

# Kirchner elektronik

The screenshot displays the ATB QC software interface, which is used for audio testing and analysis. The main window shows a graph titled "ATB PC Demo (Bass-test)" with the Y-axis labeled "dB" (ranging from 0 to 60) and the X-axis labeled "Hz" (logarithmic scale from 20 to 20k). The graph shows a red curve (B) and a blue curve (MH) that intersect around 40 Hz. A green curve (BMH) is also visible, showing a sharp drop around 40 Hz. The interface includes a top menu bar with options like File, Quality, Measure, Input, Output, Settings, Extras, and Info. On the left, there are panels for Input (SigmaTel Audio 013 Mikrofon), Output (SigmaTel Audio), Signal (BMH), and Diagram (DVD-PCM). A "Settings" panel on the left includes "System-correction" (none), "Mic-correction" (as set: intern. cardcorr), and "loaded file". A "Step" window is open, showing a graph titled "ATB-PC" with the Y-axis labeled "ATB-PC" (ranging from -180 to 180) and the X-axis labeled "ms" (ranging from 0 to 24). The graph shows a green curve that rises sharply from 0 ms and levels off around 12 ms. A "Values" window is also open, showing "Phase [°]" with X=1062 and Y=0. The bottom status bar shows "autoexport: none".

# ATB PC Pro HANDBOOK

H A N D B O O K

## **ATB PC Pro**

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## 1. AN INTRODUCTION TO THE SYSTEM

The ATB PC Measuring program combined with the computer's soundcard forms an exact and extensive Audio Analyser.

The following measurements are available:

Electric and acoustic amplitude frequency response, SPL

Electric and acoustic phase frequency response

Impedance measurement

Step response

Distortion THD, IMD and nonlinear distortion

The measuring system itself consists of a Surround-Test DVD for measurement signals, a PC with (USB) soundcard for the recording and valuation of sound production and the ATB PC measurement program to carry out measurements. Using the most modern measurement technology allows handling as simple as with an analyser.

The Windows ATB PC measuring program contains all the properties for frequency characteristic measurement featured in the professional ATB precision program.

The DVD includes signals for audible surround sound control.

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 interiors.

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 soundcard we have developed a new measurement technology which optimises instrumentation.

For Windows XP there is an internal generator.

For Windows 98 the measurement signal generation using a CD or DVD player bypasses the low quality multiplex operation of the soundcard which simultaneously tasks play and record. Only the record function of the soundcard is used in conjunction with system compatible speech recognition technology. Notebooks and external USB soundcards prove to be ideal for this field of work. The frequency characteristic errors of the soundcard are avoided with a correction curvature. With the help of the adapter the correction curvature is easy to create. The frequency errors of the soundcard, due to sample rate inaccuracy and fluctuation, are bypassed with a new correlation method. The new correlation method allows an acoustic measurement independent from distance definition. Thus making a time window superfluous, this results in handling as easy as operating a real time analyser.

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

The ATB PC Pro program works with Window XP, 7, Vista, 8.1, 10 32bit or 64bit

## 2. GETTING STARTED

### 2.1 INSTALLATION OF THE USB DONGLE

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 + I  
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" → In search box "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.

It could be that the yellow triangle is not to seen. It can be behind multimedia, input devices, audio, USB devices and other. Click at it to see the triangle.

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"

## 2.2 INSTALLATION OF THE USB DONGLE FOR WINDOWS XP

In order to operate the ATB PC II program a USB Dongle has to be installed.  
To install the Dongle, first insert the Dongle into a free USB slot.  
Important: The chosen USB slot is to be kept whilst operating with the program.

Inserting the dongle should activate the Hardware Wizard

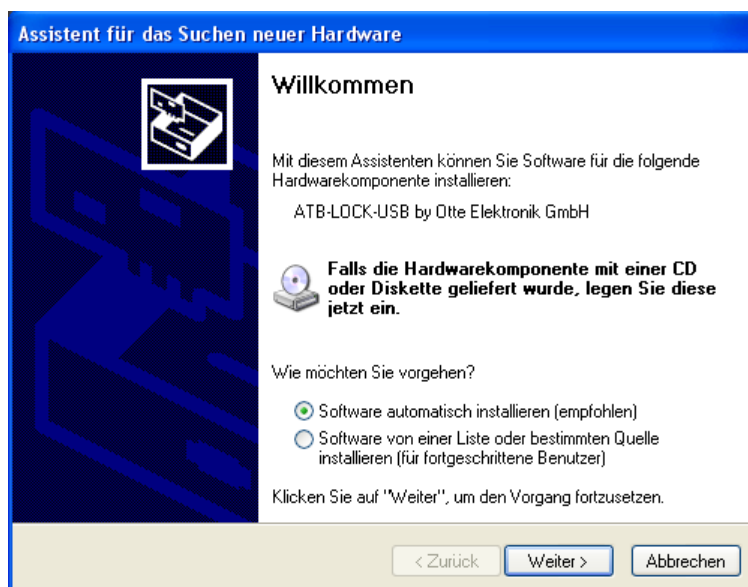


After clicking on the message the following menu will appear (it may also appear automatically)

If there is the message "Unknown device" the computer is not compactable with the dongle.

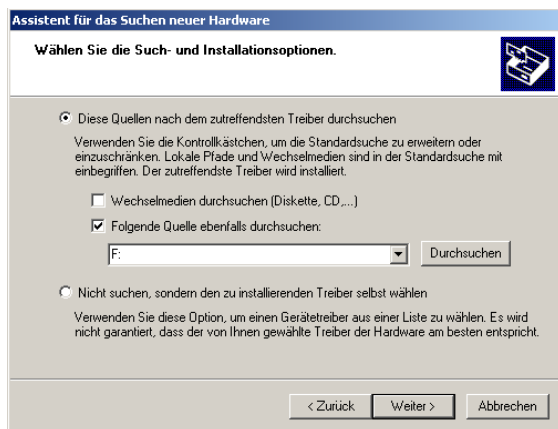
In this case you have to look for another USB input.

Please choose the **manual** installation.

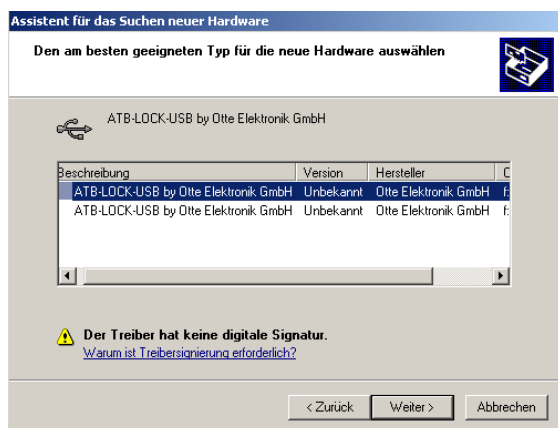




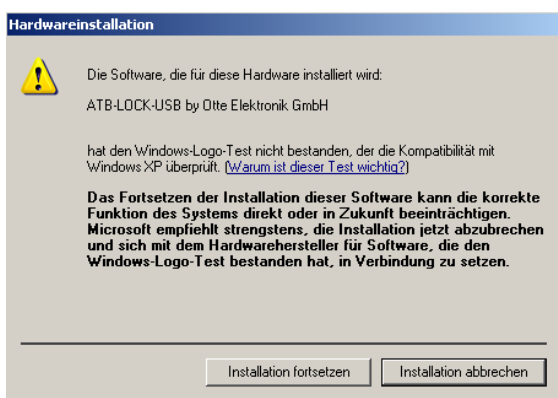
The driver is located on the ATB PC Software CD-ROM. Please choose the correct drive letter (eng. F:).



For Window XP choose the file **ATBUSXP**.



The driver has not been Microsoft certified



Please select **Continue Installation**.



Click on the **Finish** button.

**You are ready to go!**

With the short instruction of the Handbook you will get your first measurement.

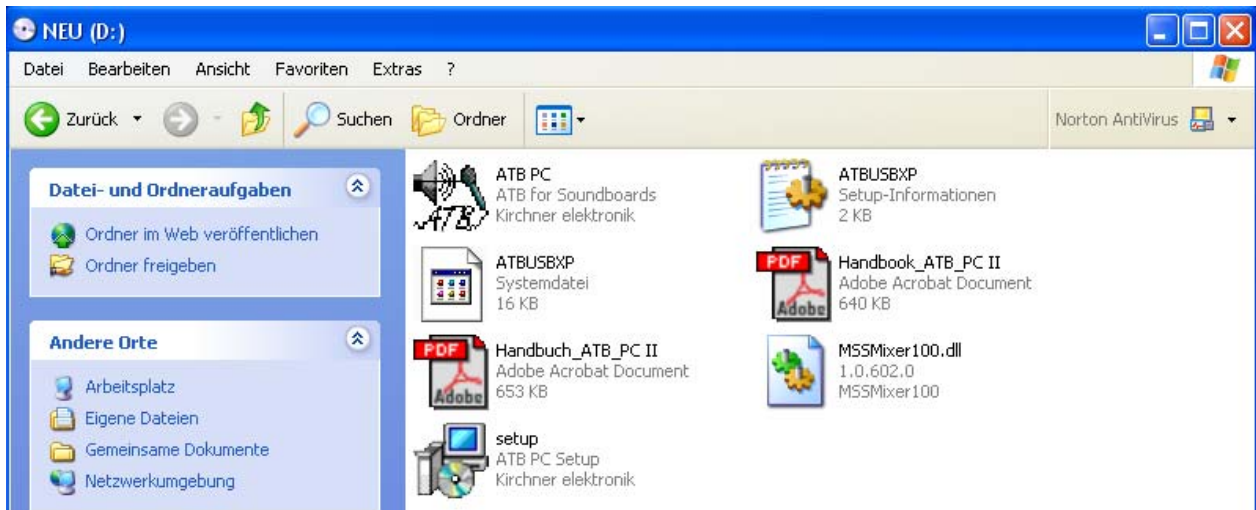
If you want to check the correct installation of the dongle then open SYSTEMCONTROL in the START menu then open SETTINGS choose SYSTEM from the list then HARDWARE and then DEVICE MANAGER where you will find a list containing various devices including the BUS controllers :



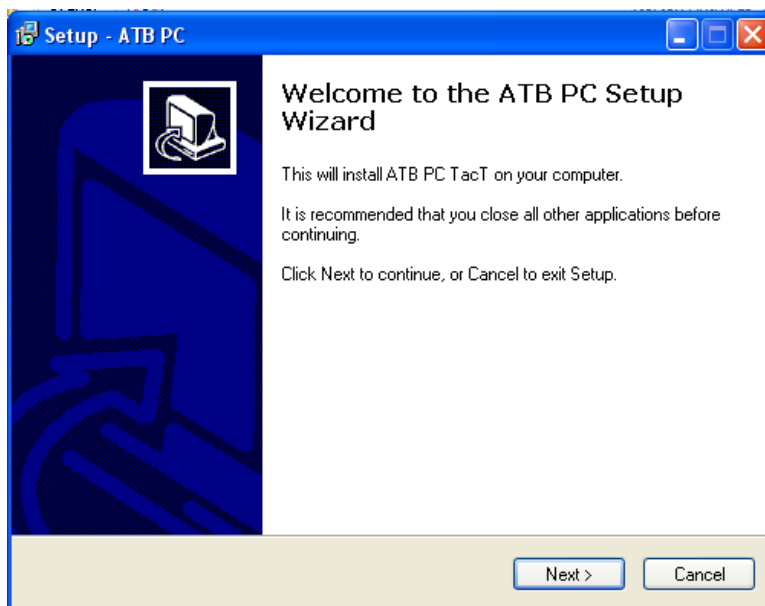
“ATB-LOCK-USB by Otte Elektronik GmbH” should be listed.

## 2.3 INSTALLATION OF THE PROGRAM

To install the program place the CD “ATB-PC Software” into the hard disk drive.  
Double click on the workplace icon on the Windows desktop. Open the disk drive e.g.  
“NEW (D :)”  
Double click on the CD icon “ATB-PC”



A double click on the Icon SETUP.EXE will start the installation.



To continue click on the NEXT buttons. Follow by clicking on the INSTALL button and complete the installation by clicking on the FINISH button.

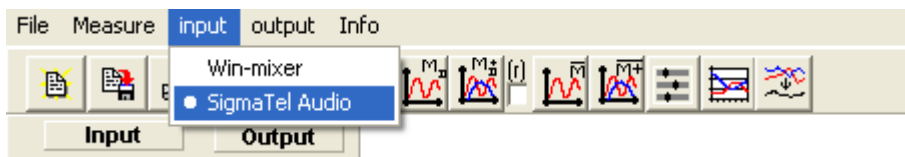
### 3. SOUNDCARD

The program runs with all newer soundcards with a sample rate capacity of 48kHz. If 48kHz is not possible an error message will be given at the start of the program

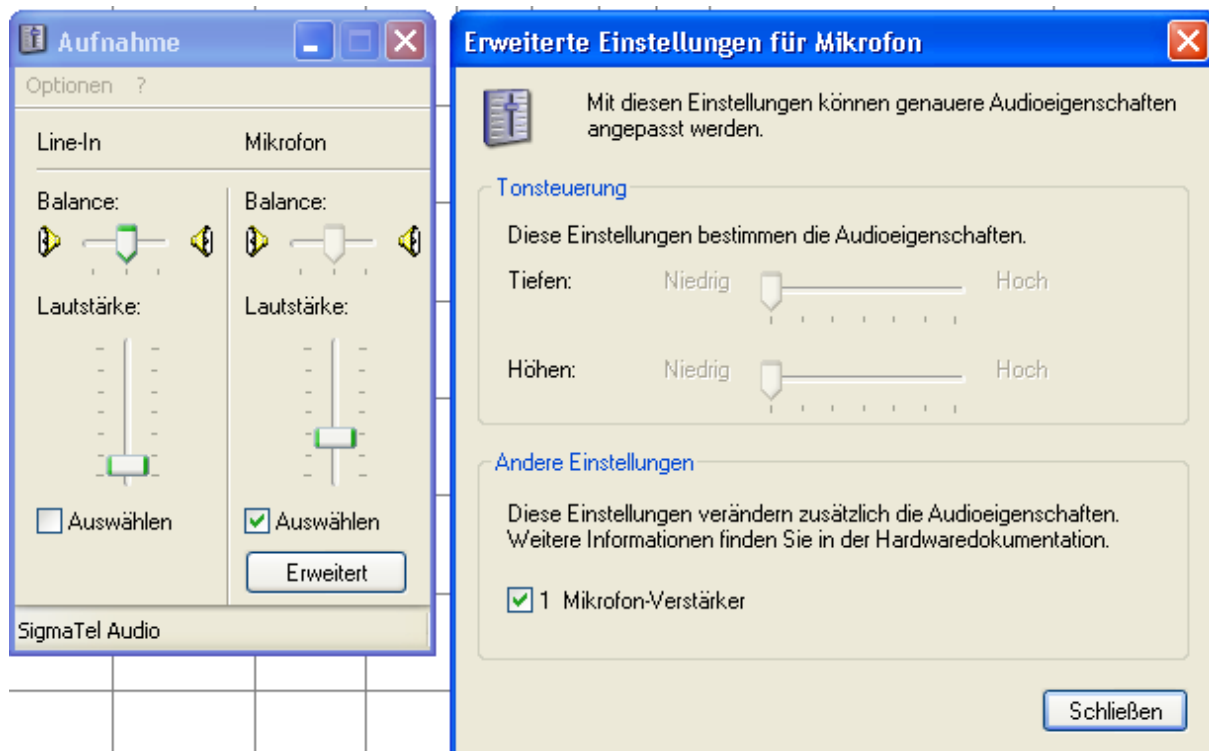
#### FEHLER BEI BEREICHSPRÜFUNG

or the measurement curvature will tend to fluctuate extremely whilst operating in system correction (is not a constant line).

#### 3.1 SETTING UP THE SOUNDCARD INPUT FOR XP

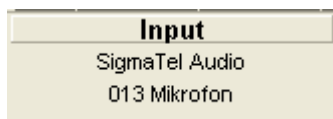


With one soundcard on the computer, it will be chosen from the program. With more than one soundcard, choose one by clicking in the menu.



In the next step the Win-mixer is called up. In the RECORD menu the microphone is chosen. With the button EXPAND the menu is EXPANDED and OPTIONS for microphones is opened. Here you click on microphone amplifier. The options are also called Mic-boost or +20dB.

When using a microphone with preamplifier or when doing electrical measurements, LINE-IN is chosen. If a digital input is available, that can also be used.

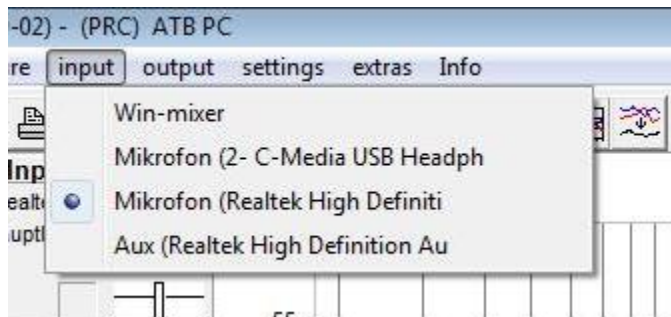


The name of the soundcard and the input is shown in the main menu.

### 3.2 SETTING UP THE SOUND CARD INPUT FOR VISTA, 8.1, 10

For the Windows Vista there is to note:

1. Please use the on board soundcard.
2. There are computers with an identification of the connected sockets. Please connect the test-box first to the IN and OUT of the soundcard. Then start the program. After exchanging the test-box with the microphone and amplifier input, please start the program again.
3. For a computer with front and back sockets use the back sockets. The front sockets could have a noise spectrum. If there is a setting for dividing the front from the back sockets, please do not use this.



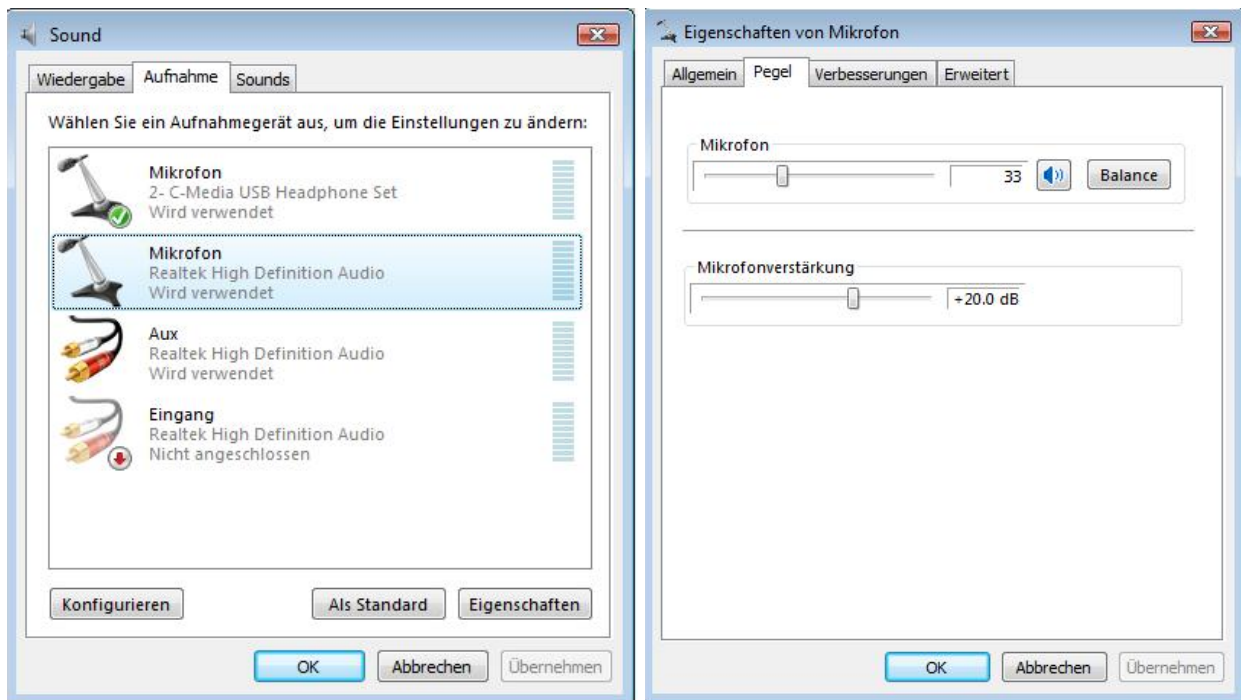
In the INPUT menu there is the setting of the microphone and the soundcard.

There is no win-mixer in the Vista program.

