

SCIENCE &
RESEARCH
EDITION

VIAS[®]

VERSATILE INSTRUMENT ANALYSIS SYSTEM
FOR BRASSES AND STRINGED INSTRUMENTS



MANUAL

English

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IMPEDANCE
FOR
WIND INSTRU-
MENTS

ADMITTANCE
FOR
STRINGED
INSTRUMENTS

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Versatile I nstrument A nalysis S ystem

VIAS was originally a software to measure and evaluate the acoustical properties of bowed instruments. In its new extended form, it is a tool for measurement and analysis of any musical instrument, hence its name: Versatile Instrument Analysis System.

What can VIAS be used for?

1. for extended and more accurate impedance measurement and analysis, including audio tests of the measured instrument based on the measured or calculated impedance curve
2. for admittance measurements and evaluation of bowed and string instruments such as violins, cellos, harps, guitars, pianos, and percussion instruments, with help of the impulse hammer method
3. to generate a good calibration matrix for the BIAS measuring head, which is automatically imported into VIAS and BIAS.

It is for these reasons that this manual is separated into several sections:

Section 1: Measuring the input impedance of brass instruments

Section 2: Measuring the admittance of string instruments

Section 3: Calibration

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IMPEDANCE MEASUREMENT

What hardware is required?

The only hardware that is needed, is what comes when buying a BIAS-System. The BIAS measuring head should certainly have a USB connector.

VIAS can in principle also work with older analog BIAS measuring heads. In this case, the excitation signal and the data acquisition are performed via your computer's sound card. It is worth noting that not all sound cards offer sufficient sound quality to be used as a measuring apparatus.

It is possible to connect an external USB sound card between the BIAS measuring head and your computer, which is available in the market at a reasonable price as computer accessory.



BIAS USB -Measuring head

ADMITTANCE MEASUREMENT

What hardware is required?

To be able to perform an admittance measurement with VIAS, you will need:

- an impulse hammer with integrated force sensor
- an accelerometer and
- a charge amplifier.



PCB 086C03 Impulse hammer

If you already have these components, you can use them with VIAS in most cases without any problems. The only thing to keep in mind, is that the accelerometer must have the least mass possible, to avoid affecting the oscillating properties of the object to be measured.



ROGA Plug.n.DAQ
USB charge amplifier

In case you do not have these devices yet, we suggest the following configuration:

- PCB 086C03 Impulse hammer with integrated force sensor
- PCB352C23 Ceramic miniature accelerometer 0.2 g
- PCB030C10 Teflon cable
- ROGA Plug.n.DAQ USB charge amplifier

You can purchase these components with the BIAS or VIAS systems, or order them directly from the manufacturer.

Should you need to take measurements from several accelerometers simultaneously, a feasible alternative would be to use a PCB 4 channel amplifier. With this device, it is possible to take measurements with the impulse hammer and 3 accelerometers simultaneously. However, the support that VIAS offers in the usage of more than one accelerometer is at best very basic. In any case, you would need extra software to allow recording of more than 2 audio channels simultaneously.

Disadvantage: The output of this amplifier is analog, which means that you would also need extra hardware to digitise the signals.



Recommended configuration

Introduction

Main differences between BIAS and VIAS



- All measurements are stored and managed in the integrated Microsoft Access Database.

- Nature of impedance measurement: Excitation with a chirp signal (a very fast frequency sweep).

The measured signal is recorded, and a Fourier Transform is performed to it.

This spectrum is then used to calculate the impedance, together with the calibration data and the reference spectrum.

ADVANTAGE: Very short measuring time, quick results.

DISADVANTAGE: Bad signal to noise ratio (SNR). Fixed resolution. It is not possible to have a higher resolution for optimisation tasks.

- Display options: Both axis linear.

Display of impedance magnitude.

- There is no integrated database. Each measurement is stored in its own file with the extension ".via", but it can also be exported into the BIAS database.

- Nature of impedance measurement: Excitation with a sinusoidal sweep.

The measured signal is filtered with a narrowband lock-in filter tuned to the frequency of the excitation signal. The output level of the filtered signal is displayed as the impedance curve, taking into account the reference channel (e.g. force sensor) and the loaded calibration curve.

ADVANTAGE: Higher energy in the excitation signal, hence better signal to noise ratio (SNR). Optional time duration and frequency range allow to meet particular needs of resolution and accuracy.

The sensitivity to interference from the acoustic field is reduced by the use of the lock-in filter

DISADVANTAGE: Long measuring time.

- Display options: Axis can be linear or logarithmic.

It is possible to display the result's magnitude, phase, real part, imaginary part, spectrogram, RMS, harmonic centroid, and a virtual glissando assuming a typical excitation spectrum (sawtooth for bowed instruments, pulse sequence for wind instruments).

Introduction

Why the need of an extra software for BIAS?

BIAS was written for a specific audience: People with no technical background, without any previous knowledge of acoustics or of how wind instruments work, should be able to use the program. A measurement should be performed easily, but still deliver good and valid results.

Nevertheless, simplicity has its price: Restrictions had to be met as well as many settings fixed, as these are only meaningful under particular circumstances, or simply because deeper background knowledge would be needed for the results to be interpreted correctly.

The freedom to choose settings and analysis possibilities would result in mal-operation by the non-expert, and the magnitude of the training effort to ensure that the BIAS is fully functional and operated correctly would be for us unmanageable.

BIAS is nowadays also employed for scientific investigations by numerous universities and institutes in Europe, Asia and the USA. Also, more and more instrument makers have acquired the expertise that was deprived from them in their original education.

The coverage of “Power User” has become bigger and bigger in the last years. This is why we have decided to release a “Science&Research Software Package”, which was previously based on bowed instruments, and that has also in our institute proved to be versatile in numerous applications.

What are the new features in VIAS?

The nature of the excitation signal can be freely chosen, measuring errors in the impedance curve can be corrected after having finished the measurement, and there are many new possibilities for the analysis, including the option to play the instrument virtually based on the measured impedance curve.

New capabilities while measuring

- Free selection of excitation signal (it can also be a .wav file)
- Free selection of frequency range
- Free selection of time duration (given in milliseconds)
- Display in real time of excitation and measured signals
- Changes in level during the measurement
- Measured signal can be stored as .wav file

New capabilities in the post processing

- Averaging of several measurements
- Easy correction from measuring errors (glitches)
- Curve filtering
- Curve smoothing
- Curve trimming
- Curve resampling
- Curve multiplication
- Curve division
- Continuous curve shifting along the Y-axis
- Measurements and analysis results can be exported

Introduction

New analysis capabilities

- Display of magnitude, phase, real and imaginary parts
- Calculation of power cross-spectrum and coherence spectrum
- Generation of a virtual sound, of which the number of harmonics can be selected
- Termination impedance calculation
- Transfer function calculation
- Air column simulation
- Sound analysis:
 - Spectrogram Display
 - Line display
 - Area display
 - Total RMS level display
 - Harmonic centroid display (sound balance point)
- Display:
 - X-axis and Y-axis linear or logarithmic
 - Free selection of frequency range
 - Free selection of dB range
 - X-axis in Hertz or musical notes

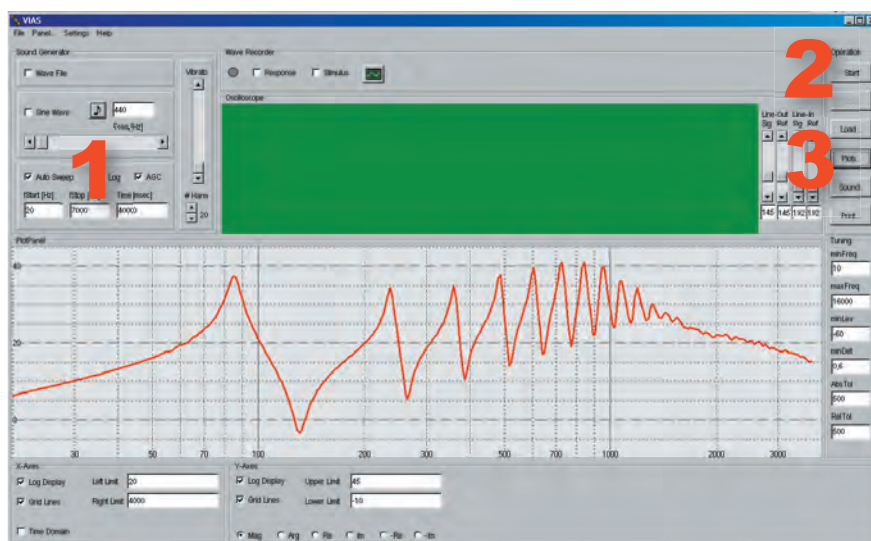
**VIAS works with Windows XP
Windows 7 and 8.1 and Windows 10**

**It runs on Apple Computer with the help of the free
Apple "Bootcamp" or commercial software like
"Parallels Desktop"**

QUICK START FOR WIND INSTRUMENTS

Short introduction for impedance measurement

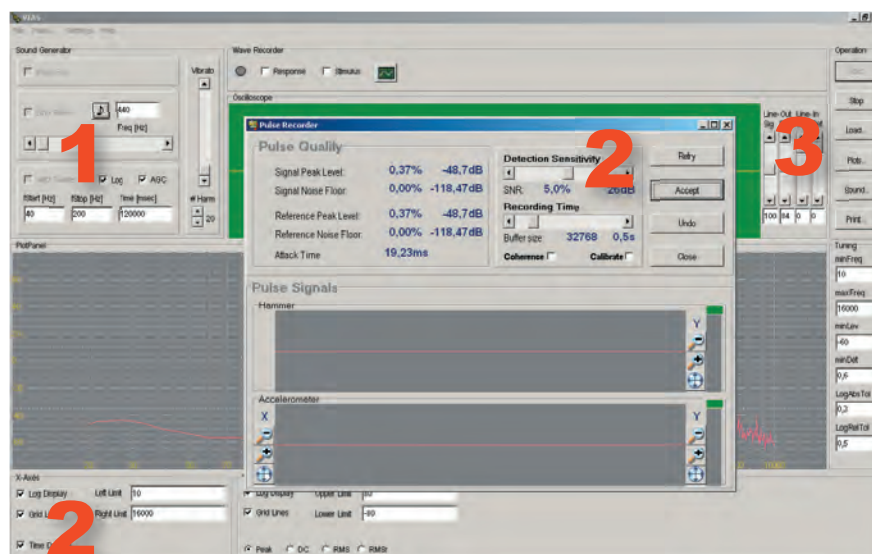
- Connect the USB measuring head
 - Start the program
- 1** Specify the start and stop sweep frequencies: Double click on the fields “fStart” and “fstop” (top left corner), and type the desired frequencies
 - In the field “Time”, specify the duration of the sweep in milliseconds (e.g., 1 minute = 60000)
 - 2** Click the button “Start” (top right corner). If you do not see the progress of the curve during the measurement, extend the range of the Y-axis, and if required, also of the X-axis (input fields are under the display area)
 - 3** To save the measurement: Click on the button “Plots” (top right corner). A list of recent curves will appear.
 - Type the name of the instrument, intonation and fingering on the field “Description”
 - Click on the diskette icon, and choose the name of the file, and the directory where it will be saved



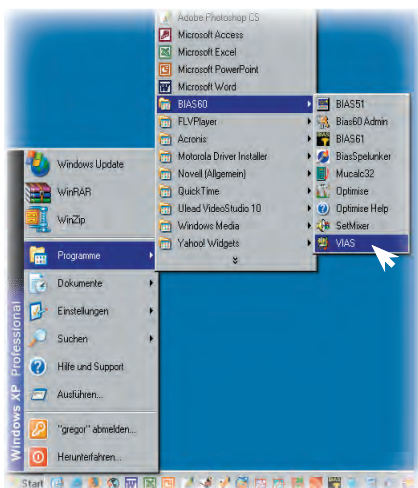
QUICK START FOR STRING INSTRUMENTS

Short introduction for admittance measurement

- Connect hardware (see page xxx)
 - Start the program
- 1** Specify the start and stop sweep frequencies: Double click on the fields “fStart” and “fstop” (top left corner), and type the desired frequencies
 - 2** Check the box “Time domain” (bottom left corner)
 - Excite with the impulse hammer
 - If the measurement fails, click on the button “Retry” (top right corner of the measurement window) and measure again. If the measurement succeeds, click the button “Accept”.
 - After having done all the measurements, close the measurement window by clicking the button “Close”.
 - 3** To save the measurement: Click on the button “Plots” (top right corner). A list of recent curves will appear. Type relevant comments on the “Description” field. Click on the diskette icon, and choose the name of the file, and the directory where it will be saved.



Starting the program

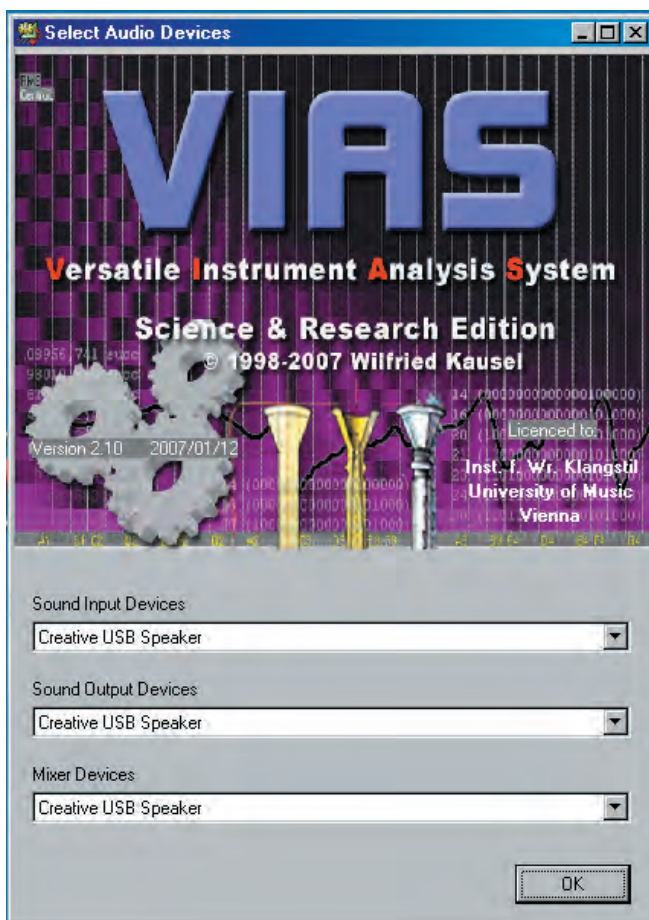


You can start VIAS either

- via the Windows Start Menu (Start -> Programs -> BIAS60 -> VIAS)



- or via the menu item Tools -> VIAS in the BIAS window.



After starting the program the splash screen will appear, where the sound devices to be used are specified.

- If the BIAS measuring head was connected to the computer before starting the program, VIAS should have recognised it as a "Creative USB Speaker", and will appear as shown in the figure on the left.

- If that was not the case, you have to choose it in the three drop down menus.

Note that the name of the device associated with the USB measuring head can depend on the Windows version you are using.

- If no measurements are to be done, but only analysis and calculations, there is no need to connect the measuring head.

Click "OK", and after the License agreement the main window will appear.

Main Window

Oscilloscope window: Real time display of excitation and measured signal (time domain)

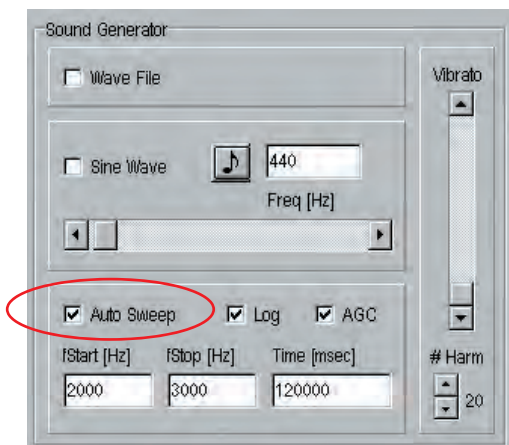
The screenshot shows the VIAS software interface with several key components highlighted by red boxes and labeled with red arrows:

- MENU BAR:** Located at the top left, containing 'File', 'Panel...', 'Settings', and 'Help'.
- Sound Generator:** A panel on the left containing controls for 'Wave File', 'Sine Wave' (with a frequency of 440 Hz), 'Auto Sweep', 'Log', and 'AGC'. It also includes a table for frequency ranges:

fStart [Hz]	fStop [Hz]	Time [insec]
2000	3000	120000
- Wave Recorder:** A central panel with 'Response' and 'Stimulus' buttons, and an 'Oscilloscope' window showing a green display area.
- Level selection for the excitation and measured signal:** A vertical control on the right side of the Oscilloscope window.
- File management:** A panel on the far right with buttons for 'Start', 'Stop', 'Load...', 'Plots...', 'Sound...', and 'Print...'. Below it is a 'Tuning' section with parameters like minFreq, maxFreq, minLev, minDet, LogAbsTol, and LogRelTol.
- PlotPanel:** A large central plot area displaying a red waveform. Below it are 'X-Axes' and 'Y-Axes' settings, including 'Log Display', 'Grid Lines', and 'Time Domain' options.
- Box for admittance measurement with the impulse hammer:** A red box at the bottom left of the PlotPanel.
- Display area "Plot-Window":** The main plot area showing the waveform.
- Selection of display between magnitude, phase, real and imaginary part:** A row of radio buttons at the bottom center, including 'Mag', 'Arg', 'Re', 'Im', '-Re', and '-Im'.
- Trigger and lock-in filter properties for the acquisition of the measured signal:** A red box at the bottom right of the PlotPanel.

MEASUREMENT: Selection of excitation signal

Suggestion:



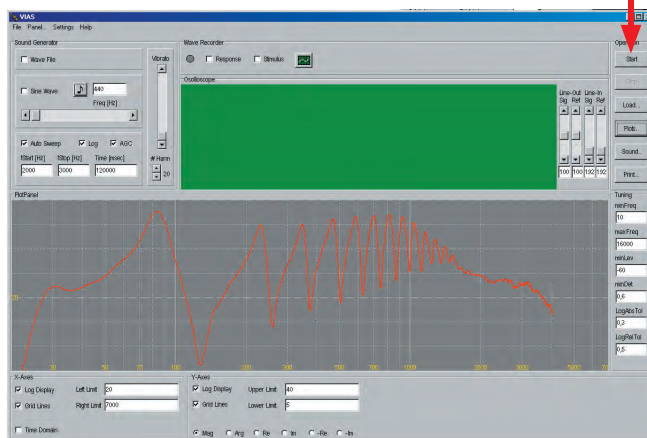
- Normally, a sinusoidal sweep will be used as excitation signal. In this case, the box “Auto Sweep” should be checked.

- Choose between a logarithmic or linear sweep

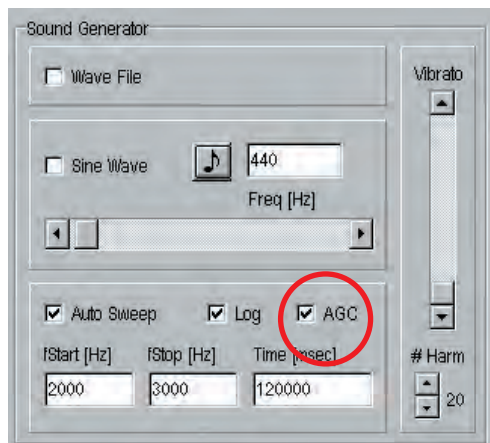
This selection depends on the desired result: A logarithmic sweep is usually recommended, as it gives enough time for the generation of standing waves in lower frequencies, and it is musically relevant, since it corresponds to the way pitch and musical intervals are perceived.

If you need a specific accuracy in high frequencies, the linear sweep is recommended – In any case, you should allow enough time to achieve the desired resolution: e.g.: 5 minutes = 300,000 ms

- Type the start and stop frequencies in both fields **fStart** and **fStop**, in the range from 0 to 20,000 Hz. In the shown picture, the measurement will be done between 2 kHz and 3 kHz.
- Type the duration of the sweep (in milliseconds) in the field **Time**. In the shown picture, the duration of the measurement is 2 minutes (=120,000 ms) for a range of 1 kHz
- For short sweeps (less than 5 minutes for 0 – 20 kHz), the AGS sometimes does not work properly.
- Click on the “Start” button to start the measurement.



MEASURING – Automatic Gain Control

**AGC - Automatic Gain Control**

The AGC is a program that controls the level of the excitation signal automatically, so that the input signals do not saturate. The AGC can only detect and regulate one of two types of saturation. The AGC cannot detect a saturation from the reference microphone.

To be active, the AGC box has to be checked before the start of the measurement!

Which kinds of saturation can occur?

1. There is danger of saturation the analog to digital conversion (ADC) stage. In case the measured signal in certain frequency range becomes saturated, it will be automatically displayed in **RED** in the measured signal display.

● **Solution:** You can either lower the level of the excitation signal and repeat the measurement (see page 20), or measure with a lower excitation signal only the frequency range shown in red, and paste the new measurement in the original curve with the command "Patch Previous" (see page 52).

2. There is danger of saturation also in the reference microphone inside the measuring head. This microphone is 20 mm away from the speaker membrane, and is used to acquire the reference signal. It is exposed to a very high acoustic pressure.

● **Solution:** It is recommended to observe the measured signal on the oscilloscope window during the measurement. In this case, saturation of the sinusoidal excitation signal can be seen immediately. Alternative: Make a test measurement with a quick sweep, to determine if there is danger of saturation.

MEASURING – Automatic Gain Control

How does the AGC work?

The program monitors the measured signal, which should have an amplitude between -32768 and +32768. As soon as the amplitude of the measured signal exceeds 25000, the amplitude of the excitation signal is decreased by 6 dB. When the amplitude of the measured signal stays for certain time under the limit, the amplitude of the excitation signal is increased again.

Some periods of the signal are necessary for this adjustment to take place. The AGC is therefore most effective for slow sweeps!

The AGC operates independently from the level controls located in the display area. It is therefore wise to turn up the level controls to the maximum

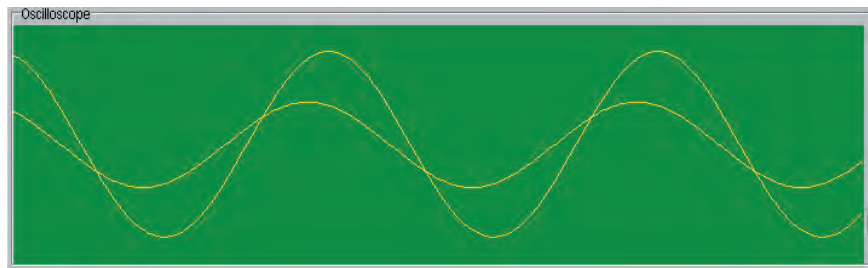
It is recommended in general to leave the AGC active, also for fast sweeps, as the signal from the reference microphone should always be monitored on the oscilloscope window for possible saturation

MEASURING – Automatic Gain Control

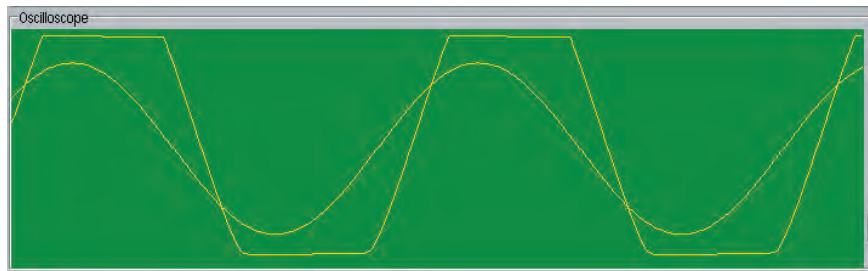
How is saturation in the measured signal detected?

1. During the measurement, one of the signals (which should have a sinusoidal form) in the oscilloscope window looks truncated on top and/or bottom (the measured signal is displayed yellow and the reference signal white).

Correct level



Saturated



2. The frequency ranges where the ADC had significant saturation are shown in **RED** in the Plot window, where the measured curve is displayed.



MEASURING – Manual Gain Control



The BIAS measuring head will be recognised and handled by Windows as a USB sound card.

It has 2 output channels (Line-Out) where the excitation signal is delivered, and 2 input channels (Line-In) for the measured signal.

Depending on your Windows driver, you will be able to move each level (by adjusting the corresponding scrolling bar) independently, or only both together. In some cases these levels cannot be changed by the user.

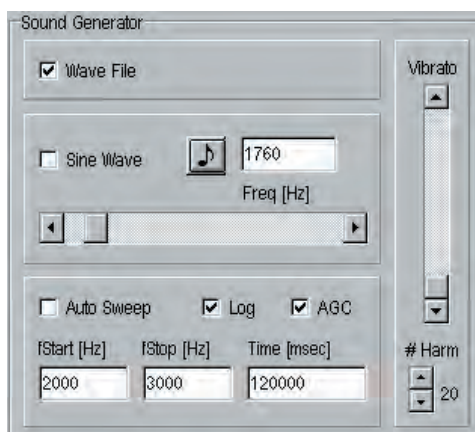
It is also possible that you only get two scrolling bars, one for Line-In and one for Line-Out.

Whichever configuration you get, for the manual gain control please only use the Line-Out scrolling bars, and always set both levels at the same value!

- It is also possible to change the Line-Out level during the measurement. This change will take effect immediately.
- The easiest and fastest way to adjust the level is to click and drag the scrolling bar with the mouse.
- If it is possible to set both levels independently from one another, it is recommended to set one with the mouse, and type on the field of the second scrolling bar the number from the field under the first scrolling bar. This is to ensure that both levels are set at the same value.

MEASURING – Selection of excitation

Types of excitation:



- An arbitrary sound file

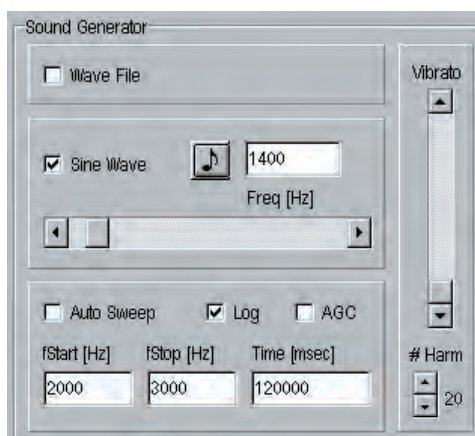
With the mouse check the box **Wave File**. A dialog window will open, where you can choose an arbitrary file in .wav format

The .wav file should have a sine wave, as the lock-in filter should be able to detect the frequency. As long as no frequency is detected, the display of the measured signal will be disabled.

- **To disable the lock-in filter:**

- ▶ In the menu "Settings"
- ▶ click on "Filter"

If the lock-in filter is disabled, the amplitude will not be measured at each frequency. It will be measured as the RMS level over the whole frequency range. The frequency detection should work anyway, because otherwise the program would not know at which point in the frequency axis to plot that particular amplitude. Polyphonic audio recordings are therefore unsuitable as stimuli, and will not lead to the display of a measured curve. 😞



- **A fixed, stationary frequency**

You can choose this frequency by:

- ▶ Typing the frequency in the field beside the note symbol, or
- ▶ clicking the note symbol and typing the name of a note (e.g. A4 for 440 Hz), or
- ▶ setting the frequency by moving the scrolling bar

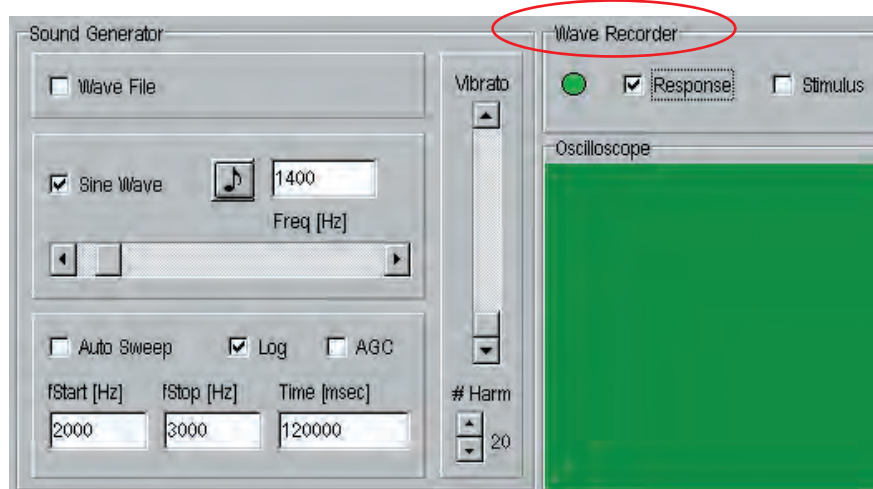
Such a measurement creates a frequency point in the impedance window, which can be broadened to the left or right by decreasing or increasing the frequency respectively.

Extremely sharp resonances can be measured this way with maximum accuracy. In this case, it is recommended to do a "resampling" before saving with a correspondingly high frequency resolution, so that the multiple measured frequency points do not inflate the result unnecessarily.

MEASURING – The wave-recorder

Show measured signals in Microsoft .wav format:

Excitation and measured signals can additionally be recorded and saved as .wav files. Therefore, they can be processed as sound files, used for the calculation of a coherence-, power cross- or auto- spectrum (see pages 107-108), used for other purposes, or simply played back.



- By checking the box “Response”, the wave recorder is activated, and a new window opens where you can choose the name of the file and the directory where the displayed measured signal will be saved.

This results in a stereo .wav file, one of the channels being the reference signal from the microphone over the loudspeaker membrane, and the other the measured signal from the microphone inside the mouthpiece.

- By checking the box “Stimulus”, the wave recorder is also activated, but this time it will record the excitation signal in a mono (one channel) .wav file. In the dialog window you can choose the name of the file, and the directory where the displayed excitation signal will be saved.

START THE MEASUREMENT



- Click on the “Start” button to start the measurement
- You can stop the measurement at any time by clicking on the “Stop” button
- Set the colour and line style of the new measurement: Right click anywhere on the display area (see page 33)

During the measurement, look at the time function of the excitation and measured signals on the oscilloscope window. At the same time, the measured curve will be displayed in the “Plot-Panel”. The measured signal is displayed in yellow, and the reference signal (as long as its display has not been disabled) in white.

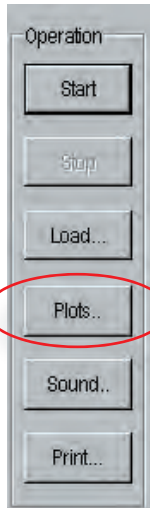
When the measurement is finished, click on the button “Plots” to save the curve or to continue working.

