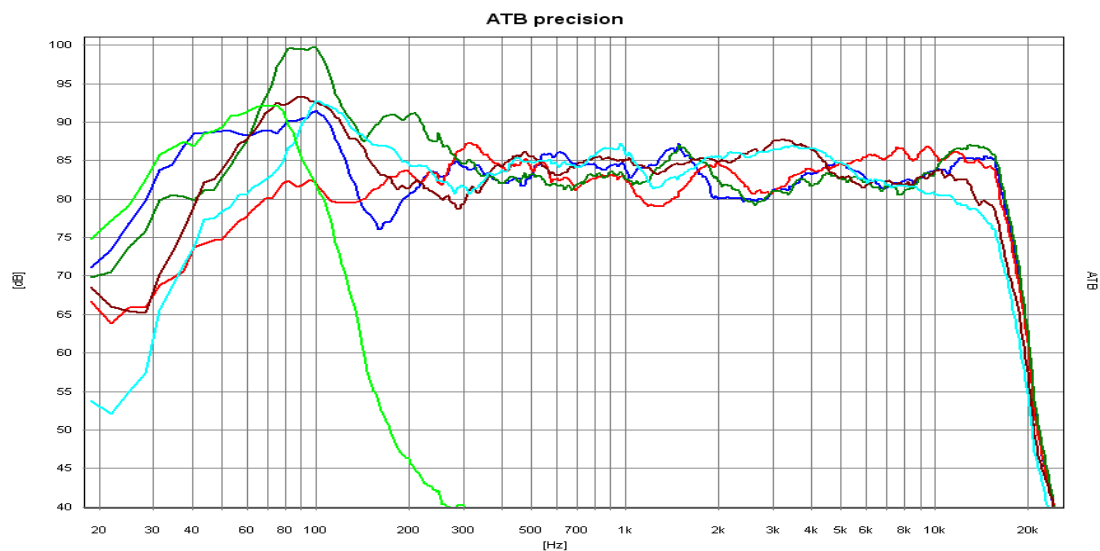
The single measurements of the loudspeakers shown in one diagram
light green = SW, red = C, green = FR, blue = FL, brown = SR, light blue = SL

The volume complies with the height of the sound pressure curve. After this FR is measured with M+ and the adjustments in the amplifier menu are changed so that the plots lie over each other. The same is done with the centre and the surround sound loudspeakers. The room influence should be considered when adjusting the volume of the Subwoofer. Although an elevation is measured seating position it is still possible to attain a steady sound when the mean average of the Bass in the room is steady. The correct measurement is carried out by comparing the Bass curve with the curves of the large (Bass corrected) FL, FR loudspeakers. The SW curve should have the amplitude of the large loudspeakers. The smaller FL, FR loudspeakers are to be given a ca. 6dB louder Bass, because it has to take over the low frequency range of the FL and FR loudspeakers.

27.5   CUMULATIVE SIGNALS

The measurement of the cumulative signals from FL and FR shows the collaboration of the loudspeakers in the room. Seen measurement technically there are special problems to be conquered. Because the sound waves of the loudspeakers overlay each other (having the same signal) interferences are created. Drops in the frequency characteristics appear, depending on the microphone position. These interferences are audible when moving the head (the sound of the noise changes). To be able to measure the correct sound pressure curve it is necessary to move the microphone (held in the hand) within a perimeter of 50cm around the seating position. This curve is a statement for the sound. The measurement is carried out with M *mittel.*
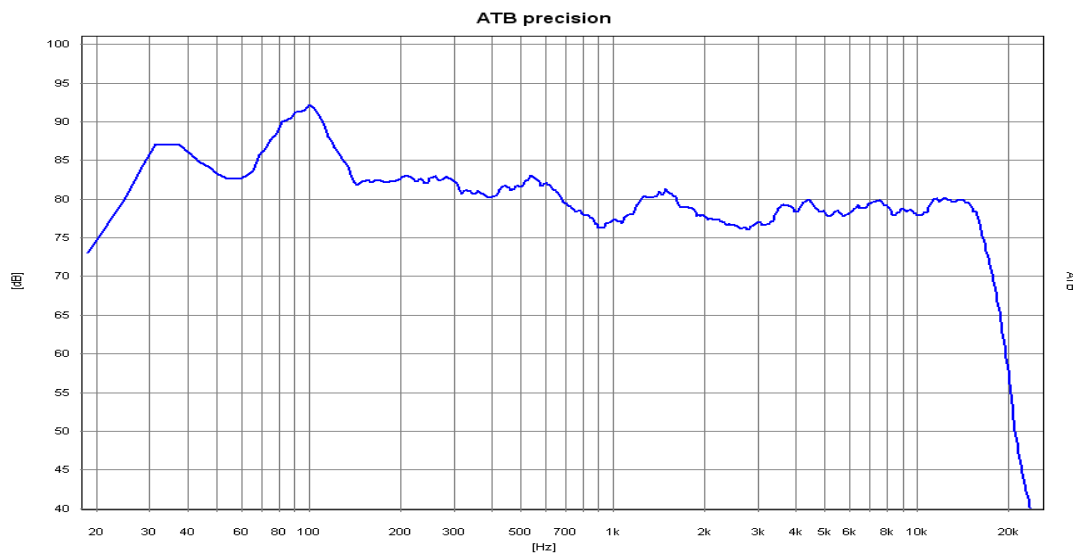
Cumulative signals for the front loudspeakers