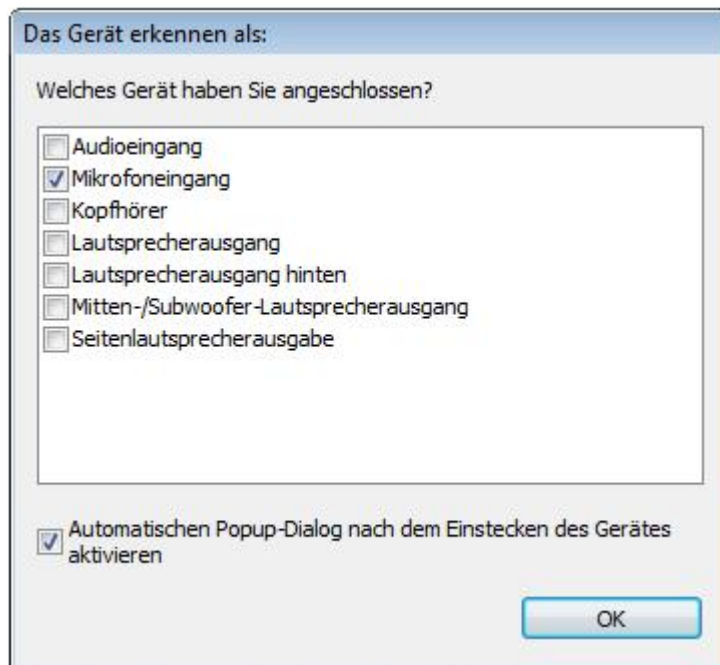
After the selection of the microphone there is to set the microphone amplification. The menu is to find at

#### SYSTEMCONTROL – HARDWARE AND SOUND – SOUND

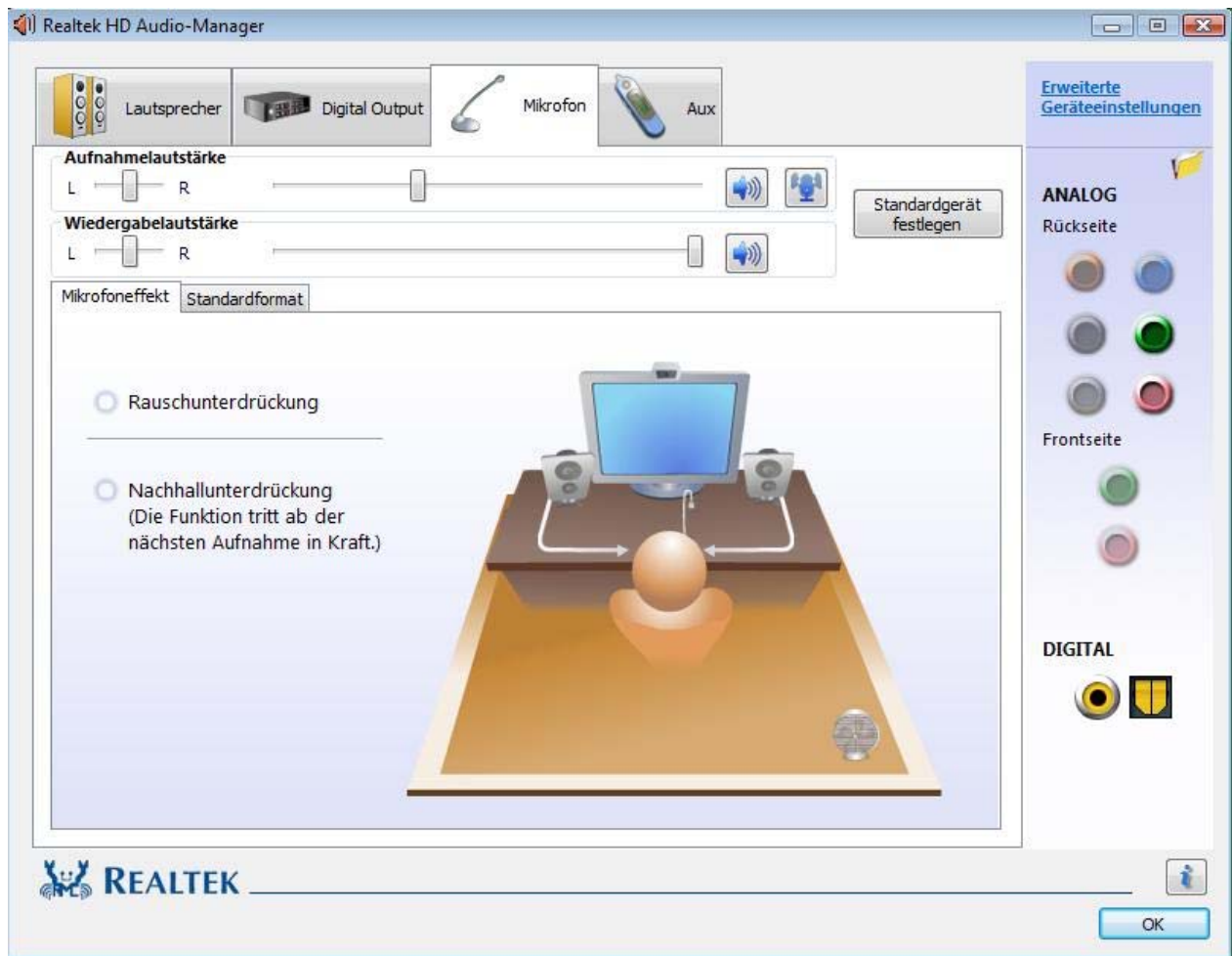
In the SOUND menu there is to select the microphone. There is to set for the system-correction the amplification 20dB. For the SPL measurement it is 30dB.




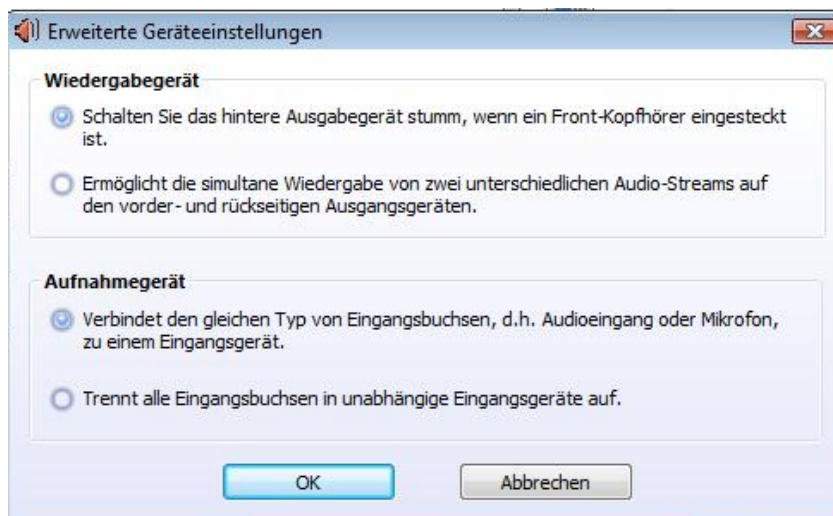
For the computer with identification of connected sockets there will come the following menu after connecting the test-box.



After selecting the microphone you will see the microphone menu.

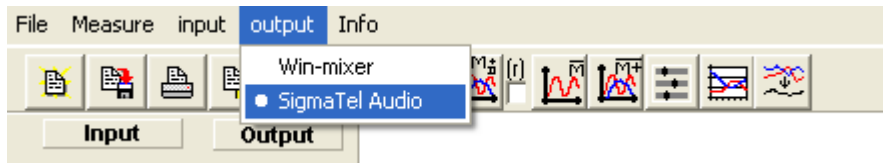


To set the microphone amplification click at this button .  
For measurement please make no other setting in the menu.

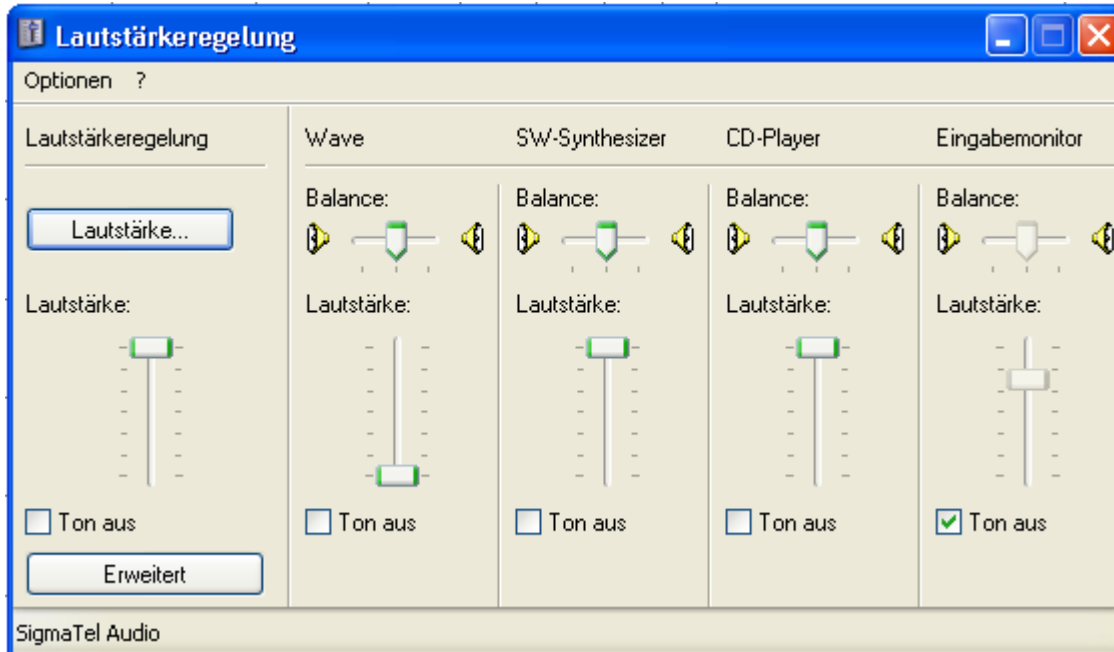


Do not change this setting.

### 3.3 SETTING UP THE SOUND CARD OUTPUT for XP



With only one soundcard on the computer that is chosen from the program. When there are more than one soundcard, you click on the preferred one in the menu.



In the menu under sound level, the sound level control is brought to the top position and sound off not activated.

Under INPUT MONITOR, SOUND OFF activated, is checked.

Under WAVE, SOUND OFF must not be activated.

Under SOUND LEVEL CONTROL you can choose under EXPANDED the digital SPDIF output.

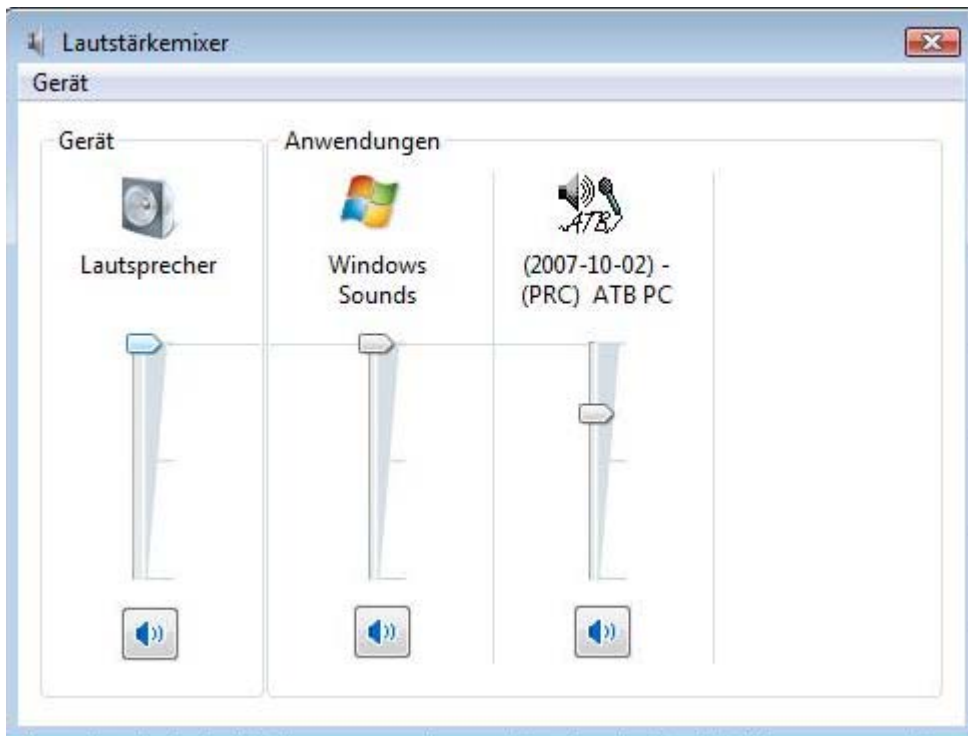


### 3.4 SETTING OF THE SOUND CARD OUTPUT FOR VISTA, 8.1, 10



In the menu OUTPUT is speaker to set.

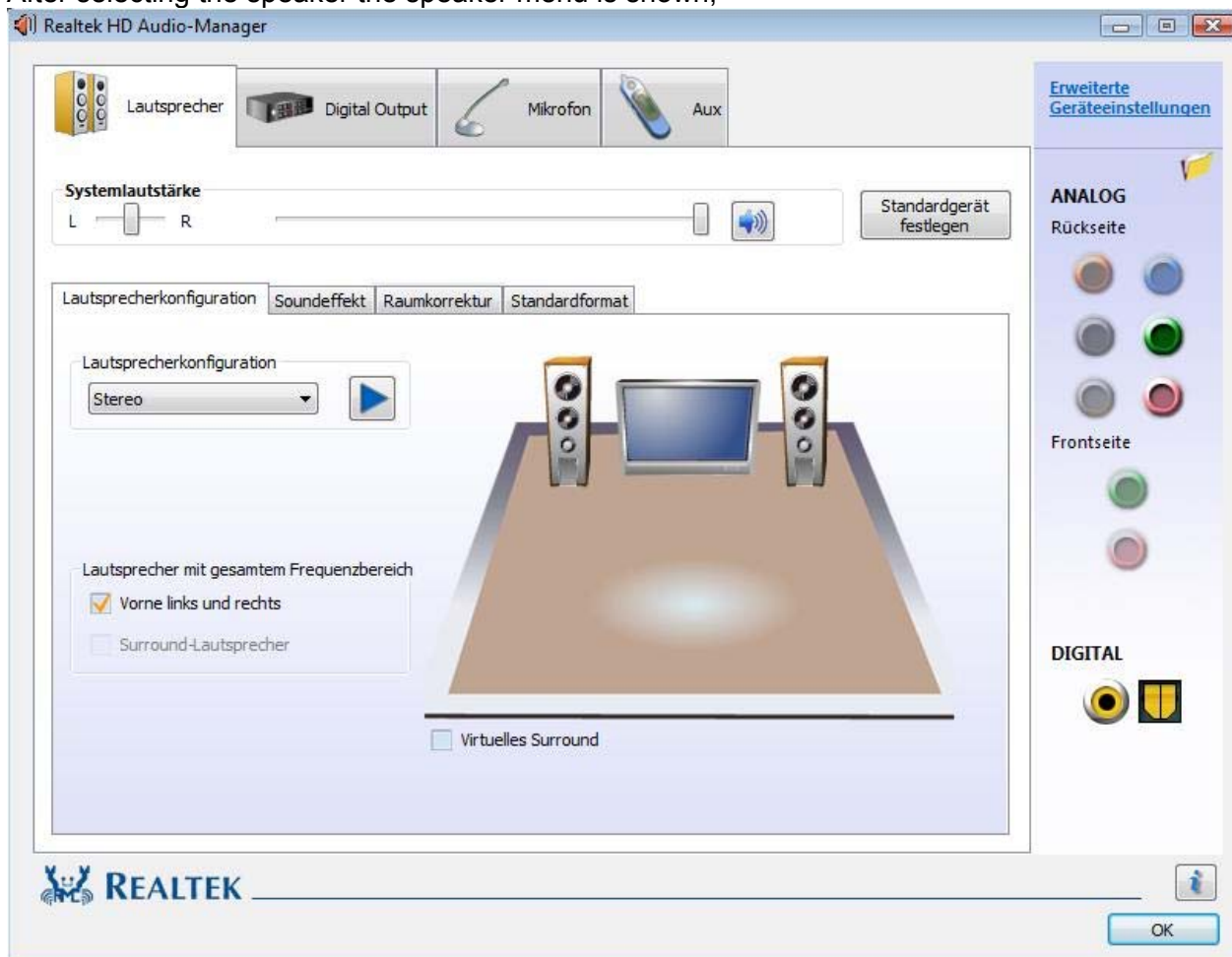
The win-mixer menu shows the setting for the output.



For the computer with identification of connected sockets there will come the following menu after connecting the test-box.



After selecting the speaker the speaker menu is shown,



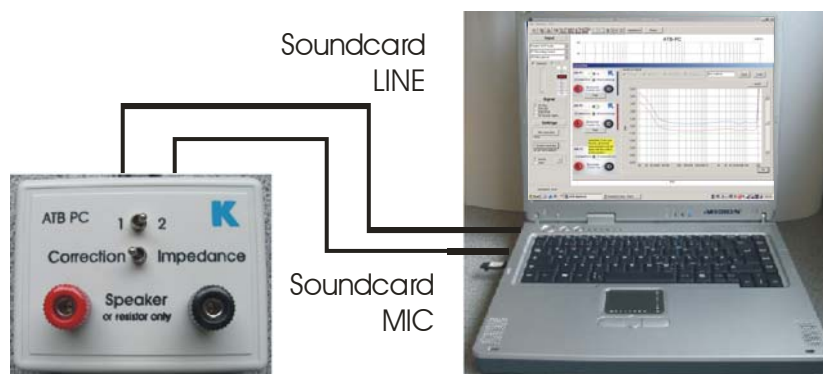
## 4. ADJUSTMENT

### 4.1 ADJUSTMENT WITH THE SIGNAL FROM INTERNAL GENERATOR

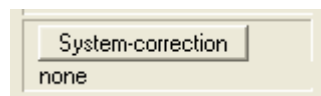
Most soundcards have a good recording function. The problem is that the microphone input channels do not possess a sufficient linear frequency characteristic required for a satisfying result when testing; therefore they need to be corrected.

#### CALIBRATION (SYSTEM-CORRECTION)

##### System-correction



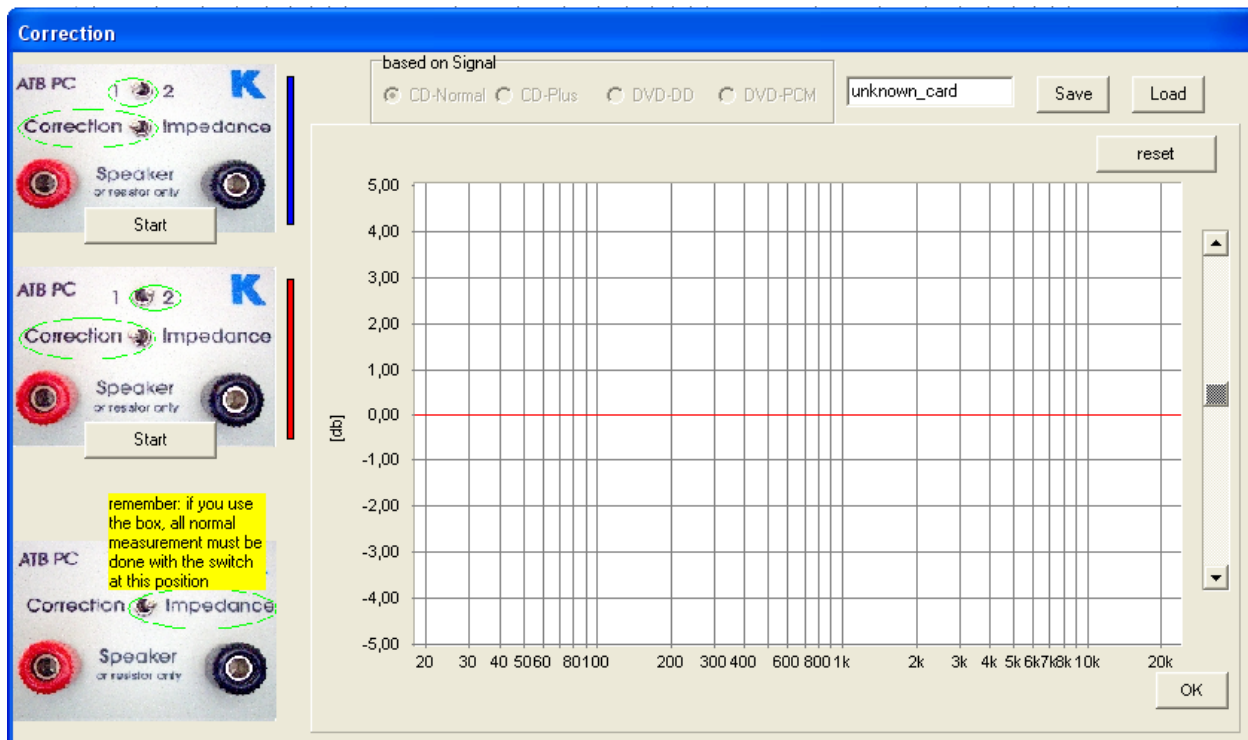
Here the system correction for the internal operation is shown.  
To do this the button in the main menu



To create the SYSTEM CORRECTION you must first connect the Testbox by inserting the cinch cable with the Cinch adapter to the headphone out of the soundcard. The end with the jack plug is connected to the Microphone IN socket (red-pink) of the soundcard.

The switches are to be set to 1 and correction.

After pressing the SYSTEM CORRECTION button the CORRECTION menu will appear.



A measurement can be initiated by pressing the top start button, switch position 1 and correction are also initiated. First of all the program will find and adjust to the optimum level. The procedure can be observed at the top left of the modulation display.

After a successful first measurement a blue frequency response curvature will be displayed. After switching the Test-box to position 2 and pressing the bottom start button, the adjustment is carried out. The result is displayed by a red curvature.

The measurement correction is herewith completed and is now available for all following measurements.

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

“Missing/weak signal” or “no signal”

Check to see if the textbox is properly connected. (The left AUDIO OUT socket of the DVD/CD player is connected to the MICROPHONE IN of the soundcard.

When using the surround-test DVD in conjunction with a DVD player check that the left hand loudspeaker is selected.

When using a CD player, track 1 or 2 should be selected from the CD.

It is also possible that the microphone needs to be amplified (boosted). To enable this, SETTINGS are to be selected from the START menu on the Windows desktop, from there on select SYSTEMCONTROL then SOUNDS/AUDIO DEVICES. From the card select AUDIO and click on RECORDVOLUME, the record mixer should appear. Click on the EXTENDED option in the microphone window and activate the microphone boost.

“signal too strong”

Set the soundcard settings in the main menu to RECORD or WAVE IN and MICROPHONE.

Switch the Testbox to CORRECTION.

Turn the microphone amplifier (booster) off. To do this SETTINGS are to be selected from the START menu on the Windows desktop, from there on select SYSTEMCONTROL then SOUNDS/AUDIO DEVICES. From the card select AUDIO and click on RECORDVOLUME, the record mixer should appear. Click on the EXTENDED option in the microphone window and deactivate the microphone boost.