LOAD A MEASUREMENT



Click on the “Load” button, and a new dialog window will open, where you can load a measurement that was saved previously. It will be displayed in the “Plot-Panel”, along with other current measurements.

FILE MANAGEMENT



The "Currently Displayed Curves" window will open after clicking on the "Plots" button, or automatically after performing a new measurement.



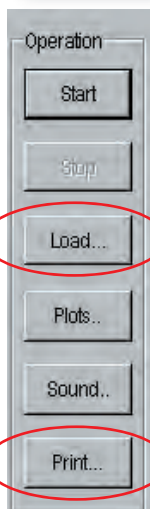
This window shows the currently displayed curves. Apart from being able to save or delete measurements, this window also:

- gives detailed information about each displayed measurement,
- allows you to manipulate the plots,
- is the starting point for multiple analysis, and
- gives you the capability to manipulate and post process the data



START THE SOUND ANALYSIS

- To open the sound analysis window, click on the "Sound" button (see Sound Analysis, page 36).



LOAD FILE

- Click on the "Load" button to load, display and further manipulate previously saved VIAS or BIAS files, as well as .wav files.

PRINT

- Use this command to print the contents of the "Plot" window

FILE MANAGEMENT "PLOTS"

This window opens automatically after:

- performing a new measurement ??
- loading an existing measurement ??
- clicking on the "Plots" button

SAVE

- New measurements are saved
- If the curve has already been saved or loaded, all the changes will be saved after clicking "OK" in the confirmation dialogue.

DELETE

- New non-saved measurements will be deleted!
- Already saved or loaded curves will be removed from the Plot window

PLAY BACK SOUND

Generates a sound based on the measured curve, according to the properties set in the window where the excitation signal was chosen

Date and time when the measurement was done

Number of measured points in the curve

	Date	Time	Points	Line	File Name	Description
	26.07.07	14:11:39	405		D:\WVK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" kein Ventil gedrückt
	26.07.07	14:50:22	410		D:\WVK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 2. Ventil
	26.07.07	14:26:04	410		D:\WVK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 3. Ventil

Line style, colour and thickness

File name, including path

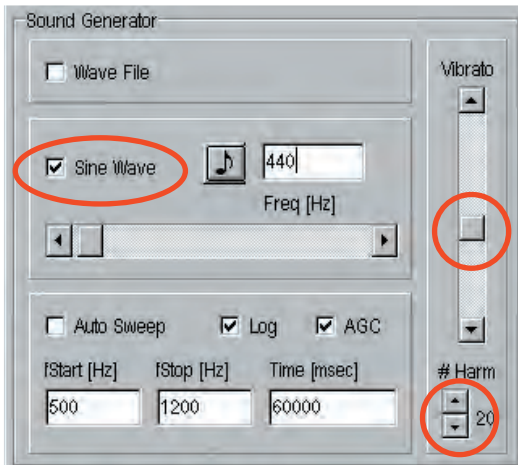
Editable comments

SHIFTING

Shifts (without changing the data) the curve along the Y-axis in the Plot window, for ease of comparison

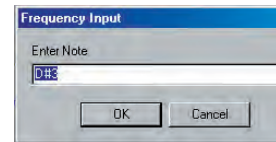
FILE MANAGEMENT "PLOTS"

SOUND PLAY BACK

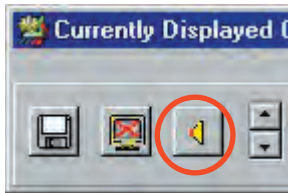


To choose the sound parameters:

- Click on the box "Sine Wave" if you would like a tone, or in the box "Auto Sweep" if you would like a frequency sweep (the current Start- and Stop frequencies apply, as well as the time duration settings)
- If you click on the musical note icon, you can write the name of the note (CAUTION: American notation! **A4 = a¹**)



- The degree of vibrato can be set with the "Vibrato" scrolling bar.



- With the arrow boxes marked "**# Harm**" under the vibrato scrolling bar, you can choose the number of harmonics (1-99) that the sound will have
- Click on the speaker icon to hear the curve. Click again to stop the sound generator.

NOTE:

You can choose the number of harmonics the sound should have, but bear in mind that if you choose e.g. 60, only the first 22 will have an effect on the sound, if the impedance curve only had 22 maxima (peaks)

HINT:

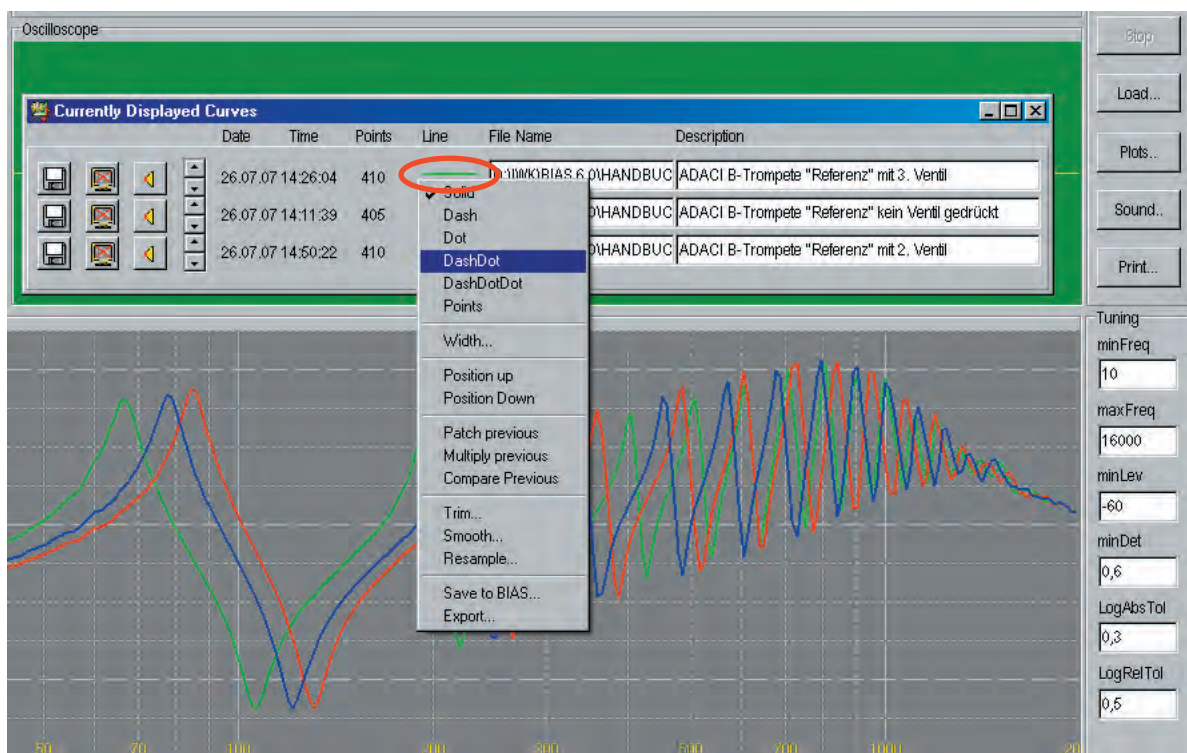
Usually, VIAS sends the excitation signal to the measuring head and the "virtual" sound to the sound card of your computer. Whether this works depends mainly on which sound card drivers are installed, as well as how the BIAS measuring head is recognised!

In case the sound is mistakenly played through the measuring head, choose the following command: Panel ... -> Options -> Properties. Here you can set the sound card as output device. If this does not help, disconnect the BIAS measuring head from the computer, so that the sound card is the only remaining output device in your computer

FILE MANAGEMENT "PLOTS"

CHANGING THE CURVE PROPERTIES

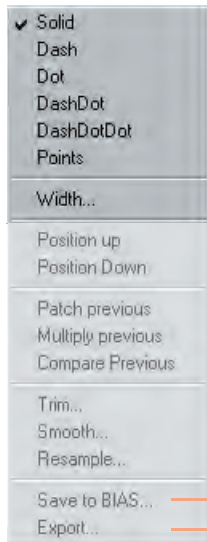
- **Right click** on the colour line of a file in the "Currently displayed Curves" window to open the **Context Menu**.
- All following actions apply only to the curve that belongs to the colour line you clicked



DATEI -MANAGEMENT "PLOTS"

CHANGING THE CURVE PROPERTIES

The context menu is divided in several areas:



Curve properties:

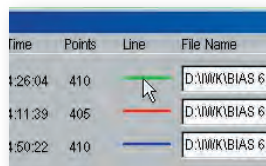
Line style (solid, dotted, dashed, etc.). The width is given in numerical value

Data manipulation for further analysis (see pages 50-57)

Export the curve into the BIAS database

Export the curve in ASCII format

● **To change the colour of a curve:**

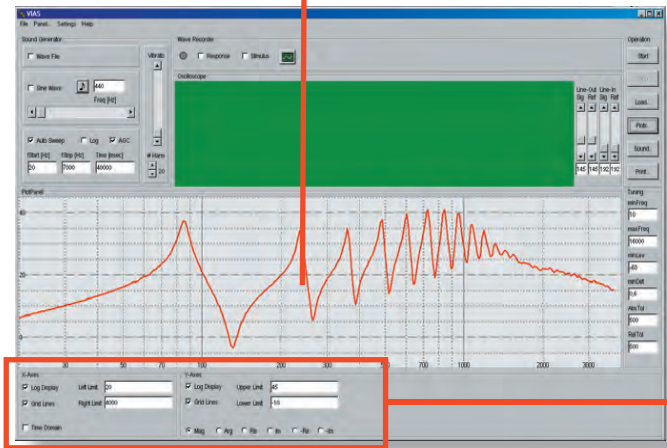


Simply (**left**) click on the colour line of the corresponding file in the "Currently Displayed Curves" window. A window will appear where you can choose (or generate and add) any colour.



DISPLAY AREA - PLOT WINDOW

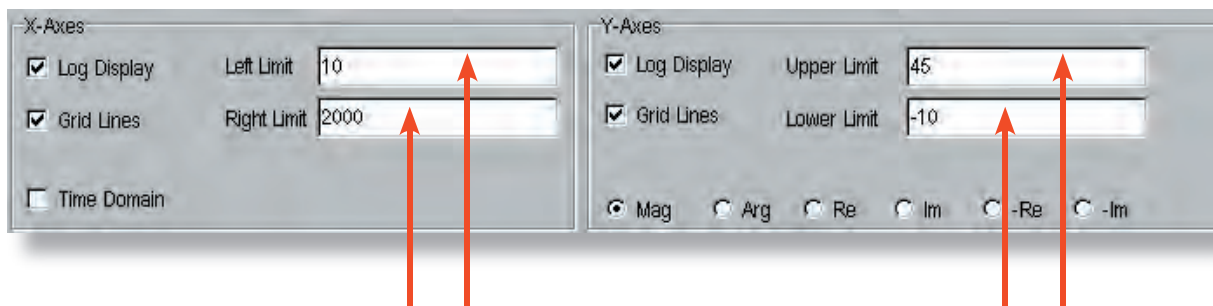
Display area (Plot window)



All parameters can be fast and easily changed by typing the appropriate values in the setting fields located below the display area

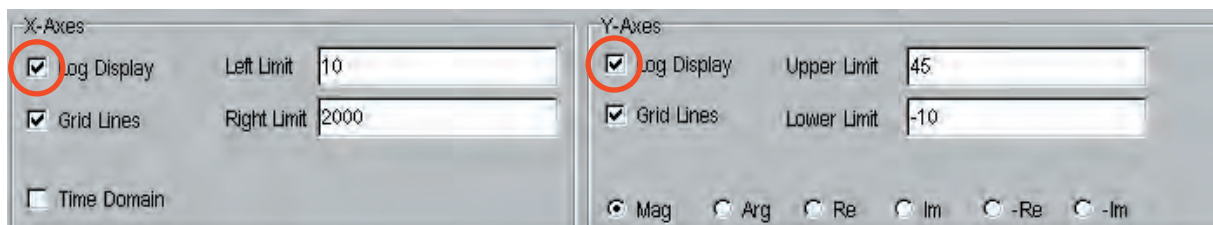
Area to set the display parameters

- Setting the frequency range (X-axis) and the impedance values (Y-axis)



Simply double click in the field, and type the desired frequency and impedance range

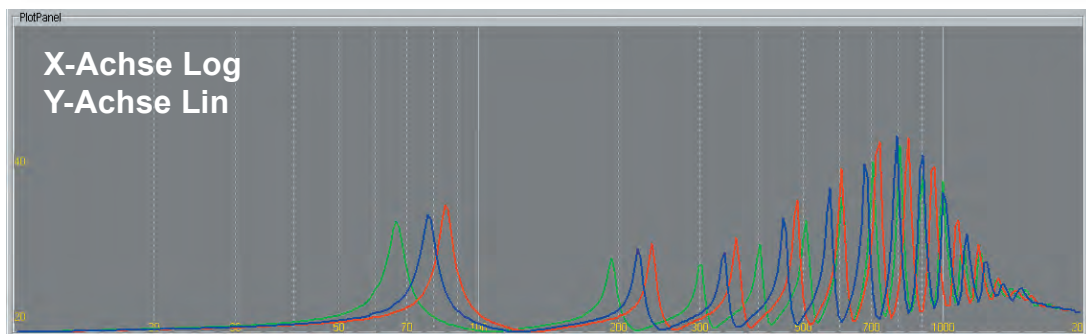
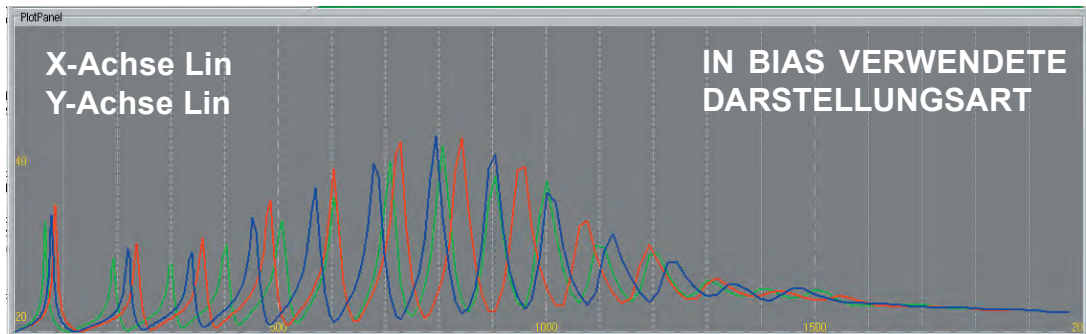
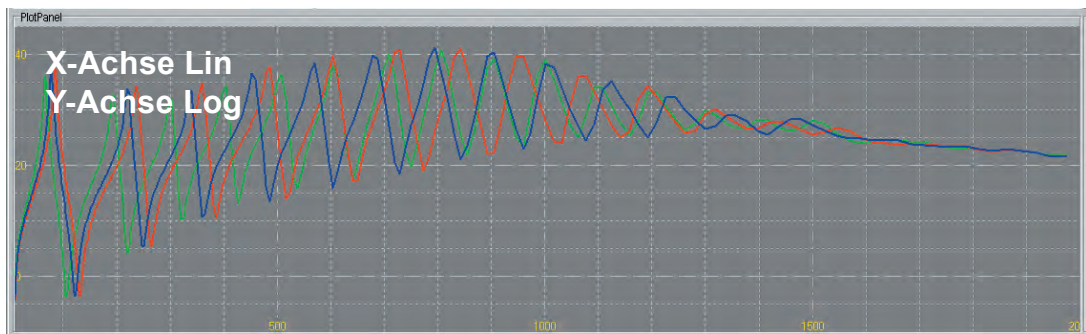
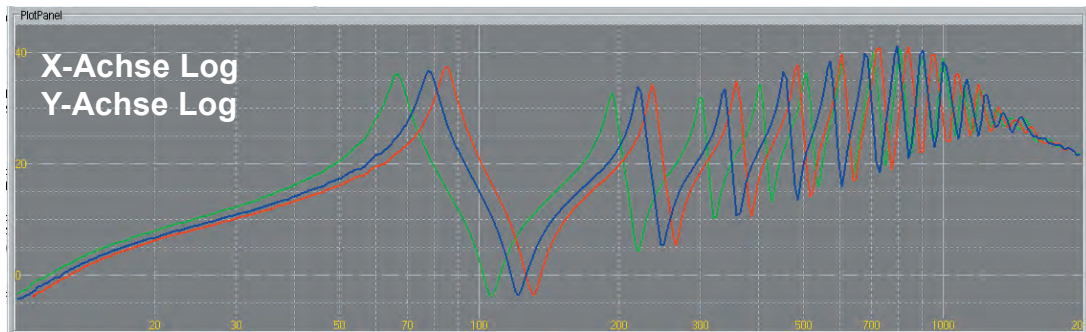
- Alternative between logarithmic and linear scale for both axes



Check or uncheck the box to turn the logarithmic scale on or off

DISPLAY AREA - SCALING

Scaling alternatives: Logarithmic – Linear



DISPLAY AREA - CURSOR

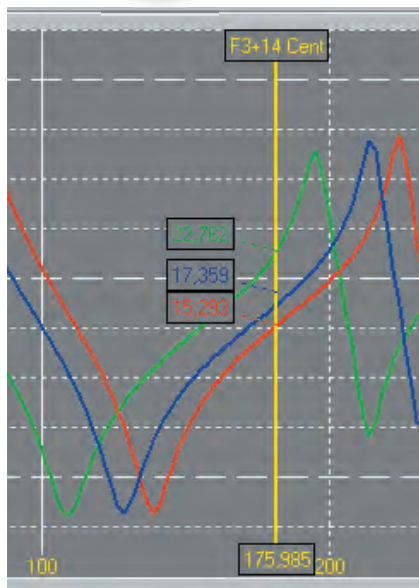
Retrieve the cursor:



Alternative 1: Move the mouse pointer to the left border of the display area, until it becomes a double arrow. Then left click and drag to the right: The cursor will appear.



Alternative 2: Strike the right arrow key (on the keyboard) several times. The cursor will appear in the left border of the display area



- The cursor reads all the impedance values of the displayed curves at the same time, and shows them in the colour of the curve to which each value corresponds

- The bottom field gives the current frequency value, the top field the corresponding musical note (in american notation $a^4 = a^1$) and its deviation in cents from an equally tempered scale, tuned at 440 Hz.

The reference frequency of 440 Hz (standard pitch) can be adjusted with the scrolling bar (below the input field). Clicking on the corresponding arrows will change the frequency in 1/10 Hz steps.

- The cursor can also be moved using the arrow keys from the keyboard

If the curve is rough (has glitches or is affected by noise), the cursor will not switch to a curve maximum or minimum, but will stay somewhere in the middle.

Solution: Smooth the curve (page XXX)



Move the cursor to the right or left with these two keys



Press the **"up arrow" key** to move the cursor to the next impedance **maximum** (peak)

Press the **"down arrow" key** to move the cursor to the next impedance **minimum** (valley). The movement of the cursor is always from left to right! If you would like to detect the maxima or minima, you should start with the cursor on the far left.



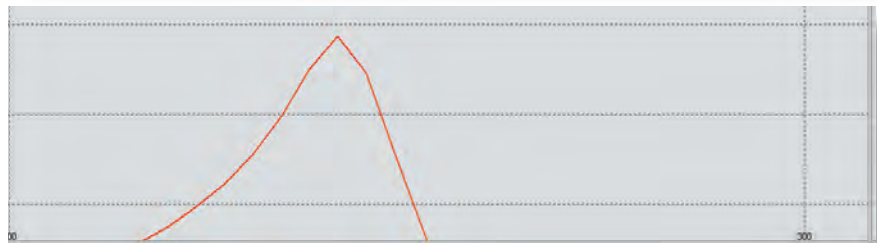
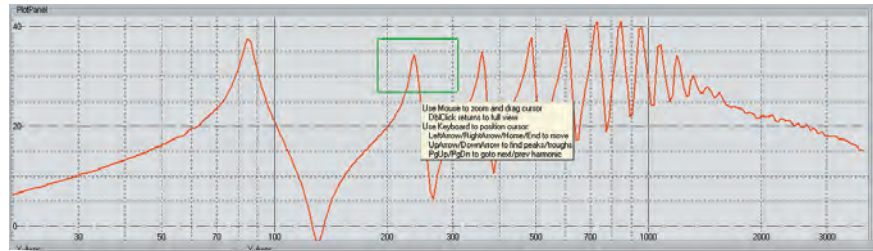
Press the **"Page up"** and **"Page down"** keys to move the cursor to 2, 3, 4, etc. times the output frequency. This is useful to detect harmonicity of the impedance peaks

All values read by the cursor will be copied automatically in the windows Clipboard. You can simply paste all maxima, minima or other values on arbitrary points of the curve in a text editor or spread sheet for further work.

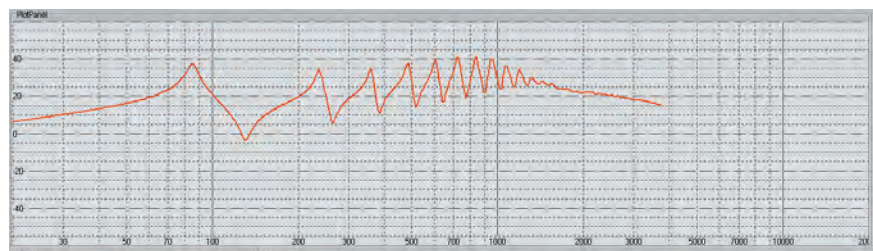
DISPLAY AREA - ZOOM

● ZOOM IN

Use the mouse pointer to drag a rectangle around the region into which you want to zoom.

**● ZOOM OUT**

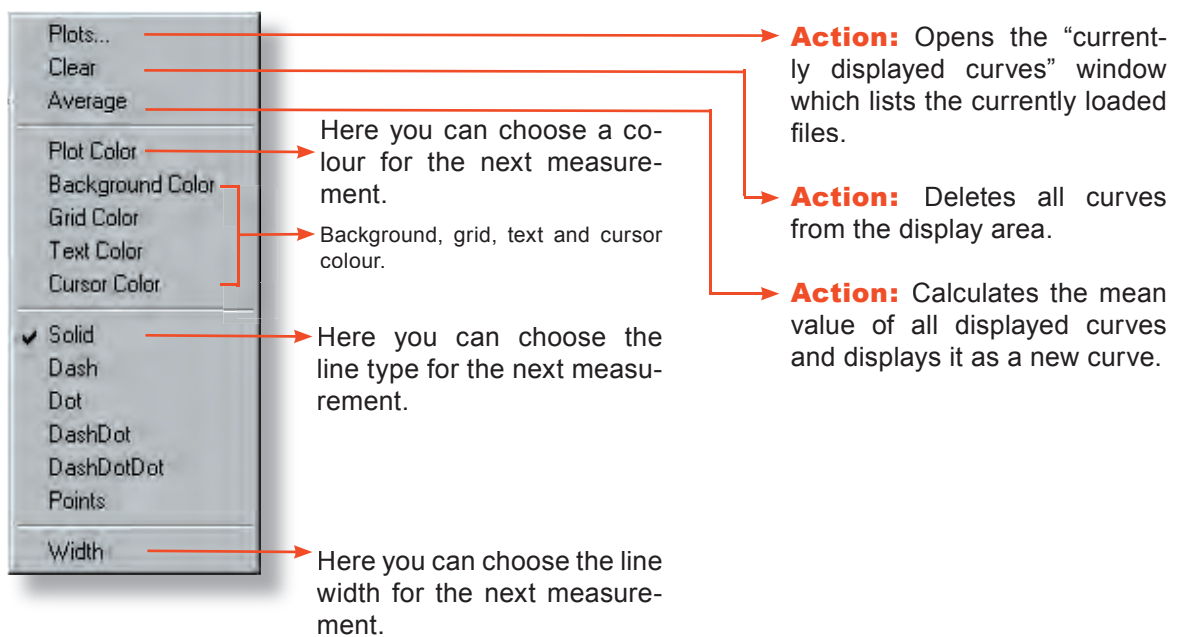
Double click on the display area, to restore the display back to the standard setting.



DISPLAY AREA - CONTEXT MENU

Right click anywhere in the display area to open the context menu.

The first three commands in the context menu refer to the curves currently displayed in the display area. All other commands refer to the curves of future measurements.

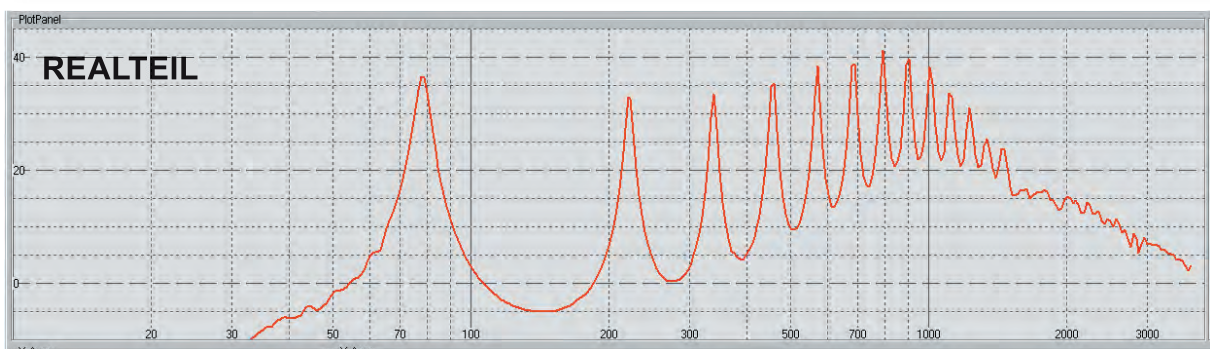
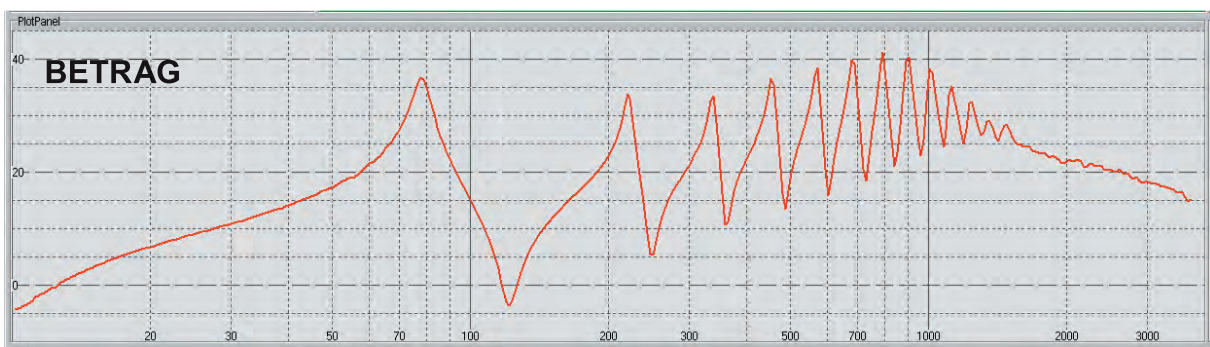
**Note:**

Changes of the axes' scaling, background colour, line type, etc. can be done during a measurement without affecting the result.

IMPEDANCE DISPLAY

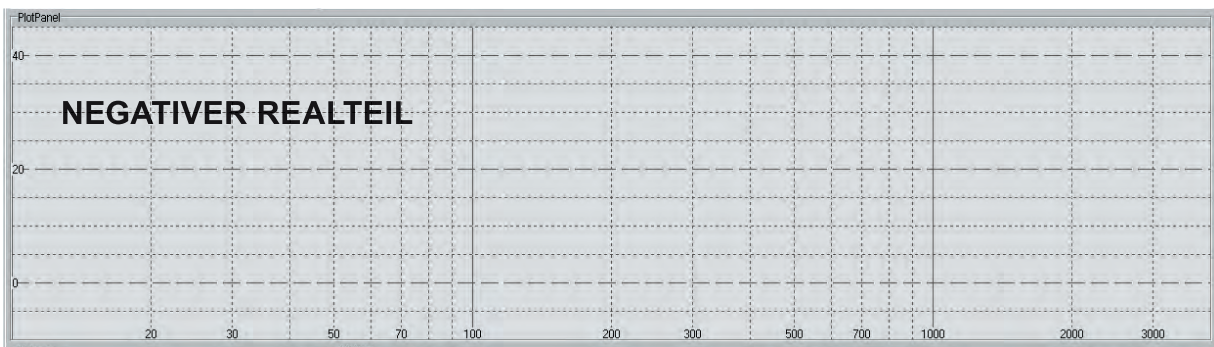
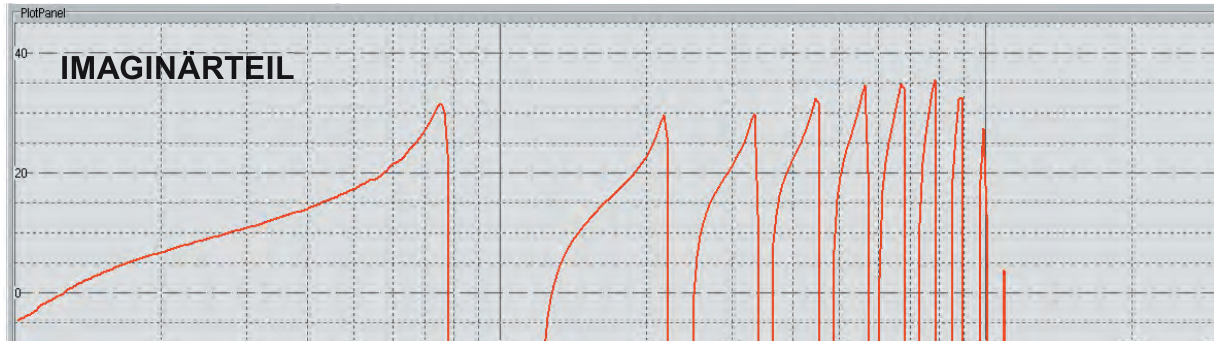


Click on the appropriate radio button to choose the desired display mode.



"**Mag**" displays the magnitude spectrum,
"**Re**" displays the real active component (= radiated or dissipated part),
"**Im**" displays the imaginary reactive component (= part that is not radiated or dissipated)

IMPEDANCE DISPLAY

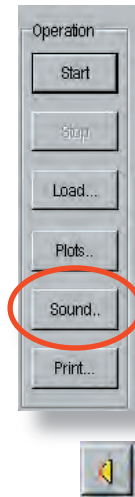
**Note:**

Normally an input impedance curve does not show a negative real part. If this ever occurs, effective power flows from the measured system back into the exciter. However, the reason for this is usually that there is some problem with the measurement conditions.