The movement of the microphone causes a drop when reaching the higher frequencies. This drop should be steady. The measurement FL+C+FR is important here.

The Equalizer
Some instruments have an equalizer to adjust the loudspeakers to the room. Cumulative signals prove best for adjusting equalizers because the acoustic properties of the room are shown. To adjust the equalizer properly a few rules should be followed. The problem with equalizers is that when linearising the frequency characteristics, the Impulse- and phase behaviour changes simultaneously. The time allocations of the signals are changed. When the equalizer is used for the FL and FR loudspeakers the adjustments should be identical for both loudspeakers. To adjust the low frequency ranges the cumulative+bass signals are used.

27.6   CUMULATIVE SIGNALS AND BASS

The Subwoofer
The bass of a surround system is transmitted by the subwoofer or the large loudspeakers FL and FR. It is contained in the Dolby Digital Signal as an independent channel. In Dolby Digital the bass is defined as the lower frequencies < 100Hz. Because the recording of the bass with the microphones for FL, FR, C, SL and SR (simplified) are the same, it is separated and fed into the SUB channel. During playback the sound source for the single tone cannot be located. This is why only one loudspeaker (the subwoofer) is necessary for playback. This is a simplified observation and does not take into account that the cinema sound does not just consist of one signal with one frequency. The sound consists of many frequencies with independent time allocations (the Phase). This is why it cannot be concluded from this simple observation that a subwoofer can be placed in many different positions in a room. The following measurements will prove this. The subwoofer cannot be seen as a single instrument in a system because the frequency range around the splitting frequency 94Hz from Subwoofer and Front/Surround loudspeakers is transmitted simultaneously. When sound waves are overlaid the phase position of the individual waves (in conjunction with one another) are decisive for the sound pressure. That is why Subwoofer has a phase switch or even a controller. The correct adjustment of the amplifier and subwoofer are almost impossible without measuring technology. The measurement signals must come from the DVD to correctly compensate for the properties and adjustments of the Dolby Digital Decoder.

Measurement of the cumulative signals of the front loudspeaker
The subwoofer is turned off for this measurement the amplifier menu remains active. The sound pressure is measured with the cumulative signal FL+C+FR+SW with M *Mitteln*.
Because for this measurement only the lower frequencies are considered. It is not necessary to move the microphone as in the previous measurement.