**“insufficient record level”**

It is also possible that the microphone needs to be amplified (boosted). To enable this, SETTINGS are to be selected from the START menu on the Windows desktop, from there on select SYSTEMCONTROL then SOUNDS/AUDIO DEVICES. From the card select AUDIO and click on RECORDVOLUME, the record mixer should appear. Click on the EXTENDED option in the microphone window and activate the microphone boost.

**“same level”**

For the second measurement turn the Testbox to position 2

**“level too high”**

Repeat the measurement, use position 1 on the Testbox switch for the first measurement and position 2 for the second measurement.

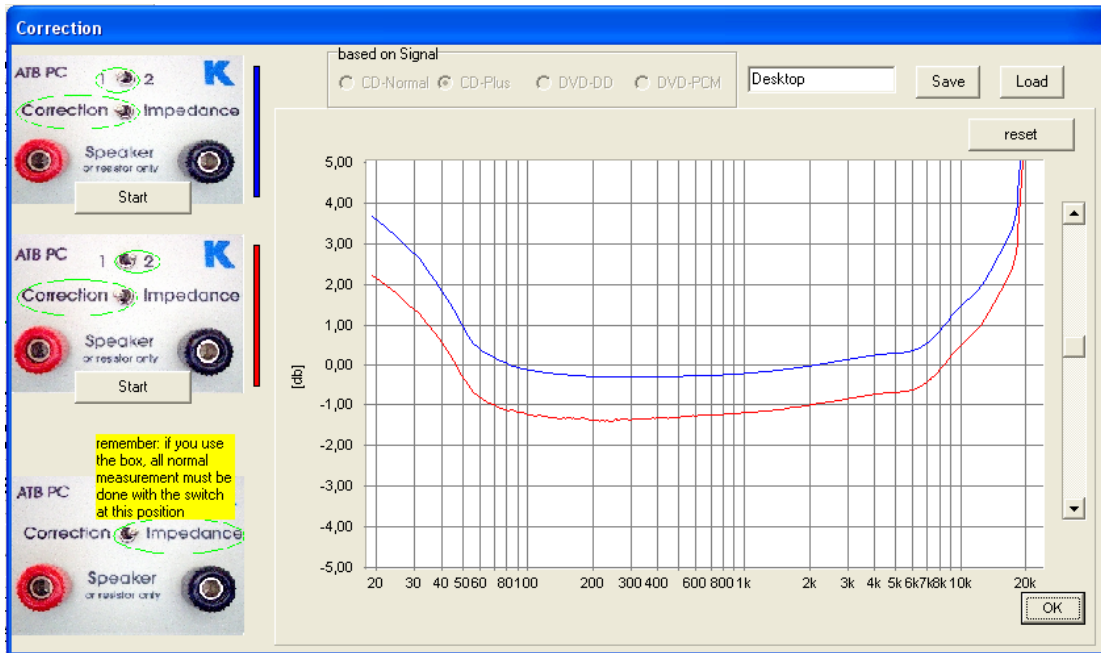
When the measurement curvatures correspond with the figure below then the correction measurement was successful.

After successful completion the curvature should be saved along with the name of the soundcard, input and DVD/CD player.

After leaving the menu with OK, a test measurement can be taken. A test signal is used in compliance with the system correction measurement. The measurement is started with the button M. When the correction curvature has been correctly measured then the result is a straight line that descends steeply at 20 kHz.

the System-correction menu is called up.

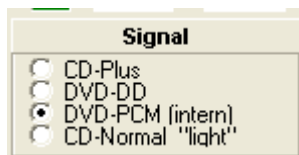
With START the measurement is started. Here the signal strengths are automatically setup. Following image appears, that suggests a successful calibration.



After saving, the correction curve is available for all following measurements and is automatically loaded at the start of the program.

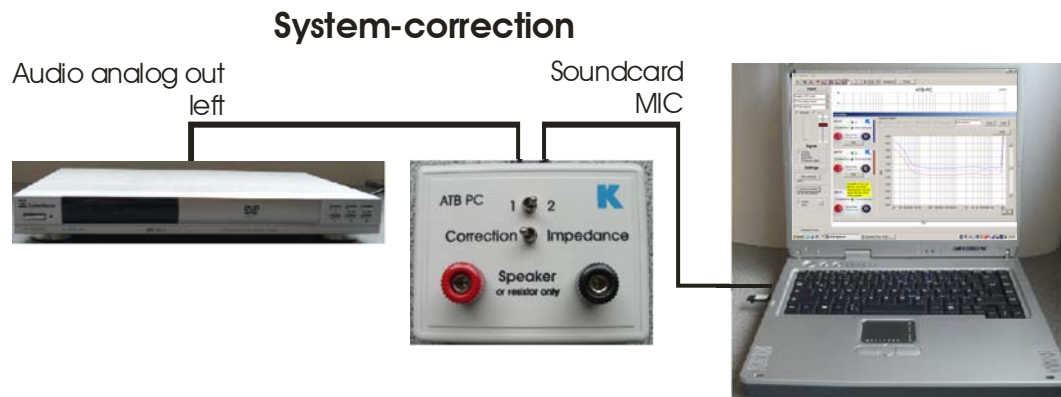
## 4.2 ADJUSTMENT WITH THE SIGNAL FROM CD/DVD PLAYER

### CHOOSING THE SIGNAL



At this point you have the choice between taking measurements with the Surround-Test DVD or the Auto-Test CD. For the Surround-Test DVD the option DVD-DD is chosen (as shown above). CD-Plus is chosen for the CD.

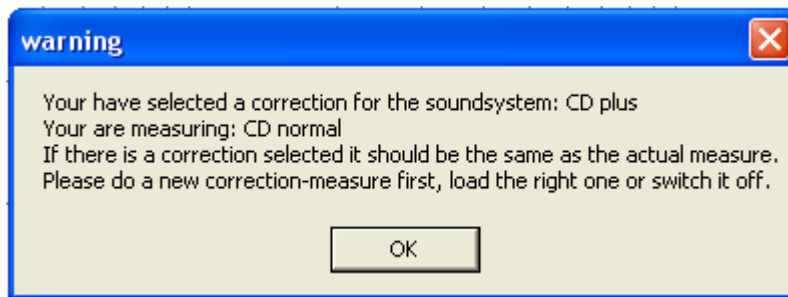
### ATTACHING THE TEST BOX



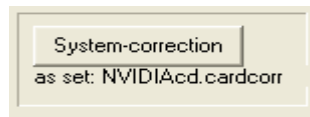
Connect the Test box by inserting the cinch cable to the left (channel) analogue socket of the DVD player. The end with the jack plug is connected to the Microphone in slot (red-pink) of the soundcard. The switches are to be set to 1 and correction.

When undertaking a correction the DVD player is also corrected. The Dolby Digital and the PCM signals also require different correction curvature. Even the measurement space can be entailed in the correction. Therefore as many card correction curves as desired can be used.

If you have started a measurement without using already measured or previously saved system corrections an error message will appear.



The message tells you that you have selected either a wrong or no system correction. Therefore you must either select or create an appropriate system correction. After closing the message by clicking on the OK button the menu system correction will appear.  
(The menu can also be opened under SETTINGS).



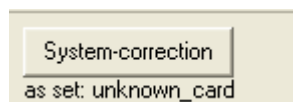
## STARTING THE DVD PLAYER

The Surround-test DVD is to be placed into the DVD player and "Single signal" is to be selected in the Surround-test DVD's menu. In the following summary the left front loud-speaker should be flash on and off (DVD track 3).

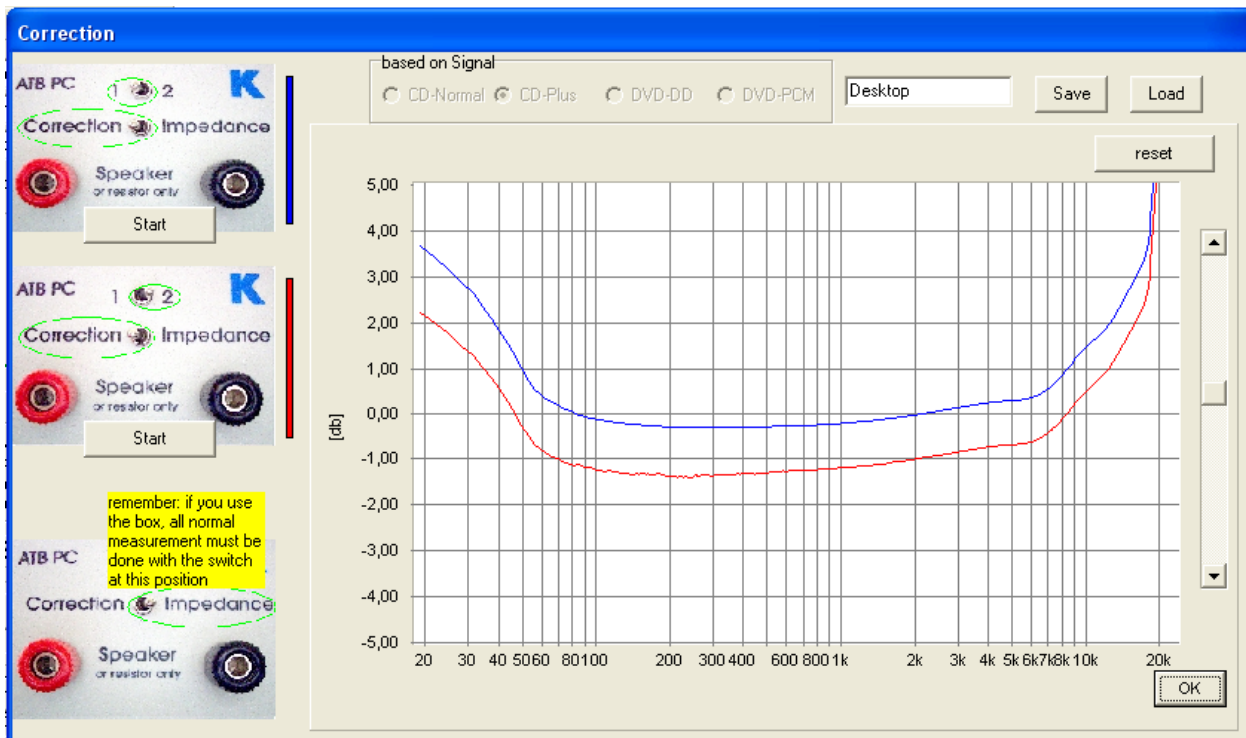
When using the CD please choose CD track 2.

## SOUNDCARD ADJUSTMENT

After pressing the button "system correction"



The correction menu will appear:



A measurement can be initiated by pressing the top start button; switch position 1 and correction are also initiated. First of all the program will find and adjust to the optimum level. The procedure can be observed at the top left of the modulation display.

After a successful first measurement a blue frequency response curvature will be displayed. After switching the Test-box to position 2 and pressing the bottom start button, the adjustment is carried out. The result is displayed by a red curvature.

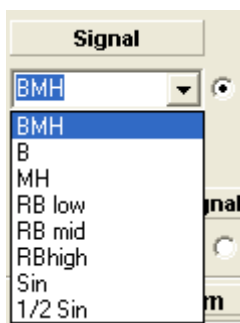
The measurement correction is herewith completed and is now available for all following measurements.



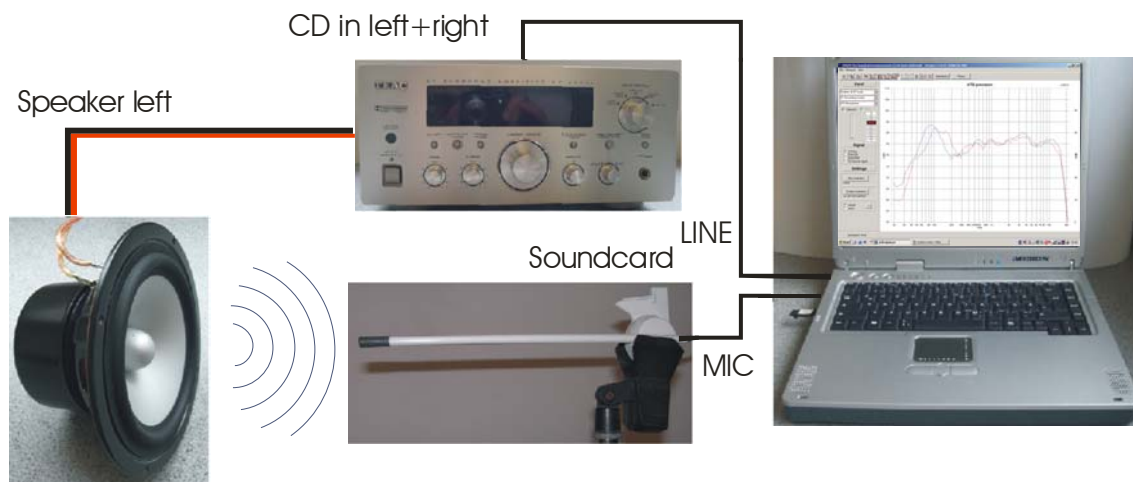
## 5. MEASUREMENT

### 5.1 MEASUREMENT WITH THE SIGNAL FROM INTERNAL GENERATOR

#### CHOOSING THE MEASUREMENT SIGNAL

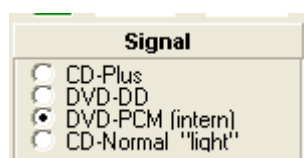


For the internal Generator the signal BMH is to set and to mark  
Before measurement, the soundcard is tested and a correction file is generated.  
The ATB PC program allows measurement using the internal generator. Here the measurement signal is generated by the soundcard. In the System-correction, a correction curve is generated for the input and output of the soundcard.



### 5.2 MEASUREMENT WITH THE SIGNAL FROM CD/DVD PLAYER

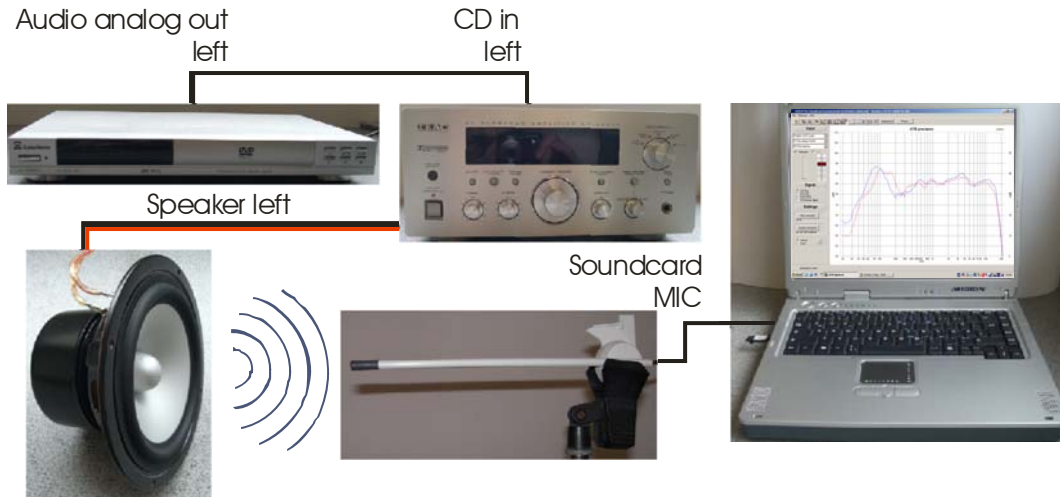
#### CHOOSING THE SIGNAL



At this point you have the choice between taking measurements with the Surround-Test DVD or the Auto-Test CD. For the Surround-Test DVD the option DVD-DD is chosen (as shown above). CD-Plus is chosen for the CD.

## MEASUREMENTS

### SPL



Please connect your DVD player back to the HiFi system.

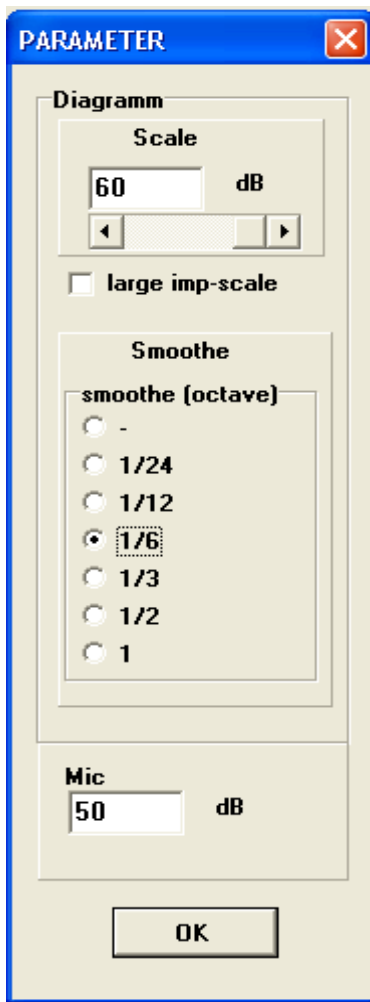
## 5.3 CARRING OUT THE MEASUREMENT

Position the microphone 1m away and in front of the left front loudspeaker and connect the microphone to the microphone socket of the soundcard. The test-box is not necessary for this measurement. The microphone should be positioned at “ear height”. Usually about 80-90 cm above the floor, as music is usually listened to whilst in a seated position (unless you are intending to dance of course!)

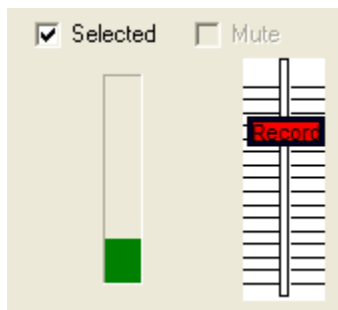
A double click on the desktop connection will activate the atbpc.exe program.



Choose “parameter” from the menu and position the smooth value to 1/6:



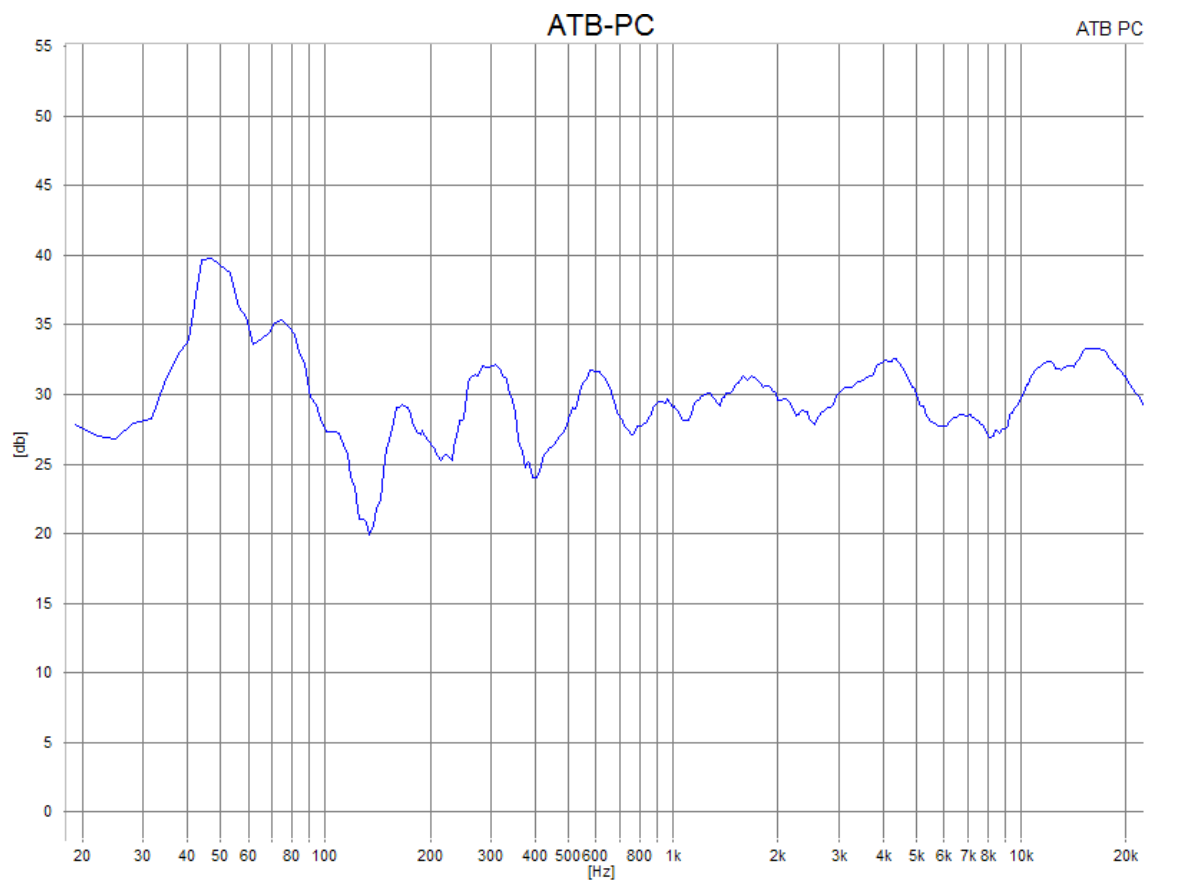
Play DVD track 3 once again (Dolby Digital left hand front loudspeaker, Menu-single signals) and activate the button “mean measure”.



When using the CD, please use CD track 2. 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 is overriding (set to a too sensitive position).

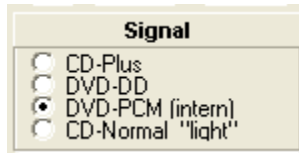


The frequency response curvature of the left hand loudspeaker should appear on the screen.

## 6. MEASUREMENT SIGNAL

### 6.1 SETTING

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



There are 2 signals on the CD and 2 signals on the DVD available for measurement purposes.

The signals CDplus and CDnormal are to be found on the CD.

CDplus is the usual test signal used in most measurement processes. It corresponds to the DVD signal in relation to its frequency distribution. The signal with its x 4 over sampling is almost analogue. This Signal is also the internal Generator Signal.

The signal CDnormal with its 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 signal with its x 4 over sampling is almost analogue.

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 the signal.

The internal BMH signal is like the PCM signal.