Possible error sources: The sweep moves too fast across sharp resonances, the polarity of a signal is inverted, there are phase shifts in an external amplifier or in the sound card.

Of course, arbitrary curves, such as the difference of two curves, can show a negative real part!

ANALYSES - SOUND ANALYSIS



- **To start the sound analysis**

click on the button "Sound"

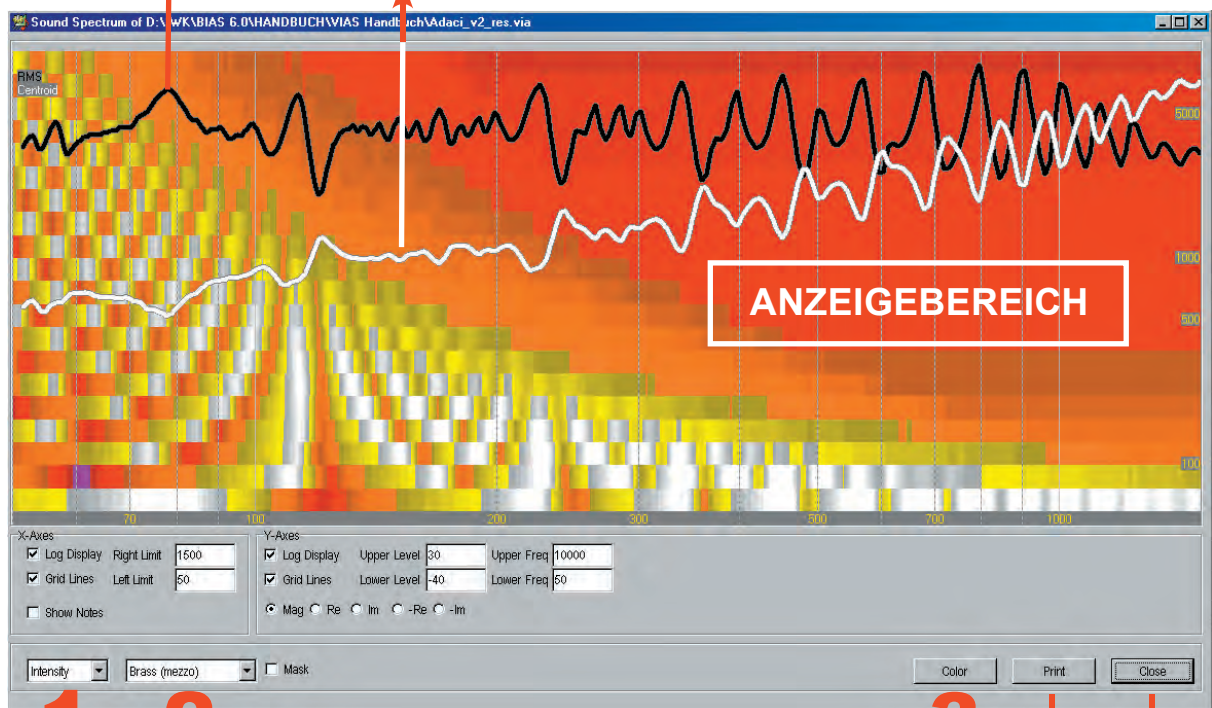
- the sound analysis will only be shown **for one curve**, no matter how many curves you are currently displaying
- the sound analysis for the **last active** curve will be displayed

- **To choose a particular measurement for sound analysis:**

In the "Currently displayed Curves" window click on the speaker icon which belongs to the measurement you wish to analyse. A subsequent single click on the "Sound" button opens the sound analysis window.

RMS: relative dB values of the radiated sound level vs. frequency or note

Harmonic Centroid (sound balance point) for every note (right y-axis)



1 2

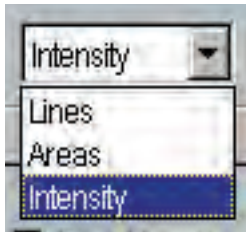
3

prints the contents of the display area

closes the sound analysis window

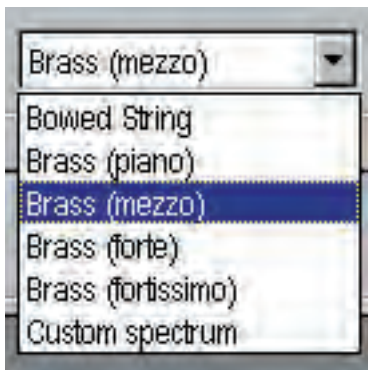
ANALYSES - SOUND ANALYSIS

1

**Display Style**

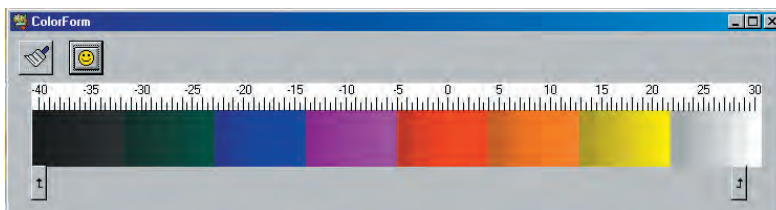
You can choose between a line spectrum ("Lines"), an area spectrum ("Areas") or a spectrogram ("Intensity").

2

**Sound Formation**

With this drop down menu you can choose the number and weight of the harmonics in the produced sound.

3

**Colour Coding**

Using the two arrows, you can assign colours to the different intensity degrees (in dB).

Axes scaling for the Display Area:

Y-Axis: Left: Partial from bottom to top (the shown picture depicts 20 partials)
Right: Frequency in Hz, logarithmic

X-Axis: Frequency in Hz or musical note
Linear or logarithmic

NOTE:

All available setting fields, e.g. Mag, Re, Im, etc., always work, but it does not always make sense to use them.

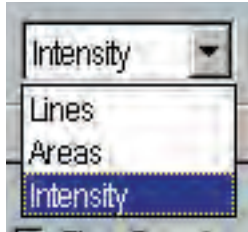
"**Mag**" displays the magnitude spectrum,

"**Re**" displays the real active component (= radiated or dissipated part),

"**Im**" displays the imaginary reactive component (= part that is not radiated or dissipated)

ANALYSES - SOUND ANALYSIS

1



Here you can choose between three **different display options** for the analysed virtual sound of the instrument.

Lines:

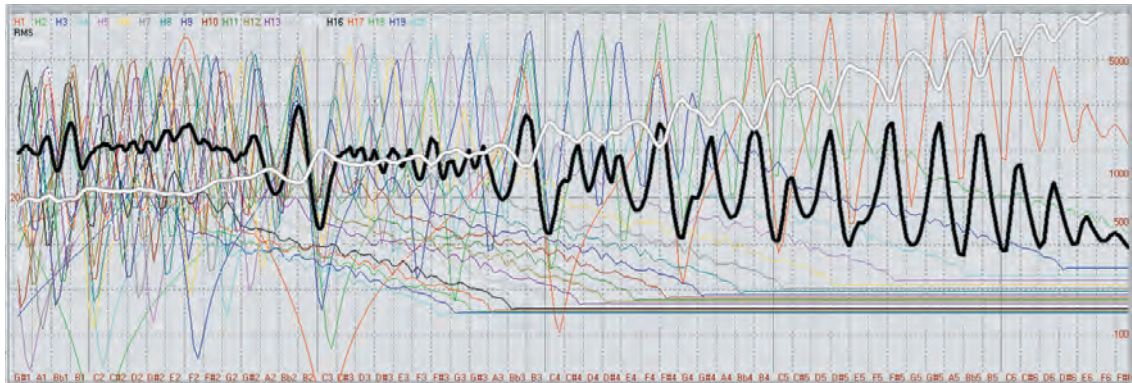
X-axis: Frequency in Hz (linear or logarithmic) or musical note

Y-axis (left): Strength of the partial in dB (relative unit)

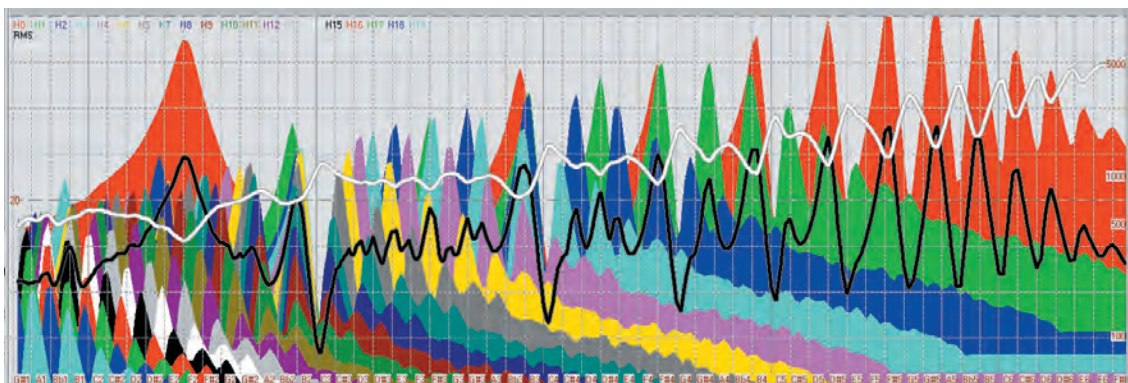
Y-axis (right): Frequency in Hz (logarithmic) of the harmonic centroid

The lines represent the contribution of each partial in the total spectrum for each frequency point or musical note.

The legends can be found on the top left corner of the display area. H1 specifies the colour of the first partial (=fundamental), H2 of the second partial, etc.

**Areas:**

Same as with “Lines”, but a colour area is displayed for each partial, instead of just a line. Higher partials are displayed on top of lower ones.



ANALYSES - SOUND ANALYSIS

Intensity:

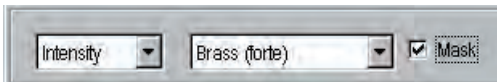
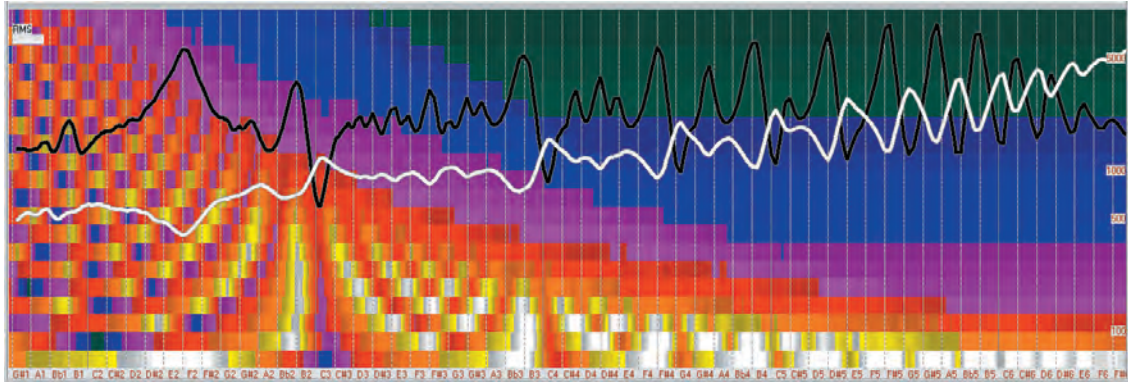
X-axis: Frequency in Hz (linear or logarithmic) or musical note

Y-axis (left): Partial (from bottom to top). First band = First partial, second band = second partial, etc.

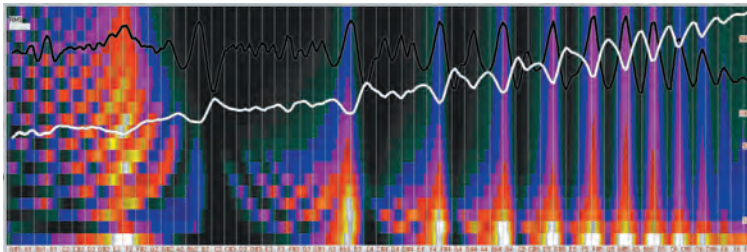
Y-axis (right): Frequency in Hz (logarithmic)

The amplitude is colour coded.

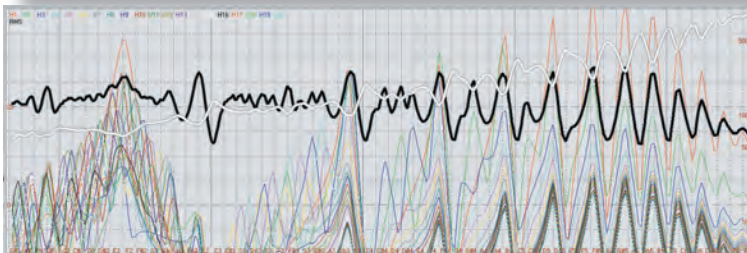
Colour grade step= 1 dB, dark means low amplitude, bright means high amplitude.



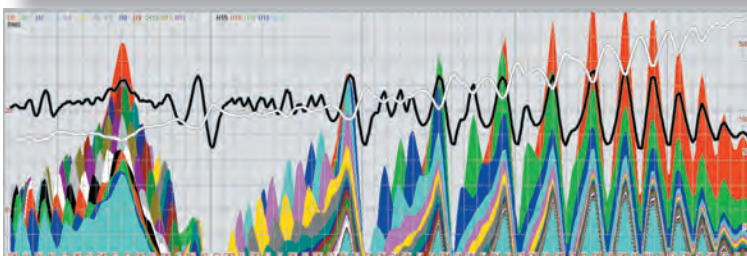
- The graphs show the amplitude of each harmonic for **every** frequency point. But in reality that is irrelevant, as notes can only be played where the impedance peaks lie. To show only the relevant areas, check the box "**Mask**": This will filter the sound plot through the impedance curve.



Intensity – masked



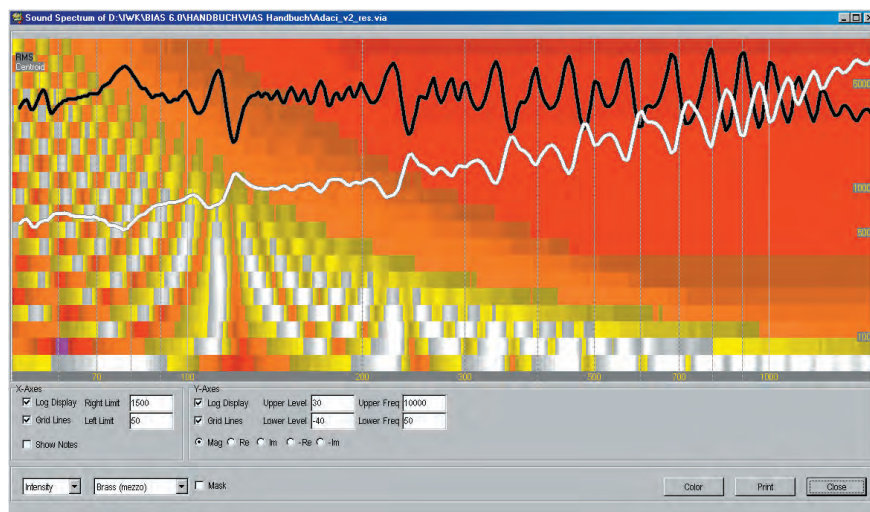
Lines – masked



Areas – masked

ANALYSES - SOUND ANALYSIS

Sound level = black curve
Harmonic centroid = white curve



Sound Level

The black curve displays the sound level radiated by the instrument, for every frequency, or for every note if the box "Show Notes" is checked.

Harmonic Centroid

The white curve displays the harmonic centroid for every frequency or note. The harmonic centroid determines the sound balance point. For example, if the lower frequencies or the fundamental are pronounced then the harmonic centroid is located in a lower frequency, and vice versa.

The frequency value of the harmonic centroid can be determined from the right Y-axis of the display area.

ANALYSES - SOUND ANALYSIS

2



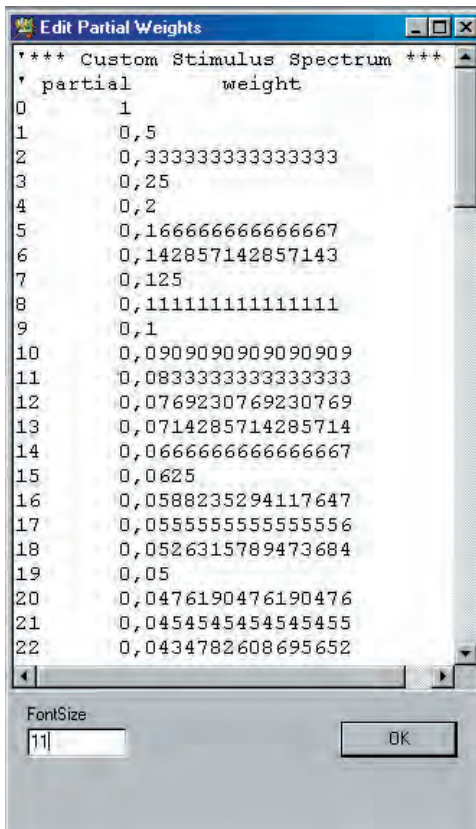
It is well known that the **sound of a brass instrument** strongly depends on the dynamic at which it is played. Here you can choose between different dynamics.

The radiated sound of an instrument can be calculated using its "Transfer function" and the excitation spectrum: The radiated sound is the result of the excitation spectrum filtered through the Transfer function of the instrument.

As the individual excitation spectrum depends on the physiology of the player, it is important that its assumptions are as general as possible.

The radiated sound of a brass instrument is calculated by modeling the excitation spectrum with the weighing curves implemented in BIAS (see "Weighing" in the BIAS user's manual).

The radiated sound of a bowed string instrument is calculated using an excitation spectrum that corresponds to the well known saw tooth waveform.



- To be able to generate a sound that is as natural as possible, a "Custom spectrum" function is available, whereby an **individual excitation spectrum** can be modeled.

The left column corresponds to the harmonic number, and the right column is the weight of that harmonic in the sound spectrum.

You only need to edit the given text guide.

1. The strongest harmonic has a weight of 1 (NOTE: The strongest harmonic does not have to be the fundamental, as it is shown in the example).

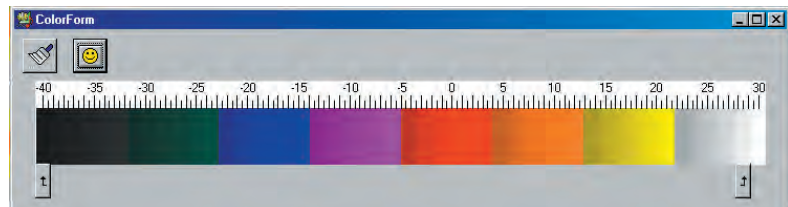
2. The amplitudes of all other harmonics are given as a percentage of the strongest harmonic.

Example in the left picture: The amplitude of the 3rd harmonic constitutes a fourth of that of the strongest harmonic (in this case, the fundamental). Therefore, its weight is 0.25. The amplitude of the 19th harmonic constitutes 5% of that of the strongest partial. Therefore, its weight is 0.05.

If for instance, the desired excitation spectrum consists of only 20 harmonics, the weight of the harmonics higher than 20 must be "0".

ANALYSES - SOUND ANALYSIS

3



Click on the brush icon to generate a colour gradient between the two colour fields marked with the arrows. The colour of all the fields that lie in between these two marked fields will be overwritten.

- The arrows can be dragged and dropped with the mouse
- Each field can be colourised individually. Click on the desired field to open a colour dialog window
- The typical procedure is as follows:
 1. Place the arrows on the start and end of the colour gradient
 2. Colorise the fields marked by the arrows
 3. Click on the brush to generate the colour gradient in the marked interval



The "Smiley" icon restores the standard colour gradient.

ANALYSES - SOUND ANALYSIS

Remarks on making the data “audible”

The audible sounds generated by VIAS have only the stationary part of the sound, which lacks the characteristic attack and decay parts of the sound of a real instrument. That is the reason why they still sound synthetic and unnatural. The attack and decay can certainly be modeled as well, however, these tasks are highly complex, and the program would be much more complicated to use.

Furthermore, a real musical instrument sound has characteristic noise components produced by the excitation mechanism (bowing noise in bowed instruments, blowing noise in flutes, etc). The program does not simulate these noise components.

The goal of this analysis is not to produce an exact reproduction of how the sound develops in time, in the same way as when the player plays the instrument, but to make the **characteristic sound colour** of the instrument audible. Such data is for instance suitable to compare the sounds generated by different curves.

Of course, the composed "Sound" (i.e. musician + instrument) is strongly influenced by the musician. The excitation signal produced by the musician can weaken or change the characteristic sound colour of the instrument.

Remarks on making the sound "visible"

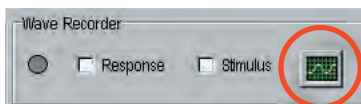
The goal is to present the typical characteristics of the instrument, which in reality can be weakened or changed by the musician.

ANALYSES - COHERENCE

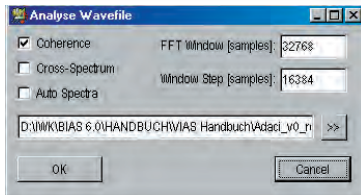
The coherence function is a measure of the degree of linear dependence between two time signals, $x(t)$ and $y(t)$, in the frequency domain. Mathematically, it is nothing more than the square of the magnitude of the average cross-power spectrum and is calculated as follows:

$$\gamma_{XY}^2(f) = \frac{|\langle G_{XY}(f) \rangle|^2}{\langle G_{XX}(f) \rangle \cdot \langle G_{YY}(f) \rangle}$$

If the coherence between an input signal $x(t)$ and an output signal $y(t)$ of a oscillating system in a frequency range is not 1, it is an indication that linear signal analysis theory is not applicable.



- With the coherence button you can check the validity of a VIAS measurement:



- Record the "Response" with the wave-recorder (see page 22)
- Click on the shown icon to start the coherence function
- Load the recorded response-signal and click on the "OK" button

Set the window length (in samples) in the field marked **FFT Window**.

Set the step size (also in samples) in the field marked **Window Step**

You can find more information about Coherence in Wikipedia, under the following link:

http://de.wikipedia.org/wiki/Koh%C3%A4renz_%28Signalanalyse%29



If you are reading the PDF version of this user's manual, simply click on the Internet Explorer icon (left) to go to this webpage.

ANALYSES - CROSS SPECTRUM

To be able to calculate the coherence spectrum, it is necessary to calculate as intermediate results the **cross power spectrum** as well as the **auto power spectrum**. These results can also be plotted by clicking the corresponding boxes.

The cross power spectrum is calculated by using the following equation:

$$S_{XY}(f) = X^*(f) \cdot Y(f)$$

The equation to calculate the auto power spectrum is as follows:

$$S_{XX}(f) = X^*(f) \cdot X(f) = |X(f)|^2$$

In both equations, the star sign (*) denotes complex conjugate.

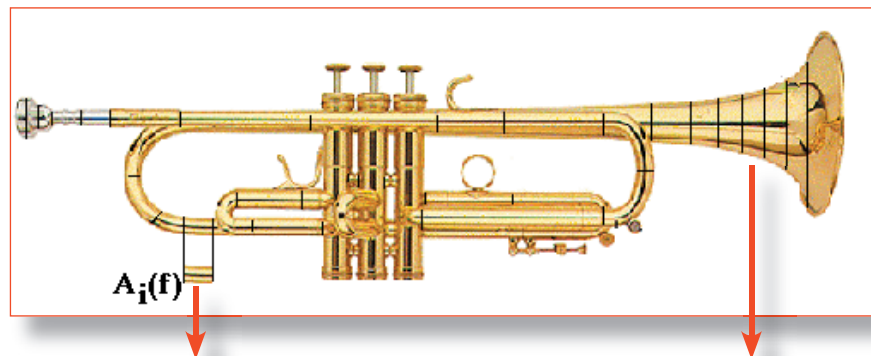
ANALYSES - SIMULATION

The input impedance calculation is done using a Transmission Line model, as described in the paper by Dan Mapes Riordan [Horn modeling with conical and cylindrical transmission line elements, Journal of the Audio Engineering Society, Vol. 41, Nr. 6, Montclair/NJ: AES, 1993, pp. 471-483].

This model is based on electric analogies that are so often used in acoustics. The instrument is seen as a lossy electric transmission line. The sound pressure corresponds to the electric voltage and the sound flow to the electric current. Viscothermal losses are modeled with a corresponding complex resistance.

The transmission line is terminated with a termination impedance (Z_t), which causes a partial reflection of the waves in a real instrument.

To perform the calculation, the bore of the instrument is first represented by a series of conical and cylindrical sections. The acoustical transmission properties of these simple elements are given by frequency dependent (f) matrices ($A_i(f)$).



Example of a cylindrical section

Example of a conical section

The impedance of such an element is:

$$A_i(f) = \begin{bmatrix} a_{i11}(f) & a_{i12}(f) \\ a_{i21}(f) & a_{i22}(f) \end{bmatrix}$$

Pressure (p) and volume flow (u) before and after the i th element are related as follows:

$$\begin{bmatrix} p_i(f) \\ u_i(f) \end{bmatrix} = A_i(f) \begin{bmatrix} p_{i+1}(f) \\ u_{i+1}(f) \end{bmatrix}$$

The impedance (Z_i) at i is calculated with

$$Z_i(f) = \frac{p_i(f)}{u_i(f)}$$

The transmission properties of the whole instrument are obtained by calculating the product of all matrices: $A(f) = \prod_i A_i(f)$

ANALYSES - SIMULATION

$$a_{11} = \frac{x_1}{x_0} \left[\cosh(\Gamma L) - \left(\frac{1}{\Gamma} \right) \sinh(\Gamma L) \right]$$

$$a_{12} = \frac{x_0}{x_1} Z \sinh(\Gamma L)$$

$$a_{21} = \left[\left(\frac{x_1}{x_0} - \left(\frac{1}{\Gamma x_0} \right)^2 \right) \sinh(\Gamma L) + \left(\frac{\Gamma L}{(\Gamma x_0)^2} \right) \cosh(\Gamma L) \right]$$

$$a_{22} = \frac{x_0}{x_1} \left[\cosh(\Gamma L) - \left(\frac{1}{\Gamma x_0} \right) \sinh(\Gamma L) \right]$$

Matrix elements of one section

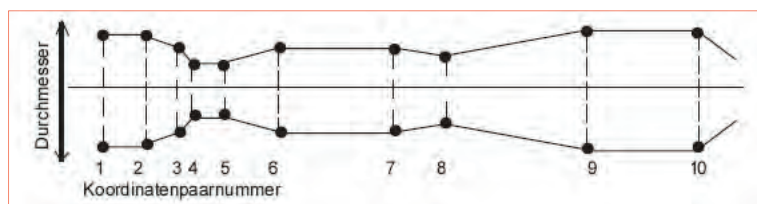
$$Z = \frac{\rho c}{\pi R^3} \left(\frac{(k R)^2}{4} + j0.613 \mathfrak{X}(k R) \right)$$

Termination impedance

● To create a list with bore dimensions:

For each element, write an X value (= distance from a reference point, e.g. the outer edge of the mouthpiece) and a Y value (diameter at this point), both in mm. The x and y values should be separated by an empty space or a tab.

Example:

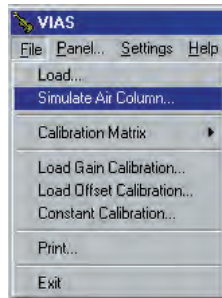


The list for this geometry would read as follows:

Nr.	X Value	Y Value
1	0	10
2	10	20
3	18	14
4	21	10,78
5	29	10,78
6	49,3	13,6
7	78	13,6
8	88	12,1
9	91	20,6
10	111	20,6

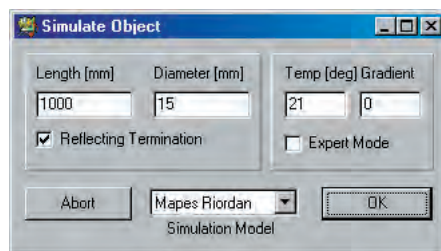
ANALYSES - SIMULATION

- To calculate the input impedance of a tube with arbitrary bore profile and length:



On the menu "File" (located on the top left corner of the main window), click on "Simulate Air Column..."

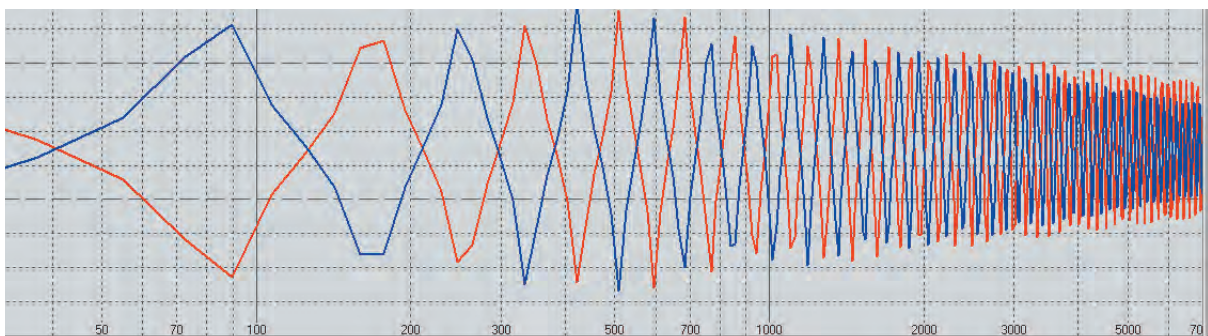
A window will open (see below), where you can calculate the impedance of a cylinder of arbitrary length and diameter.



The simulation window is already filled with default values. If you click on the OK button, the impedance of a closed cylindrical pipe of 1 m length and 15 mm diameter, and a temperature of 21°C is calculated and displayed.

After entering the length and diameter, you can enter the **air temperature in °C (degrees Celsius)**.

If the **Reflecting Termination** box is checked, the impedance will be calculated with a closed end (complete reflection). If not, the radiation impedance will be that of an open cylindrical duct terminated by an infinite flange.



Blue curve: Impedance of an open ended pipe ("Reflecting Termination"). Red curve: closed ended pipe. Both curves correspond to a pipe of 1 m length and 15 mm diameter.