Measuring the Subwoofer
The following measurement is carried out with the single signal SW. The measurement is started with M+*D*.
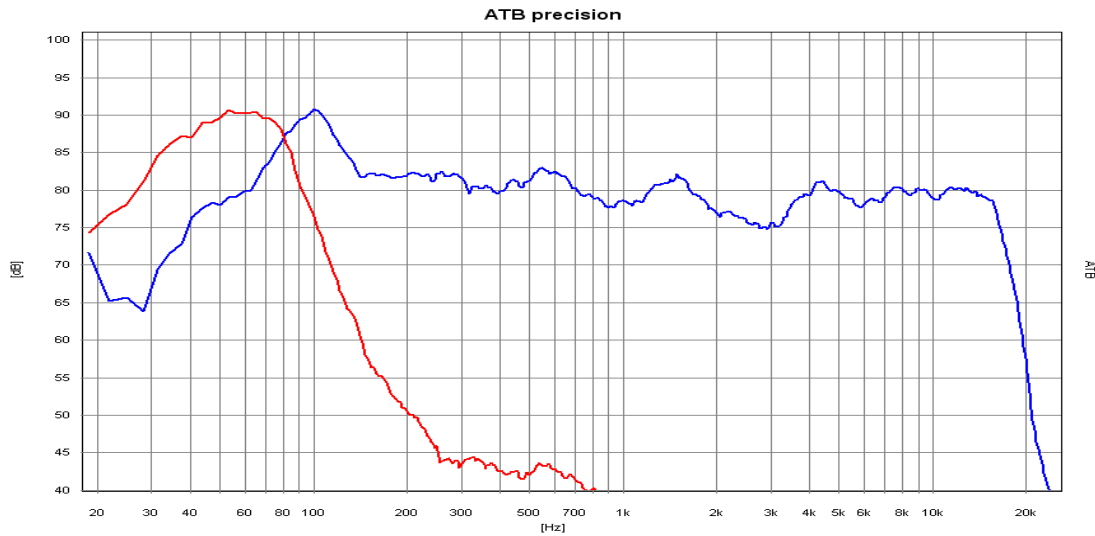
The picture shows the plots for the front loudspeaker and SW

Whilst the test is running the controller of the subwoofer (for the splitting frequency) is to be operated until the plot for the front loudspeaker and the bass plot are symmetrically mirrored with one another. The steepness of the bass curve can be adjusted with the controller or switch for the Filter steepness to be adapted to the steepness of the front loudspeaker curve. The point of intersection of both plots should be –3dB in reference to the front loudspeakers curve. The bass curve can be 4db higher than the front loudspeakers curve.

Measuring the cumulative+bass signals

Taken from the adjustment work described before it is seen that the cumulative and the Bass curves (measured separately) fit together. Now all we have to do is test if the sound waves overlap. This is determined with the phase behaviour of the single loudspeakers. By constant phase position the sound waves add up. With a phase difference of 180° the sound waves are cancelled out and a drop in the frequency characteristic curve is displayed. The phase positions can be adjusted with the phase controller. The measurement is started with the FL+C+FR+SW Signal and M+*D*.

ATB precision
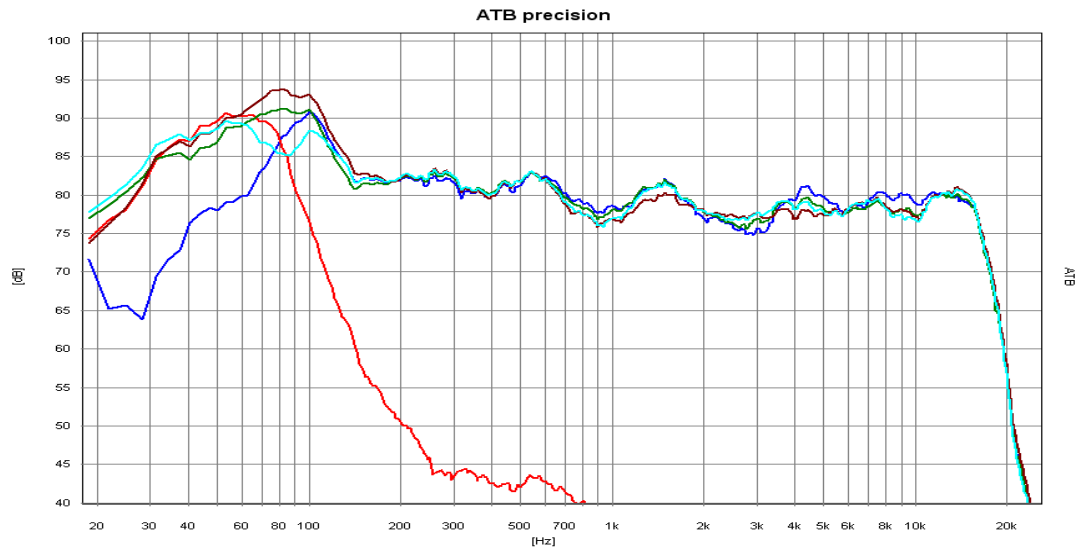
The plots show the cooperation of front loudspeakers and subwoofer and the influence of the phase controller. Light green, green and brown show the influence of the phase controller. Die brown curve is the adjustment without deletion in the low frequency range. The inconsistency of the middle and high tone ranges are brought forth through persons moving in the room.