The PCM signal is also used for digital measurements.

#### IMPORTANT!

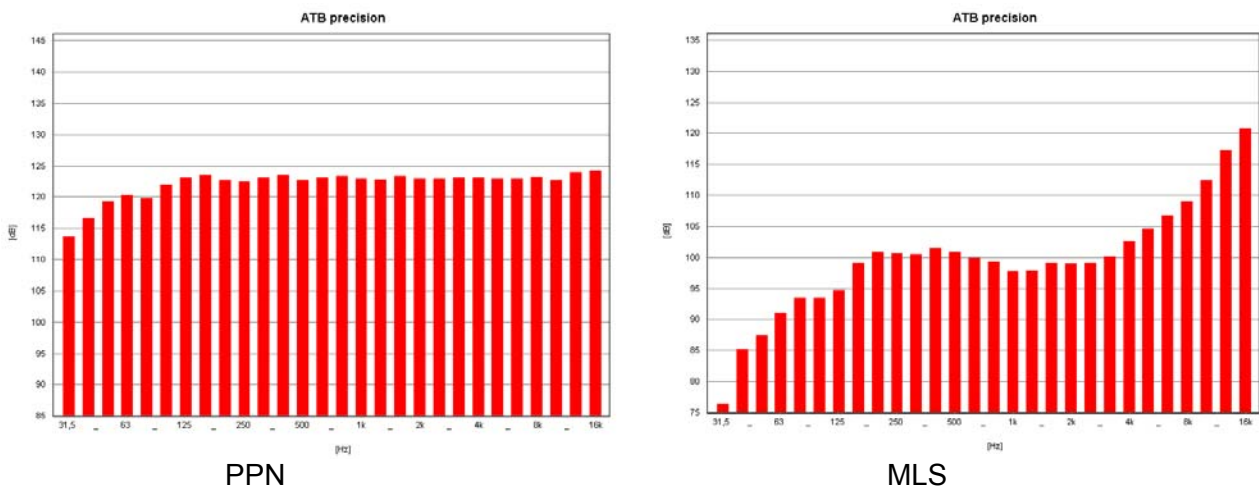
The adjustments and system correction must comply with the test signal.

## 6.2 THE M-PN SIGNAL

For measuring the M-PN (Metzner-Periodic-Noise) signal is used. It is a deterministic noise. Through the function of 4X over sampling it is an analogue 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 loud speaker 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.



The pictures show the analyser measurements of the PPN and MLS signals. Both signals are a deterministic noise (a frequency mix). The PPN 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  $-45\text{dB}$ . 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 same signal will be recorded by the microphone used. 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 PPN 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).

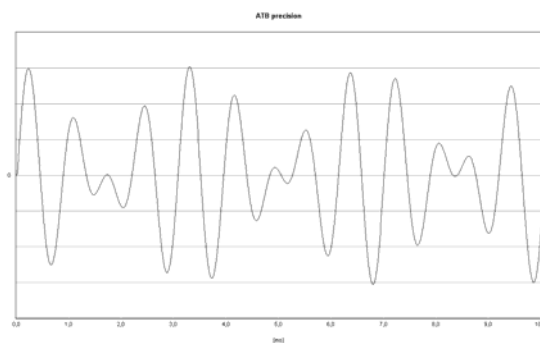
### 6.3 THE SIGNAL 13

The Signal 13 is for the distortion measurement. For the THD measurement is the sine wave signal. The Signal 13 is also for nonlinear distortion measurement.

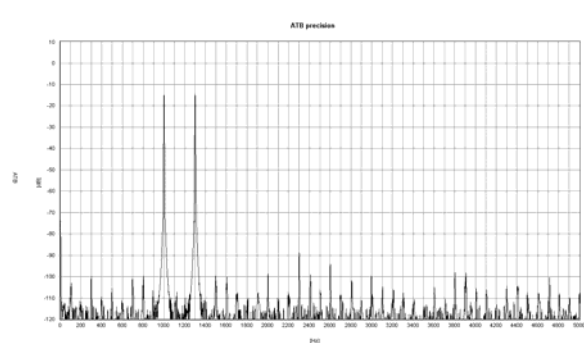
Wolfgang Klippel, Germany, describes the distortion from nonlinear systems.

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. THE MICROPHONE

### 7.1 MICROPHONE ADJUSTMENT

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.

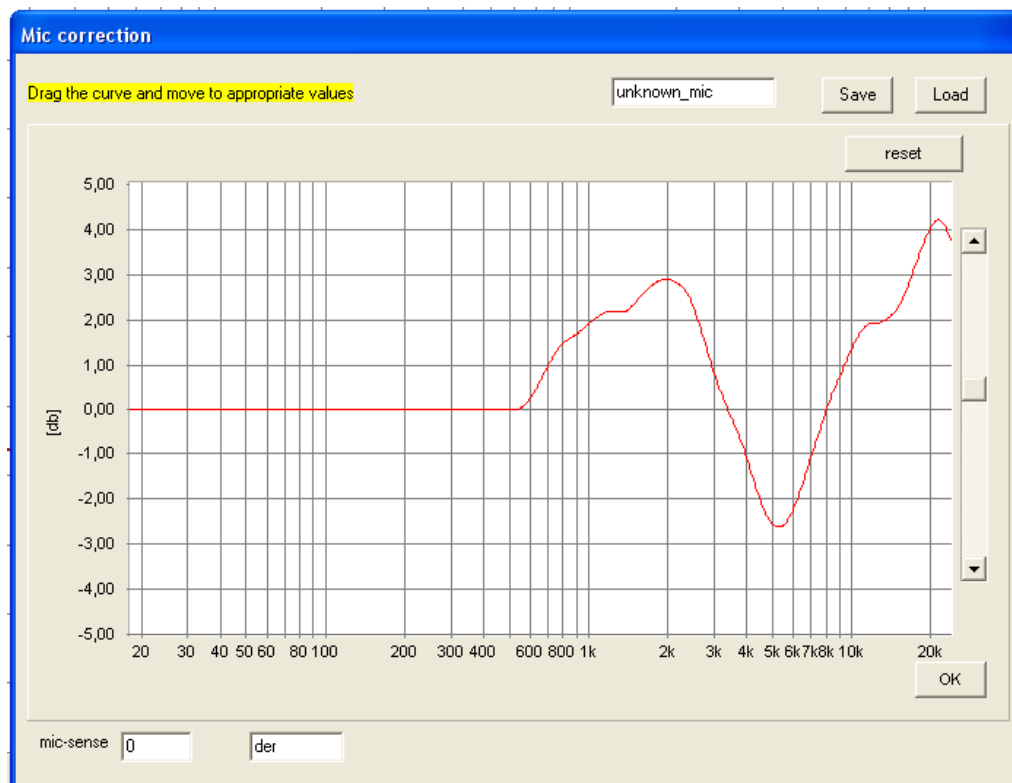
The ATB PC measurement program makes the adjustment of as many microphones as desired possible. This function allows many other adjustments:

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

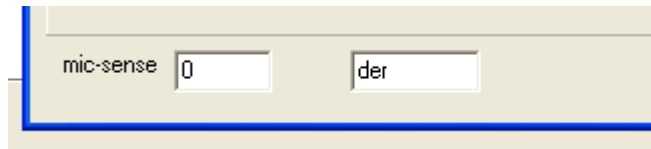


adjusted with MIC-CORRECTION. Thus only the straight frequency characteristic must be taken into consideration. Audible adjustment is the appropriate adoption. Open MIC-CORRECTION in the SETTINGS menu.

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).

## 7.2 MICROPHONE SENSITIVITY

MICROPHONE SENSITIVITY is necessary for SPL measurements. Because of the differing input sensitivity of soundcards the measurement cannot be directly applied. Thus a correction factor is used. The correction factor is to be found 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.



The image shows a software interface with two input fields. The first field is labeled 'mic-sense' and contains the value '0'. The second field is labeled 'der' and contains the value 'der'. The interface has a light beige background with a blue border.

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

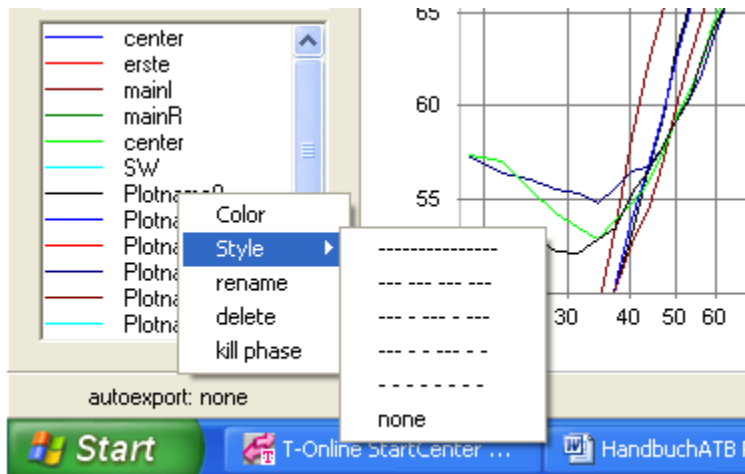
### Measuring the correction factor

The value for the correction factor is to be ascertained with instrumentation. The microphone controller of the soundcard is to be put to full volume and a loudspeaker, whose SPL curve has been measured by the manufacturer, is placed 1 metre away. The input voltage of the speaker should be 2.8V for a 8Ω or 2V for a 4Ω speaker. After the measure the SPL values are compared for a part being well balanced and in both curves identical. The difference in dB between both curves is noted in MIC-SENS. After testing both curves should have the same dB values.

## 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.



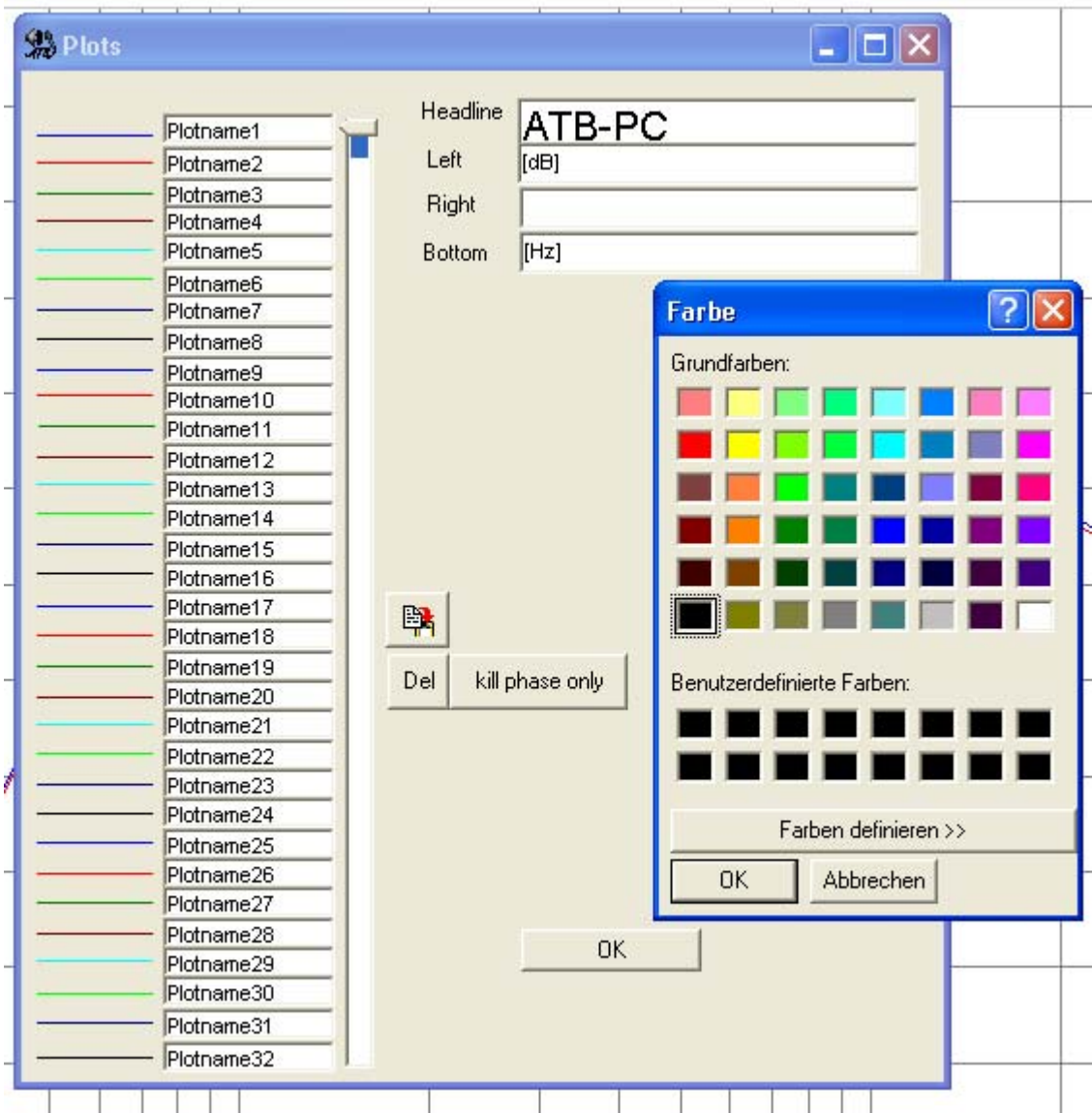
With a right mouse click at the plot name there is to activate the menu.  
The menu performs following tasks:

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 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:

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:

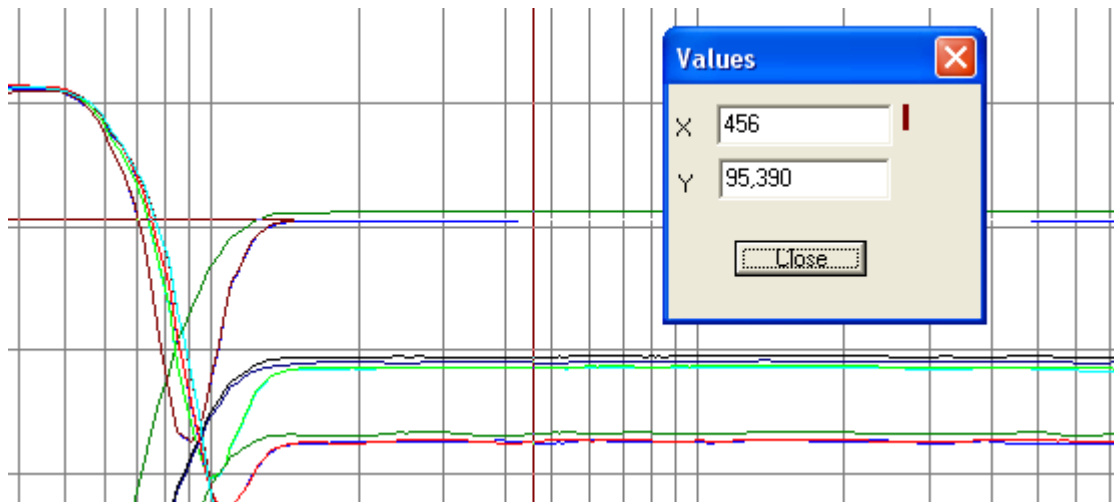
Actuation of the DEL button erases the graph.

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:

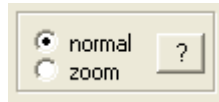
X = Frequency

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 optitest basic 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

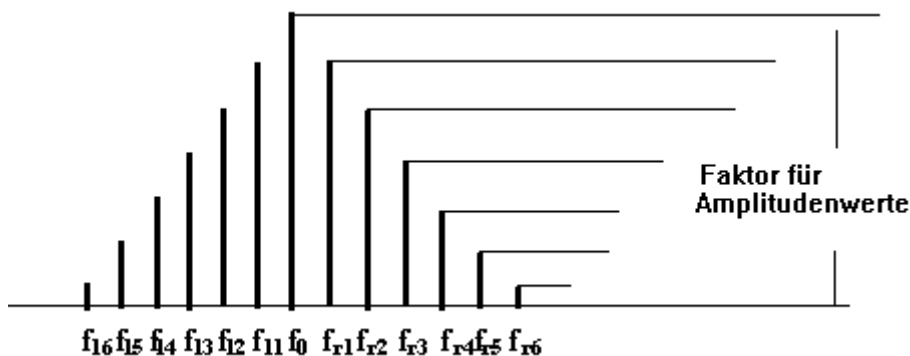
The parameter SMOOTH is there to smooth out a line described by a plot curve.

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

The smoothing option compares with a digital filter



The amplitude values to the left and to the right of  $f_0$  are multiplied by a factor in compliance with the illustration and added, including  $f_0$ . The result will be divided and indicated for  $f_0$ .

The advantage of this smoothing procedure lies in the fact that the graph characteristics are maintained.

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

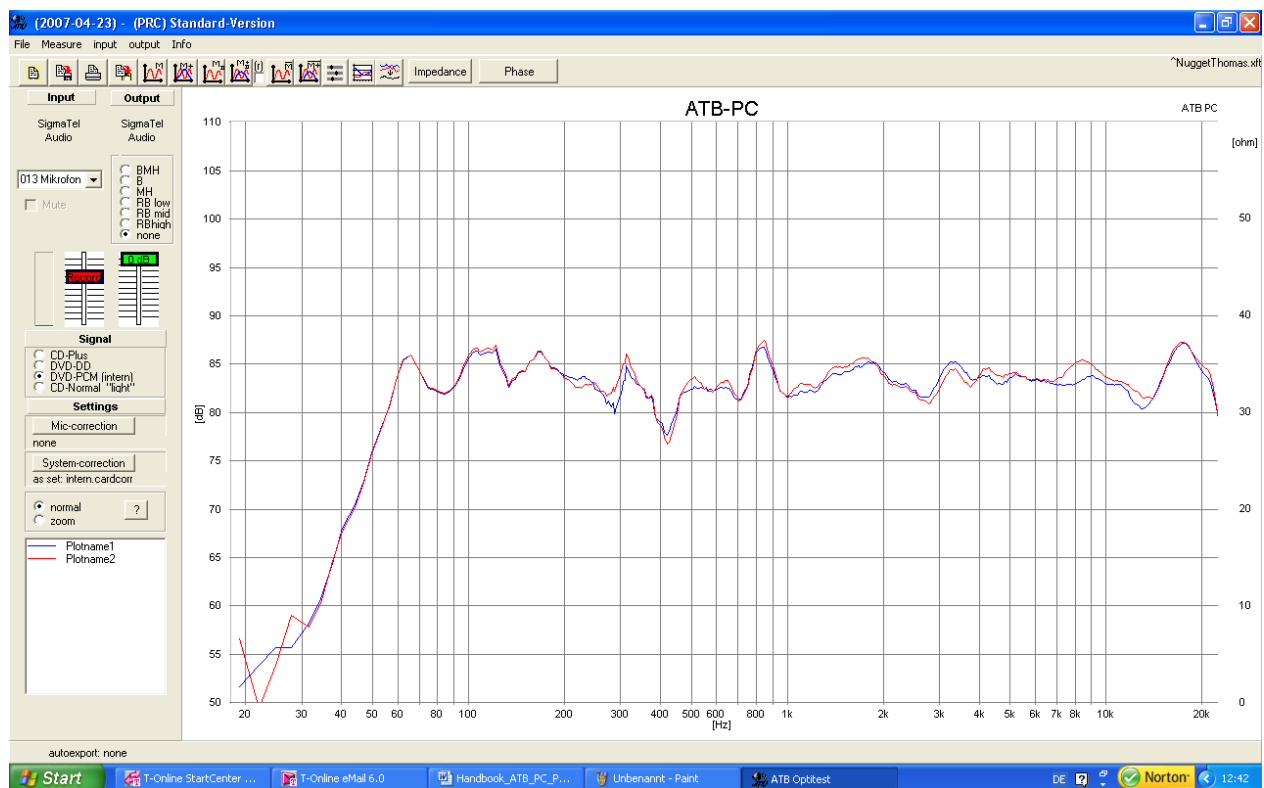
### 10.1 STARTING ATB PC Pro

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 with a double click on the program symbol:



The ATB user interface for Windows:

Program operation complies with the Windows standard operating method.





## 10.2 USER DEFINED MEASUREMENT PROGRAM

When leaving the ATB precision program the actual adjustments are saved. When reopening the program the saved adjustments will automatically be loaded, the last measurement carried out to be started immediately. Data is saved in the INI file - ATB.INI.

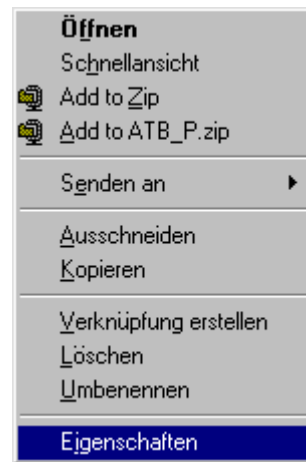
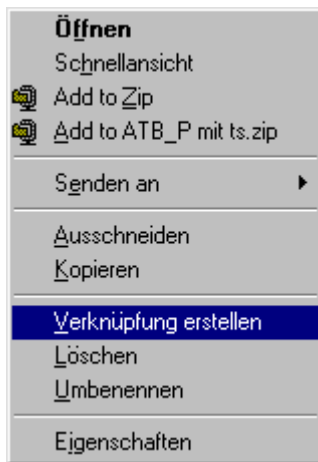
Because more than one user often operate with the same system or if one user has a lot of different operating sequences with corresponding parametric adjustments, it is possible to keep user defined parametric records (Basic adjustments) and load them when required.

This is activated with a program link to a \*.INI file. The INI file data is saved when leaving the program.

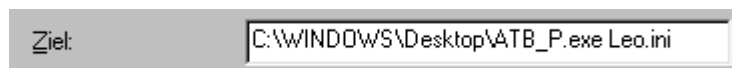
Creating a test program with a user defined \*.INI file:



After a right mouse click on the ATB program icon the following menu will appear, where CREATE LINK should be chosen (left mouse click).