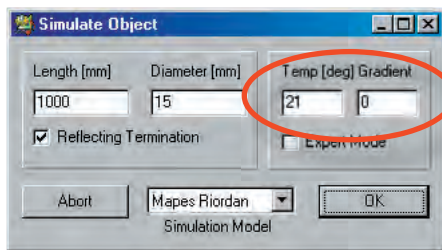
ANALYSES - SIMULATION

● **To take the temperature into account:**

The ambient air temperature is of importance because it influences the speed of sound, and therefore the tuning of the instrument.

There are three important factors:

- The room temperature, e.g. 21 °C
- the breath temperature, approx. 36 °C
- the temperature gradient (how much the air cools down in the instrument)



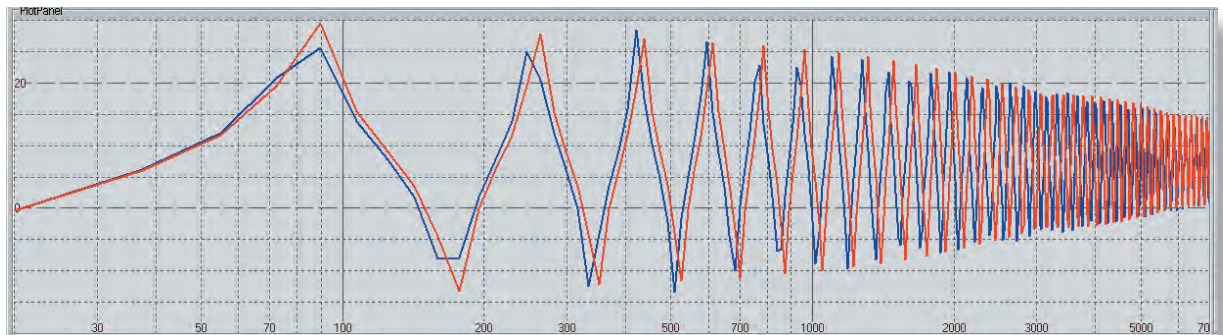
You have two options:

1. If you enter the temperature (**Temp [deg]**) and set the **Gradient** to "0", the impedance for the room temperature, e.g. 21 °C, is calculated.
2. If you enter a value in the field "**Gradient**" then the value in the field Temp [deg] denotes **breath temperature** and **not room temperature!**

The value of the gradient determines how many degrees (°C) the temperature in the instrument drops per metre (e.g. a gradient of 2.5 means that the temperature of the instrument drops by 2.5°C per metre).

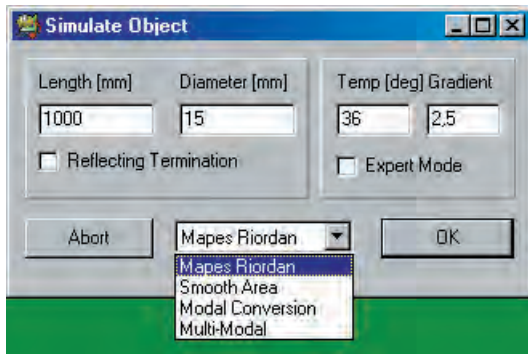
The gradient depends, on the one hand, on the difference between breath and room temperature, and on the other hand, on the thermal conductivity and thickness of the walls of the instrument.

It is well known that the temperature influences the tuning of an instrument. However, there are no scientific studies about it. In the future, our research group will conduct research in this area to provide reliable and useful values.



Blue curve: Impedance for a room temperature of 21 °C without gradient. Red curve: Impedance for a breath temperature of 36 °C and a gradient of 2,5 °C per metre. Both curves correspond to a pipe of 1 m length and 15 mm diameter.

ANALYSES - SIMULATION

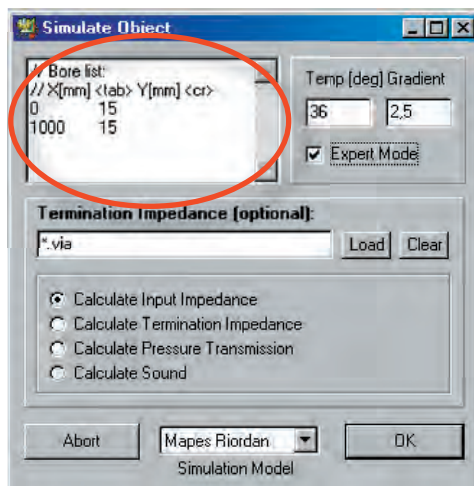


You can choose between different simulation models with the drop-down menu "Simulation Model".

So far, only the Mapes Riordan model has been implemented!

Our goal is to implement a multi-modal model which, additional to the fundamental frequency, also takes the higher harmonics into account.

- To model a more sophisticated geometry (like the bore profile of a real instrument) with varying diameter, check the box "**Expert Mode**".



Enter the bore dimensions of the piece to be simulated in the editor field shown in the picture.

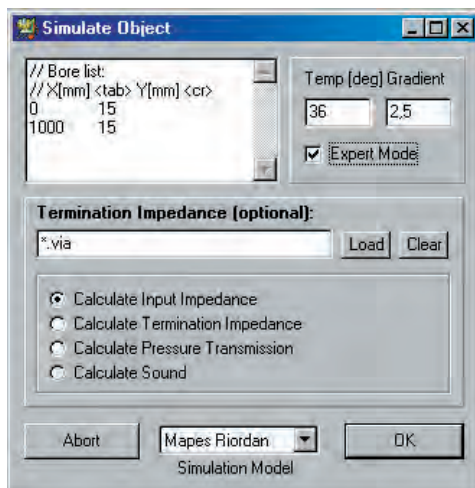
Recommendation

Create the list of bore dimensions in a text editor or in a spread sheet, and then paste it into the text field.

Whether you need to type a comma or a full stop (English setting) as a decimal point depends on the country settings of your Windows.

In German speaking countries a comma is used while in English speaking countries, a full stop is used. If necessary, open the file in a text editor and use the function "Search and replace".

ANALYSES - SIMULATION



- Further simulations in Expert Mode

Calculate Input Impedance: With this option, the input impedance curve of an instrument with the user specified bore profile is calculated – in the same way as if it was being measured

Type the bore dimensions in the editor field, and click on the “OK” button.

If you have already calculated and saved the termination impedance, you can load it before performing the calculation. If not, the model implemented in BIAS will be used (therefore this field is optional).

The termination impedance specifies essentially the acoustical properties of the instrument.

The implemented model is good, but is also subject to improvement. Work is being done in this area.

The formulae of the model are presented in pages 46-47.

Calculate Termination Impedance: With this option, you can calculate the termination impedance from the given bore profile. If you already have calculated and saved the impedance curve of the instrument described by the bore dimensions specified in the editor field, you can load it to get a more accurate calculation. If not, the implemented model based on the radiation through the end cross section specified in the editor field will be used.

Calculate Pressure Transmission: With this option, the **transfer function** of the instrument is calculated – the relationship between the sound pressure in the mouthpiece and the sound pressure at the end of the instrument. Put it another way, what the instrument does with a pressure excitation signal. In this case, it is necessary to load an input impedance curve!

Calculate Sound: *The curve displayed with this option is the result of calculating the product of the input impedance times the transfer function (pressure transmission), and is essentially an intermediate step in the sound calculation. Loading an input impedance curve (either calculated or measured) is required.*

$$\begin{array}{c} \text{Sound} \\ = \\ \text{Input impedance} \\ \\ \times \\ \\ \text{Transfer function} \\ \text{(Pressure Transmission)} \end{array}$$

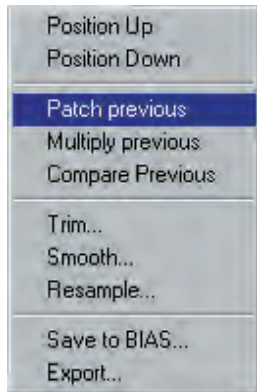
As the pitch of the instrument depends on the velocity of sound, which in turn depends on the air temperature, the latter must be entered in degree Celsius.

DATA POST PROCESSING

ERROR CORRECTION

To correct measuring errors:

Right click on any icon corresponding to the curve that you would like to paste into the previously listed curve. The context menu will open. Click on the command "Patch previous" (the current curve will be inserted in the previous curve).

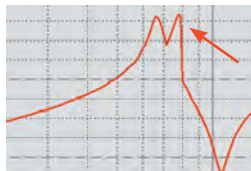


Currently Displayed Curves						
	Date	Time	Points	Line	File Name	Description
	26.07.07	14:11:39	405	—	D:\WIK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" kein Ventil gedrückt
	26.07.07	14:50:22	410	—	D:\WIK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 2. Ventil
	26.07.07	14:26:04	410	—	D:\WIK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 3. Ventil

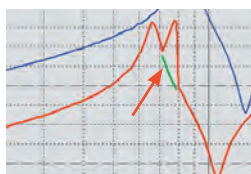
NOTE:

With the "Patch previous" command, the data in the previously listed curve is replaced with the data from the current curve. Any curve which needs to be improved, must **always be listed just before** the curve which will be used to improve it.

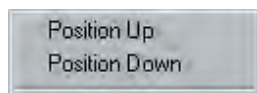
Example:



The red curve (see picture) has an error between 70 and 80 Hz (second peak). You have repeated the measurement only in this frequency range, which is correct (green curve).



You would like to improve the red curve with the green curve. For this purpose, **the red curve must be listed just before the green curve**. As this is not the case (see picture), you will need to either shift the green curve upwards or the red curve downwards.



This is done with the command "Position Up" or "Position Down" in the context menu. Let us shift the green curve upwards:

410		D:\WIK\BIAS
§		D:\WIK\BIAS
405		D:\WIK\BIAS

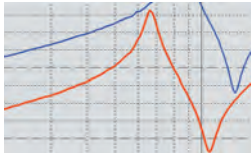
- Right click on e.g. The speaker icon that corresponds to the green curve
- Click on "Position up"

Now the curves are in the required order.

DATENBEARBEITUNG - Post Processing

ERROR CORRECTION

Now we can perform the error correction:

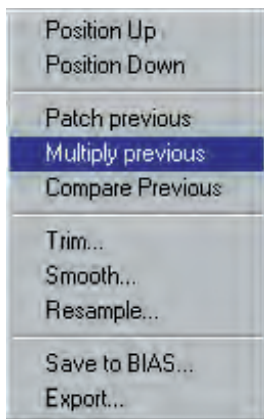
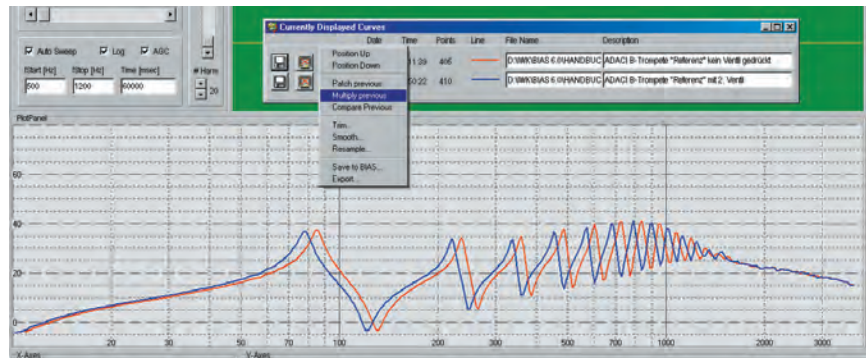


- Right click on any icon that corresponds to the green curve
- Click on "Patch previous"

Done!

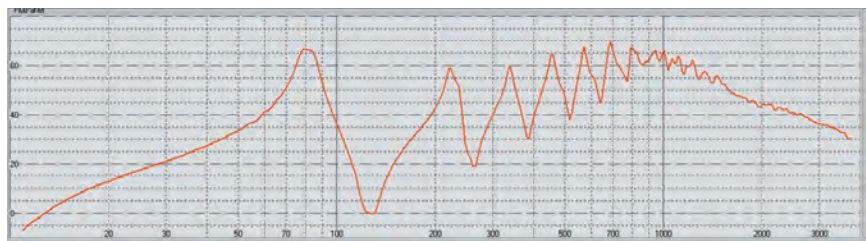
DATENBEARBEITUNG - Post Processing

CURVE MULTIPLICATION - ADDITION



To multiply the red curve times the blue curve:

- Right click on any icon that corresponds to the blue curve
- Click on the command “Multiply previous”

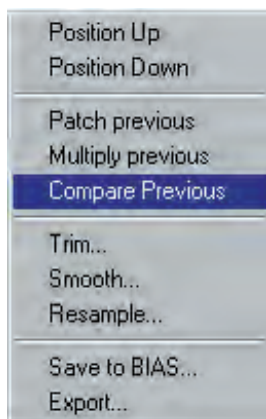


RESULT

For the commands “Multiply” and “Compare”, the two curves do not have to have the same length, or frequency range, or sampling frequency.

DATENBEARBEITUNG - Post Processing

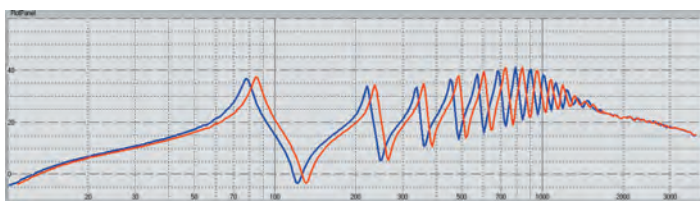
CURVE COMPARISON - DIFFERENCE



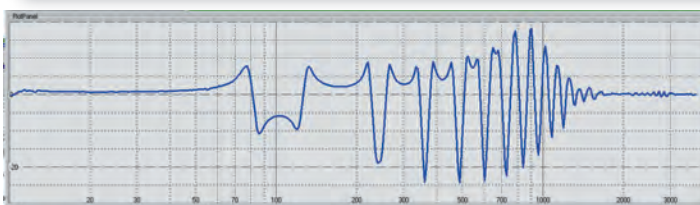
To compare the red curve with the blue curve:

- Right click on any icon that corresponds to the blue curve
- Click on “compare previous”

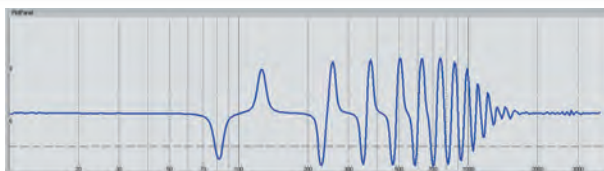
Curves to be compared



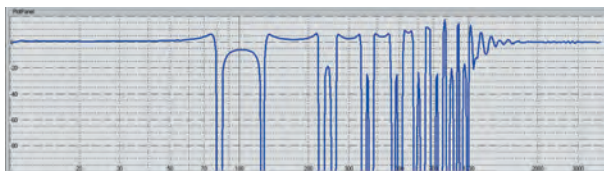
Result



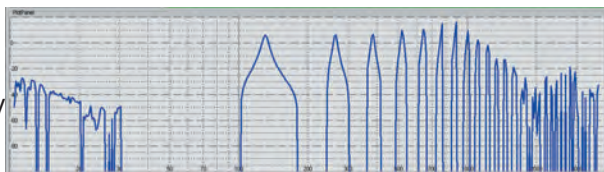
Result – phase



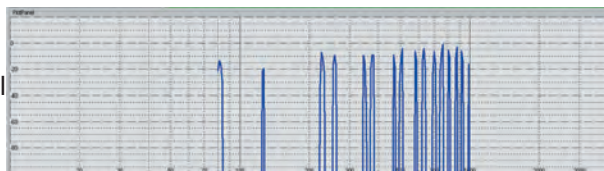
Result – Real part



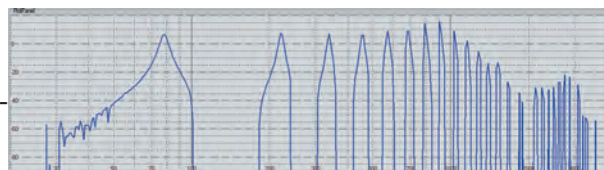
Result – imaginary part



Result – negative real part

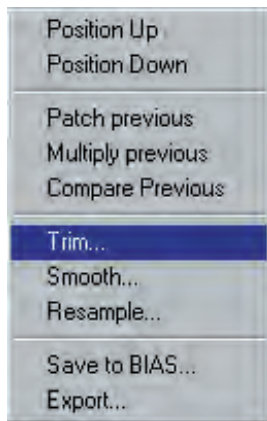


Result – negative imaginary part



DATA POST PROCESSING

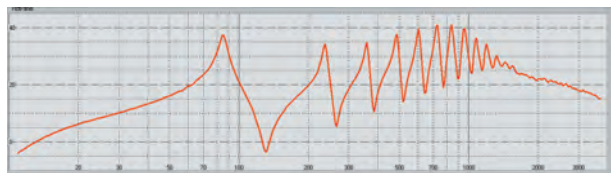
CURVE TRIMMING



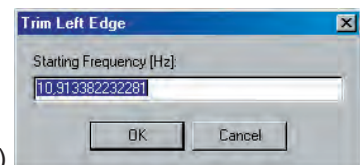
- To crop a curve, use the command “Trim” on the context menu

All data under the start frequency and over the stop frequency will be cut off

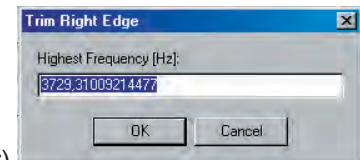
Curve to crop



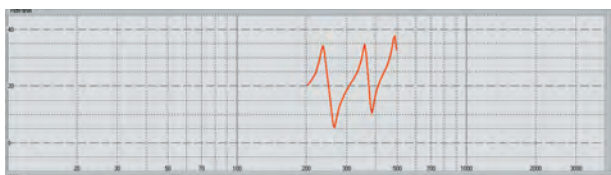
- Type in the lower frequency (e.g. 200 Hz)



- Type in the upper frequency (e.g. 500 Hz)



Result

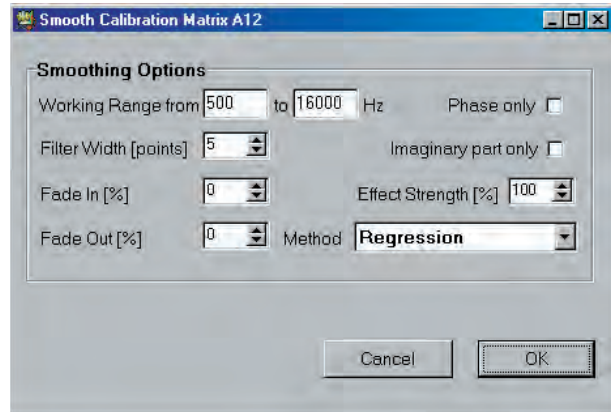
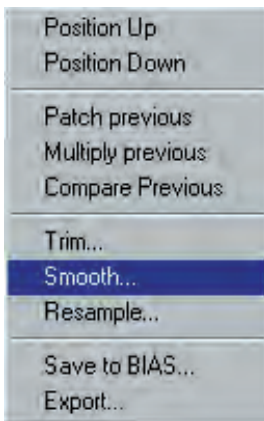


DATA POST PROCESSING

CURVE SMOOTHING

There are many reasons to smooth a curve: For instance, when detecting maxima / minima points by pressing the arrow keys does not work properly, or when the curve has got ripples, etc.

- **To define the smoothing parameters:**
In the context menu, Right click on the command “Smooth”.

**Parameters:**

Working Range: Here you can specify the frequency range where the smoothing will take place.

Filter with [points]: Filter (smoothing window) width

Fade in [%] and **Fade out [%]:** If you would like to avoid having an abrupt transition between the unsmoothed and the smoothed parts of the curve, you can specify a transition range. The value 10 means e.g. that the transition range amounts to be 10% of the working range. In the picture above, it would be a width of 1,550 Hz.

Phase only and **Imaginary Part only:** The smoothing parameters will only be used for smoothing the phase or the imaginary part.

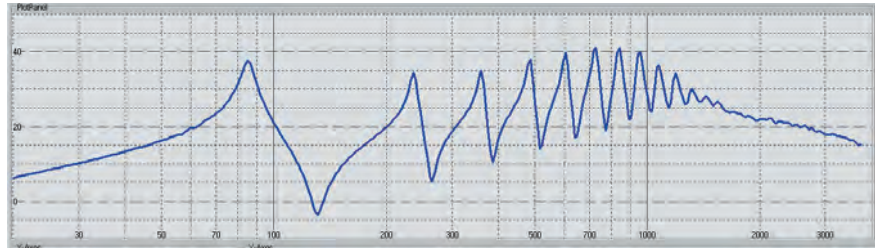
Effect Strength [%]: E.g. 50% means that the curve in the working range consists of a 1:1 mixture of filtered and unfiltered values.

Method: Here you can choose the smoothing method: “Running average”, “Gauss-filter” and “linear regression” methods are available.

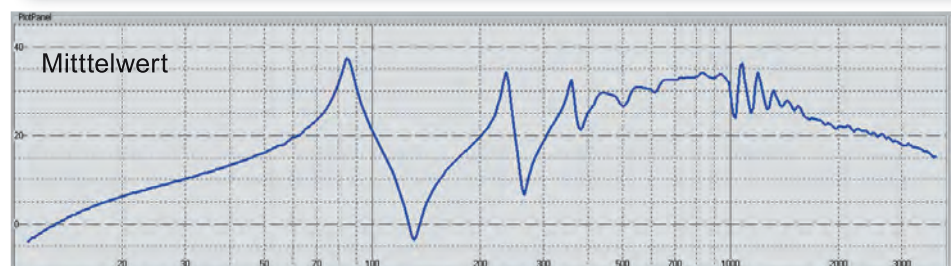
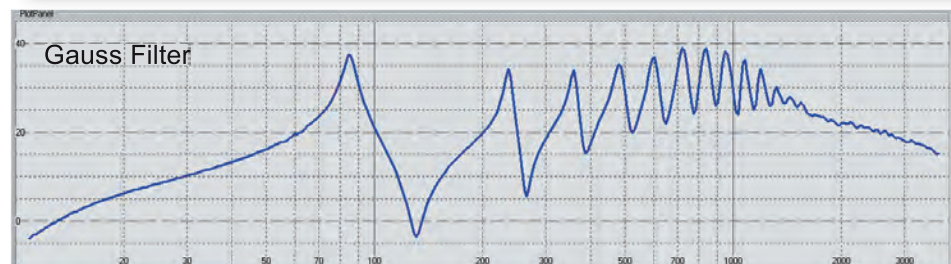
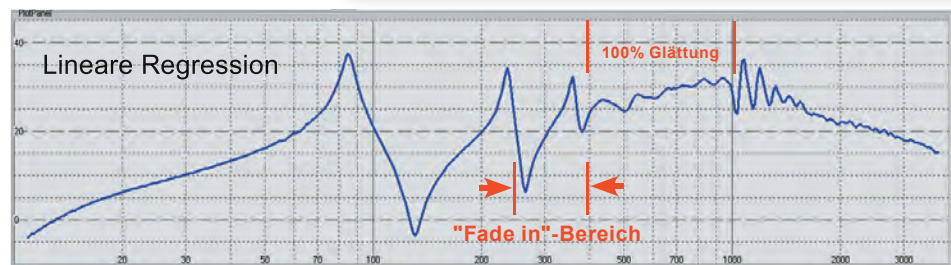
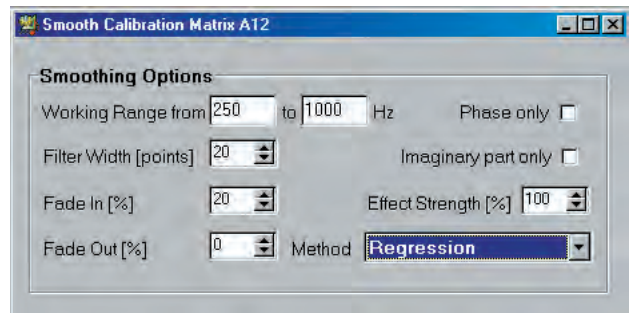
DATA POST PROCESSING

CURVE SMOOTHING

Here is a radical example:

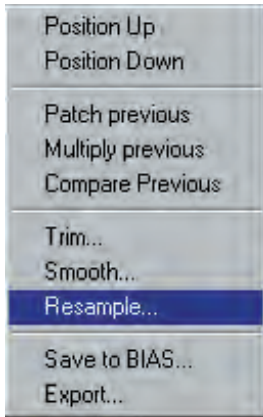


Working range: 250-1000 Hz
 Filter width: 20 points
 Fade in range: 150 Hz
 No Fade out
 Method: Linear regression

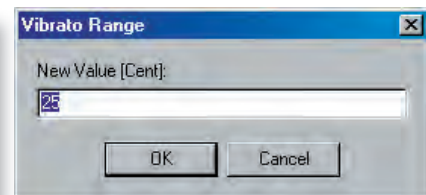
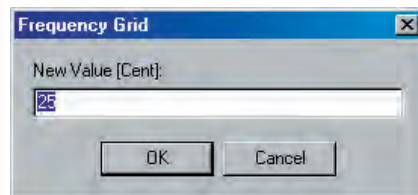


DATA POST PROCESSING

CURVE RESAMPLING



In some cases, having a higher or lower resolution than what was actually measured is required. This can be changed with the command “resample” in the context menu. Two parameters are required: GRID and VIBRATO RANGE.



GRID: With this parameter, the total number of points in the curve (hence the resolution), as well as the distance between the points in the frequency axis can be changed.

EXAMPLE:

We have a measurement from 100 Hz to 7 kHz with 600 points.

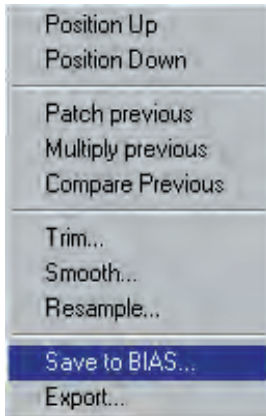
- This measurement encompasses approximately 6 octaves:
1st octave: 100-200 2nd octave: 200-400 3rd octave: 400-800
4th octave: 800-1600 5th octave: 1600-3200 6th octave: 3200-6400
- One octave has 1200 cents, and has 100 points in this measurement
- Therefore, 1 point corresponds to 12 cents
- A new grid of 25 cents per point would have half the resolution, and the curve would have 300 points
- If in this case, a value of 6 cents per point were selected, the curve would have 1200 points.

VIBRATO RANGE: Any new non-existent point must be interpolated. In this case, 25 cents means that the interpolation will take into account all points that fall within 25 cents of the new interpolated point, and not only the two closest neighbours.

Engineers need to get used to the unit **Cent** instead of Hz. But in this case the terminology is musically relevant, as the logarithmic scale corresponds to the way pitch and musical intervals are perceived.

It is for this reason that the X-axis uses the unit cent. This ensures that the frequency resolution corresponds to how we hear.

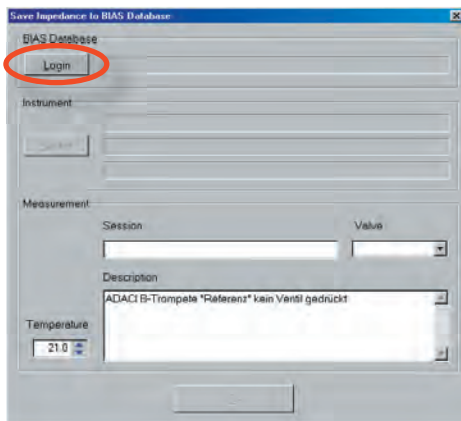
EXPORTING MEASUREMENTS



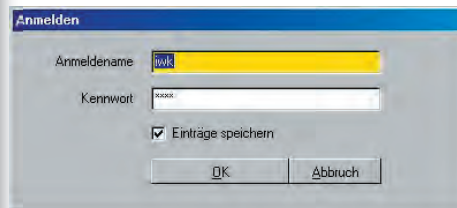
● **Export directly to the BIAS 6.0 database**

Every measurement can be exported directly into the BIAS database, with the help of the integrated OLE functions. The format and sampling rate are automatically adjusted to match the BIAS format.

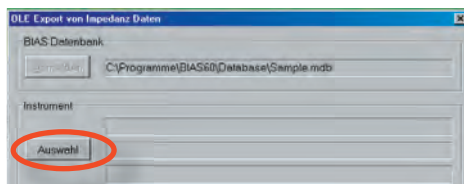
Click on **“Save to BIAS...”** to open the following window:



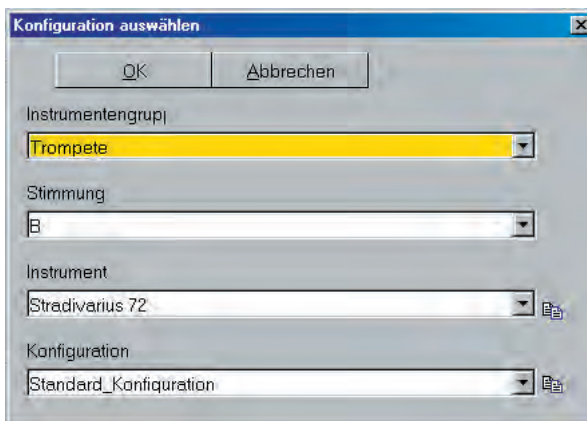
Click on “Login” and sign in to BIAS. The current BIAS database will open in the background. Here you have to perform specific steps, to integrate the measurement in the BIAS database correctly.



BIAS Sign-in window



After you have successfully signed in to BIAS, the database that was used last will be open, and you will be prompted to choose the instrument type and the intonation. See picture on the bottom left.



At the same time, all the available instruments will be listed in the “Instrument” drop down menu. Choose the instrument that belongs to the measurement you are exporting.

If you would like to create a new instrument type, click on the icon that is displayed beside the drop down menu.



It is advisable to use the Standard Configuration for this purpose.

EXPORTING MEASUREMENTS

To create a new instrument:

Type any instrument name

Type the year in which the instrument was manufactured. Click on the blue arrow to insert the current year.

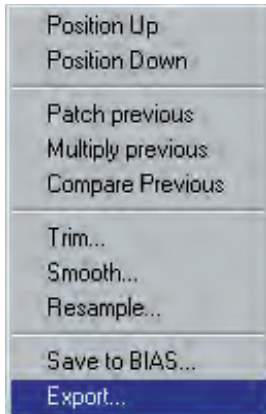
Type the name of the owner, or click on the blue arrow to choose from a list

Type the name of the manufacturer, or click on the blue arrow to choose from a list

The only mandatory field is that of the instrument name, all other fields are optional. There is also the possibility of importing a photograph of the instrument.