Whilst measuring the phase controller is to be operated until there is no drop to be seen in the curve. For a subwoofer with phase switch is it adjustment with the smallest drop to be applied. If there is a drop in both adjustments then the subwoofer needs to be repositioned.

27.7   THE AUDIO TEST

With the audio test the quality of a surround sound system can be simply tested. The signal for the audio test is a noise circling the room. After the initial measurement it can also serve the fine adjustment of the system in the amplifiers menu CHANNEL BALANCE. With an optimal system the noise moves freely within the room without the sound source being detected. At the same time the noise should sound steady. When fine adjustment is to be carried out, first of all the centre volume is adjusted. This can be adjusted within a range of 1-3 dB. If the centre is to be heard with the front running signal then the volume is to be lowered. If the noise is too quiet in the middle then the centre volume must be raised. After this the same procedure is carried out with the surround loudspeakers. In opposite to the FL and FR loudspeakers they may have different volumes.

The second signal runs diagonally through the room. This signal puts the highest demands on the system. It serves to test the systems of recording studios.

28. CALIBRATING THE HIFI STEREO SYSTEM

28.1 MICROPHONE

This chapter is for the loudspeaker developers out there! The ATB PC program offers the function continuous measurement with averages, a tool that no developer can miss. When using a DVD player the measurement signal used is the PCM signal on the surround test DVD. If a CD player is the signal source then the CD plus signal is used on the auto test CD.

The usual frequency characteristic measurements are carried out with a predefined microphone position on a certain axis and with a predefined angle. This enables the developer to optimise the frequency characteristic for this position. To say that this diagram (results) is a lie would be too hard a statement but the truth is that loudspeakers with almost identical frequency characteristics sound completely different with this method. A more satisfactory result is attained when the reflected energy is measured. This corresponds to the audio impression heard with the ear because indirect sound plays a larger role in living rooms. The sound waves that are reflected from floor, walls and ceiling must have a volume that is proportional in relation to the music played. The professional journal "HiFi Vision" has not so long ago completed a series of tests with a rotating microphone and ascertained very satisfying results. Thanks to the measuring program ATB PC the true frequency characteristic can be simply determined using a personal computer. The continuous measurement function using averages is the key to correct results. Whilst the measurement is being carried out the microphone is moved slowly, with a distance of 60 cm from source, in a circle with a radius of 60 cm. Practice shows that the frequency characteristics are independent from this type of movement and thus enables the determination of a frequency characteristic error.
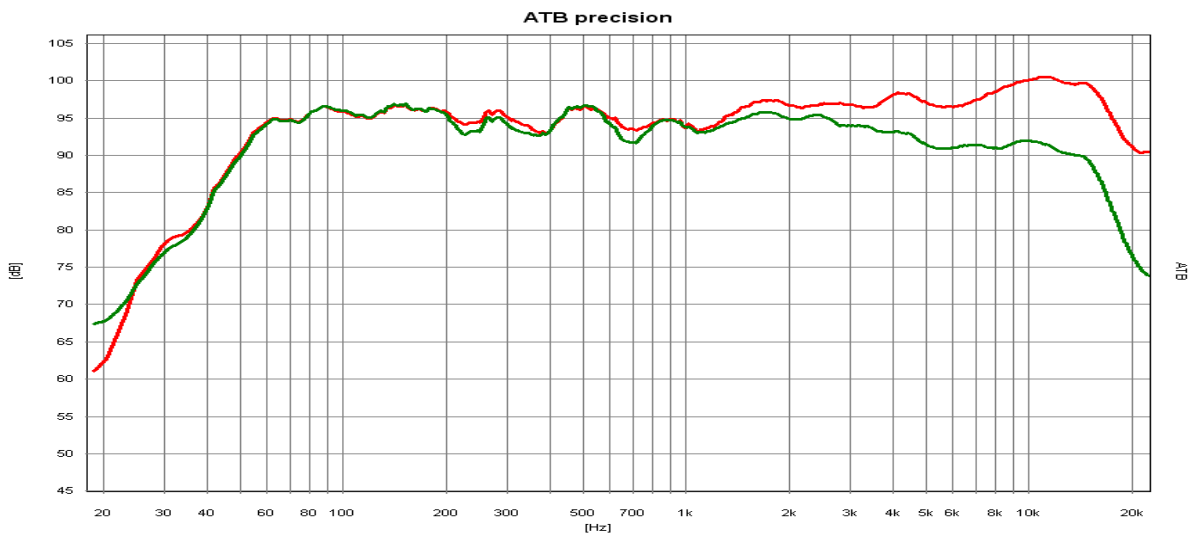
The frequency characteristic of a loudspeaker measured with a predefined axis red and its proper frequency characteristic green.