The Icon „Link with ATB\_P, should appear, a right click on the icon opens the next menu where you should left click on the option “PROPERTIES”. The menu PROPERTIES OF LINK WITH ATB\_P in LINK under TARGET (GOAL) first press „space” on your keyboard and choose a name extension for the file e.g. for user Leo the name LEO.INI.

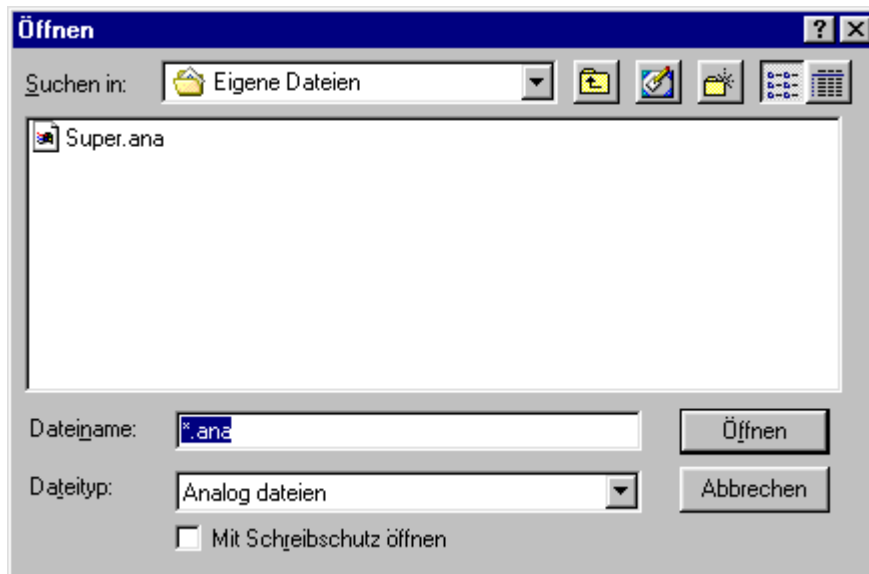


## 11. OPENING DATA FILES

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



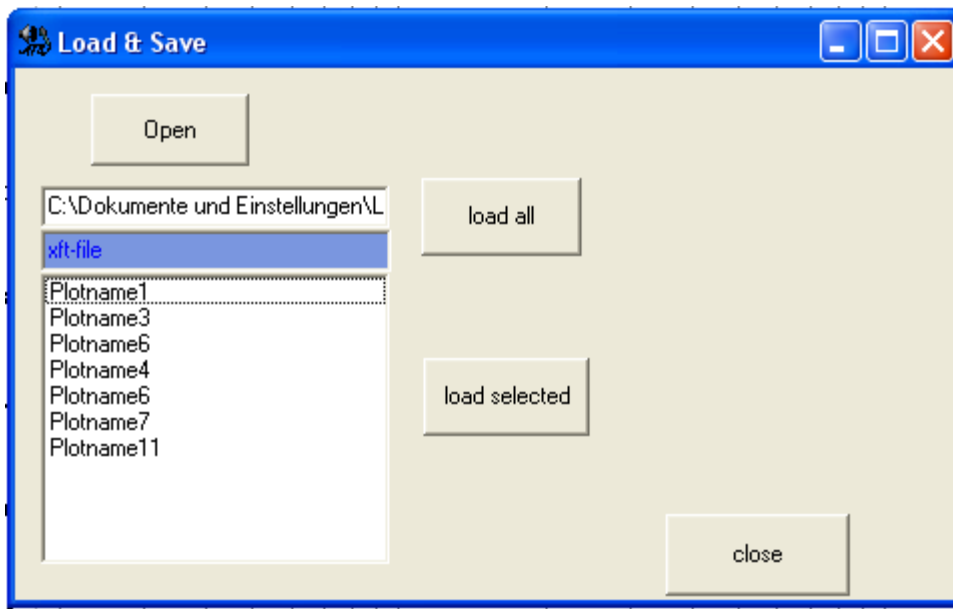
The following pop up window will appear:



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 DATA 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.

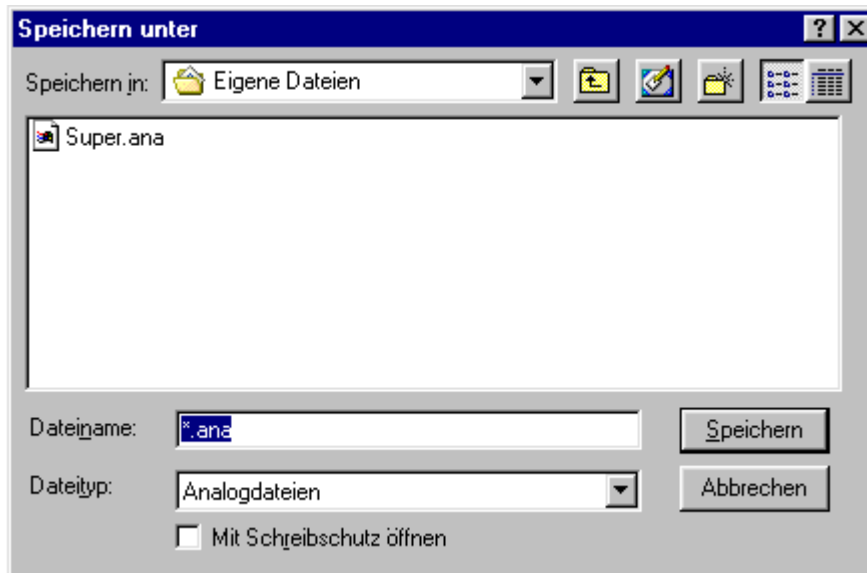
DVD Summe+BassS\W\no.xft

## 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:



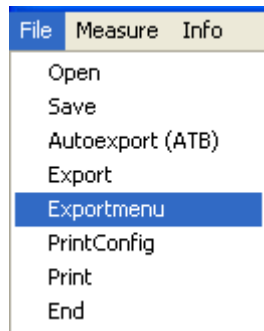
The name of the data to be saved should be written in the box data name. It is suggested that the name extension .ana be used. Im Feld SPEICHERN IN wird Laufwerk und Ordner gewählt, in dem Ihr Dokument gespeichert werden soll. The pop up window is closed by clicking SAVE.

### 13. EXPORTING DATA

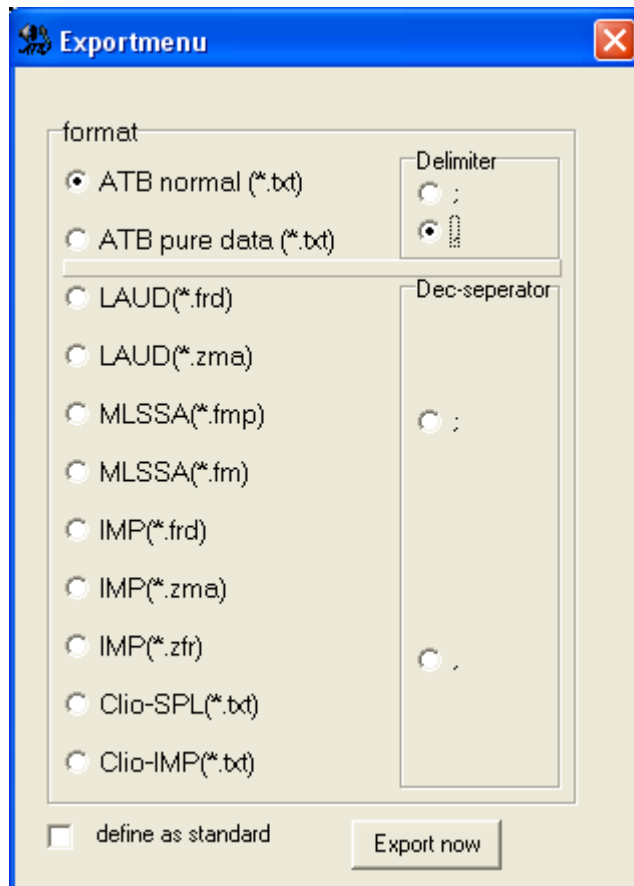
The ATB program has many export formats so that data can be exported to other programs. It is an ASCII text format. This allows the data to be read by many programs.

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



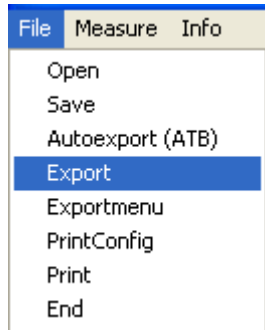
In this menu there is to set the format.



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.

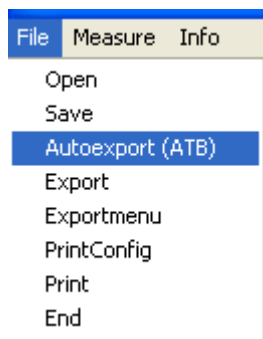
After measuring the menu PLOTS is opened. Pick out the plots to be saved with the pointer. After that press the clipboard button:



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.



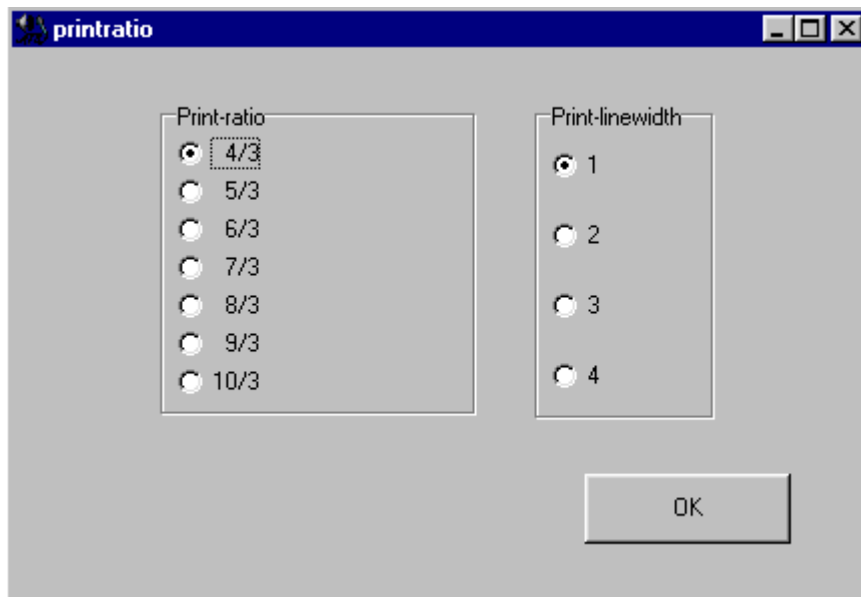
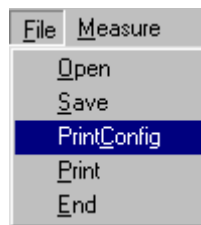
In the context menu the target path can be chosen. The separator/slash can be selected using the EXPORTMENU.

## 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

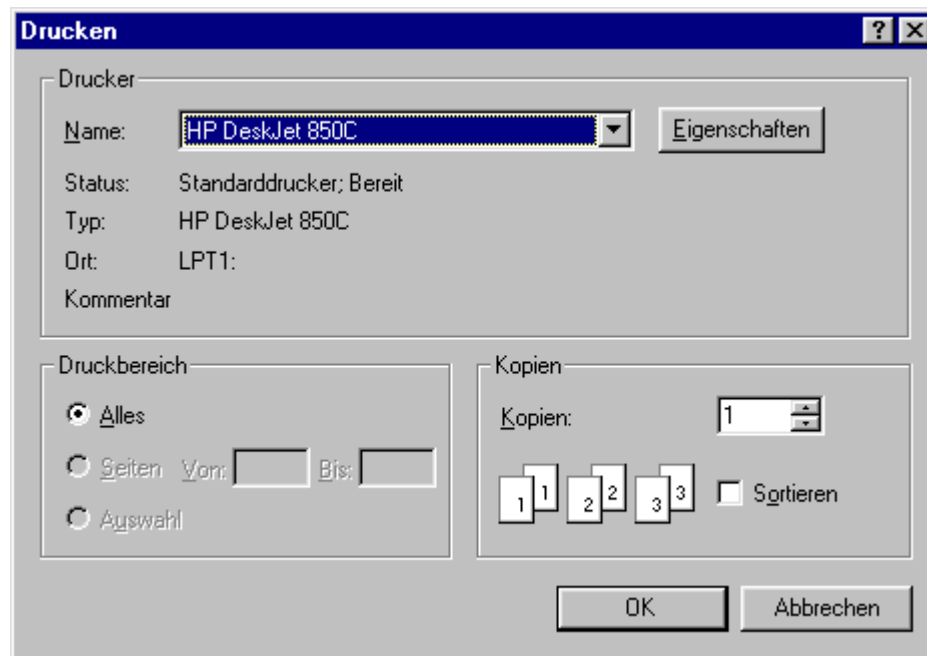
By opening printconfig in the menu FILE you can choose the printratio (length/height) and the print line width



## 14.2 PRINT START

The button  prints the whole document.

If PRINT is activated in the menu FILE the following 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.

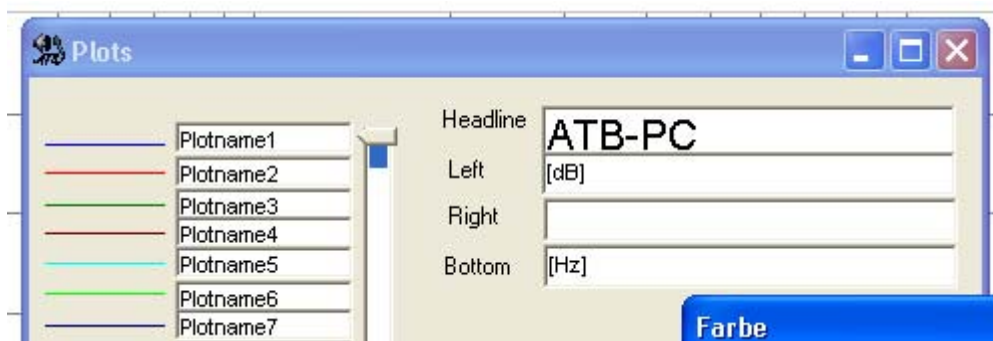
## 16. DIAGRAM

### 16.1 DIAGRAM INSCRIPTION

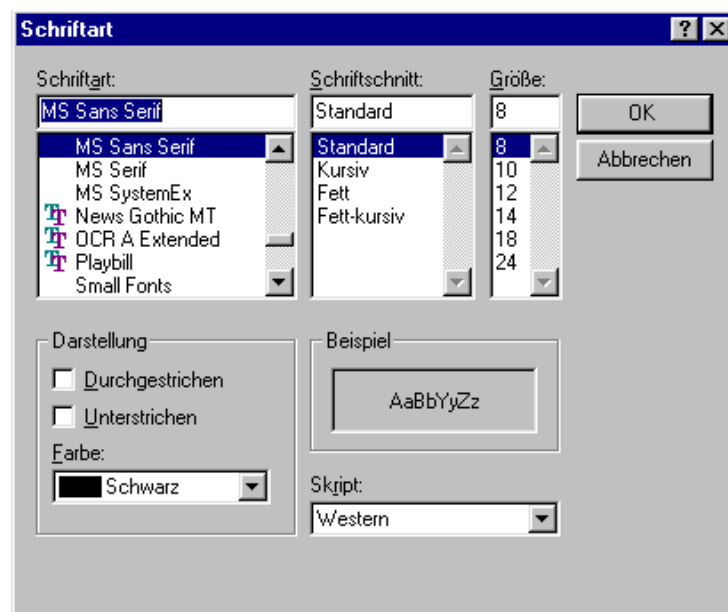
The diagram can be marked in the window PLOTS. PLOTS is opened with



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) in question. 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

The screenshot shows a dialog box titled "PARAMETER" with a close button (X) in the top right corner. The dialog is divided into three main sections. The top section, labeled "Diagramm", contains a "Scale" field with the value "60" and the unit "dB", and a "large imp-scale" checkbox which is currently unchecked. The middle section, labeled "Smoother", contains a "smoother (octave)" field with a list of radio button options: "-", "1/24", "1/12" (which is selected), "1/6", "1/3", "1/2", and "1". The bottom section, labeled "Mic", contains a field with the value "50" and the unit "dB". At the very bottom of the dialog is an "OK" button.

#### SMOOTHING

Smoothing radius in octaves, 1/24, 1/12, 1/6, 1/3 = Terz, 1/2, 1

#### SCALE (RESOLUTION)

Field of scale from 5 to 60dB

#### MIC

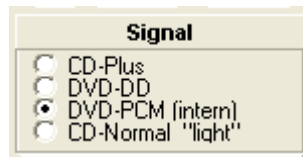
In this window there is to set the lowest dB level for the dB scale.

#### 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

## 17.2 SIGNAL

The program has the capability to task various assignments.



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

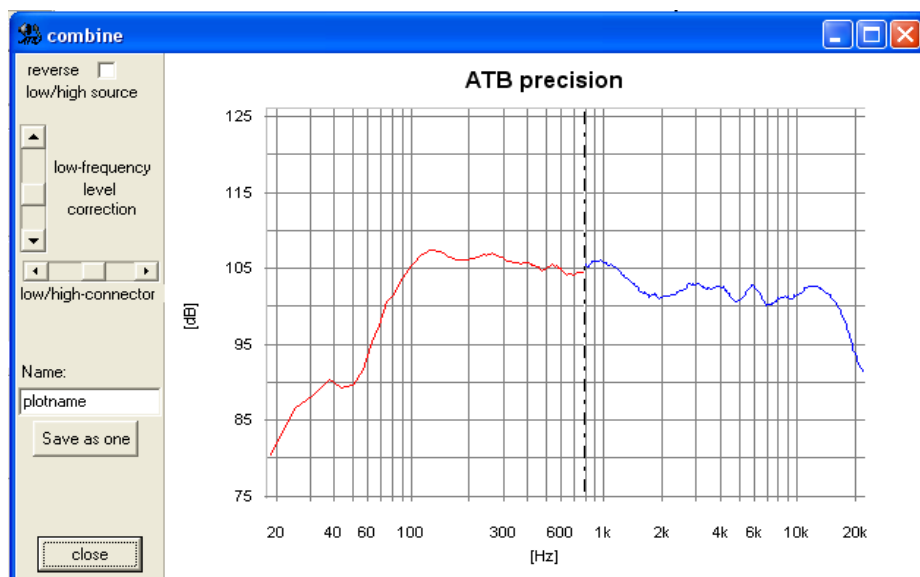


testing is started and stopped. After a test new starts it is possible keep the first plot (to allow comparison) or to allow it to be deleted. This is done by marking the box next to the M+D button.

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.

## 17.5 COMBINE

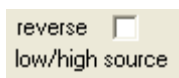
The Optitest has the special function COMBINE



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



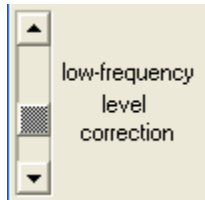
can the distribution of near field and distance (SPL) plots be changed.

The next step is to set the cut frequency between distance (SPL) and near field plots with the



low/high connector.

Low/high-connector 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 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 amplifiers output-voltage is set to 2,8V for 8 $\Omega$  loudspeakers and 2V for 4 $\Omega$  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 new correlation type. The test signal can come from either a CD or DVD or from the PC.

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. AMPLITUDE-FREQUENCY RESPONSE

Amplitude-frequency response is the difference between the electronic measurements of the LINE-IN and the acoustic measurements, SPL, of the MIC input.

### 18.1 LINE-IN

Using this measurement the frequency response of amplifiers, electronic switches and the equalizers of CD and DVD players can be tested. Depending on the soundcard it is also for the impedance measurement. For this test the device's audio output is connected to the LINE IN of the soundcard.

System-correction for LINE IN soundcard:

The left AUDIO OUT channel of the CD/DVD player is connected to Testbox. The jack plug is connected to LINE IN of the soundcard. The switch position of the Testbox is 1 and impedance. There should be no devices connected to the banana jacks. The soundcard should be set to RECORD or WAVE IN and LINE IN. The signal to be used is CD plus or, for studio purposes, PCM.

After starting track 1 on the CD player or with left loudspeaker in the DVD player's menu, SYSTEM CORRECTION is to be activated in the program menu. An impedance measurement cannot be carried out when using LINE IN therefore only one measurement is necessary. This measurement is carried out with the lower start button. A red curvature is rendered. The soundcard modulation, top left in the program, should be green and be in within  $\frac{3}{4}$  of the indicating range. If this is not so then the level controller is to be adjusted and the correction to be repeated. The measurement is to be saved with the name LINE and is always to be used when measuring with LINE IN.

The device to be tested is to be connected to the left AUDIO OUT channel of the CD/DVD player.

Studio devices can also be digitally connected using the DIGITAL OUT socket.

The LINE IN socket can also be used for SPL measurements when using microphones with pre-amplifiers.

### 18.2 MIC-IN

SPL frequency response measurements are carried out in conjunction with the MICROPHONE IN socket. Depending of the soundcard it is also for the impedance measurement. SYSTEM CORRECTION is carried out in correspondence with the correction.