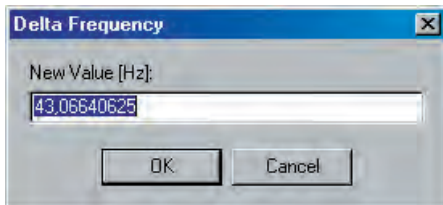
It is advisable to type as much information as possible, e.g. in the "Description" field, as with time, useful information tends to get lost.

EXPORTING MEASUREMENTS

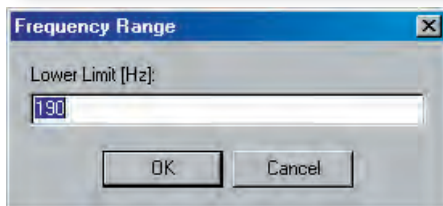


- To export a measurement in .dat format (to use with other applications):

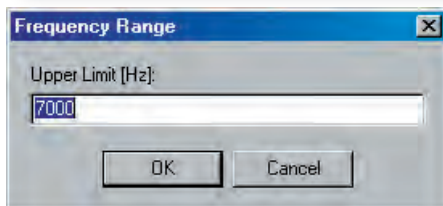
Right click on any icon of the curve to be exported in the “Currently displayed Curves” window to open the Context Menu, and click on the command “Export”.



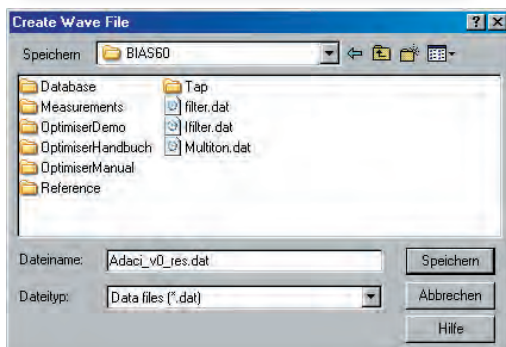
The value given in this field specifies the frequency grid, e.g. a value of 0.5 means that an impedance point will be written every 0.5 Hz



Here you specify at which frequency the curve should begin.



This value specifies the end frequency of the curve



Choose the name (and extension) of the file and the directory where it will be saved.

EXPORTING MEASUREMENTS - SUMMARY

Which alternatives are there to export data, and to use it with other applications?

It is possible to:

- Export a measurement in the BIAS 6.0 database (see page XXX)
- Save a measurement in .dat format to be used e.g. In excel (see page XXX)
- Export the maxima and minima points of a curve with frequency, pitch and pitch deviation from an equally tempered scale, and relative impedance values with the help of the NumDisplay command (see page XXX)
- Make all the points of a curve available in text format with the help of the NumDisplay command:
 - Set the detection threshold to the value "0"
 - Copy all the content of the window, and paste it in e.g. an empty excel spreadsheet (see page XXX)
- Select the values shown by the cursor. All current values of the cursor will be inserted automatically into the windows clipboard.
 - Open a text editor and paste the current values with Ctrl-V (see picture below and page XXX)

HINT:

A VIAS file is a text file with a header and 5 columns with the following values:

1. Frequency
2. Magnitude
3. Phase
4. Real part
5. Imaginary part

It is possible to read any VIAS measurement with any text editor for further work.

The screenshot shows a text editor window titled 'Unbenannt - Editor' with a menu bar (Datei, Bearbeiten, Format, Ansicht, ?) and a table of data. Red arrows point from labels below to specific columns in the table.

302,326Hz	D#4-50 Cent	31,948	19,009	21,750	31,948
338,388Hz	E4+45 Cent	17,232	26,299	33,501	17,232
360,024Hz	F#4-47 Cent	21,840	34,904	10,767	21,840

Labels and arrows pointing to columns:

- Frequency (points to the first column)
- Pitch and pitch deviation (points to the second column)
- Value 1st curve (points to the third column)
- Value 2nd curve (points to the fourth column)
- Value 3rd curve (points to the fifth column)
- Value 4th curve (points to the sixth column)

SETTINGS - LOCK-IN FILTER

Here you can change, when appropriate, the frequency detection parameters for the display and lock-in filter.



Tuning

minFreq
10

maxFreq
16000

minLev
-60

minDet
0,6

LogAbsTol
0,3

LogRelTol
0,5

minFreq and **maxFreq:**

The frequency range in which the signal detection of the lock-in filter will operate is set in these two fields.

minLev: Specifies the minimum level in dB. Levels below this will not be detected.

minDet: In the example shown, if more than 60% of the captured periods are consistent, the value will be shown on the graph, otherwise the display of the measured curve will be disabled, until the given number of periods are again consistent. For fast sweeps, choose a lower value.

LogAbsTol: The entered value must be between 0 and 1. It determines the frequency starting range.

LogRelTol: In a logarithmic sweep, the frequency distance between two consecutive measuring points should not vary arbitrarily. For very fast sweeps, choose a higher value.

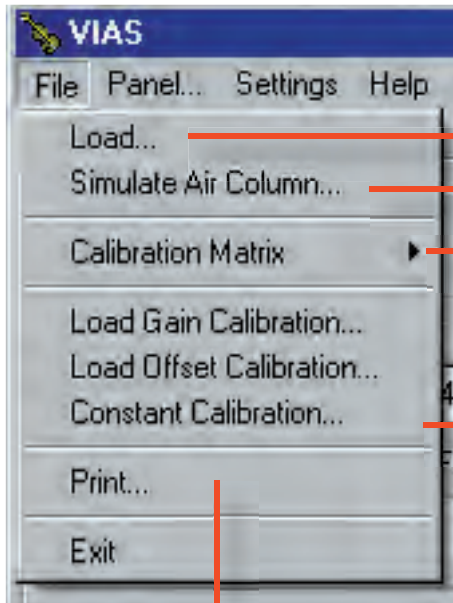
Absolute tolerance is related to frequency differences in Hz, whereas relative tolerance is related to frequency differences in percentage.

To deactivate the lock-in filter:

- Choose "Settings" from the context menu
- -> Filter

COMMANDS IN THE MENU BAR

The “File” menu:



Load data

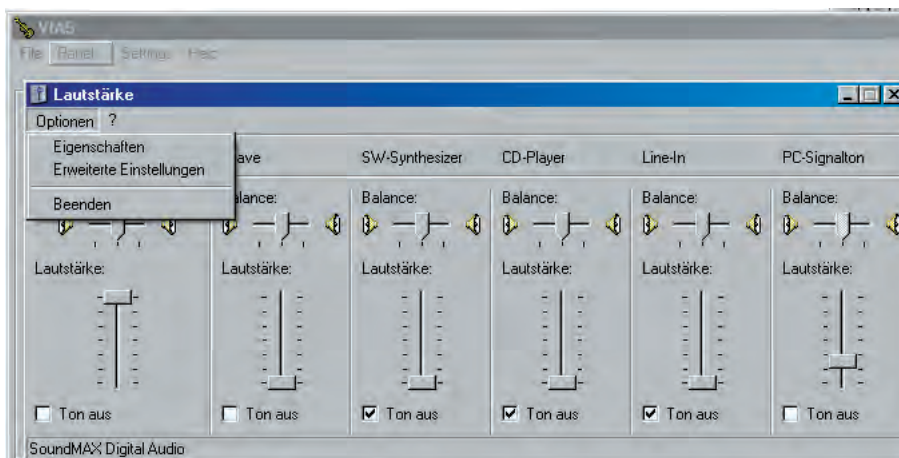
Start the “Physical modeling” tool (page XXX)

These commands are used to calibrate the BIAS measuring head. The first time that VIAS is run, the BIAS default calibration data will be automatically loaded. **We suggest you keep this data!** Details on how to calibrate are found at the end of this user’s manual.

With help of the VIAS program, it is also possible to obtain calibration data from two high precision (uncalibrated!) measurements of two known volumes, which can be later exported into BIAS.

This procedure requires some experience, and will be described in another section.

Prints the content of the display area with legends.



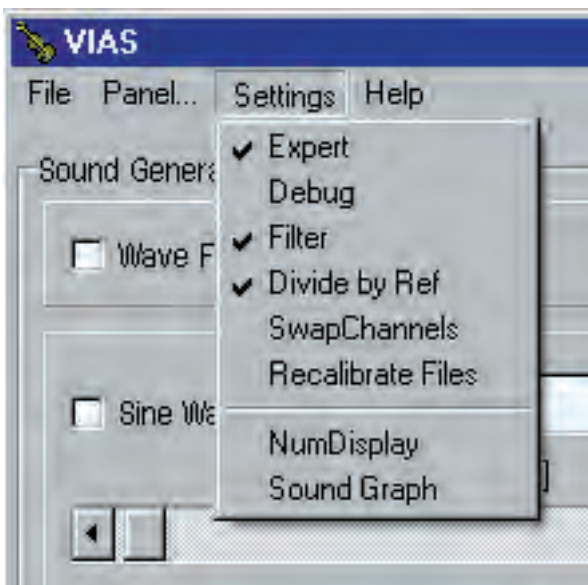
The “Panel...” command:

This command opens the Windows sound card window, where you can change the settings of your sound devices.

COMMANDS IN THE MENU BAR

The “Settings” menu:

In this menu, several operations can be turned on/off, so that other measuring settings can be used as well.



The picture shows the standard settings.

Expert enables access to all functions of the program

Debug opens a debug window, where intermediate data is shown

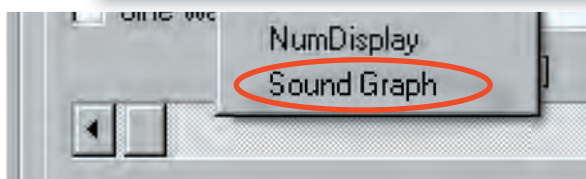
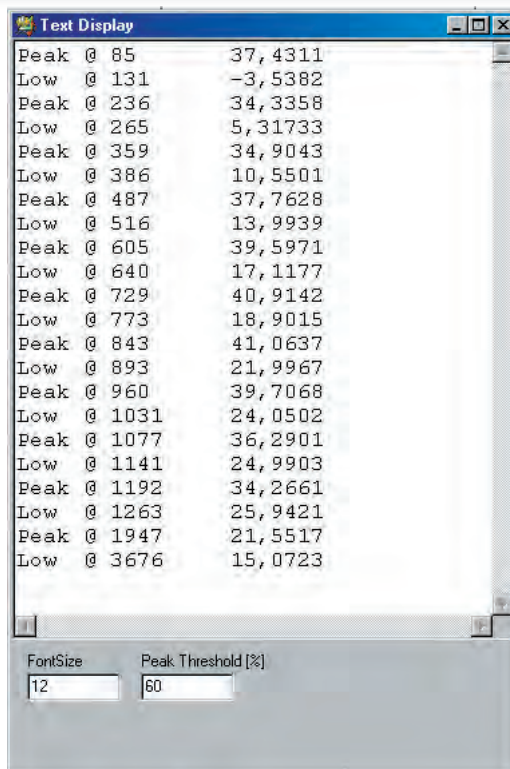
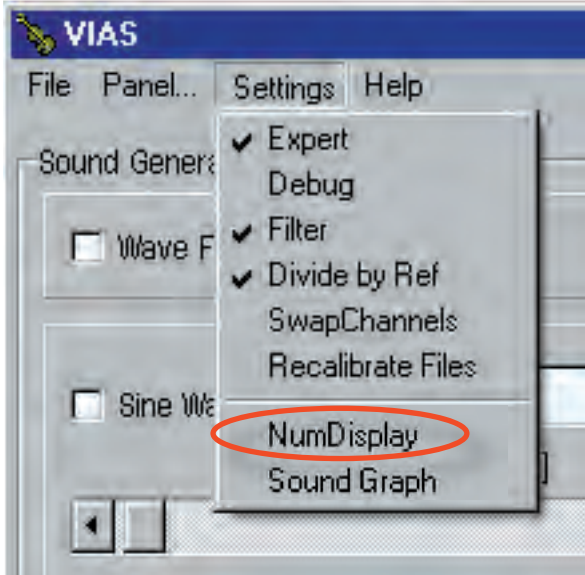
Filter turns on/off the lock-in filter. In the picture, it is turned on. If for any reason your .wav excitation file does not contain purely sinusoidal signals, the lock-in filter should be turned off, as in this case the frequency detection is not possible, and the measuring level display is disabled.

Divide by Ref: To eliminate the influence of different excitation levels on the resulting measurement, the measuring signal will be divided by the shown reference signal (actual level on the speaker membrane in the measuring head).

SwapChannels swaps the reference and measuring signal channels in the sound card. When this command is used together with the division command, the reference signal can be displayed and studied. E.g. to determine if the excitation signal is saturated, turn off the “DividebyRef” command, and turn on the “SwapChannels”. The reference signal will be displayed and plotted instead of the measured signal.

Recalibrate Files: This command loads and already saved (uncalibrated) curve, and calibrates it automatically.

COMMANDS IN THE MENU BAR



NumDisplay shows the frequencies and impedance values of the maxima and minima of a curve, which can be copied and pasted into other applications for further use.

- Turn on NumDisplay
- Choose detection threshold e.g. 50% or 60% (-> maximum or minimum should be at least 60% higher than the surrounding values).
- If the threshold is set to "0" the whole curve is displayed in numerical values. The first column contains the frequency, and the second the impedance.
- Choose the font size, if necessary
- Load a curve: The result will be shown in a text window (see picture). In the picture, a threshold of 60% was chosen. This is why only the frequency and the impedance of minima and maxima are displayed (Peak = maximum, Low = minimum).
The values can be copied to other applications for further use.
- Single rows or the whole contents of the window can be marked and deleted.
- If you load further curves while the NumDisplay window is open, the values for the new curves are appended at the end of the list.

The command **Sound Graph** starts the sound analysis



THE BASICS

Possible uses:

- Admittance measurement with the impulse hammer method

If the bridge of a traditional string instrument is excited with an impulse hammer on one end and the accelerometer is attached at the opposite end, an admittance curve can be obtained within a few seconds. The resulting curve displays almost all acoustical properties of the instrument.

For the admittance measurement you need:

- an impulse hammer with integrated force sensor. Recommendation: PCB 086C03



- an accelerometer, preferably with a small mass. Recommendation: PCB 352C23 Ceramic miniature accelerometer 0.2 g together with a PCB 030C10 teflon cable.

- a charge amplifier. Recommendation: ROGA Plug.n.DAQ USB charge amplifier



- Oscillation measurement at arbitrary points

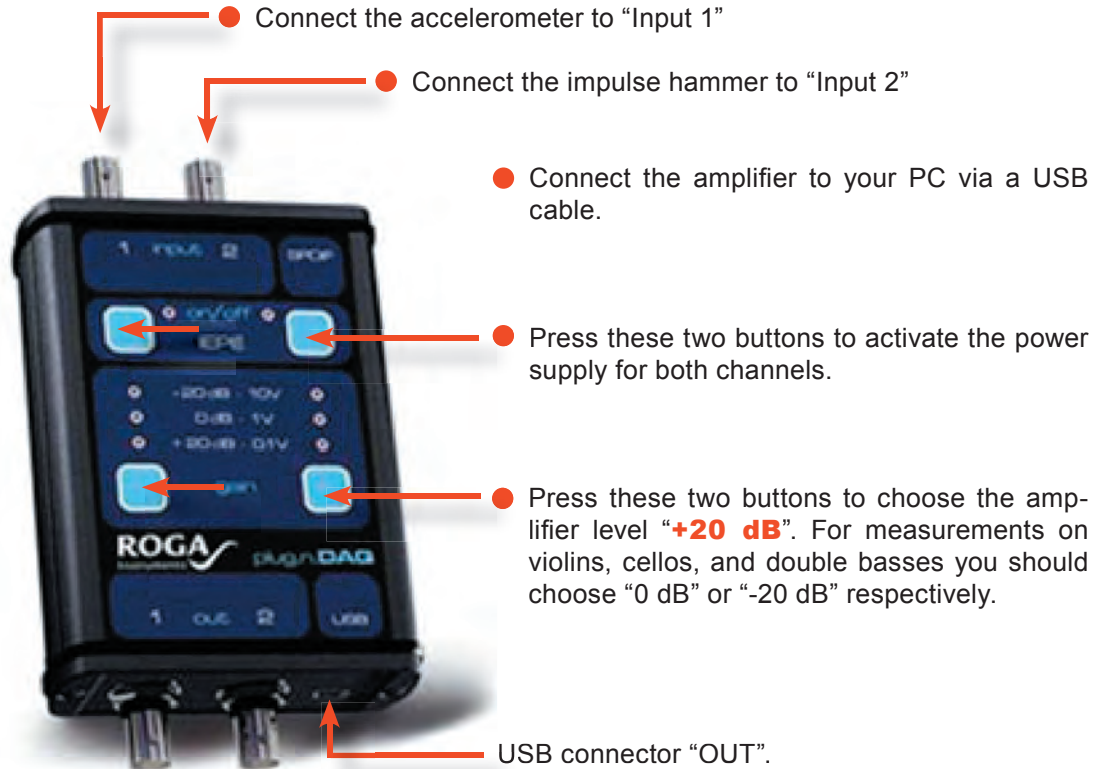
With the hardware mentioned above you can perform arbitrary oscillation measurements. This equipment is particularly useful as data acquisition set for performing modal analyses.

- Oscillation measurements with a sinusoidal sweep and shaker excitation.

VIAS works under Windows 2000 and Windows XP

We give no guarantee that VIAS will work under older Windows versions, like Windows 98, or newer versions, like Windows Vista.

MEASUREMENT SETUP FOR THE IMPULSE HAMMER

**Connection points of the sensors and excitation point:**

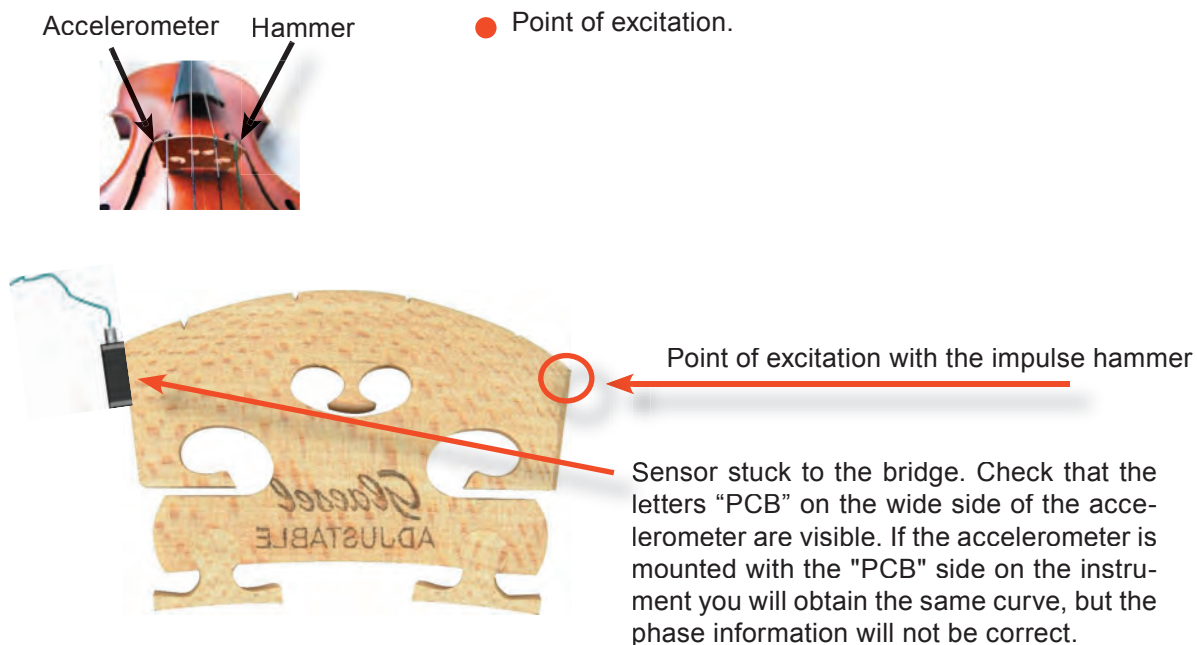
In most cases it is advisable to perform an admittance measurement directly at the bridge of the instrument, in particular in the case of traditional string instruments.

- To mount the accelerometer on the instrument:
Use the delivered beeswax to mount the accelerometer, for instance, to the left upper border of the bridge of a violin (as seen by the player)

REMARK:

It is advisable to use beeswax because it does not leave marks on the instrument. Furthermore, it is commonly accepted by the owners or players even of very precious instruments. The less beeswax you put on the wide side of the accelerometer, the better the contact. Unfortunately, the adhesiveness of the beeswax is reduced with high ambient temperatures.

MEASUREMENT SETUP FOR THE IMPULSE HAMMER

**IMPORTANT:**

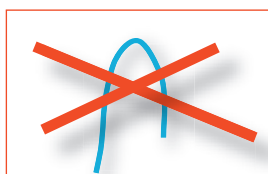
From an electronic engineering point of view, the amplifier is a charge amplifier rather than a voltage amplifier. Since every cable has a capacitance, every movement of the accelerometer cable (during a measurement) leads to substantial measuring errors.



● To avoid possible measuring errors and to provide strain relief, we recommend to fix the accelerometer cable. In most cases, this can be done by clamping the cable in between the fine tuners or by fixing it with adhesive tape on the tail piece.



● Please make sure that the accelerometer cable is not flexed or bent in a radius of less than 4cm. A displacement of the core relative to the coating can change the capacitance of the cable.



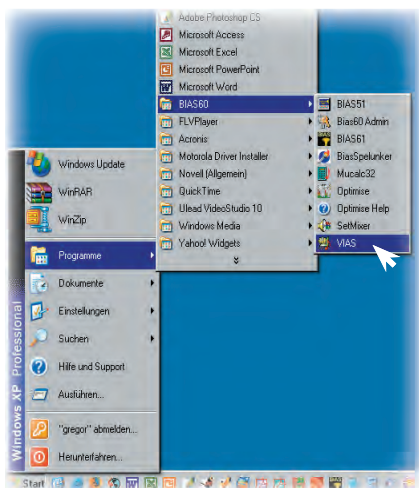
MEASUREMENT SETUP FOR THE IMPULSE HAMMER

IMPORTANT:

The strings of the instrument have to be damped with a piece of foam. If this is not done, energy is withdrawn from the system at the resonance frequencies of the strings. This results in minima in the admittance curve that distort the final result!



STARTING THE VIAS PROGRAM

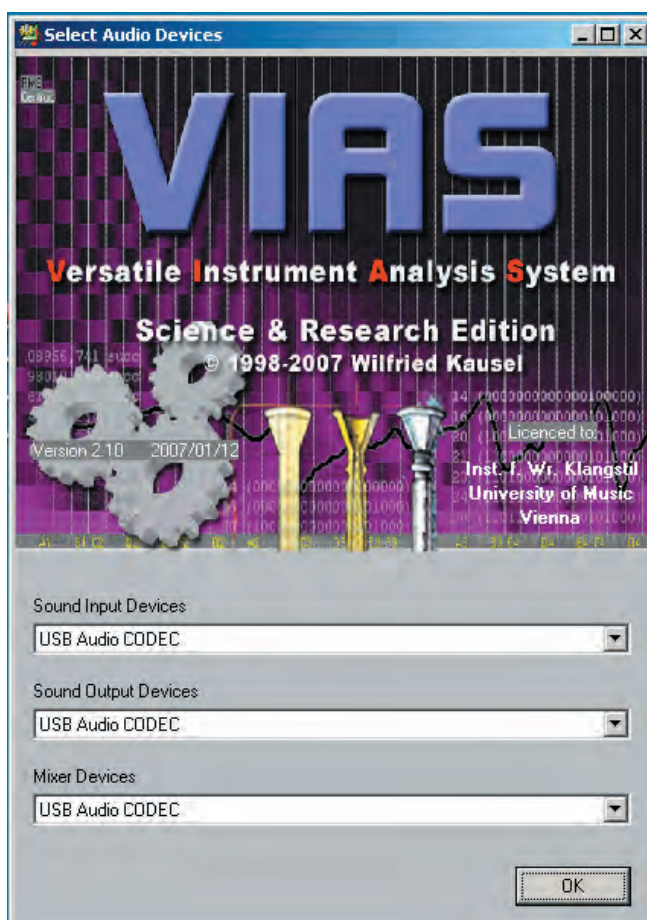


You can start VIAS either:

- via the Windows Start menu (Start -> Programs -> BIAS60 -> VIAS)



- or via the menu item Tools -> VIAS in the BIAS window.



After starting the program the splash screen will appear, where the sound devices to be used are specified.

- If the USB amplifier was connected to the computer before starting the program, VIAS should have recognised it as a “**USB Audio CODEC**”, and will appear as shown in the figure on the left.

- If that was not the case, you have to choose it in the three drop down menus.

Note that the name of the device associated with the USB amplifier can depend on the Windows version you are using.

- If no measurements are to be done, but only analysis and calculations, there is no need to connect the amplifier.

Click “OK”, and after the License agreement the main window will appear.

THE MAIN WINDOW

Oscilloscope window: Real time display of excitation and measured signal (time domain)

Selection of excitation signal

Menu bar

Settings for the virtual sound

Level selection for the excitation and measured signal

File management

Operation

Start

Stop

Load...

Plots

Sound...

Print...

Tuning

minFreq 10

maxFreq 16000

minLev -60

minDet 0,6

LogAbsTol 0,2

LogRelTol 0,5

Y-Axes

Log Display Left Limit 20

Grid Lines Right Limit 7000

Y-Axes

Log Display Upper Limit 40

Grid Lines Lower Limit 5

Time Domain

Mag Arg Re Im C-Re C-Im

Display area "Plot-Window"

Scaling and settings for X-axis and Y-axis

Selection of display between magnitude, phase, real and imaginary part

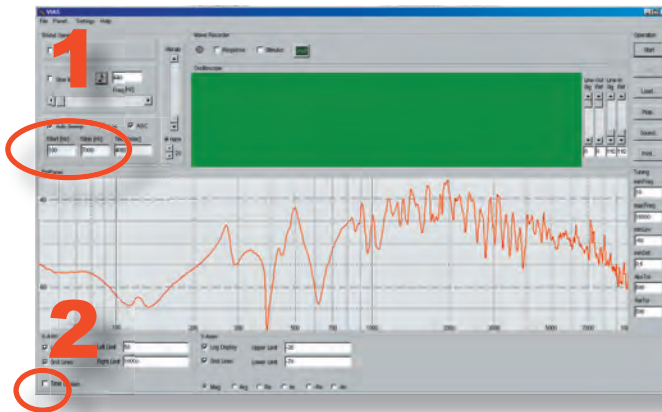
Check-box for admittance measurement with the impulse hammer

Trigger and lock-in filter properties for the acquisition of the measured signal

Note for first measurements:

For the time being, do not change any settings but use the defaults. In most cases the default settings are sufficient.

START AN IMPULSE HAMMER MEASUREMENT



- To start a measurement:

- 1 Determine the frequency range for your measurement by choosing the start and end frequency.



- 2 Check the box "**Time Domain**" to open the measurement window. VIAS is now ready for an impulse hammer measurement in the time domain. Hit the excitation point with the hammer and you are **DONE!**

Numerical details of the signals

Set the the detection threshold

Choose the duration of the measurement

Repeat a measurement

Pulse Recorder

Pulse Quality

Signal Peak Level:	0,0%	-100dB
Signal Noise Floor:	0,0%	-100dB
Reference Peak Level:	0,0%	-100dB
Reference Noise Floor:	0,0%	-100dB
Attack Time	0,0ms	

Detection Sensitivity

SNR: 5,0% 26dB

Recording Time

Buffer size: 32768 0,5s

Coherence Calibrate

Buttons: Retry, Accept, Undo, Close

Pulse Signals

Hammer: **ANZEIGEBEREICH**
Zeitfunktion des Hammersignals

Accelerometer: **ANZEIGEBEREICH**
Zeitfunktion des Meßsignals

Labels and Arrows:

- Accept a measurement (pointing to Accept button)
- Delete an already accepted measurement (pointing to Undo button)
- Stop measuring (pointing to Close button)
- Saturation indicator (pointing to green bar on Hammer plot)
- Peaklevel-meter (pointing to Y-axis scale on Hammer plot)
- Zoom out on Y-axis (pointing to - zoom button)
- Zoom in on Y-axis (pointing to + zoom button)
- Show everything on Y-axis (pointing to globe zoom button)
- Zoom out on both x-axes (pointing to - zoom button)
- Zoom in on both x-axes (pointing to + zoom button)
- Show everything on both x-axes (pointing to globe zoom button)

START AN IMPULSE HAMMER MEASUREMENT

- 3** As soon as the measurement window is open, the **system is ready** for recording and expects a signal. Use the impulse hammer to knock lightly on the bridge. The trigger is released and the signals are recorded.

The measurement lasts about 1 second.


NOTE:



The force sensor on the impulse hammer is very sensitive. It is possible that a trigger impulse is already produced by merely picking up the hammer from a table.


If this happens, delete the measurement with the “Undo” button and click “Retry” to activate the system again.

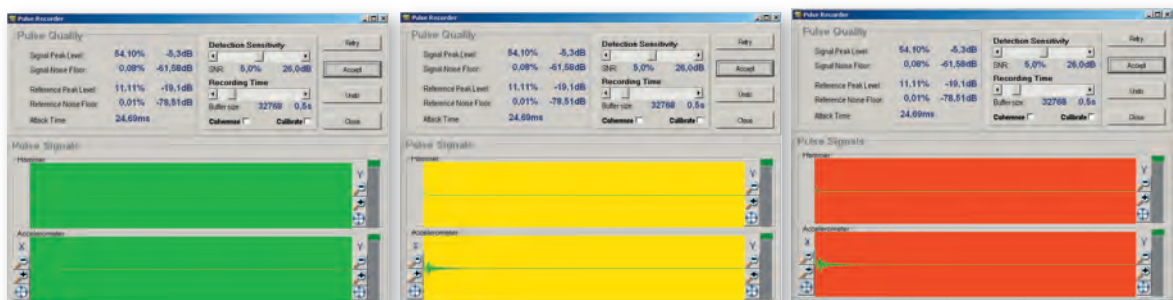
● **To check whether a measurement is useful:**

VIAS uses a colour coding to easily check the signals during a measurement. During a measurement the backgrounds of both display areas is:

 **GREEN** if the signal strength is just about high enough to be useful. The impulse was too low and therefore the SNR is not optimal.