The separate curvatures of a well-constructed loudspeaker will only differ in the drop when coming towards the high tones, as depicted. A correct measurement is also well balanced. The drop in the high tone region is also an advantage in living quarters because the near walls amplify high tones anyway.

28.2 POSITIONING LOUDSPEAKERS AND THE SEATING POSITION

Chapter 17.4 has already dealt with the positioning of the loudspeakers. A stereo sound system demands a higher sound quality and special balance.

First of all the relationship between the distance to the walls and loudspeakers should be investigated.
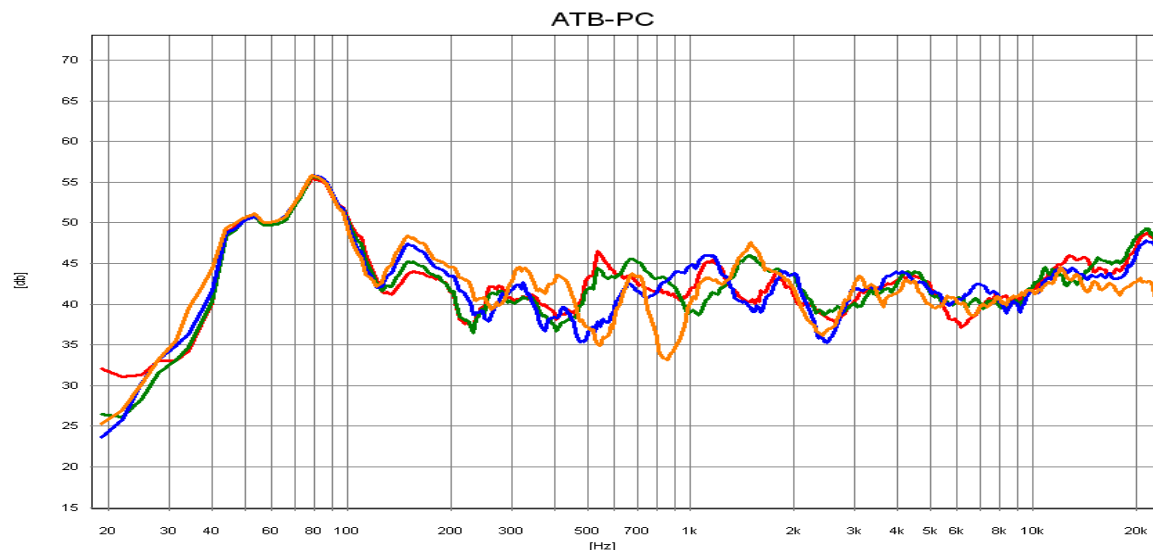

The curvatures display the frequency characteristics of a loudspeaker in relation to the distance from a lateral (side) wall. Orange has a distance of 8 cm from the wall, the jagged curvature is easy to recognize. The peaks and falls are caused by the interferences created by the sound reflected from the wall. With blue the distance has been enlarged by 16 cm. The curve has become a little more balanced out. The difference between green 24 cm and red 32 cm is not very large at all. This distance should be chosen for this loudspeaker. When the distance becomes greater a large ascent in the higher tones can be detected, this proving that the all-round emitting spherical high tones are not the ideal answer for living quarters. The properly measured frequency responses of the last chapter with their high tone drop prove to be ideal for living quarters.

If there is simply not enough room then it is possible to reduce sound reflection with wall carpets or curtains.