### 18.3 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.

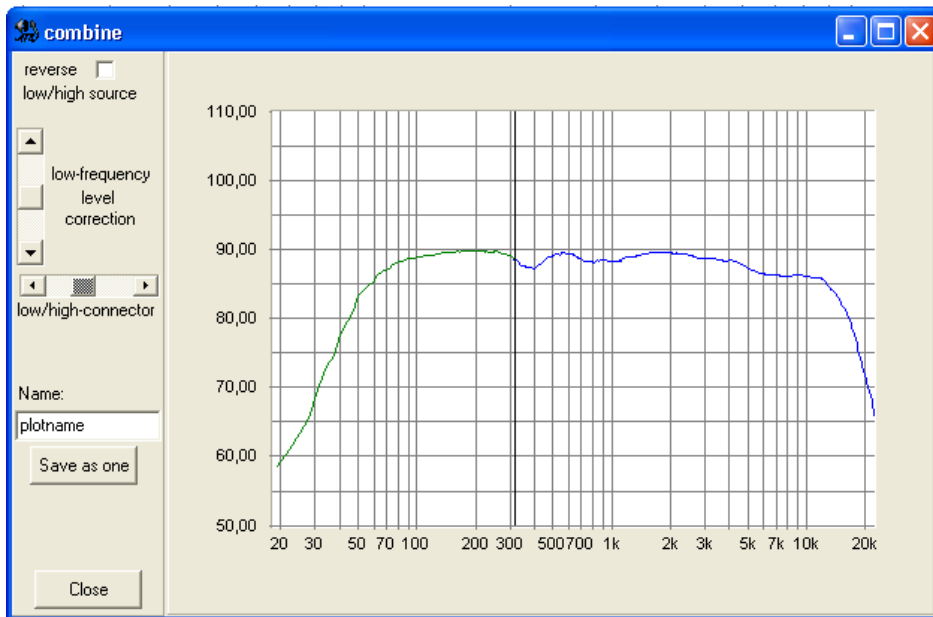


Fig. 18.1

Fig. 18.1 shows 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:

$$\text{Pressure} \sim \text{Power} \times \text{Area}$$

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 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.



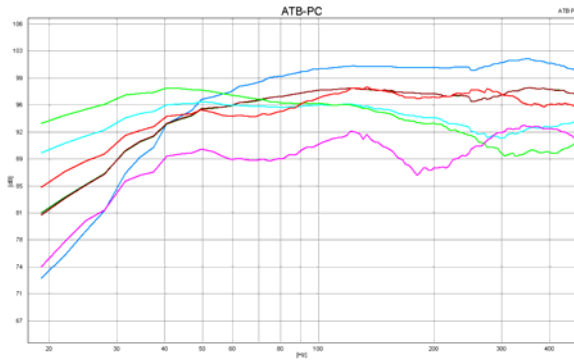


Fig.18.2

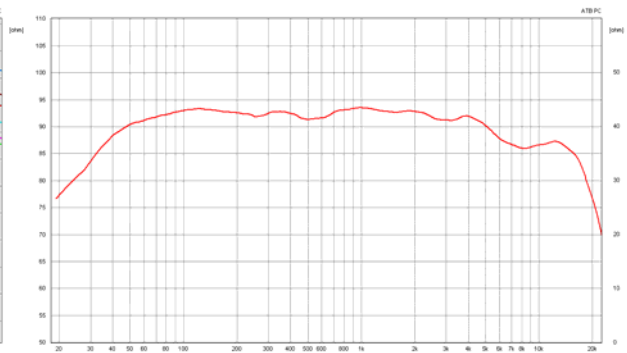


Fig. 18.3

Fig. 18.2 shows the measurement of the bass reflex opening at different distances. The follow up of distance to the rim of the basreflex 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. 18.3 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.

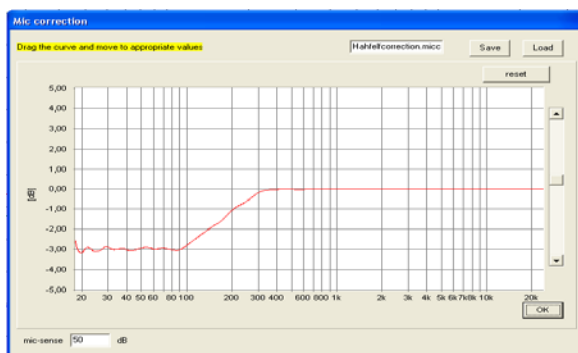


Fig.18.4

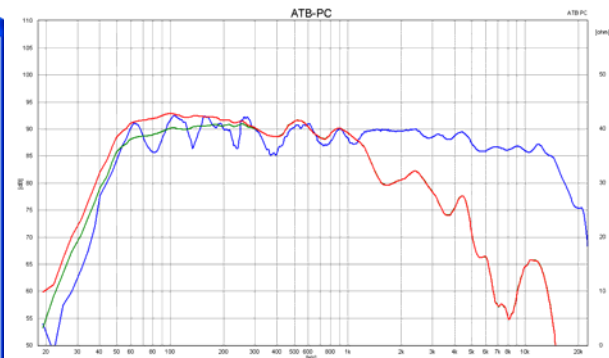
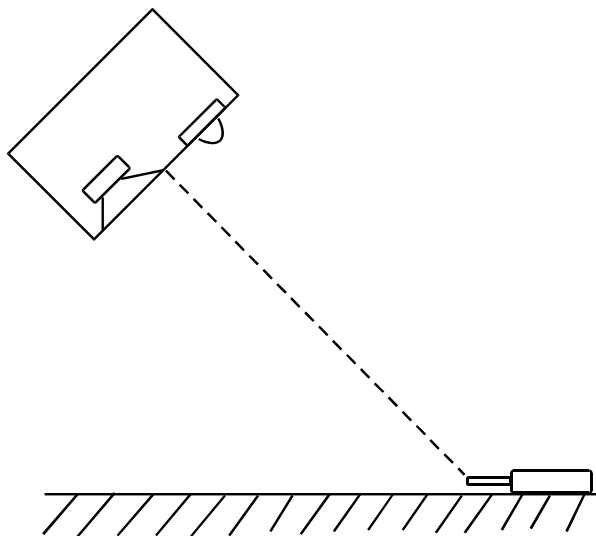


Fig. 18.5

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. 18.4 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.18.5, 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.

## 18.4 THE BORDER AREA MEASUREMENT

By the border area measurement the measurement microphone is operated as border 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 border 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.

## 19. IMPEDANCE MEASUREMENT

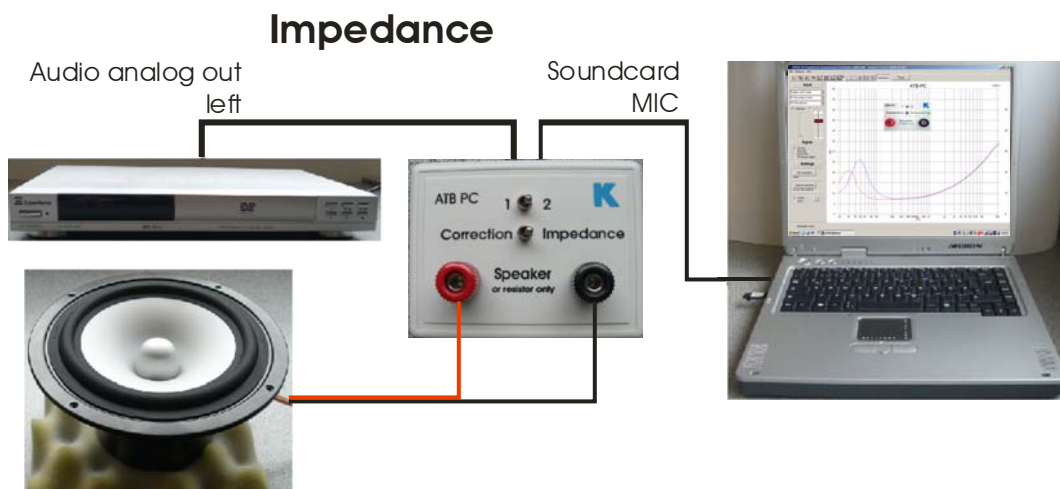
### 19.1 THEORY

Impedance is the frequency dependent resistance of alternative currents. It consists of the value  $Z$  in  $\Omega$  and the phase  $\varphi = ^\circ$ . The measurement displays both values above the frequency range. The phase  $\varphi$  displays the values of reactive and active resistance which together result in the value  $Z$ .  $\varphi = 0^\circ$  consists of resistance only and  $\varphi = 90^\circ$  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  $\varphi$  reveals possible instability in amplifiers, which can create oscillation and damage loudspeakers. The phase angle should not be  $> +60^\circ$ .

The impedance measurement enables the testing of electronic devices, resistors, capacitors and spools.

### 19.2 ARRANGEMENT



For the measurement with internal generator the test box is connected to the LINE OUT of the soundcard.

### 19.3 MEASURING

The impedance measurement can use the LINE or MIC input of the soundcard.

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  $\Omega$ .

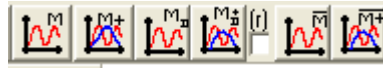
The change over between amplitude and phase measurements is initiated with the button



after changing over the usual buttons are available. The constant measurement using averages is not necessary for impedance measurements and is deactivated. When carrying out an impedance measurement the Testbox is connected to the left AUDIO OUT analogue socket of the CD/DVD player. The jack plug is inserted into the soundcard's MICROPHONE IN socket. The switch is set to 1 or 2 and impedance. When working with a CD player, track 1 on the auto test CD is selected. When using a DVD player the option left loudspeaker is selected on the surround test DVD's menu. The loudspeaker is connected to the sockets on the Testbox.

Important:

The loudspeaker is to be connected directly to the Testbox without being connected to an amplifier. An amplifier may not be connected to the sockets on the Testbox. If doing so the soundcard or computer may be seriously damaged.



The measurement is started with:

The result is displayed in  $\Omega$  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.

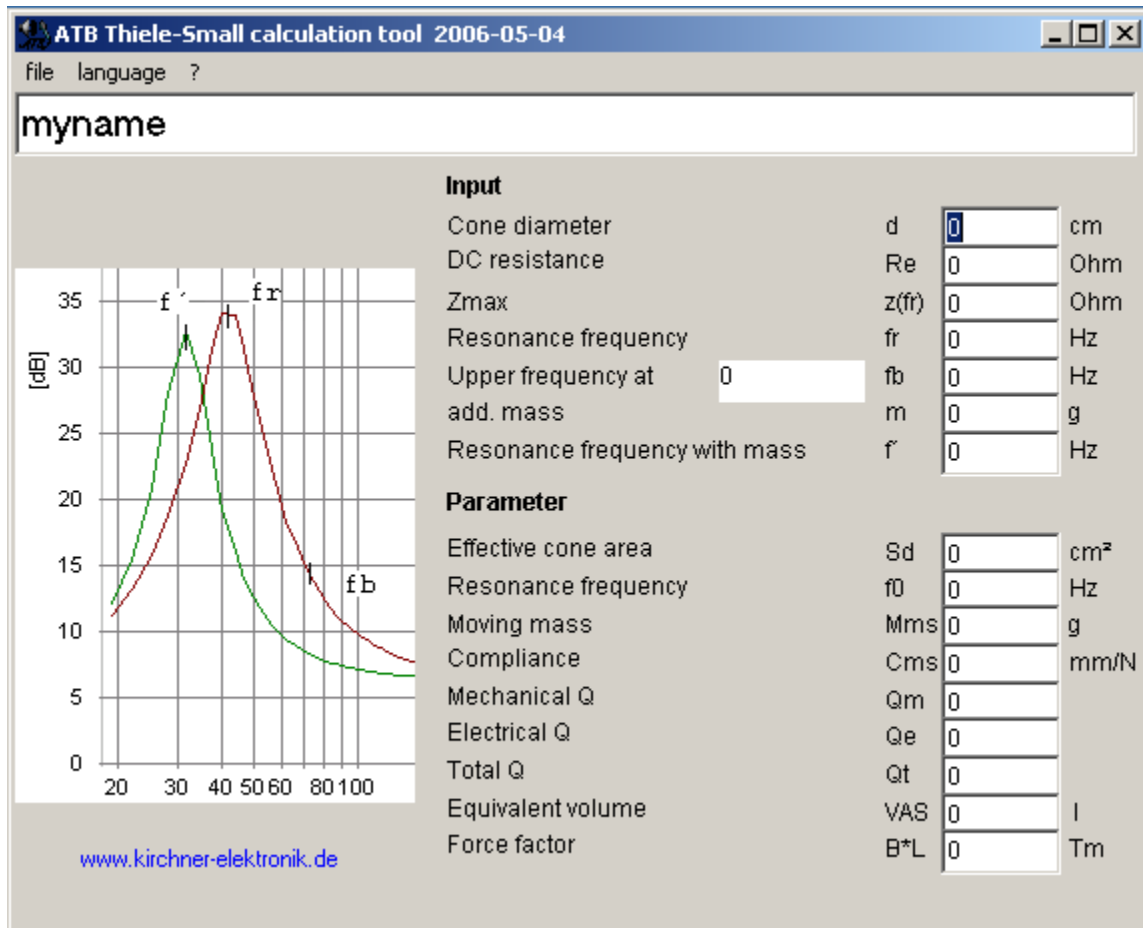
The two buttons at right are for a duration mean measurement. This measurement is important for the impedance because errors will be out of the curve.

## 2.4 THIELE-SMALL PARAMETER

Die Thiele-Small Parameter werden mit dem externen Programm

ATB\_TS\_Tool.exe

Berechnet.



Some speakers have impedance with a value greater than 50Ω. Therefore is in the PARAMETER menu following function

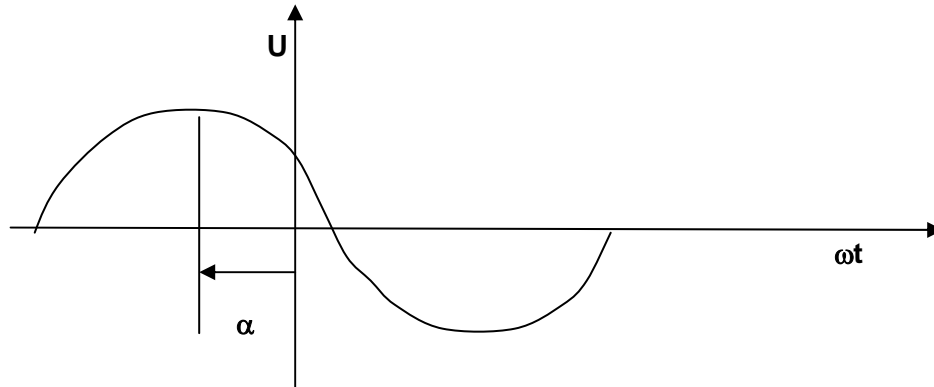
☐ large imp-scale

After activating it, the scale gets the value of 250Ω.

## 20. PHASE MEASUREMENT

### 20.1 BASICS

The phase is used when describing sinus shaped processes such as mechanical oscillation, alternate currents, radio waves and sound waves.



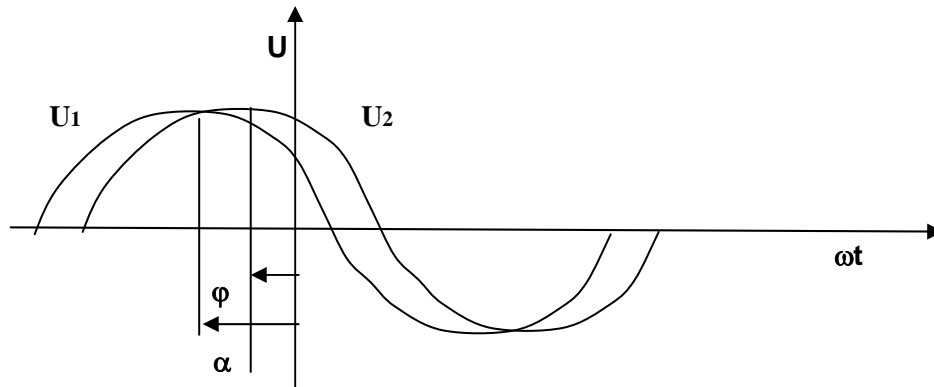
$$U(t) = U_s \cos(\omega t + \alpha)$$

$t$  Time,  $U(t)$  Instantaneous value of voltage,  $U_s$  peak value (max. Amplitude),

$f$  Frequency,  $T = 1/f$  Period duration,  $\omega = 2\pi f = 2\pi/T$  Angular frequency,

$\alpha$  Null phase angle

The phase angle  $\alpha$  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.

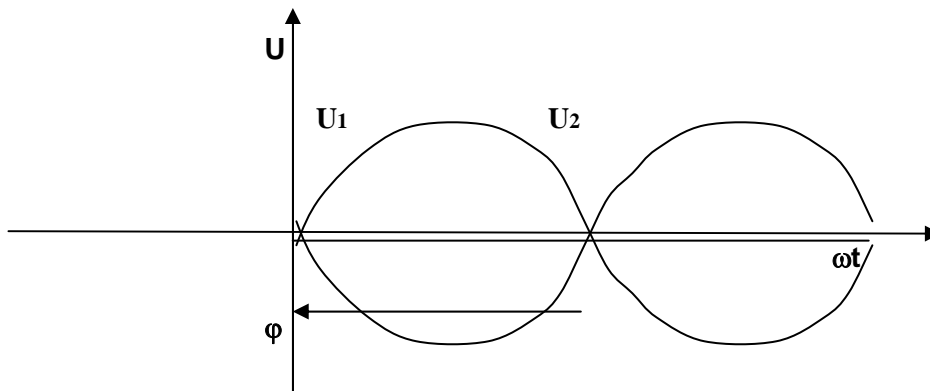


$$U(t) = U_{s1} \cos(\omega t + \alpha) + U_{s2} \cos(\omega t + \alpha + \phi)$$

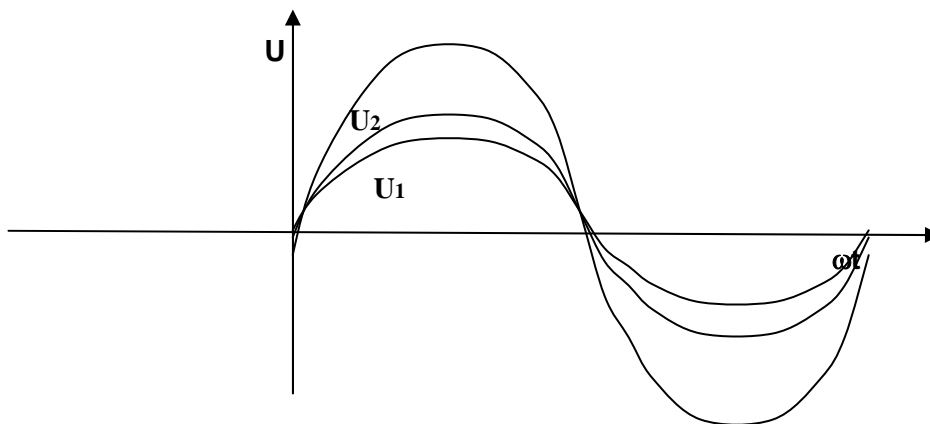
The angle  $\phi$  defines how the oscillations (waves) overlap.

Examples of how the angle  $\varphi$  effects the waves :

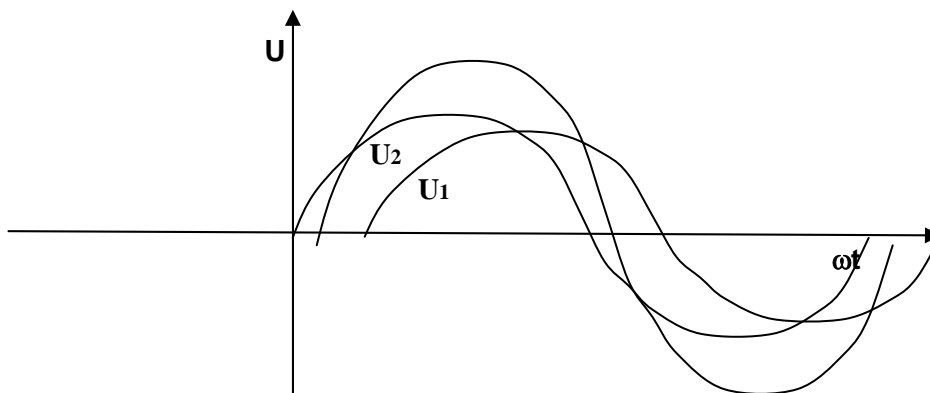
$\varphi = 180^\circ \Rightarrow$  blanked out



$\varphi = 0^\circ, 360^\circ \Rightarrow$  maximum amplitude



$\varphi = 90^\circ, 270^\circ \Rightarrow$  partly amplified, or also partly blanked out



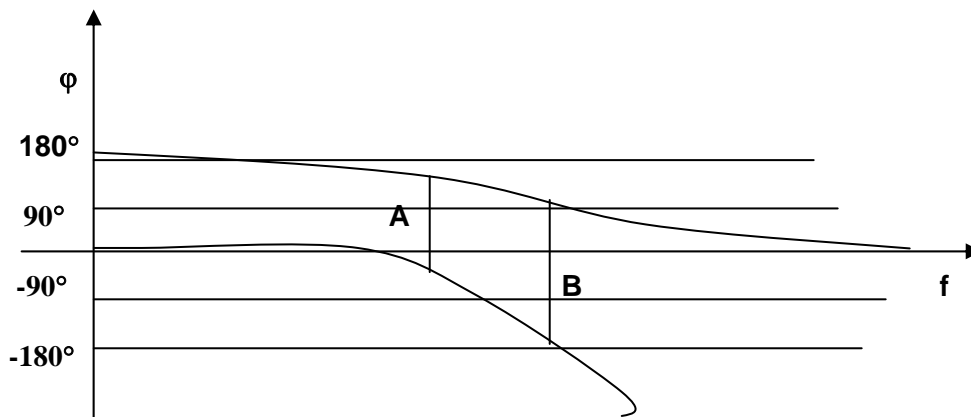
The phase and loudspeakers

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 not 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  $180^\circ$ ,  $360^\circ$  or  $540^\circ$  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  $180^\circ$  using the method described above

B = phase angle of  $270^\circ$ . 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  $90^\circ$ ,  $270^\circ$  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.



## 20.2 ELECTRICAL 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  $> 5\text{kHz}$  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.

## 20.3 ACOUSTICAL PHASE MEASUREMENT

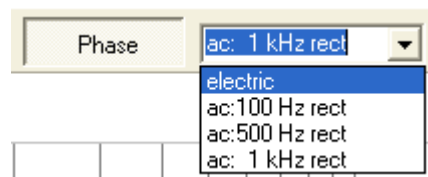
The acoustical phase measurement of a loudspeaker can only be carried out in quiet rooms. This test is more sensitive to disturbances than the frequency response measurement. Strong reflection in the in which the test is being carried out can also influence results. In small rooms the distance should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

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  $180^\circ$  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.