  **YELLOW** if the signal strengths are optimal

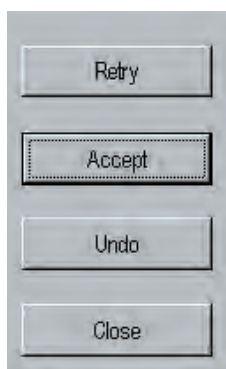
 **RED** if the measurement is saturated, as the signal strength was too high



IMPULSE HAMMER MEASUREMENT

4 If you are happy with your measurement you can click the “**Accept**” button or press the **Enter-key**. Consequently, the measurement is passed on to the VIAS main window. There you can edit or save the curve.

The **measuring window** will remain **open**, and you can continue measuring immediately



If you are not happy with a measurement you can either click on the “**Retry**” button or press the ESC-key to repeat the measurement.

The key “**Undo**” deletes a measurement that has already been passed on to VIAS with the “Accept” button. Thus, you do not have to close the measurement window to delete a measurement that has already been passed on.

The “**Close**” button closes the measurement window and brings you back to the VIAS main window

The buttons described above are useful for a one-person measurement. While you hold the violin in one hand, you can hold the impulse hammer in the other.

If a measurement is OK you can press the Enter-key with the hand that holds the impulse hammer, as the focus of the measurement window is set to the “Accept” button by default.

If a measurement is not OK, you can press the Arrow-Up-key to choose “Retry”, and continue measuring.

We recommend to perform at least 5 measurements (or more) at a time!

AVERAGING SEVERAL MEASUREMENTS

NOTE:

We recommend that you perform at least 5 measurements or more. You can use these measurements to calculate an average curve with the “Average” command in the context menu of the VIAS main window.

EXPLANATION:

The measurement method is based on the assumption that a very short impulse contains all frequencies. However, the amplitude in the high frequency range is typically very small. Therefore, the measured curve is very noisy at higher frequencies and contains amplitude fluctuations.

By computing the average from several measurements, these fluctuations are reduced and the reproducibility of the measurement is increased.

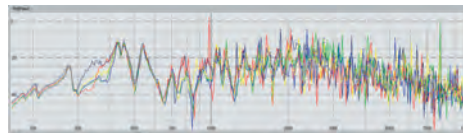
TIP:

Please note that the original curves are deleted by the “Average” command. Only the new, average, curve is displayed.

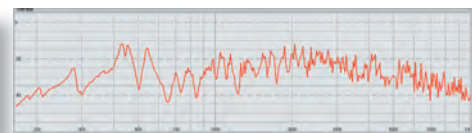
Please save the original curves before the calculating the average if they are needed for later use.

- **To calculate the average of all curves shown in the display area:**

Right click anywhere in the **display area** and select the command **“Average”**.



The 5 original curves

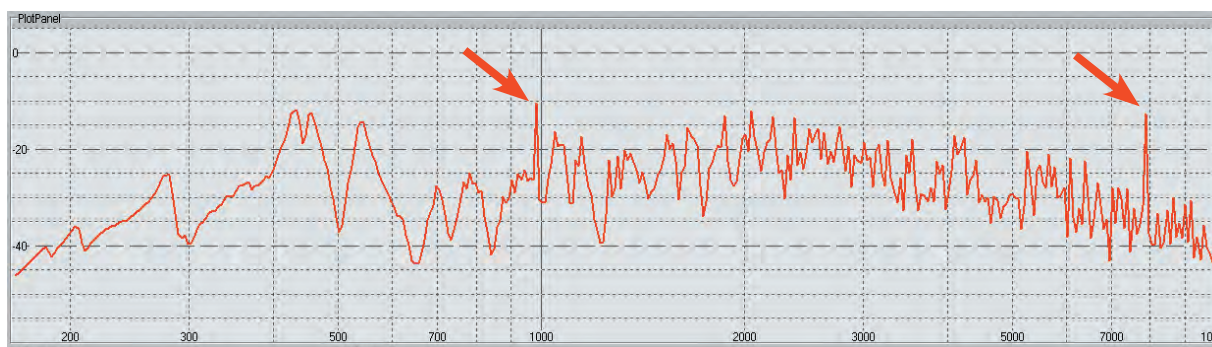


the average curve

SMOOTHING TO CORRECT ERRORS

Curve averaging and smoothing: An example

After calculating the average curve as described above, there are two measurement errors (peaks at 1 kHz and 8 kHz).

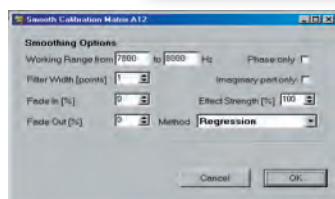
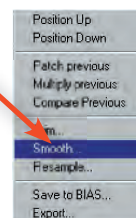


- Press the right arrow key several times to activate the cursor. Determine in which frequency range(s) smoothing has to be performed. In the above curve the first range is between 950 Hz and 1000 Hz, and the second range is between 7800 Hz and 8000 Hz.

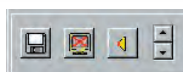


- Click on the "Plots" button (in the upper right hand corner) to list the average curve.

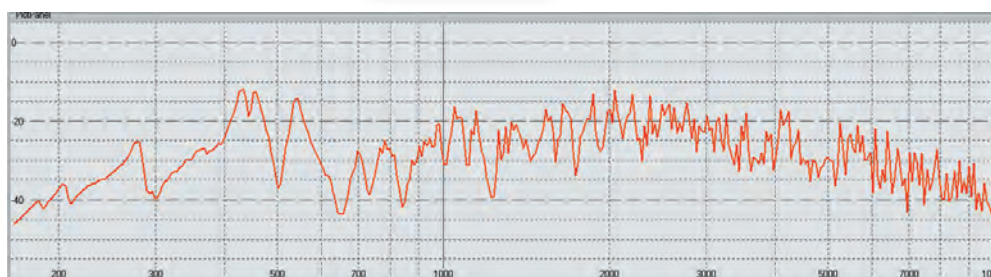
- Right click on the coloured line to open the context menu, and select the command "Smooth".



- Enter the frequency range and choose a filter width of **1** point. Then click on the "OK" button. Repeat the same procedure for the second frequency range



- Save the resulting curve by clicking on the disk icon



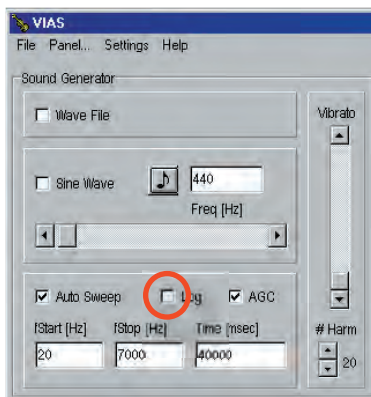
Result

IMPULSE HAMMER MEASUREMENT

Axes scaling

The X-axis (frequency) scale is by default set to logarithmic. That is: In the low frequency range there are more measuring points (higher resolution), than in the high one (lower resolution).

If for any reason you need to have the same resolution in the whole frequency range, you have to uncheck the box marked “Log” before opening the measuring window.



- **To measure with a linear frequency resolution:**

Disable the logarithmic scaling by unchecking the box “Log”.

A further click on this box will enable the logarithmic scaling and will be again the standard scaling for all following measurements.

IMPULSE HAMMER MEASUREMENT WINDOW

Pulse Quality		
Signal Peak Level:	54,10%	-5,3dB
Signal Noise Floor:	0,08%	-61,58dB
Reference Peak Level:	11,11%	-19,1dB
Reference Noise Floor:	0,01%	-78,51dB
Attack Time	24,69ms	

The details are self explanatory (see picture).

Apart from the signal levels and the SNR of the reference and measured signals, the impulse duration is also measured and specified.

Detection Sensitivity	
SNR:	5,0% 26,0dB
Recording Time	
Buffer size:	32768 0,5s
Coherence <input type="checkbox"/>	Calibrate <input type="checkbox"/>

The trigger sensitivity (from which signal strength the measuring process will start) is specified with the scrolling bar marked "**Detection Sensitivity**". The standard setting is 5% of the maximum level.

The **Recording Time** (how long the measured signal will be recorded) is set with the scrolling bar marked "Recording Time".

The standard setting of 0.5 seconds works fine for violins. For instruments whose decay time is longer (e.g. Guitars), zoom in the Y-axis to see if the signal is cut off at the end of the measurement, in which case you can just increase the recording time.

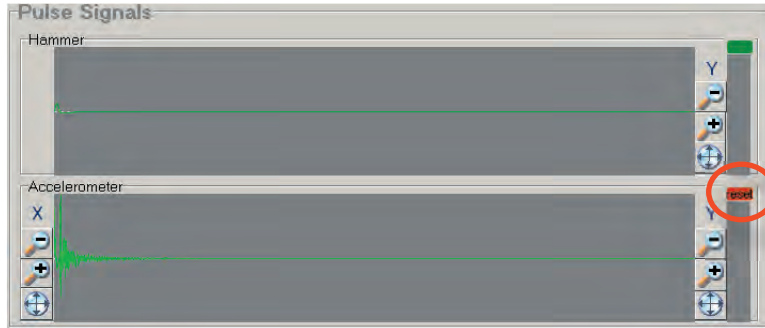
If the box marked "**Coherence**" is checked before a measurement takes place, the following curves will be generated and passed on to VIAS: two auto power spectra, a cross power spectrum, a coherence spectrum and the admittance curve (5 curves in total). This calculation takes place as soon as you press the "Accept" button.

WARNING:

This works only if you have checked the box "Response" (in the top middle of the VIAS screen, see page XXX) before opening the measuring window. This will record the signals as .wav files

Calibrate: This command is explained in an extra document. Please do not use it for the time being.

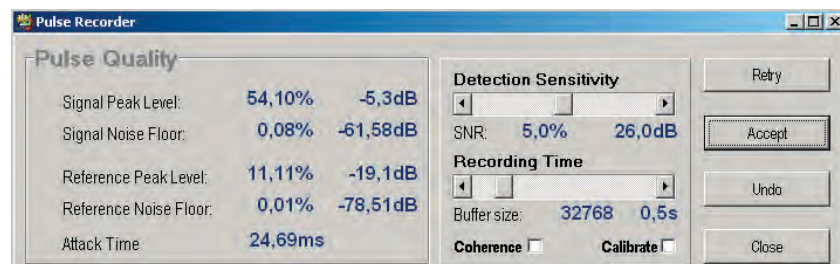
IMPULSE HAMMER MEASUREMENT WINDOW



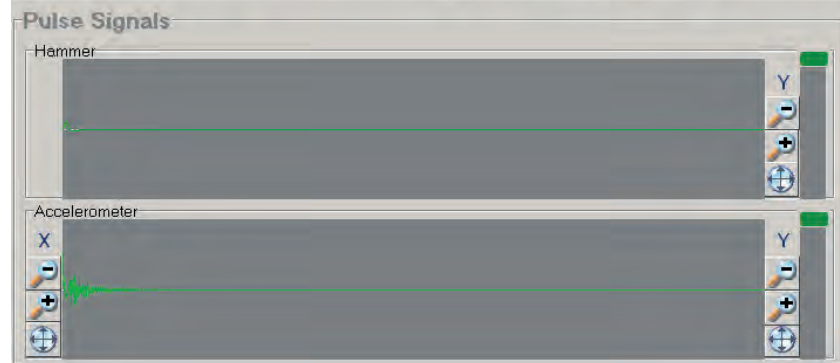
Display area

When one of the signals saturates, two things will happen: 1. the display area will be shortly coloured in red, and 2. the corresponding peak level meter will be marked red, and will stay red to show that there was saturation during the measurement (see picture).

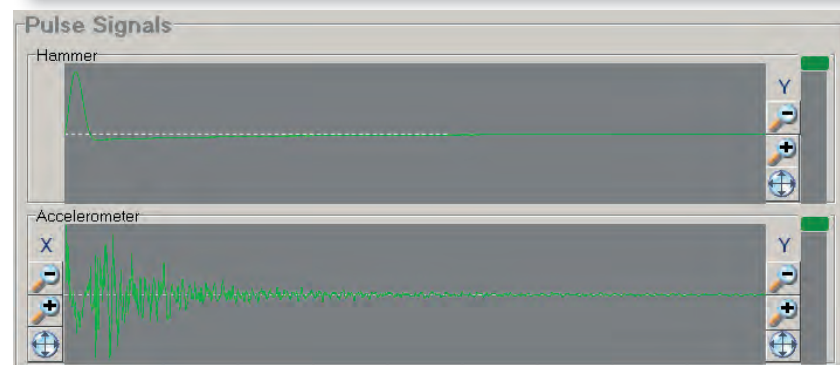
The Zoom functions are self explanatory. While zooming in the Y-axes, the X-axis settings for both channels will remain unchanged.



Normal display



Y-axis zoomed in



MEASUREMENT WITH SHAKER EXCITATION

Apart from using an impulsive excitation, a sinusoidal excitation is also possible by using e.g. a shaker. In this case, the sinusoidal sweep generated by VIAS is used.

HARDWARE SETUP

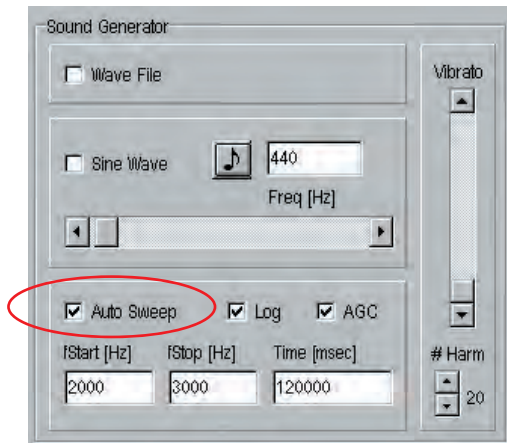
The set up is similar to that of the impulse hammer setup. You need to have a shaker (any brand) instead of an impulse hammer. The shaker receives the excitation signal (e.g. a sinusoidal sweep) from VIAS via the output 1 or 2 of the ROGA amplifier. As the voltage delivered by the ROGA is very small, you need to add another amplifier in between the ROGA and the shaker.

To get a measurement as meaningful as possible, you should add a force sensor between the shaker and the object to be measured (see picture).



MEASUREMENT WITH SHAKER EXCITATION

To select the excitation signal:



- Normally, a sinusoidal sweep will be used as excitation signal. In this case, the box “Auto Sweep” should be checked.
- Choose between a logarithmic or linear sweep.

This selection depends on the desired result: A logarithmic sweep is usually recommended, as it gives enough time for the generation of standing waves in lower frequencies, and it is musically relevant, since it corresponds to the way pitch and musical intervals are perceived.

If you need a specific accuracy in high frequencies, the linear sweep is recommended – In any case, you should allow enough time to achieve the desired resolution: e.g.: 5 minutes = 300,000 ms

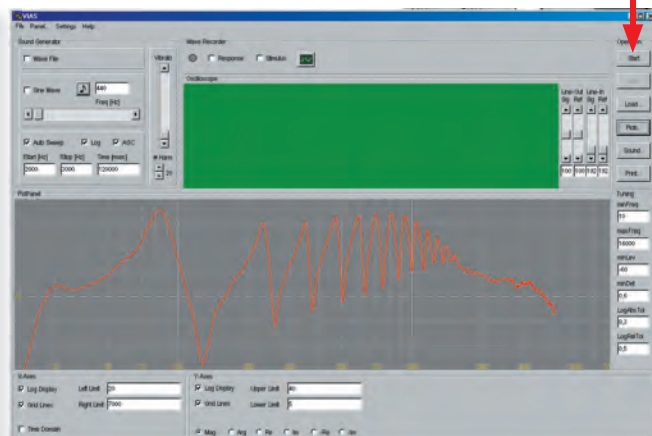
- Type the start and stop frequencies in both fields **fStart** and **fStop**, in the range from 0 to 20,000 Hz. In the shown picture, the measurement will be done between 2 kHz and 3 kHz.
- Type the duration of the sweep (in milliseconds) in the field **Time**. In the shown picture, the duration of the measurement is 2 minutes (=120,000 ms) for a range of 1 kHz
- For short sweeps (less than 5 minutes for 0 – 20 kHz), the AGS sometimes does not work properly.
- Start the measurement by clicking the “Start” button.

WARNING!

Whenever the shaker is used as exciter, the previously set calibration data will be deleted! (See page XXX).

*To perform BIAS measurements afterwards, the **BIAS .cal file** must be imported first.*

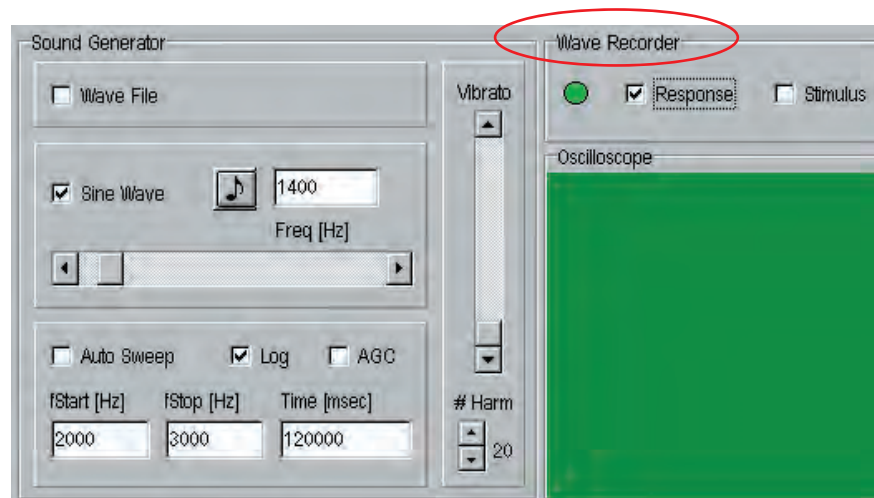
If no force sensor is to be used, the shaker input signal should be set as reference signal.



MEASURING - The Wave-Recorder

Show measured signals in Microsoft .wav format:

Excitation and measured signals can be played and saved as .wav files. Therefore, they can be processed as sound files, used for the calculation of a coherence-, cross- or auto- spectrum, used for other purposes, or simply played back.



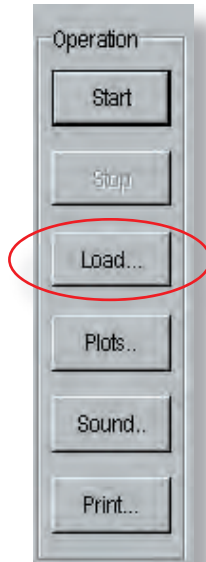
- By checking the box “Response”, the wave recorder is activated, and a new window opens where you can choose the name of the file and the directory where the displayed measured signal will be saved.

This results in a stereo .wav file, one of the channels being the reference signal from the force sensor on the impulse hammer, and the other the measured signal from the accelerometer on the bridge.

- If you wish to choose the function “Coherence” in the measuring window, the Wave-Recorder must be activated.
- The box “Stimulus” is deactivated during measurements with the impulse hammer.

FILE MANAGEMENT

LOADING A MEASUREMENT

**Load...**

Click on the “Load” button, and a new dialog window will open, where you can load a measurement that was saved previously. It will be displayed in the “Plot-Panel”, along with other current measurements.

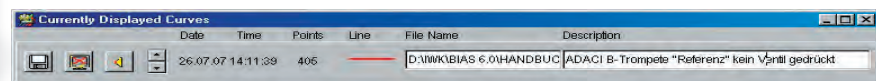
You can also load, show and work with BIAS or .wav files

NOTE: The buttons “Start” and “Stop” are only used for BIAS measurements, and have no effect for impulse hammer admittance measurements!

FILE MANAGEMENT LIST CURVES

**Plots...**

The “Currently Displayed Curves” window will open after clicking on the “Plots” button.



This window shows the currently displayed curves. Apart from being able to save or delete measurements, this window also:

- gives detailed information about each displayed measurement,
- allows you to manipulate the plots,
- is the starting point for multiple analysis, and
- allows you to manipulate and post process the data

FILE MANAGEMENT SOUND ANALYSIS

**Sound...**

- To open the sound analysis window, click on the “Sound” button (see Sound Analysis, page XXX).



FILE MANAGEMENT PRINTING

Print...

Use this command to print the contents of the “Plot” window

FILE MANAGEMENT "PLOTS"

This window opens automatically after clicking on the "PLOTS" button

SAVE

- New measurements are saved
- If the curve has already been saved or loaded, all the changes will be saved after clicking "OK" in the confirmation dialogue.

DELETE

- New non-saved measurements will be deleted!
- Already saved or loaded curves will be removed from the Plot window

PLAY BACK SOUND

Generates a sound based on the measured curve, according to the properties set in the window where the excitation signal was chosen

Date and time when the measurement was done

Number of measured points in the curve

	Date	Time	Points	Line	File Name	Description
	26.07.07	14:11:39	405		D:\NWK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" kein Ventil gedrückt
	26.07.07	14:50:22	410		D:\NWK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 2. Ventil
	26.07.07	14:26:04	410		D:\NWK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 3. Ventil

Line style, colour and thickness

File name, including path

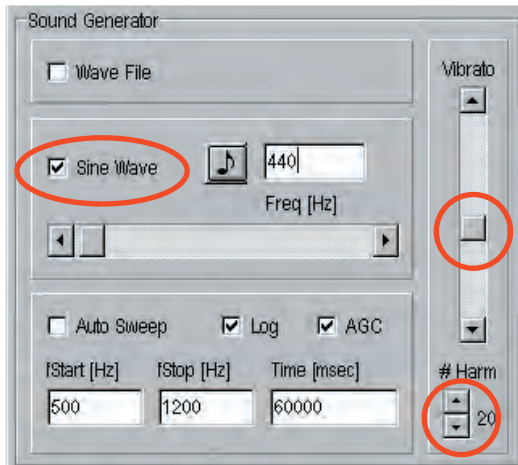
Editable comments

SHIFTING

Shifts (without changing the data) the curve along the Y-axis in the Plot window, for ease of comparison

FILE MANAGEMENT "PLOTS"

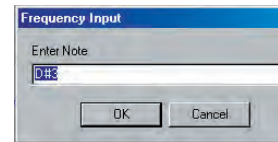
SOUND PLAY BACK



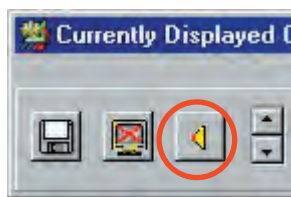
To choose the sound parameters:

- Click on the box "Sine Wave" if you would like a tone, or in the box "Auto Sweep" if you would like a frequency sweep (the current Start- and Stop frequencies apply, as well as the time duration settings)

- If you click on the musical note icon, you can write the name of the note (CAUTION: American notation! **A4 = a¹**)



- The degree of vibrato can be set with the "Vibrato" scrolling bar.



- With the arrow boxes marked "**# Harm**" under the vibrato scrolling bar, you can choose the number of harmonics (1-99) that the sound will have
- Click on the speaker icon to hear the curve. Click again to stop the sound generator.

NOTE:

You can choose the number of harmonics the sound should have, but bear in mind that if you choose e.g. 60, only the first 22 will have an effect on the sound, if the admittance curve does not show any significant data above partial 22.

HINT FOR IMPULSE HAMMER MEASUREMENTS:

To be able to hear the "virtual sound", you have to disconnect the "ROGA USB amplifier" from your computer first (simply pull out the USB cable!).

HINT FOR SHAKER MEASUREMENTS:

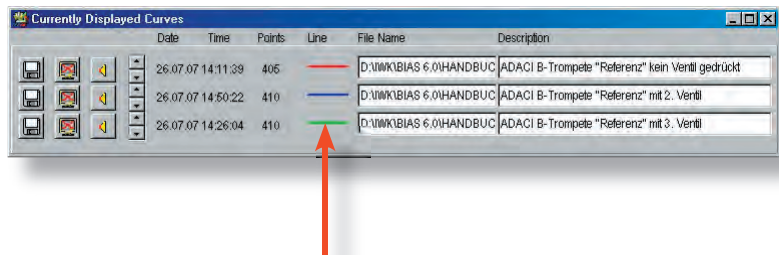
Usually, VIAS sends the excitation signal to the amplifier/shaker and the "virtual" sound to the sound card of your computer. Whether this works depends mainly on which sound card drivers are installed, and on which USB amplifier you are using.

In case the sound is mistakenly played through the shaker, choose the following command: Panel ... -> Options -> Properties. Here you can set the sound card as output device. If this does not help, disconnect the amplifier and shaker from the computer, so that the sound card is the only remaining output device in your computer

FILE MANAGEMENT "PLOTS"

CHANGING THE CURVE DIAGRAM

- To change the colour of a curve:

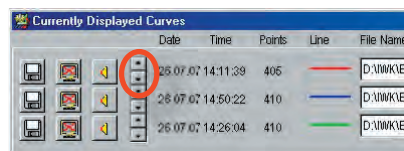


Simply **left** click on the colour line of the corresponding file in the "Currently Displayed Curves" window.



A window will appear where you can choose (or generate and add) any colour.

- To shift a curve without changing the data:



For ease of comparison, you can shift the curve up or down on the Y-axis by clicking on the arrows, without changing the data

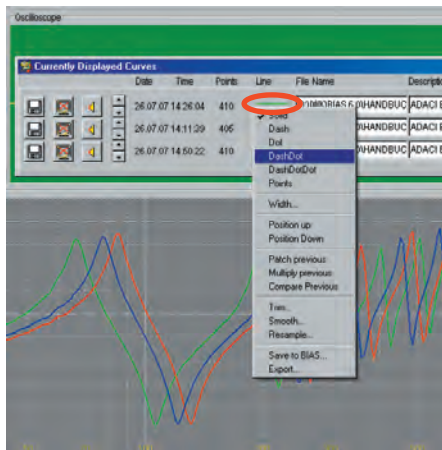
HINT:

All changes apply only to the curve that corresponds to the colour line you clicked!

FILE MANAGEMENT "PLOTS"

CHANGING THE CURVE DIAGRAM

- To open the context menu:



Right click on the colour line of a file in the "Currently displayed Curves" window

The context menu is divided in several areas:

The relevant commands needed to change the curve diagram are located on the top area of the Context Menu.

**Curve representation:**

Line style (solid, dotted, dashed, etc.). The line width is specified with numerical values.

Data manipulation (Post Processing) for further analyses (see page XXX)

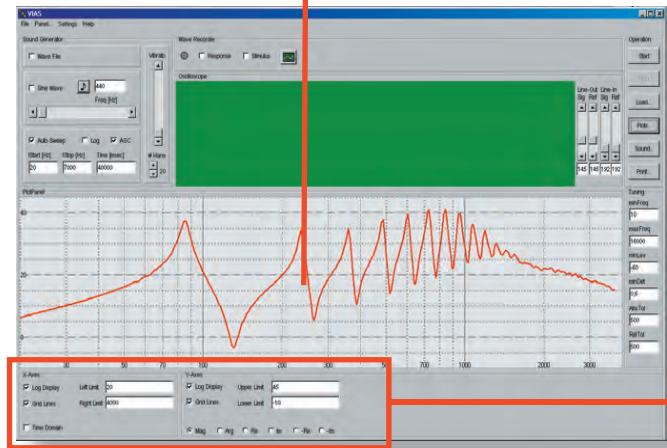
Export the curve in the BIAS database

HINT:

All changes apply only to the curve that corresponds to the colour line you clicked!