The distance from the back wall is very important for the middle tone range whilst choosing the seating position. A too smaller distance causes the middle frequency range to become jagged; this can only compensated through a larger distance to the wall or acoustic mats or wall carpets.
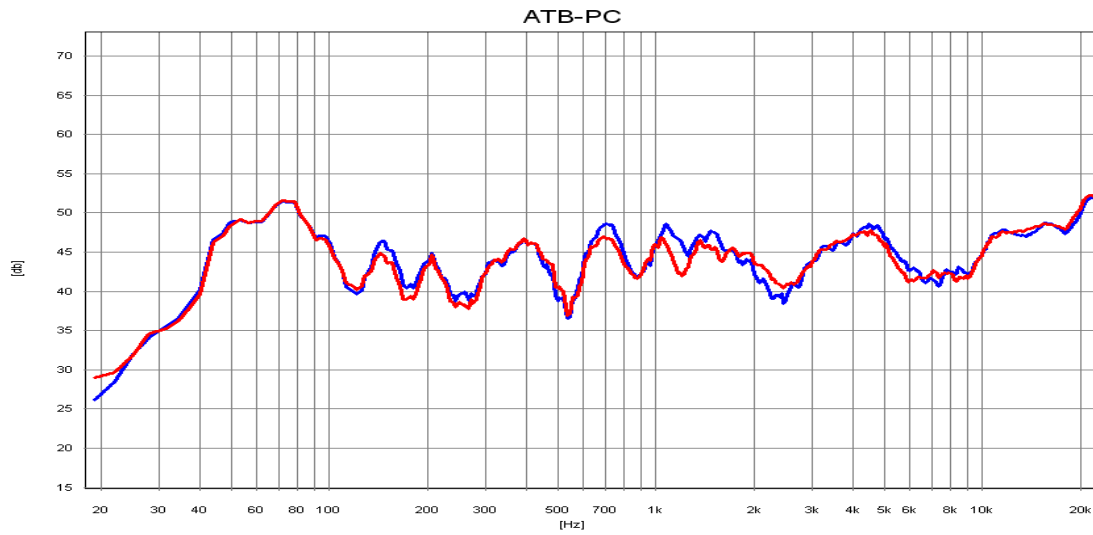
The figure displays the influence of the back wall in relation to the seating position. The waves in the blue curve are caused by the interference of direct sound and reflected sound from the back wall. The sound reflected from the wall is delayed and creates an interfering reverberation. The red curve displays the results when acoustic mats are placed on the back wall, the sound becomes more satisfying.

28.3   ROOM ACOUSTIC MEASUREMENT

Room acoustic describes the sound waves in a room. When a sound wave is created in a room then speckles are formed. Speckles are points on a sound wave that have either the greatest or the smallest amount of energy. This is especially important when working in the low frequency ranges. Everybody has had the experience that in a room there are certain areas where the deeper tones can only be heard very weakly and in other positions very loud.
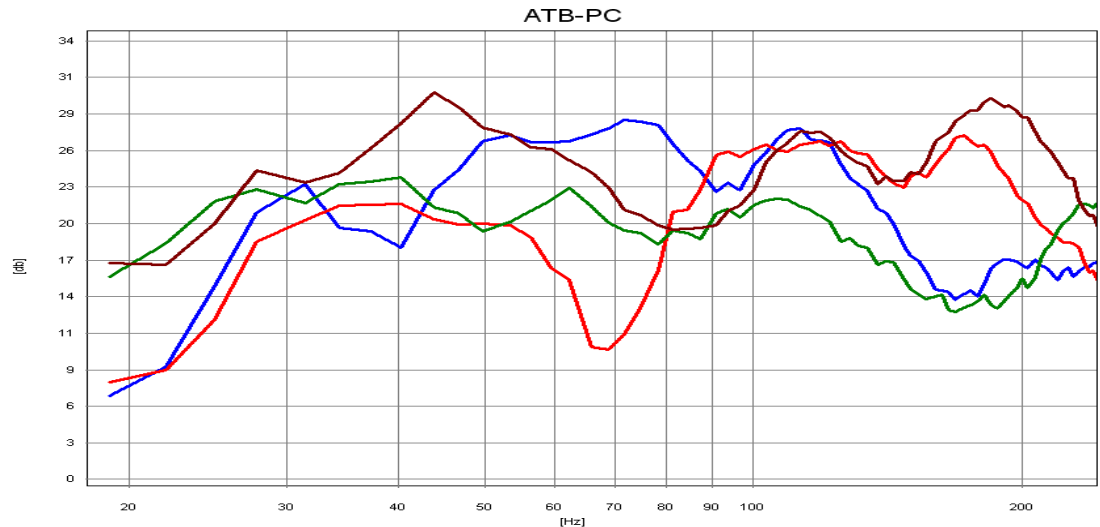
The figure shows the frequency responses in different positions in a room. The red and brown curves are speckles whilst the green curve reveals an optimal position.

These measurements display the results attained whilst the loudspeaker was between the front loudspeakers, the results display how important the positioning of the subwoofer is.