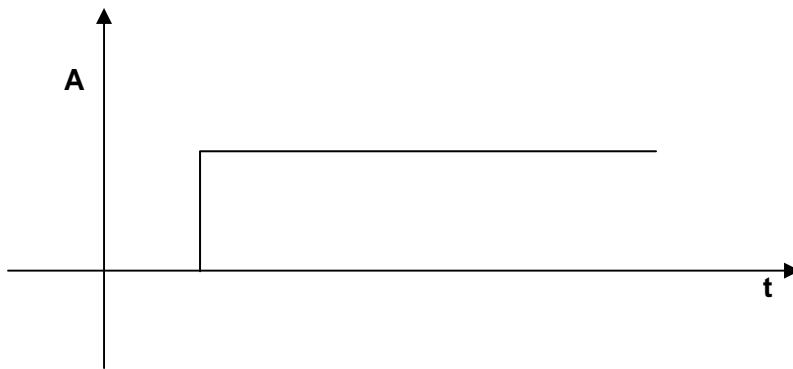
## 21. STEP RESPONSE

### 21.1 THEORY

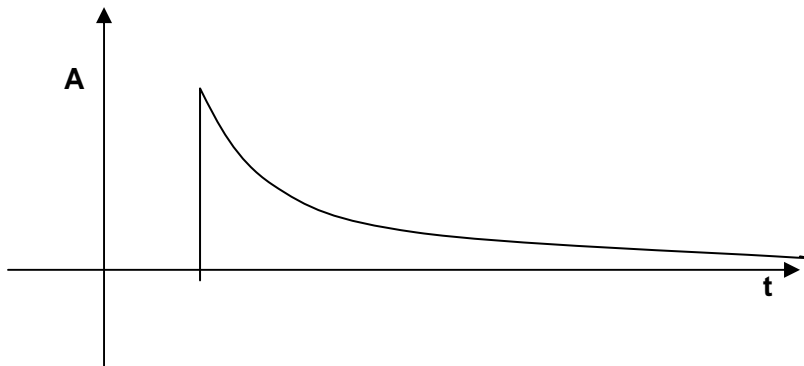
The step response shows the amplitude and the phase simultaneously. It is calculated using both measurements. The first thing done is to calculate the impulse of the sinus oscillation through a complicated addition, with value and phase. Because the impulse is a mathematical factor, ideally the amplitude 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.

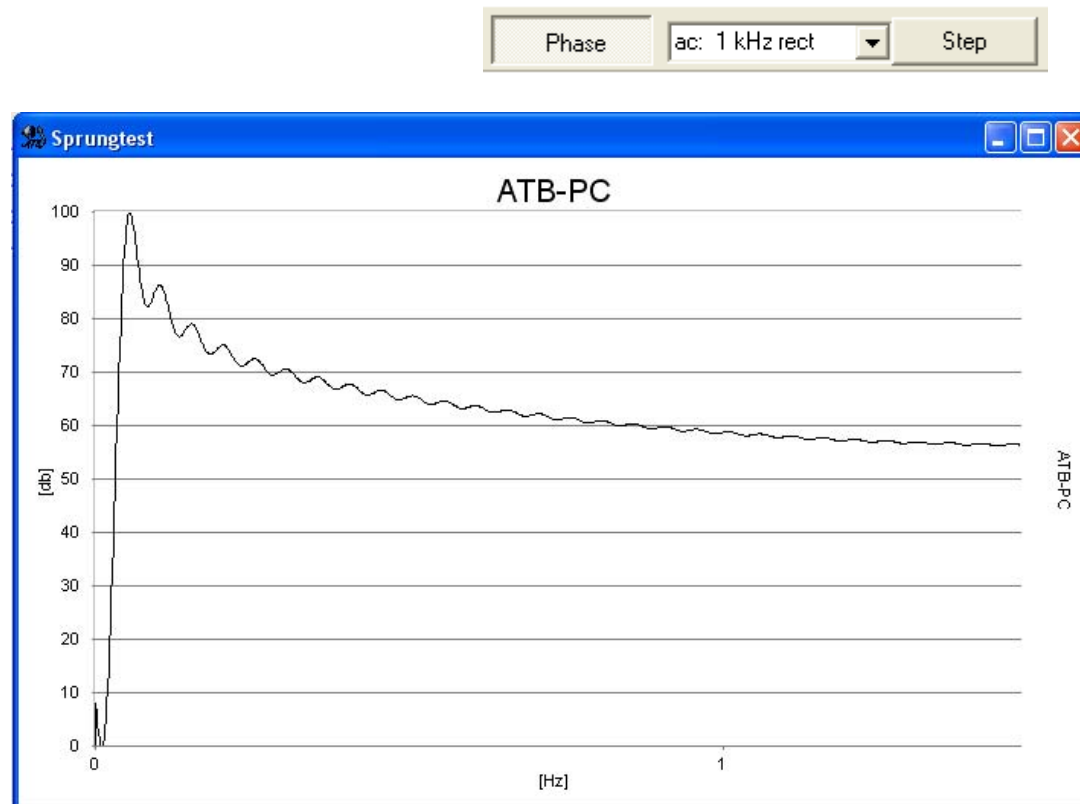
The picture shows an ideal loudspeaker. The steep ascent of the step response displays a high top frequency border. The e – function's corresponding descent shows a linear frequency characteristic. The lower frequency border can be deducted from the steepness of descent. The deciding factor in this measurement 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.

## 21.2 THE MEASUREMENT

The step response is not a measurement on it's own but an evaluation of frequency characteristic and phase.

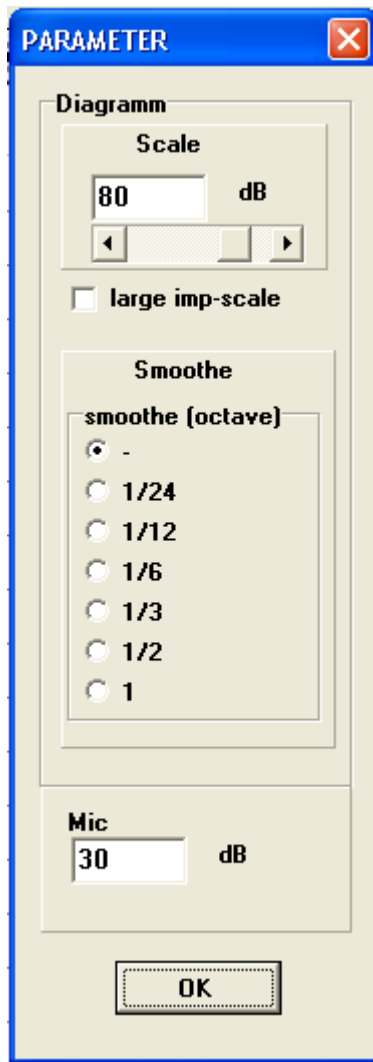
The window for the display is activated with the button:



The display for the lower and top frequencies is adjusted under the curve. The scale for the x – axis is in msec.

## 22. DISTORTION MEASUREMENT

For the distortion measurement there are to set following parameters in the PARAMETER menu.

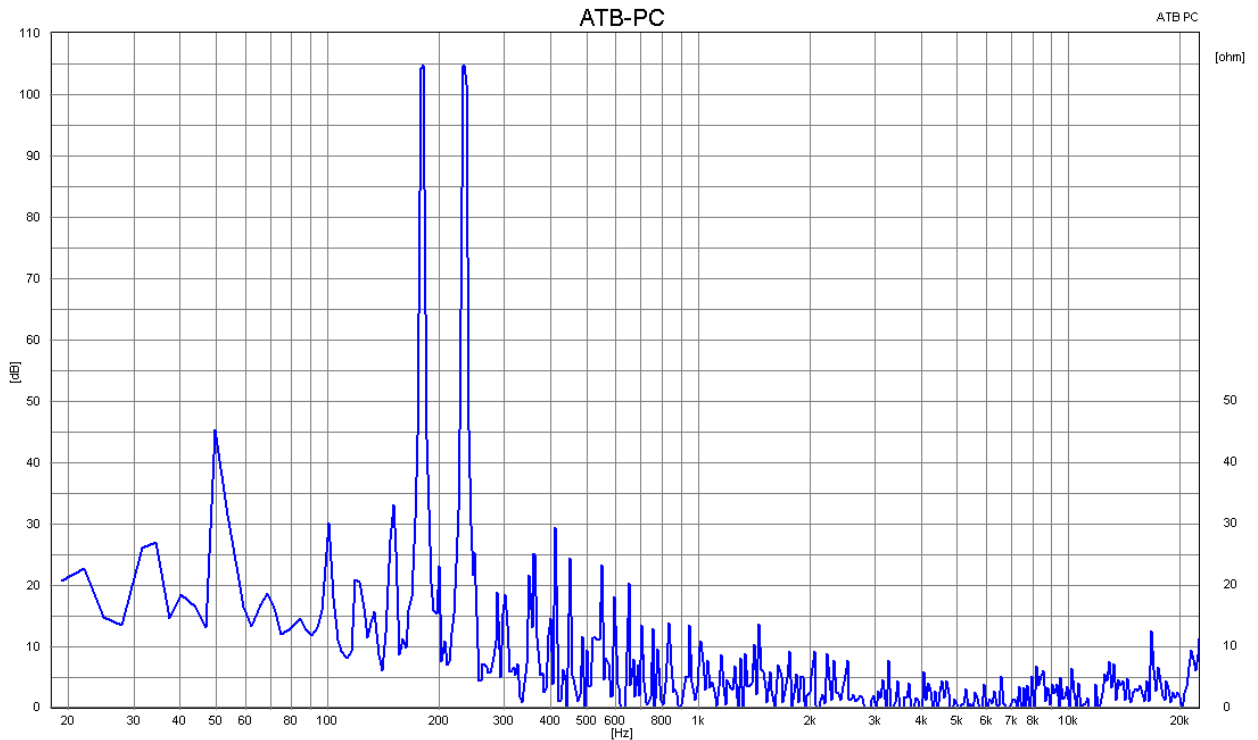


For Scale is to set 80dB, to get the completely dynamic range of the soundcard.

In Smoothe there is to set "-".

For Mic is to set 30dB

The output level attenuator is to set to 0dB. This setting gives the greatest dynamic range.  
The power for the speaker is to control with the volume of the power amplifier.



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%

## 23. MEASURING CAR INTERIORS WITH THE CD

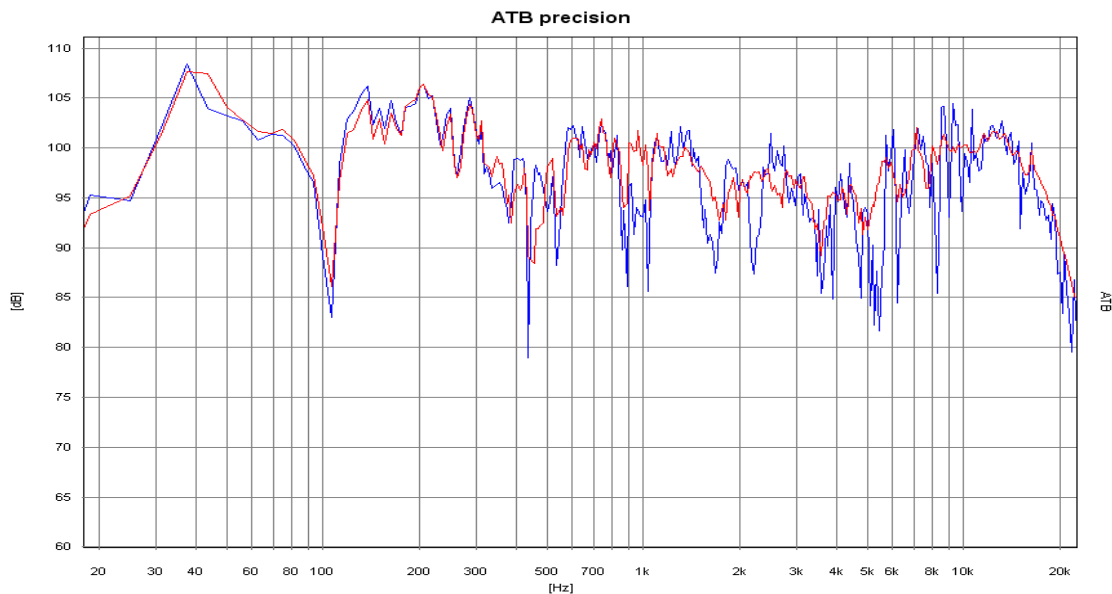
Before testing, the system-correction is to be measured once. The system-correction is for comparison of the soundcard. Thus any CD/DVD Player can be used.

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 is there a need for a new system-correction.

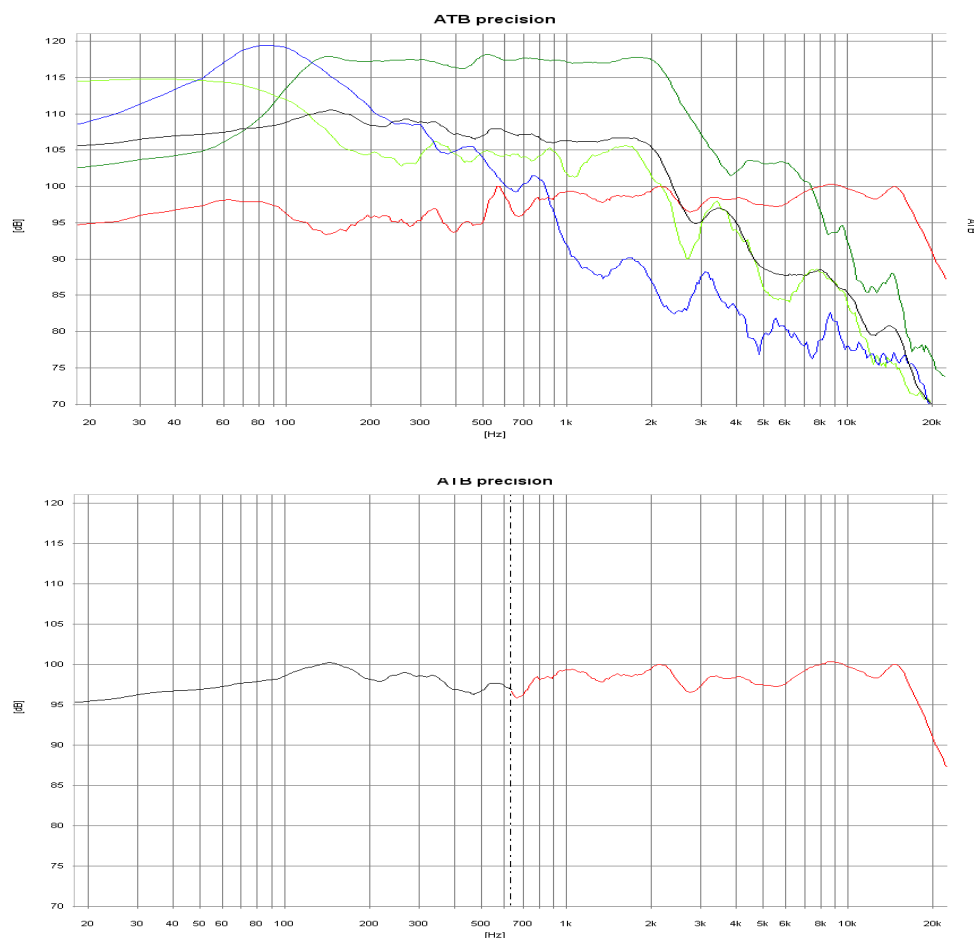
### 23.1 ROOM CORRECTION



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.

## 23.2 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.

## 24. 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 is there a need for a new system-correction.

### 24.1 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 are 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  $1\text{m} = 2.94\text{msec}$ . 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.



## 24.2. MICROPHONE

The microphone is connected to the INPUT slot of the soundcard.

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.

## 24.3 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

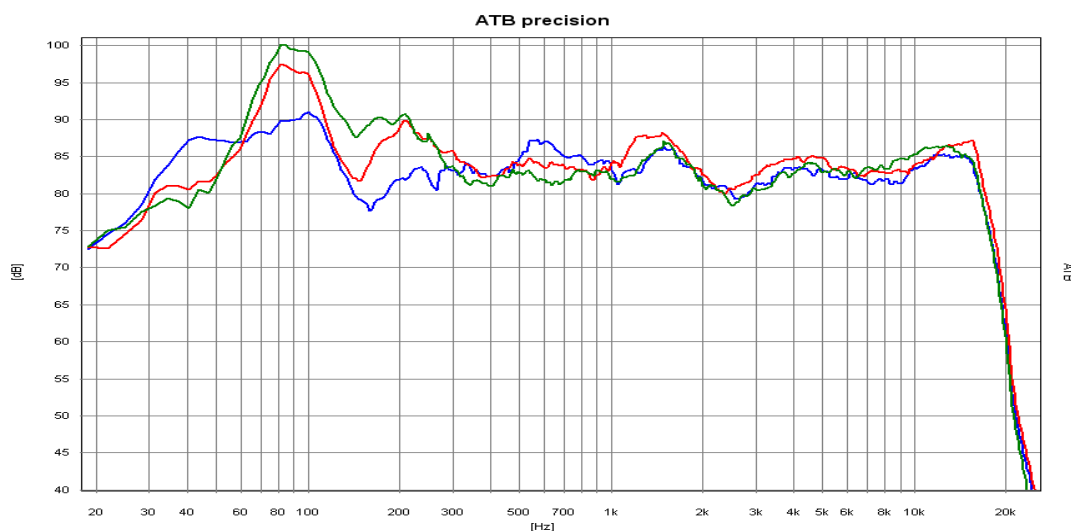
The DVD is started and the single measurements are chosen in the menu.

## 24.4 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.



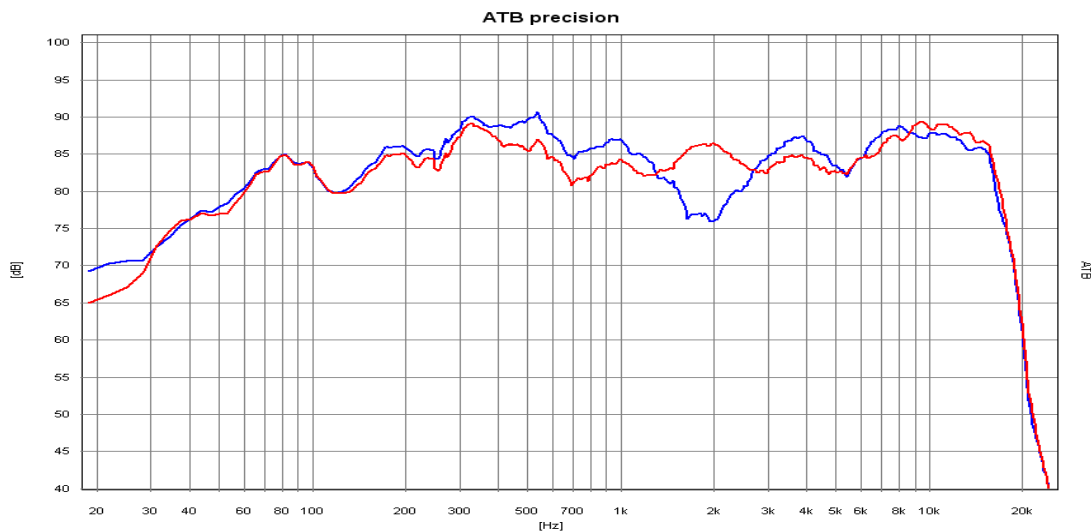
The blue curve is the FL loudspeaker, the red curve is the FR loudspeaker with large and the green curve with small distance from the corner of the room.

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

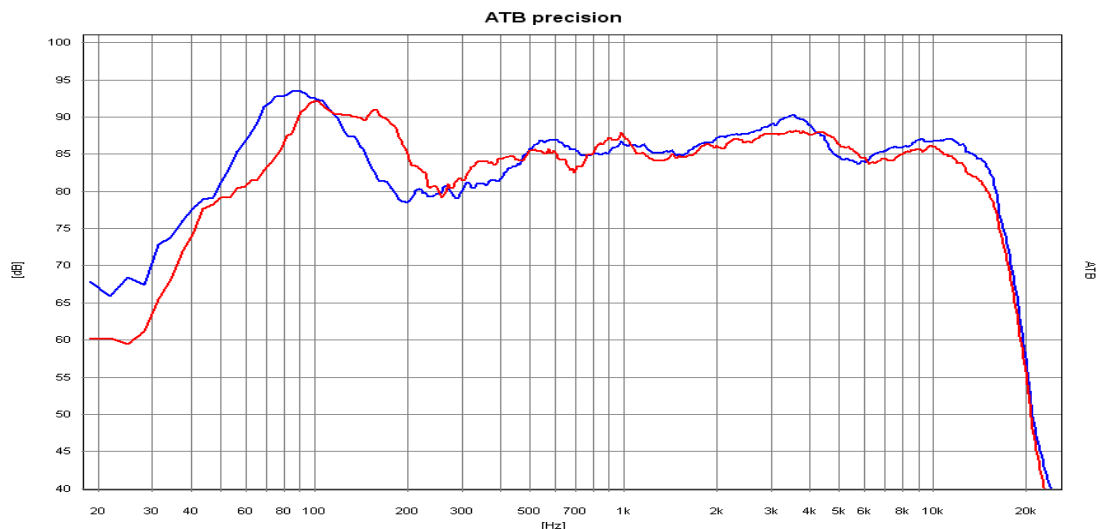
The centre is for voice playback. Therefore the centre has to have a steady frequency characteristic. The high tone playback should not be emphasized, this making the voices sound too hard. The low frequency range may drop below 100 Hz, being transmitted from either the Subwoofer or the FL, FR loudspeakers.