DISPLAY AREA - PLOT WINDOW

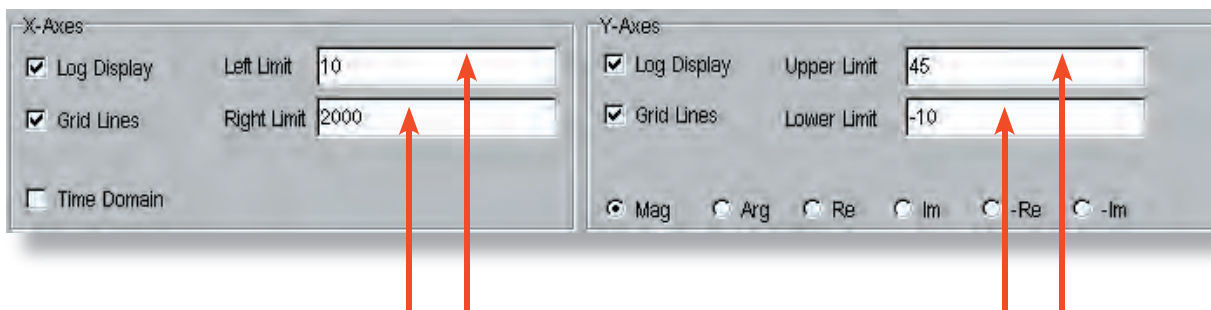
Display area (Plot window)



All parameters can be fast and easily changed by typing the appropriate values in the setting fields located below the display area

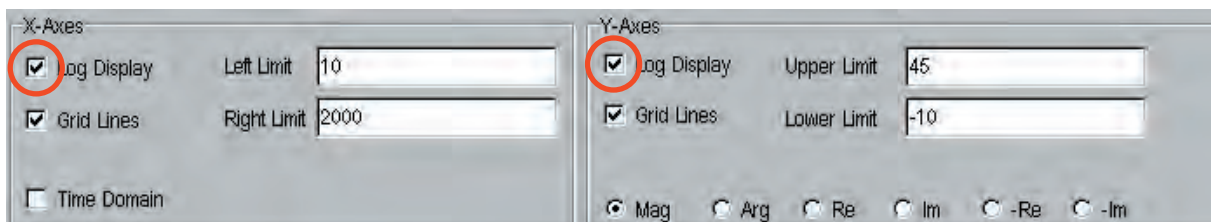
Area to set the display parameters

- Setting the frequency range (X-axis) and the impedance values (Y-axis)



Simply double click in the field, and type the desired frequency and impedance range

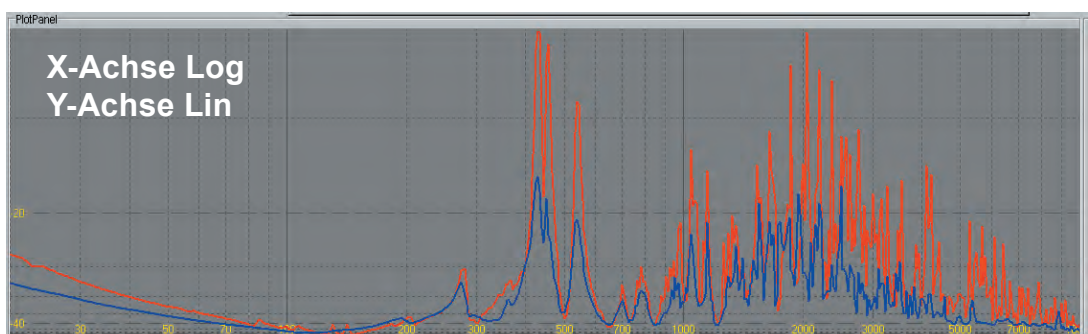
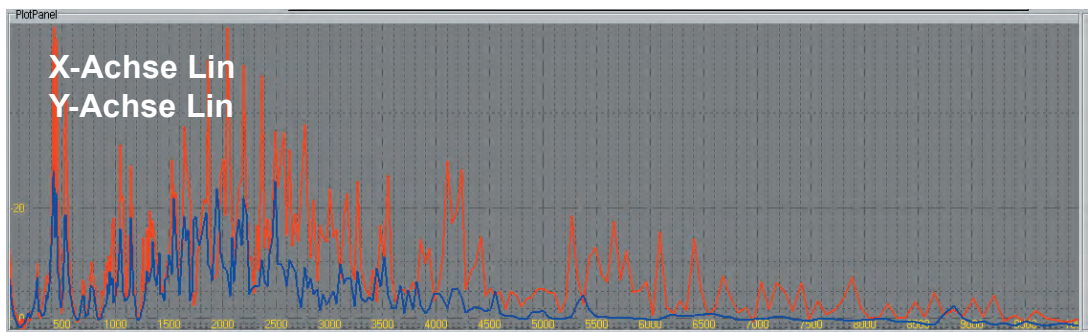
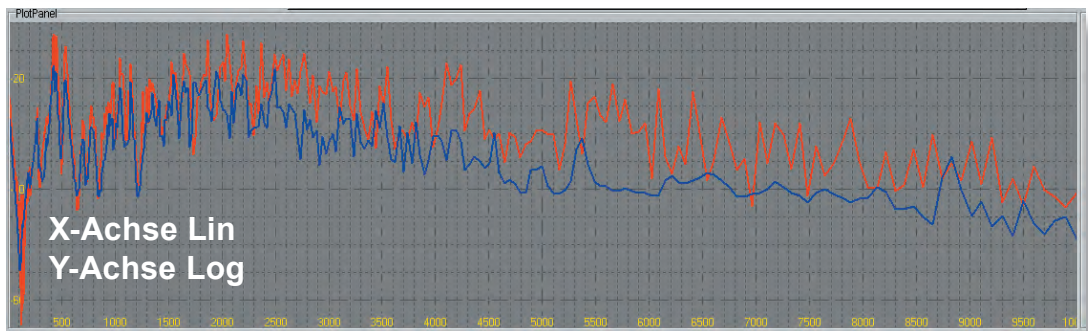
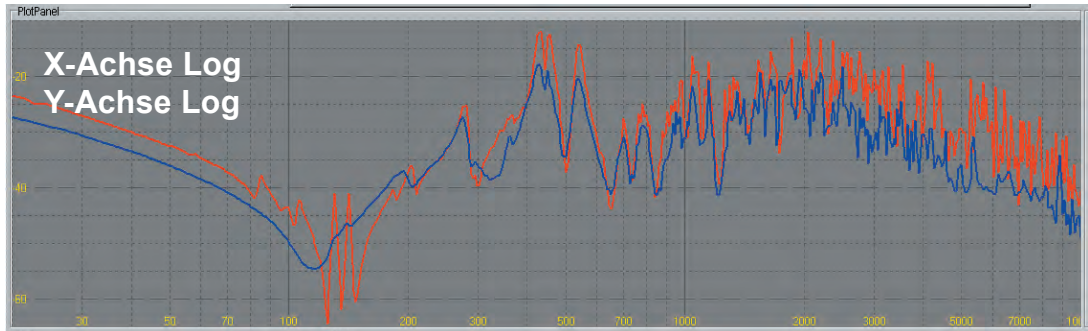
- Alternative between logarithmic and linear scale for both axes



Click on the box to turn the logarithmic scale on/off

DISPLAY AREA - SCALING

Scaling alternatives: Logarithmic – Linear



DISPLAY AREA - CURSOR

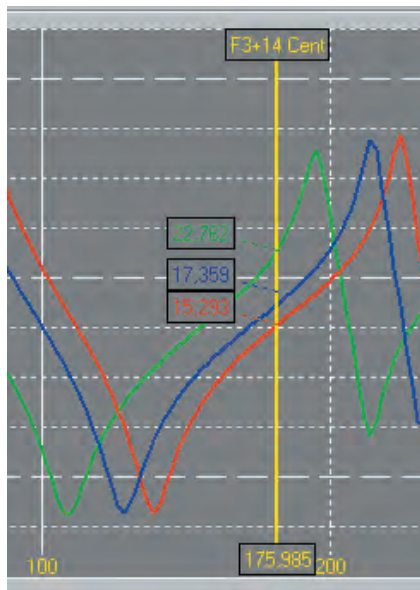
Retrieve the cursor:



Alternative 1: Move the mouse pointer to the left border of the display area, until it becomes a double arrow. Then left click and drag to the right: The cursor will appear.



Alternative 2: Strike the right arrow key (on the keyboard) several times. The cursor will appear in the left border of the display area



- The cursor reads all the impedance values of the displayed curves at the same time, and shows them in the colour of the curve to which each value corresponds

- The bottom field gives the current frequency value, the top field the corresponding musical note (in american notation $a^4 = a^1$) and the deviation in cents from an equally tempered scale, tuned at 440 Hz.

The reference frequency of 440 Hz (standard pitch) can be adjusted with the scrolling bar (below the input field). Clicking on the corresponding arrows will change the frequency in 1/10 Hz steps.

- The cursor can also be moved using the arrow keys from the keyboard

Admittance curves from bowed string instruments have pronounced maxima and minima.

If you would like to detect only the most important peaks, you should smooth the curve with the command "Smooth" in the context menu (see page XXX).



Move the cursor to the right or left with these two keys



Press the "**up arrow**" key to move the cursor to the next impedance **maximum** (peak)

Press the "**down arrow**" key to move the cursor to the next impedance **minimum** (valley). The movement of the cursor is always from left to right! If you would like to detect the maxima or minima, you should start with the cursor on the far left.



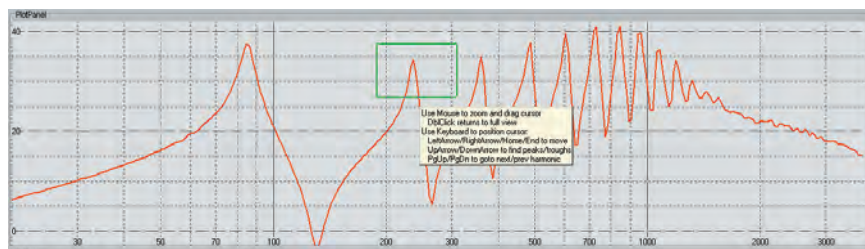
Press the "**Page up**" and "**Page down**" buttons to move the cursor to 2, 3, 4, etc. times the initial frequency. This is useful to detect harmonicity of the impedance peaks

All values read by the cursor will be loaded automatically in the windows Clipboard. You can simply paste all maxima, minima or other values on arbitrary points of the curve in a text editor or spread sheet for further work.

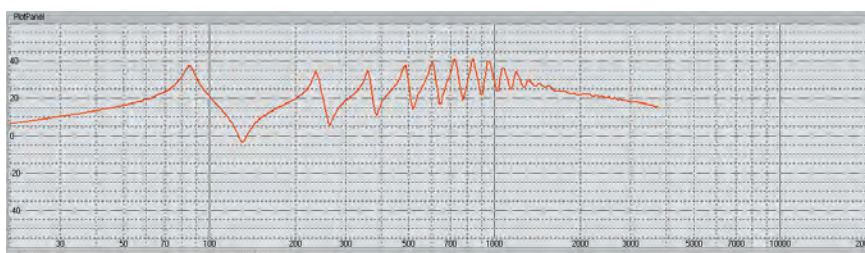
DISPLAY AREA - ZOOM

● Zoom in

Use the mouse pointer to drag a rectangle around the region into which you want to zoom.

**● Zoom out**

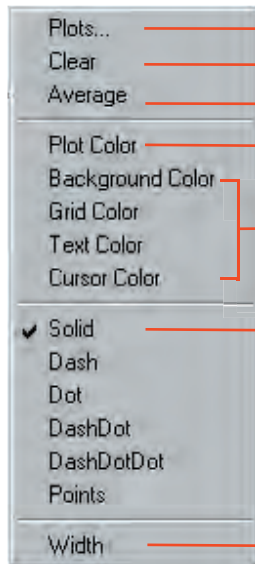
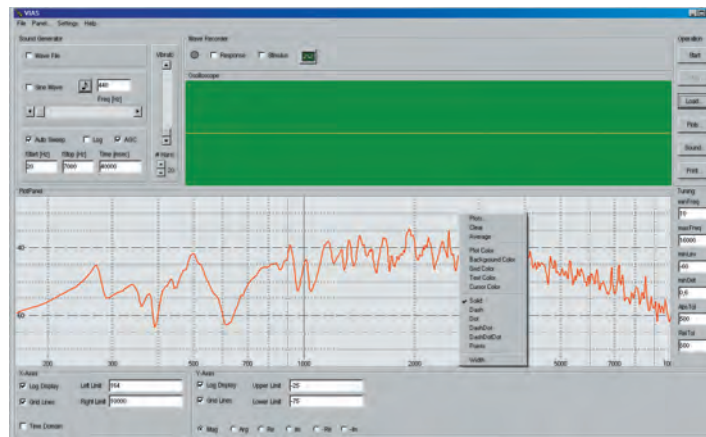
Double click on the display area, to restore the display back to the standard setting.



DISPLAY AREA - CONTEXT MENU

Right click anywhere in the display area to open the context menu.

The first three commands in the context menu refer to the curves currently displayed in the display area. All other commands refer to the curves of future measurements.



Choose a colour for the next measurement.

Background colour, grid colour, text colour, cursor colour.

Choose a line type for the next measurement.

Choose a line width for the next measurement.

Action: Opens the “currently displayed curves” window which lists the currently loaded files.

Action: Deletes all curves from the display area.

Action: Calculates the mean value of all displayed curves and display it as a new curve.

TIP:

Bear in mind that when the mean value is calculated, the original signals are deleted from memory, and only the **new averaged curve will be displayed.**

If you have not saved the original curves yet, you should do it before performing this command!

ADMITTANCE DISPLAY

X-Axes

Log Display Left Limit: 20

Grid Lines Right Limit: 7000

Time Domain

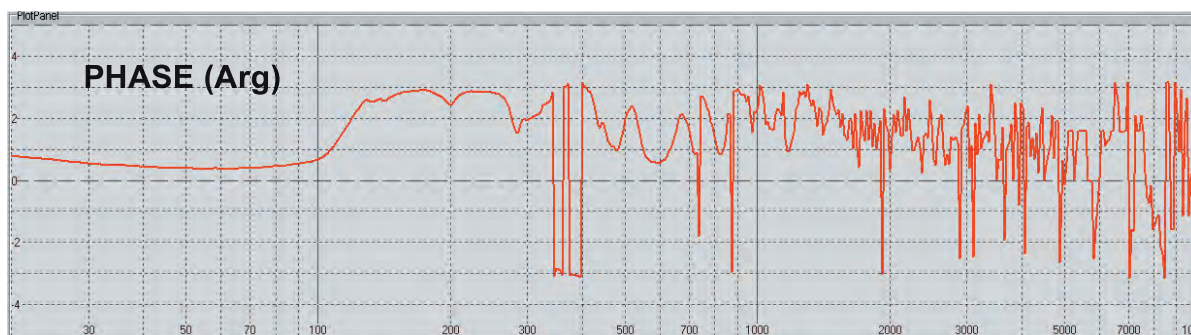
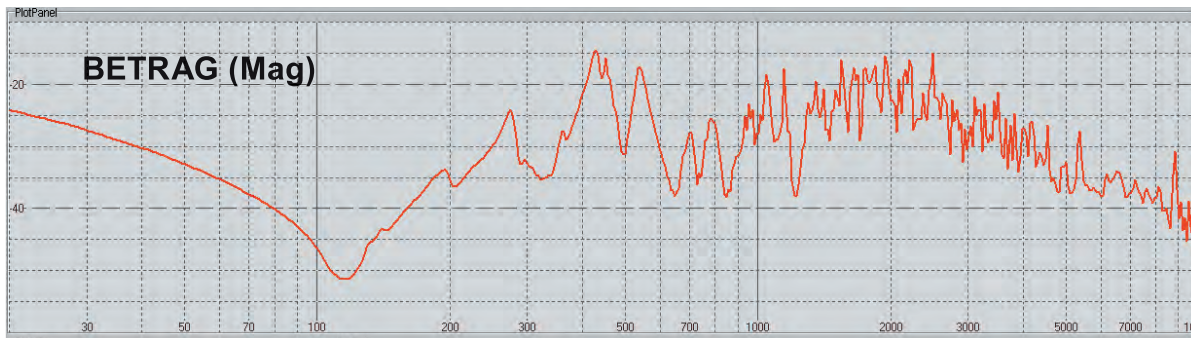
Y-Axes

Log Display Upper Limit: 40

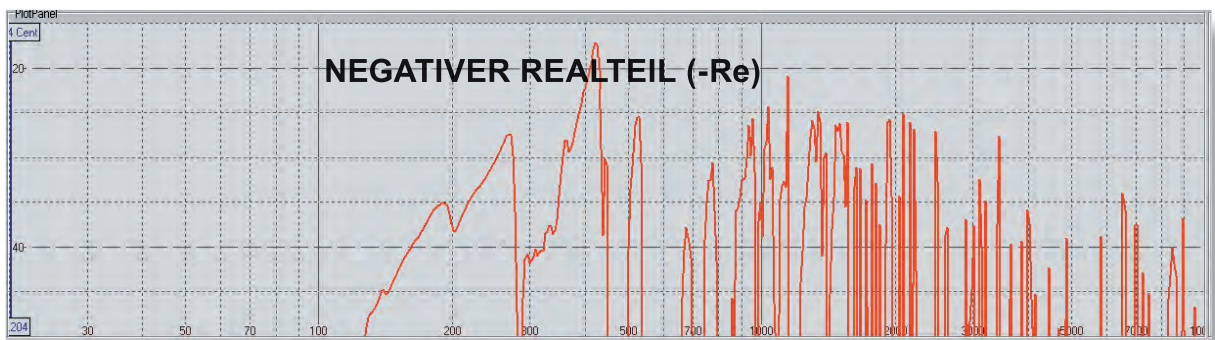
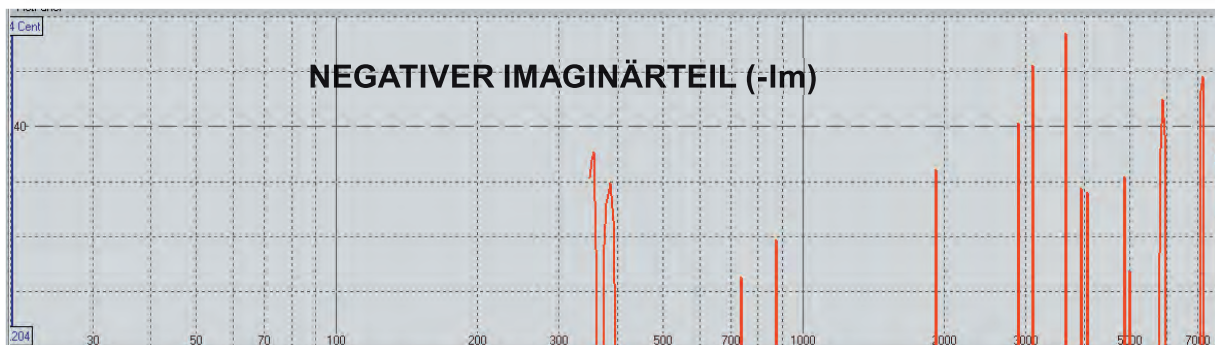
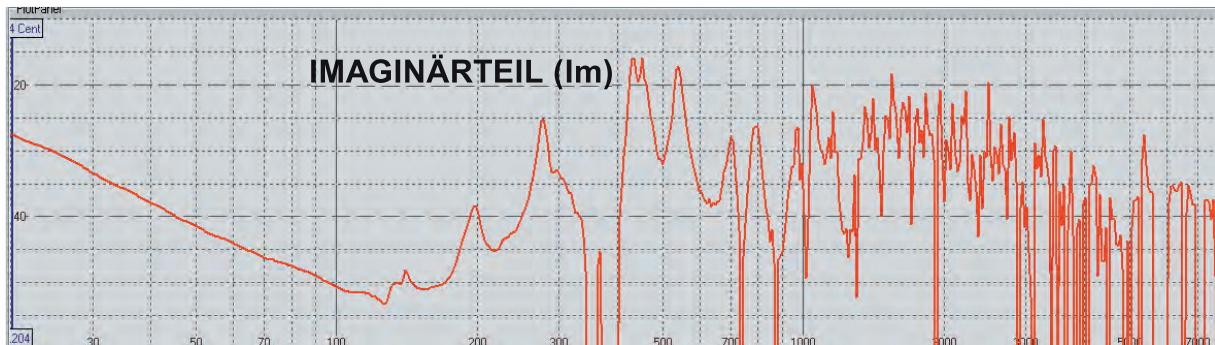
Grid Lines Lower Limit: 5

Mag Arg Re Im -Re -Im

Click on the appropriate radio button to choose the desired display mode.



ADMITTANCE DISPLAY

**NOTE:**

Normally an input impedance curve does not show a negative real part. If this ever occurs, effective power flows from the measured system back into the exciter. However, the reason for this is usually that there is some problem with the measurement conditions.

Possible error sources: The sweep moves too fast across sharp resonances, the polarity of a signal is inverted, there are phase shifts in an external signal amplifier or in the sound card.

Of course, arbitrary curves, such as the difference of two curves, can show a negative real part!

ANALYSES - SOUND ANALYSIS



- **To start the sound analysis**
click on the button "Sound"

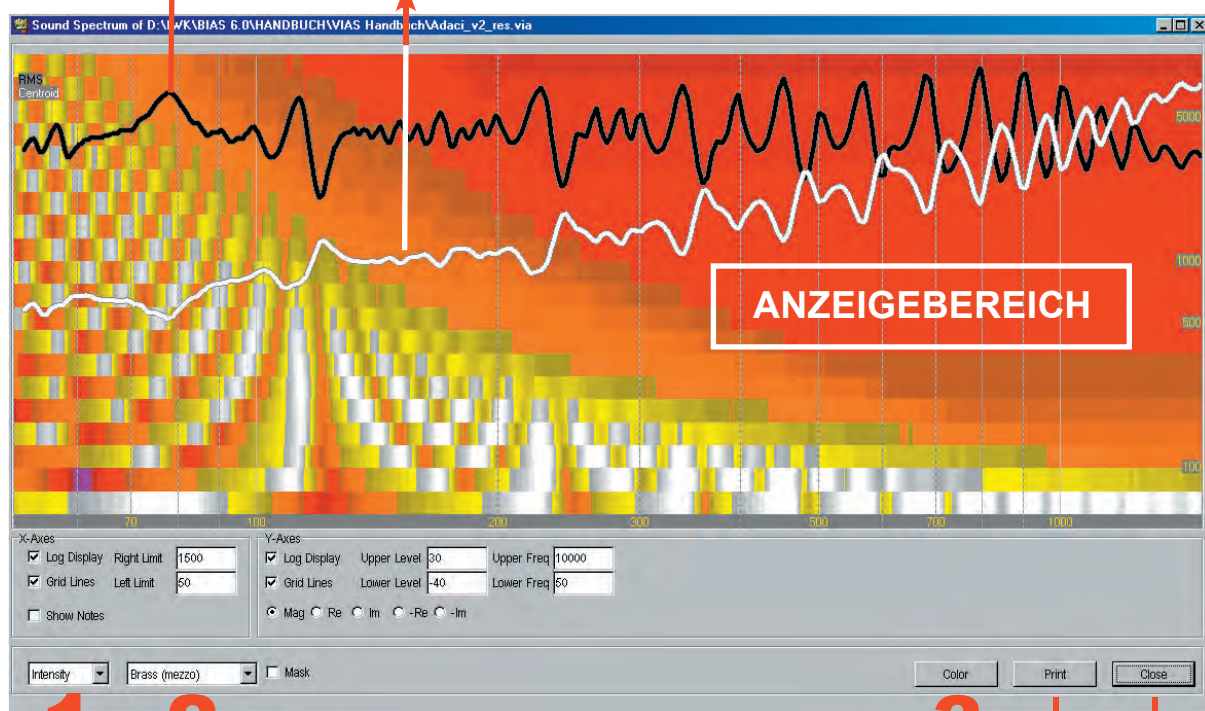
- The sound analysis will only be shown **for one curve**, no matter how many curves you are currently displaying.
- The sound analysis for the **last active** curve will be displayed.

- **To choose a particular measurement for sound analysis:**

In the "Currently displayed Curves" window click on the speaker icon which belongs to the measurement you wish to analyse. A subsequent single click on the "Sound" button opens the sound analysis window.

RMS: relative dB values of the emitted sound level vs. frequency or note.

Harmonic Centroid (sound balance point) for every note (right Y-axis)

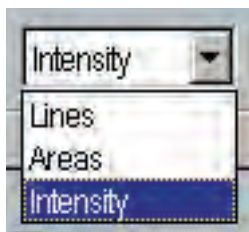


prints the contents of a display area

closes the sound analysis window

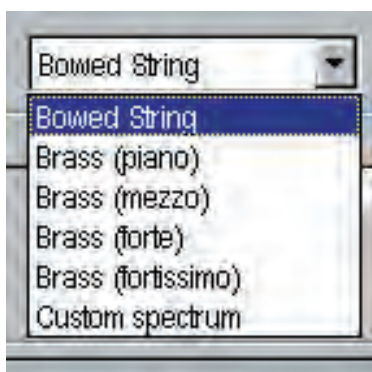
ANALYSES - SOUND ANALYSIS

1

**Display Style**

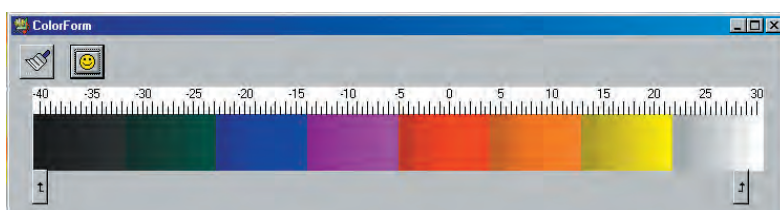
You can choose between a line spectrum ("Lines"), an area spectrum ("Areas") or a spectrogram ("Intensity").

2

**Sound Formation**

With this drop down menu you can choose the number and weight of the harmonics in the produced sound.

3

**Colour Coding**

Using the two arrows, you can assign colours to the different intensity degrees (in dB).

Axes scaling for the Display Area:

Y Axis: Left: Partials from bottom to top (the shown picture depicts 20 partials)
Right: Frequency in Hz, logarithmic

X-Axis: Frequency in Hz or musical note
Linear or logarithmic

NOTE:

All available setting fields, e.g. Mag, Re, Im, etc., always work, but it does not always make sense to use them.

"**Mag**" displays the magnitude spectrum,

"**Re**" displays the real active component (= radiated or dissipated part),

"**Im**" displays the imaginary reactive component (= part that is not radiated or dissipated)

ANALYSES - SOUND ANALYSIS

1



Here you can choose between three **different display options** for the analysed virtual sound of the instrument.

Lines:

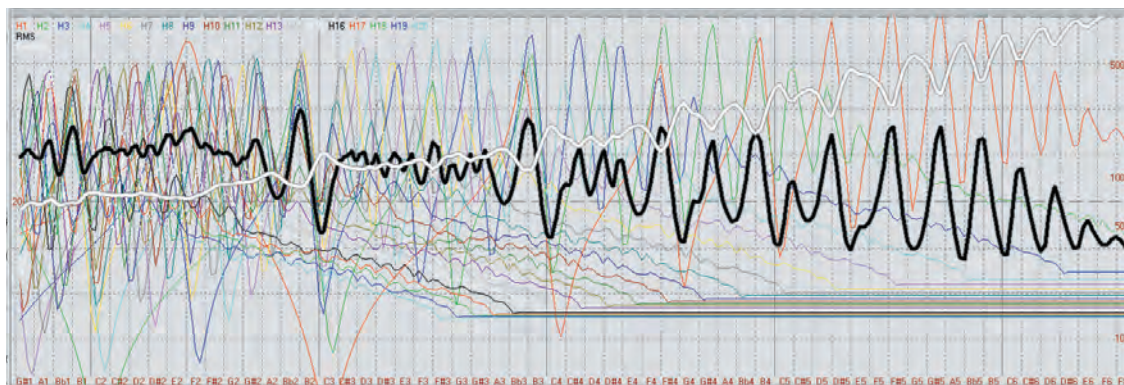
X-axis: Frequency in Hz (linear or logarithmic) or musical note

Y-axis (left): Strength of the partial in dB (relative unit)

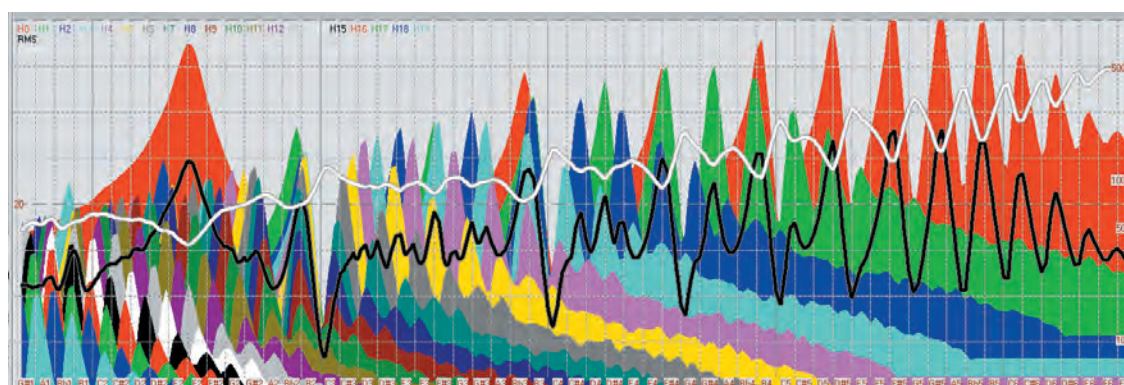
Y-axis (right): Frequency in Hz (logarithmic) of the harmonic centroid

The lines represent the contribution of each partial in the total spectrum for each frequency point or musical note.

The legends can be found on the top left corner of the display area. H1 specifies the colour of the first partial (=fundamental), H2 of the second partial, etc.

**Areas:**

Same as with “Lines”, but a colour area is displayed for each partial, instead of just a line. Higher partials are displayed on top of lower ones.



ANALYSES - SOUND ANALYS

Intensity:

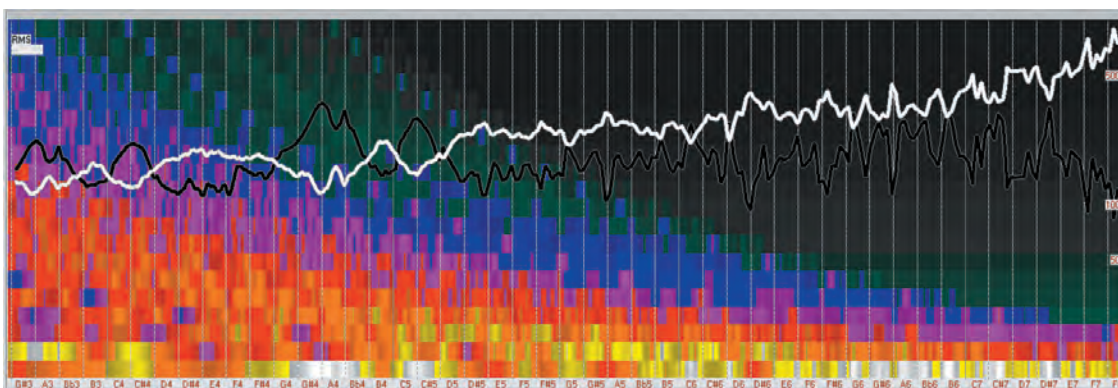
X-axis: Frequency in Hz (linear or logarithmic) or musical note

Y-axis (left): Partial (from bottom to top). First band = First partial, second band = second partial, etc.

Y-axis (right): Frequency in Hz (logarithmic)

The amplitude is colour coded.

Colour grade step= 1 dB, dark means low amplitude, bright means high amplitude.



- The graphs show the amplitude of each harmonic at **every** frequency point, in other words, the contribution of each harmonic to the sound of a playable tone.

Example: e² (E5 in American notation). The fundamental is considerably weaker than the 1st harmonic (2nd partial) and the 2nd and 3rd harmonics (partials 3 and 4) are even stronger.

Example: a¹ (A4 in American notation). Conversely, the fundamental is very strong, and the first harmonic is very weak, which is also small relative to the amplitudes of the three following partials.

- **Sound level (RMS):** The black line shows the total sound level that the instrument produces at each note
- **Harmonic Centroid:** The white line shows the sound balance point of each note. Reference scale: Frequency scale on the right Y axis of the diagram.

-  The function "Mask" is intended for wind instruments, and has **no relevance** for string instruments!

ANALYSES - SOUND ANALYSIS

Examples:

Sound level (black curve): The instrument plays very loudly between the notes G4 and C5, but the sound level falls down when the note Bb4 is played.

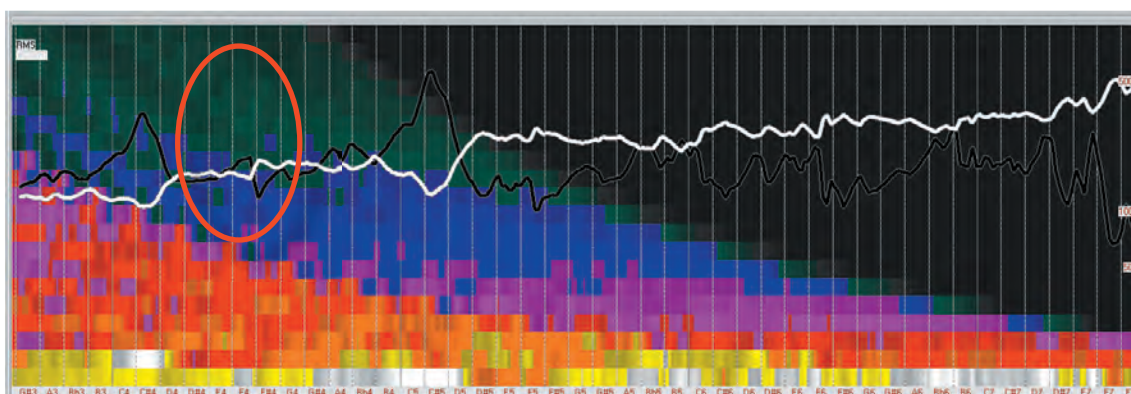
A closer look at the diagram reveals that the strong fundamental is responsible for the high RMS levels in these notes. Note that the second harmonic is weaker than the third!