Further measurements are taken with specialized room acoustic programs, these mainly measuring the reverberation period. This value is only of importance to the acoustic engineer because the reverberation period of rooms can only be corrected with constructive steps.
The ATB PC program enables thoroughly satisfying results for the HiFi stereo owner when determining loudspeaker and seating positions. The optimum position is achieved when the frequency characteristics of loudspeaker and seating position are well balanced. The room acoustic and loudspeaker are both measured when determining the seating position.

29.   Surround-Test DVD

The test signals
The curves shown are from test signals played back from a DVD Player and measured with ATB precision USB.

ATB precision
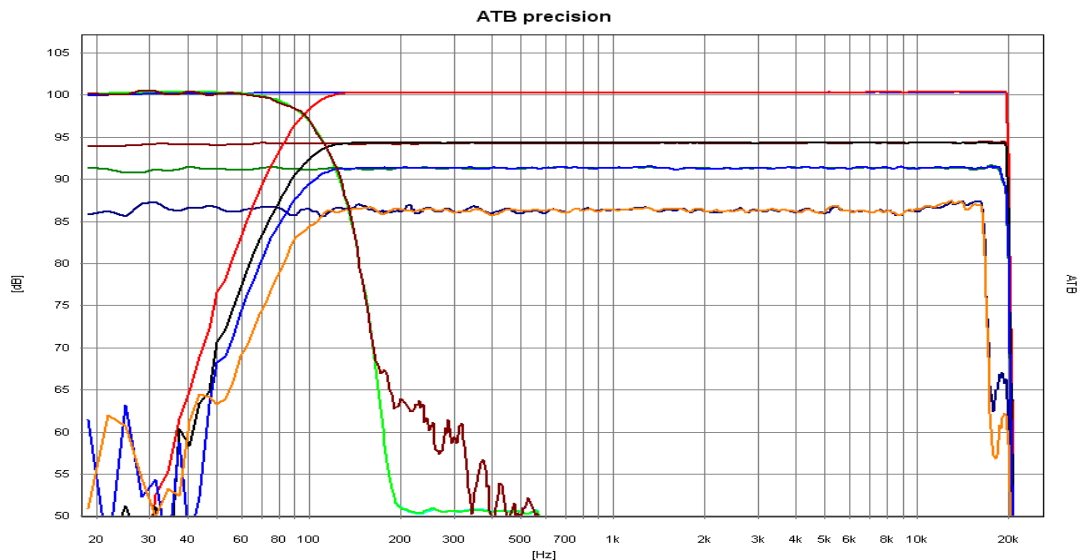
The Signals:

Blue above	= single signals FL, C, FR, SL, SR
Light green	= single signal SW, with Surround amplifier output +10dB
Brown	= cumulative signals FL+FR, SL+SR
Green	= cumulative signals FL+C+FR
Blue bottom	= cumulative signals FL+C+FR+SL+SR
Red	= cumulative signals +Bass FL, C, FR, SL, SR
Black	= cumulative signals +Bass FL+FR, SL+SR
Blue middle	= cumulative signals +Bass FL+C+FR
Orange	= cumulative signals +Bass FL+C+FR+SL+SR
Brown	= cumulative signals +Bass SW

The not quite smooth frequency characteristics are created by the high compression rates of the cumulative signals.
The PCM signal corresponds to the single signals with a border frequency of 22kHz.
The audio signals consist of round and diagonally moving noises in the room.

30.   TECHNICAL DATA

Technical data of the ATB 805
2 x
Microphone input: Frequency range   20Hz – 20kHz, +0.1dB
                     Input range   50mV – 0,7V, noise < -50dB, Distortion < 0.05%
                     Input impedance   2.2kΩ, 9V microphone-power
                     ATB 805 XLR   Imput impedance 6.8kΩ, 48V microphone power
2 x
LINE output   Frequency range   20Hz – 20kHz, +0.1dB
                     Max. level   6dBV
2 x
LINE input   Input impedance   87.0kΩ,
                     Max. Level 30V
Speaker output   Frequency range   20Hz -22kHz, +0.2dB,
                     Power   4Ω =60W, 8Ω = 30W, Distortion < 0.05%
Connection   USB 2.0
Power   230V~ / 50Hz / 120VA
Dimensions   (W x H x D) 430 x 40 x 150 mm
Program   Windows 7, Vista, 8.1, 10