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

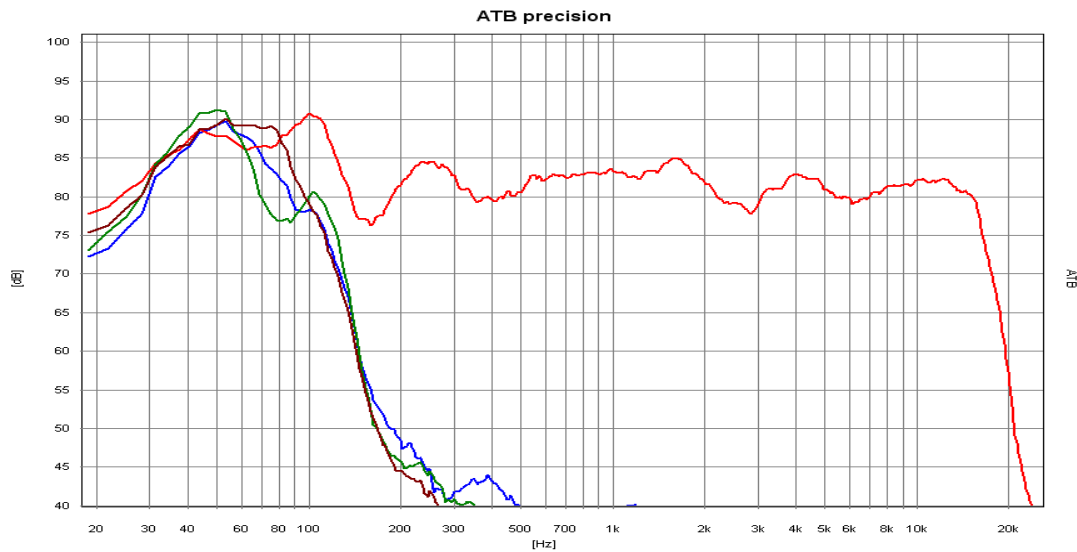
Special rules are needed for the positioning of surround sound loudspeakers. The Home Cinema owner should be aware of the fact that films are not made for the living room but for the cinema. Just as the picture format is not suitable for a television screen so has the sound been mixed for the cinema. 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.



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.

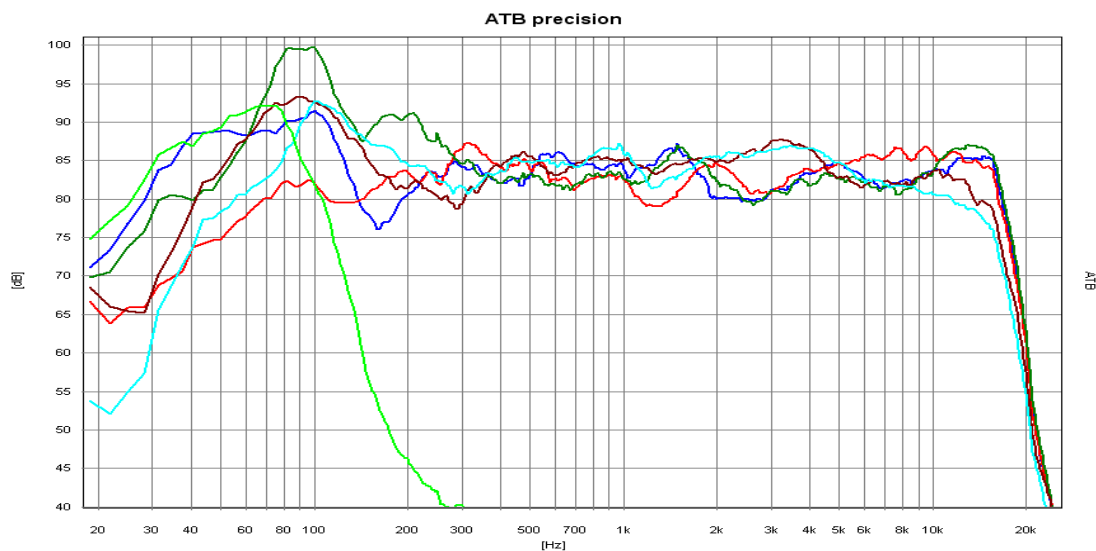


The picture shows plots for the SW in different positions. The difference to each position is 20cm. The red curve is the FL loudspeaker.

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.



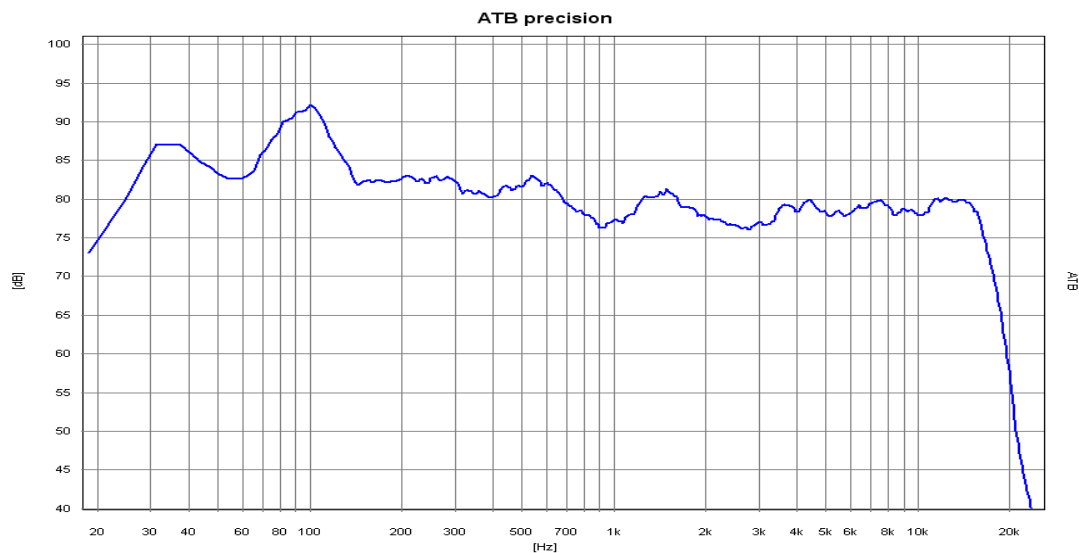
The single measurements of the loudspeakers shown in one diagram

light green = SW, red = C, green = FR, blue = FL, brown = SR, light blue = SL

The volume complies to 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.

## 24.5 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.

## 14.6 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  $< 100\text{Hz}$ . 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  $94\text{Hz}$  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.

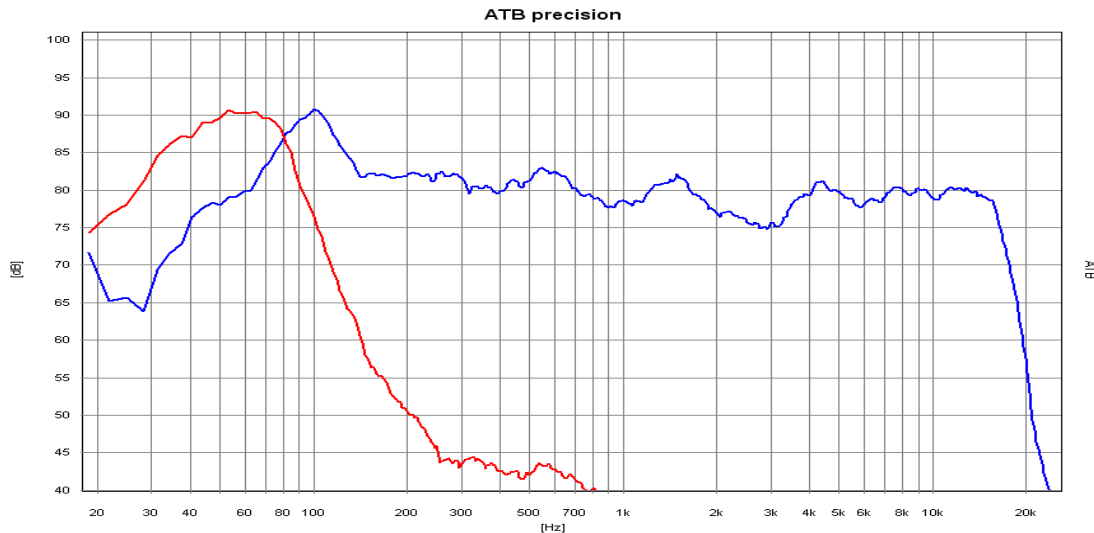
### Measurement of the cumulative signals of the front loudspeaker

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*.

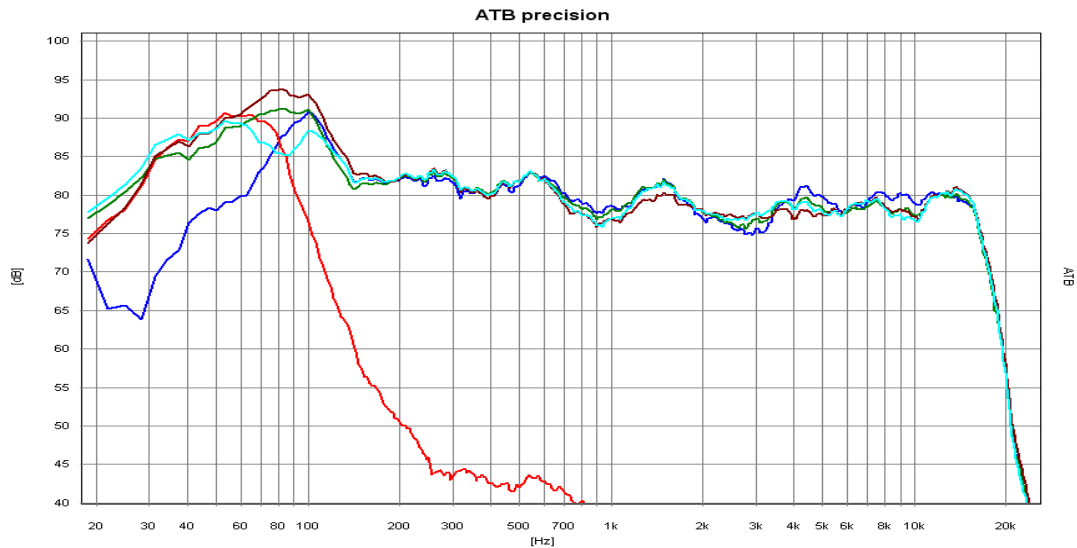


The picture shows the plots for the front loudspeaker and SW

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  $-3\text{dB}$  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  $180^\circ$  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.



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.

## 24.7 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.

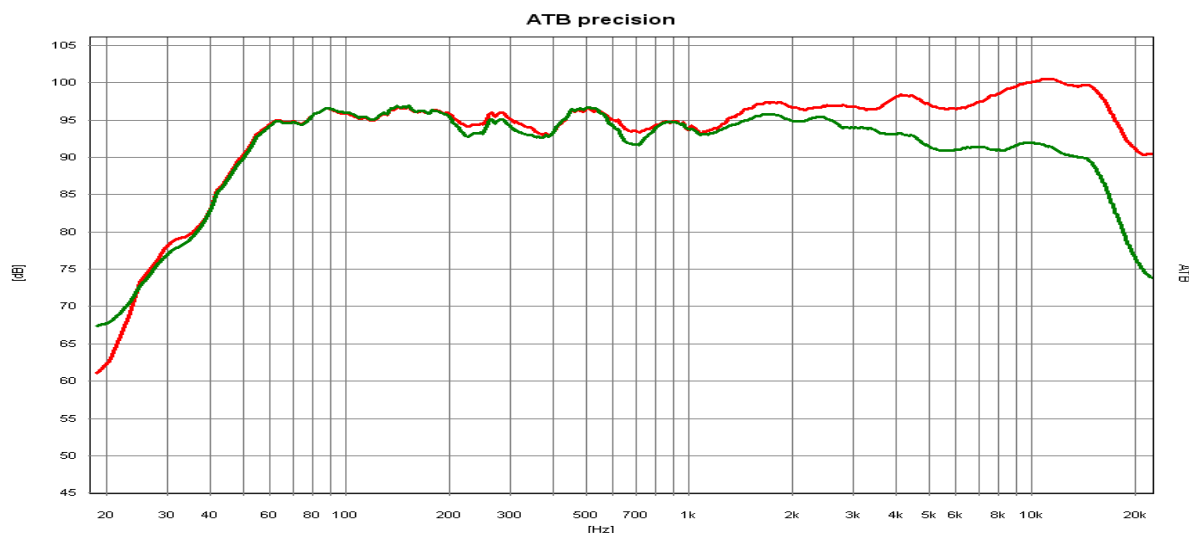


## 25. CALIBRATING THE HIFI STEREO SYSTEM

### 25.1 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.



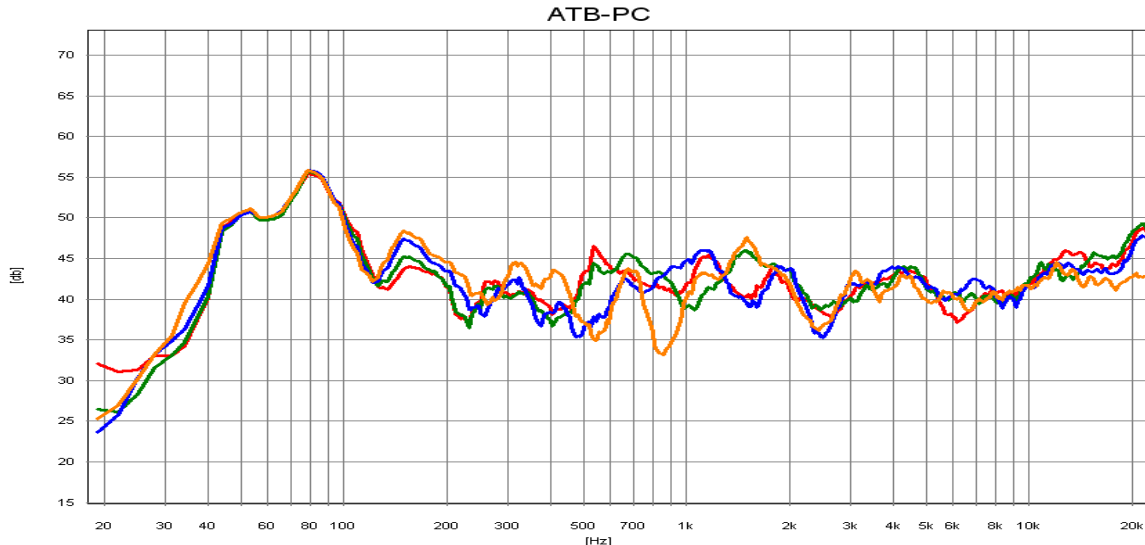
The frequency characteristic of a loudspeaker measured with a predefined axis **red** and its proper frequency characteristic **green**.

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.

### 25.2 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.

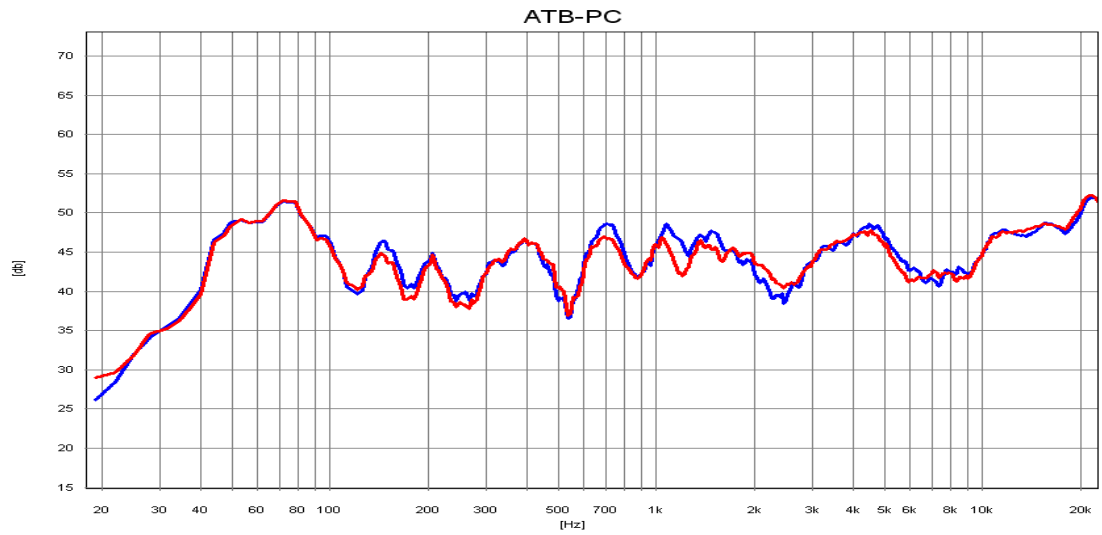
First of all the relationship between the distance to the walls and loudspeakers should be investigated.



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.

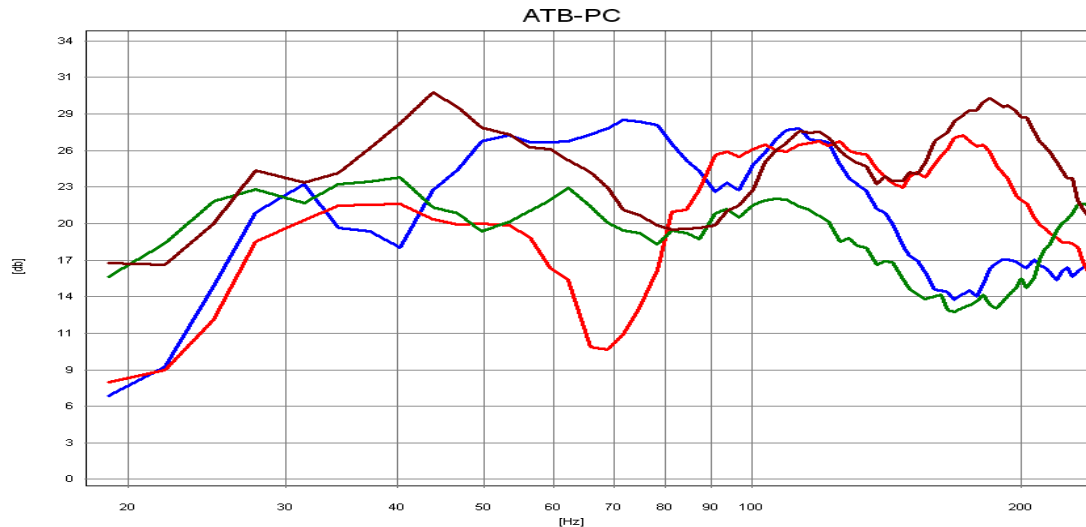
The distance from the back wall is of utmost importance for the middle tone range whilst choosing the seating position. A too smaller distance causes the middle frequency range to become jagged; this can only compensated through a larger distance to the wall or acoustic mats or wall carpets.



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.

### 25.3 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.

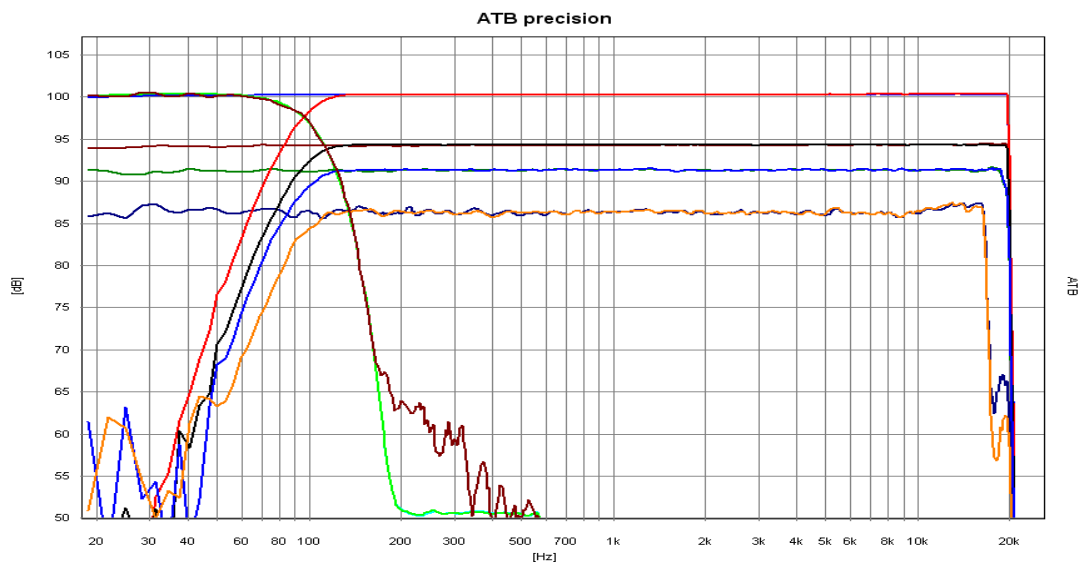
Further measurements are taken with specialized room acoustic programs, these mainly measuring the reverberation period. This value is only of importance to the acoustic engineer because the reverberation period of rooms can only be corrected with constructive steps.

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.

## 26. 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.



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.

## 27. DIGITAL MEASUREMENTS

The Optitest measurement is also for the frequency characteristic measurement of digital audio instruments like DVD Players, Digital switches, Digital loudspeakers and equalisers. The signal source serves the digital output of DVD Player or Computer. The correlation function of the program needs no time coordination of e.g. recording and playback, so that instruments with large delay between in- and output signal are able to be tested.

The PCM Signal with 48kHz is used for testing. Therefore the measurements are only suitable for studio instruments.

When measuring the digital input on the soundcard is chosen.

In opposition to the analogue signal the PCM Signal can also played digitally from a computer.

## 28. AUXILLIARY INSTRUMENTS, MICROPHONE AND ADAPTER

### AMPLIFIER

A stable amplifier is necessary for loudspeaker measurement. This should also possess enough bandwidth.

### MICROPHONE

The microphone should possess a good linearity, have a high bandwidth, on the other hand it should not be too sensitive. Microphones whose frequency characteristics need to be compensated are not suitable, because their impulse behaviour is not enough.

Delivered with the ATB PC measuring system:

- Program CD Rom
- Test-DVD
- Test microphone
- Adapter cable for balancing
- Handbook as PDF file