Sound balance point (white curve): The strong influence of the fundamental in the mentioned range (between the notes G4 and C5) can also be seen in the white curve: The sound balance point is around 1 kHz in this range, while for lower notes it was around 2 kHz!

Except of course in the case of Bb4, where the harmonic centroid is 2.6 kHz, which means it sounds very bright

Wolf note candidates:

- The combination of sound level (black) and harmonic centroid (white) curves can reveal in certain notes a tendency to produce wolf notes.

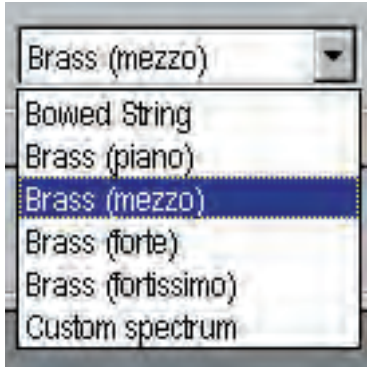


The instrument shown in this example (Schwartz 1841, Strassburg) has a wolf note at C#5

Whenever the black curve has an extreme peak, and at the same time the white curve shows a minimum, that note is a wolf note.

ANALYSES - SOUND ANALYSIS

2

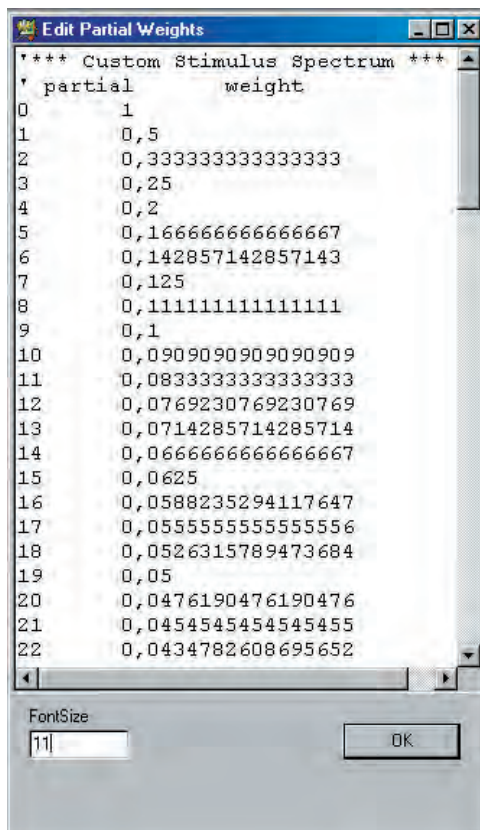


With this box a suitable excitation spectrum can be chosen.

The radiated sound of an instrument can be calculated using its “Transfer function” and the excitation spectrum: The radiated sound is the result of the excitation spectrum filtered through the Transfer function of the instrument.

The radiated sound of a **bowed string** is calculated using an excitation spectrum that corresponds to the well known **saw tooth** tooth waveform.

The “Custom spectrum” function is available, whereby an **individual excitation spectrum** can be modeled.



The left column corresponds to the harmonic number, and the right column is the weight of that harmonic in the sound spectrum.

● You only need to edit the given text guide.

1. The strongest harmonic has a weight of 1 (NOTE: The strongest partial does not have to be the fundamental, as it is shown in the example)

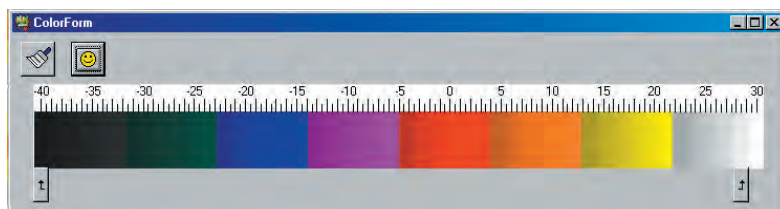
2. The amplitudes of all other harmonics are given as a percentage of the harmonic partial

Example in the left picture: The amplitude of the 3rd harmonic constitutes a fourth of that of the strongest harmonic (in this case, the fundamental). Therefore, its weight is 0.25. The amplitude of the 19th harmonic constitutes 5% of that of the strongest harmonic. Therefore, its weight is 0.05.

If for instance, the desired excitation spectrum consists of only 20 harmonics, the weight of the harmonics higher than 20 must be 0.

ANALYSES - SOUND ANALYSIS

3



Click on the brush icon to generate a colour gradient between the two colour fields marked with the arrows. The colour of all the fields that lie in between these two marked fields will be overwritten.

- The arrows can be dragged and dropped with the mouse
- Each field can be coloured individually. Click on the desired field to open a colour dialog window
- The typical procedure is as follows:
 1. Place the arrows on the start and end of the colour gradient
 2. Colourise the fields marked by the arrows
 3. Click on the brush to generate the colour gradient in the marked interval



The "Smiley" restores the standard colour gradient

ANALYSES - SOUND ANALYSIS

Remarks on making the data “audible”

The audible sounds generated by VIAS have only the stationary part of the sound, which lacks the characteristic attack and decay parts of the sound of a real instrument. That is the reason why they still sound synthetic and unnatural. The attack and decay can certainly be modeled as well, however, these tasks are highly complex, and the program would be much more complicated to use.

Furthermore, a real musical instrument sound has characteristic noise components produced by the excitation mechanism (bowing noise in bowed instruments, blowing noise in flutes, etc). The program does not simulate these noise components.

The goal of this analysis is not to produce an exact reproduction of how the sound develops in time, in the same way as when the player plays the instrument, but to make the **characteristic sound colour** of the instrument audible. Such data is for instance suitable to compare the sounds generated by different curves.

Of course, the composed "Sound" (i.e. musician + instrument) is strongly influenced by the musician. The bowing and playing technique used by the musician can weaken or change the characteristic sound colour of the instrument.

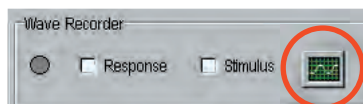
The influence that the strings have on the sound of the instrument does not have any effect on the simulated sound, as they are damped during the measurement! Only the sound characteristic of the instrument body is made audible. However, the tension of the strings affects the loudness indirectly, by exerting pressure on the top plate of the instrument. A high string pressure increases the sound level curve, and makes the sound louder.

ANALYSES - COHERENCE

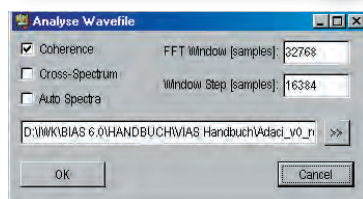
The coherence function is a measure of the degree of linear dependence between two time signals, $x(t)$ and $y(t)$, in the frequency domain. Mathematically, it is nothing more than the square of the magnitude of the average cross-power spectrum and is calculated as follows:

$$\gamma_{XY}^2(f) = \frac{|\langle G_{XY}(f) \rangle|^2}{\langle G_{XX}(f) \rangle \cdot \langle G_{YY}(f) \rangle}$$

If the coherence between an input signal $x(t)$ and an output signal $y(t)$ of an oscillating system in a frequency range is not 1, it is an indication that linear signal analysis theory is not applicable.



- With the coherence button you can check the validity of a VIAS measurement.



- Record the "Response" with the wave-recorder (see page XXX)
- Click on the shown icon to start the coherence function
- Load the recorded response-signal and click on the "OK" button

Set the window length (in samples) in the field marked **FFT Window**.

Set the step size (also in samples) in the field marked **Window Step**.

You can find more information about Coherence in Wikipedia, under the following link:

http://de.wikipedia.org/wiki/Koh%C3%A4renz_%28Signalanalyse%29



If you are reading the PDF version of this user's manual, simply click on the Internet Explorer icon (left) to go to this webpage.

ANALYSES - CROSS SPECTRUM

To be able to calculate the coherence spectrum, it is necessary to calculate as intermediate results the **cross power spectrum** as well as the **auto power spectrum**. These results can also be plotted by clicking the corresponding boxes.

The cross-power spectrum is calculated by using the following equation:

$$S_{XY}(f) = X^*(f) \cdot Y(f)$$

The equation to calculate the auto-power spectrum is as follows:

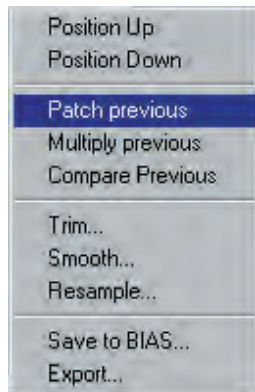
$$S_{XX}(f) = X^*(f) \cdot X(f) = |X(f)|^2$$

In both equations, the star sign (*) denotes complex conjugate

DATA POST PROCESSING

ERROR CORRECTION

To correct errors:

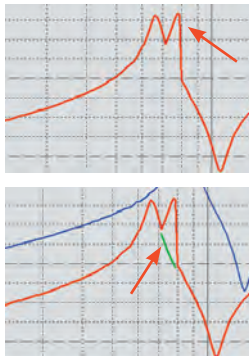


Right click on any icon corresponding to the curve that you would like to paste into the previously listed curve. The context menu will open. Click on the command “Patch previous” (the current curve will be inserted in the previous curve).

Currently Displayed Curves						
	Date	Time	Points	Line	File Name	Description
	26.07.07	14:11:39	405	—	D:\NWK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" kein Ventil gedrückt
	26.07.07	14:50:22	410	—	D:\NWK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 2. Ventil
	26.07.07	14:26:04	410	—	D:\NWK\BIAS 6.0\HANDBUC	ADACI B-Trompete "Referenz" mit 3. Ventil

NOTE:

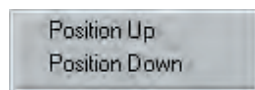
With the “Patch previous” command, the data in the previously listed curve is replaced with the data from the current curve. Any curve which needs to be improved, must **always** be listed **just before** the curve which will be used to improve it.



Example:

The red curve (see picture) has an error between 70 and 80 Hz (second peak). You have repeated the measurement only in this frequency range, which is correct (green curve).

You would like to improve the red curve with the green curve. For this purpose, the **red curve** must be listed **just before the green curve**. As this is not the case (see picture), you will need to either shift the green curve upwards or the red curve downwards.



This is done with the command “Position Up” or “Position Down” in the context menu. Let us shift the green curve upwards:

410		D:\NWK\BIAS
405		D:\NWK\BIAS
405		D:\NWK\BIAS

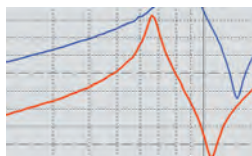
- **Right** click on e.g. the speaker icon that corresponds to the green curve
- Click on “Position up”

Now the curves are in the required order

DATA POST PROCESSING

ERROR CORRECTION

Now we can perform the error correction:

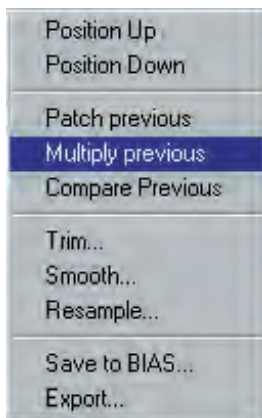
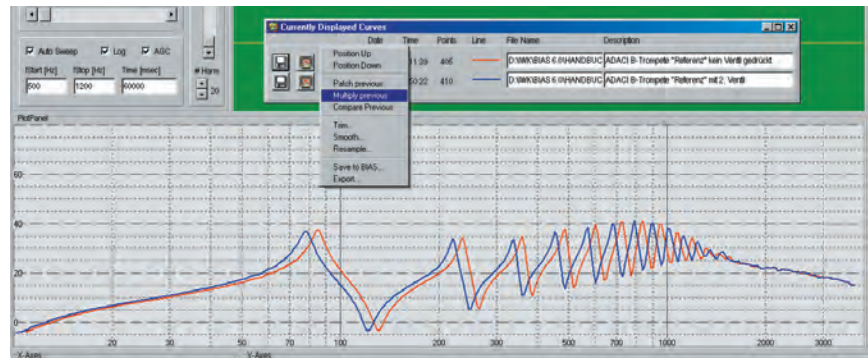


- Right click on any icon that corresponds to the green curve
- Click on “Patch previous”

Done!

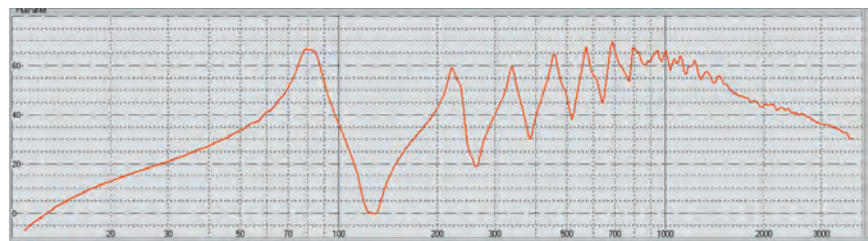
DATA POST PROCESSING

CURVE MULTIPLICATION - ADDITION



To multiply the red curve times the blue curve:

- Right click on any icon that corresponds to the blue curve
- Click on the command "Multiply previous"



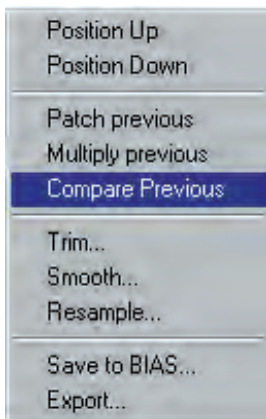
RESULT

For the commands "Multiply" and "compare", the two curves do not have to have the same length, or frequency range, or sampling frequency.

DATENBEARBEITUNG - Post Processing

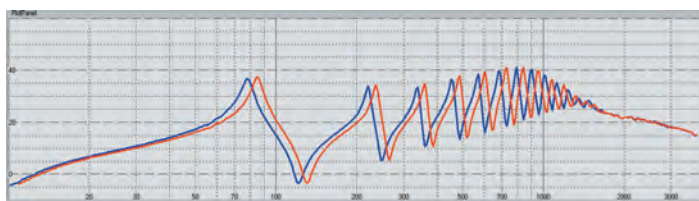
CURVE COMPARISON - DIFFERENCE

To compare the red curve with the blue curve:

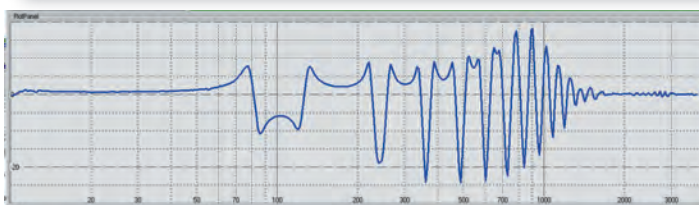


- Right click on any icon that corresponds to the blue curve
- Click on “compare previous”

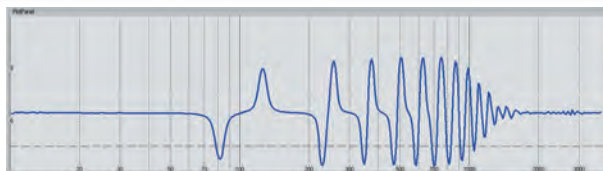
Curves to be compared



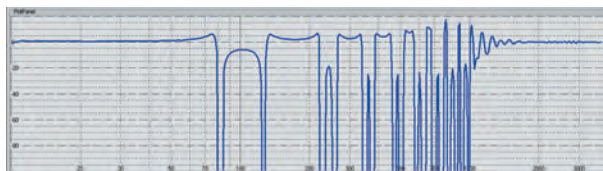
Result



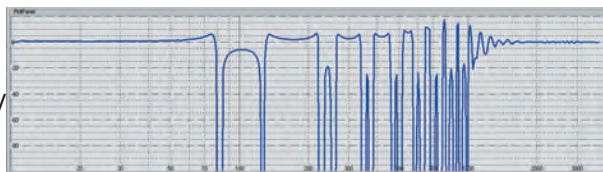
Result – phase



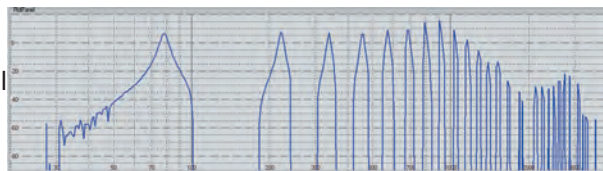
Result – Real part



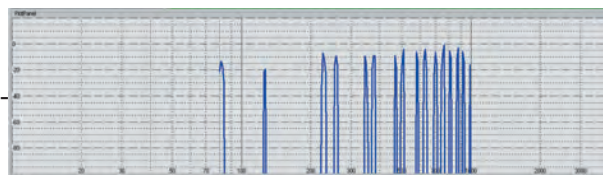
Result – imaginary part



Result – negative real part

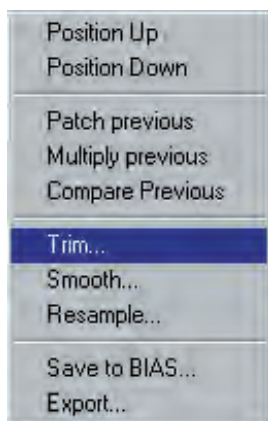


Result – negative imaginary part



DATA POST PROCESSING

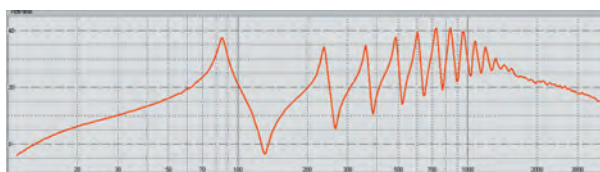
CURVE TRIMMING



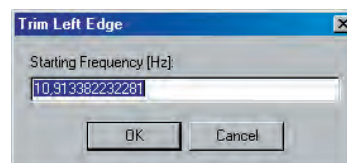
- To crop a curve, use the command “Trim” on the context menu

All data under the start frequency and over the stop frequency will be cut off

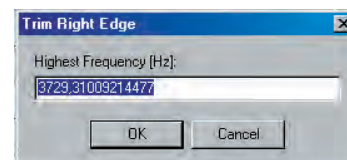
Curve to be cropped



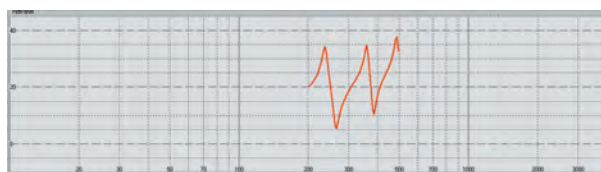
- Type in the lower frequency (e.g. 200 Hz)



- Type in the upper frequency (e.g. 500 Hz)



Result



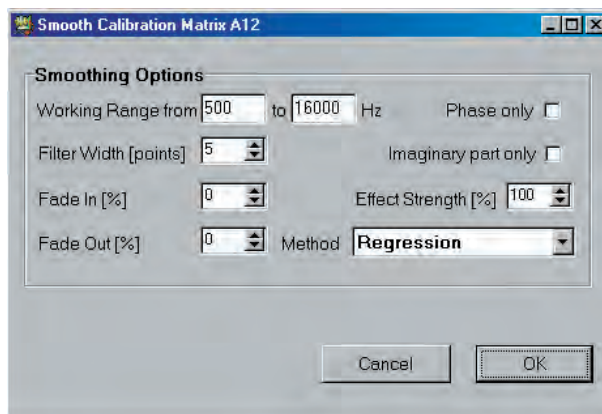
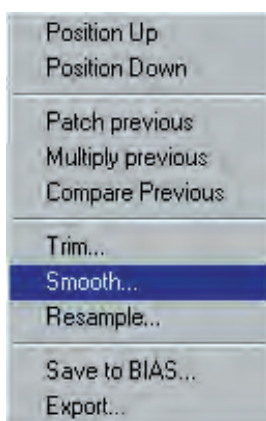
DATA POST PROCESSING

CURVE SMOOTHING

There are many reasons to smooth a curve: For instance, when detecting maxima / minima points by pressing the arrow keys does not work properly, or when the curve has got ripples, etc.

- **To define the smoothing parameters:**

In the context menu, Right click on the command “Smooth”



Parameters:

Working Range:

Here you can specify the frequency range where the smoothing will take place.

Filter with [points]: Filter (smoothing window) width.

Fade in [%] and **Fade out [%]:** If you would like to avoid having an abrupt transition between the unsmoothed and the smoothed parts of the curve, you can specify a transition range. The value 10 means e.g. that the transition range amounts to be 10% of the working range. In the picture above, it would be a width of 1,550 Hz.

Phase only and **Imaginary Part only:** The smoothing parameters will only be used for smoothing the phase or the imaginary part.

Effect Strength [%]: E.g. 50% means that the curve in the working range consists of a 1:1 mixture of filtered and unfiltered values

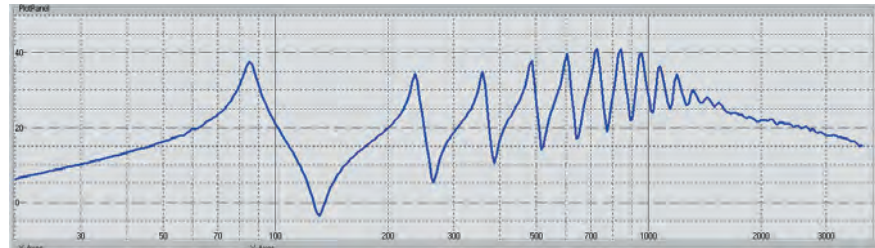
Method: Here you can choose the smoothing method: “Running average”, “Gauss-filter” and “linear regression” methods are available.

DATA POST PROCESSING

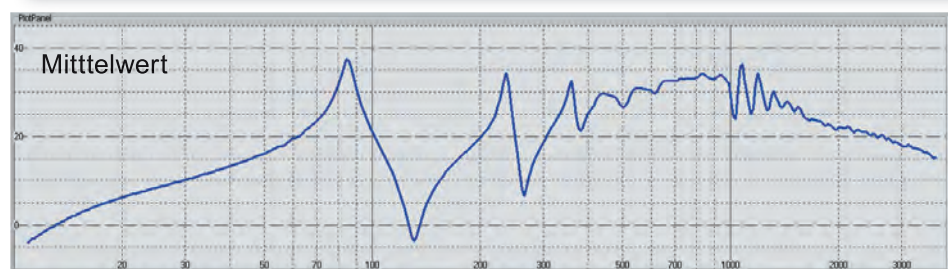
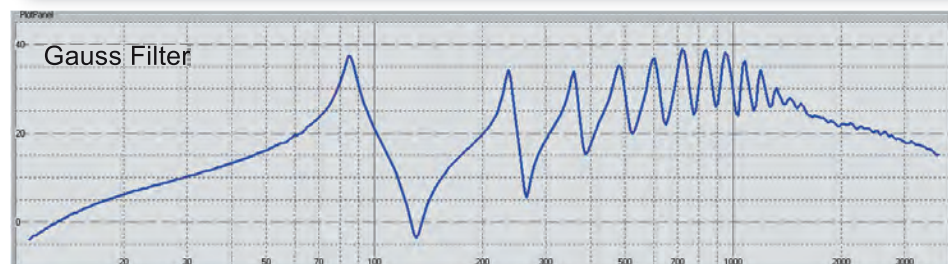
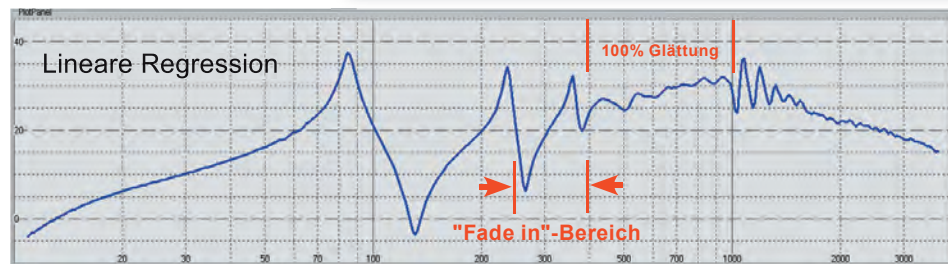
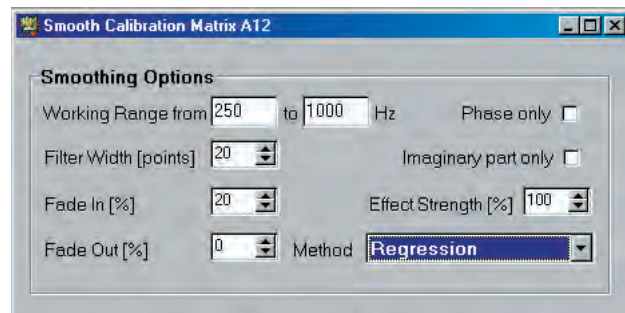
CURVE SMOOTHING

Here is a radical example:

Curve to be smoothed

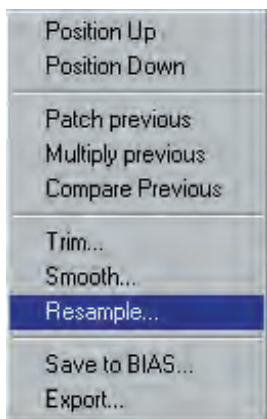


Working range: 250-1000 Hz
 Filter width: 20 points
 Fade in range: 150 Hz
 No Fade out
 Method: Linear regression

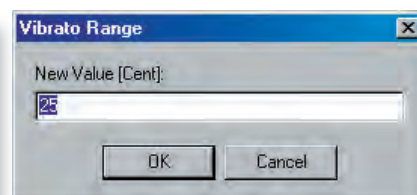
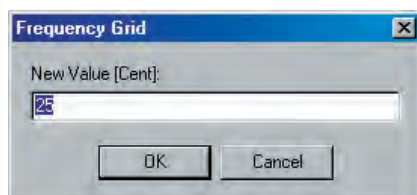


DATA POST PROCESSING

CURVE RESAMPLING



In some cases, having a higher or lower resolution than what was actually measured is required. This can be changed with the command “resample” in the context menu. Two parameters are required: GRID and VIBRATO RANGE.



GRID: With this parameter, the total number of points in the curve (hence the resolution), as well as the distance between the points in the frequency axis can be changed

Example: We have a measurement from 100 Hz to 7 kHz with 600 points.

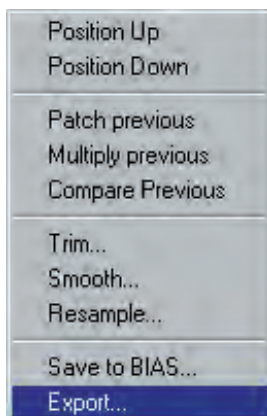
- This measurement encompasses approximately 6 octaves:
1st octave: 100-200 2nd octave: 200-400 3rd octave: 400-800
4th octave: 800-1600 5th octave: 1600-3200 6th octave: 3200-6400
- One octave has 1200 cents, and has 100 points in this measurement
- Therefore, 1 point corresponds to 12 cents
- A new grid of 25 cents per point would have half the resolution, and the curve would have 300 points
- If in this case, a value of 6 cents per point were selected, the curve would have 1200 points.

VIBRATO RANGE: Any new non-existent point must be interpolated. In this case, 25 cents means that the interpolation will take into account all points that fall within 25 cents of the new interpolated point, and not only the two closest neighbours.

Engineers need to get used to the unit cent instead of Hz. But in this case the terminology is musically relevant, as the logarithmic scale corresponds to the way pitch and musical intervals are perceived.

It is for this reason that the X-axis uses the unit cent. This ensures that the frequency resolution corresponds to how we hear.

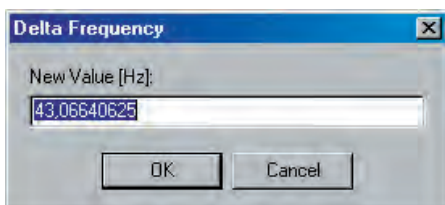
EXPORTING MEASUREMENTS



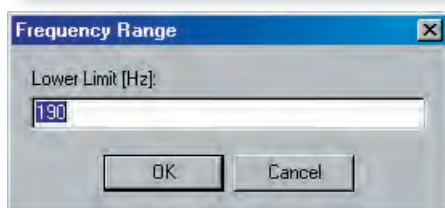
- To export a measurement in .dat format (to use with other applications):

Right click on any icon of the curve to be exported in the “Currently displayed Curves” window to open the Context Menu, and click on the command “Export”.

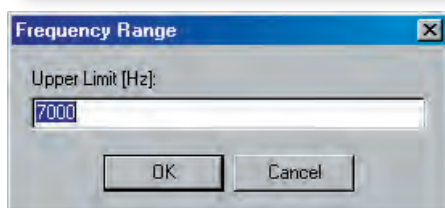
The command “Save to BIAS” is not relevant for admittance curves. If you would nevertheless like to use it, see Section 1, page XXX



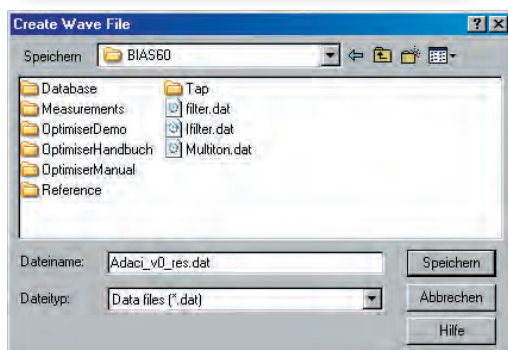
The value in this field specifies the frequency grid, e.g. a value of 0.5 means that an admittance point will be written every 0.5 Hz.



In this field, the starting frequency of the curve is specified



This value sets the end frequency of the curve



Choose the name (and extension) of the file and the directory where it will be saved.

EXPORTING MEASUREMENTS - SUMMARY

Which alternatives are there to export data, and to use it with other applications?

It is possible to:

- Save a measurement in .dat format to be used e.g. In excel (see page XXX)
- Export the maxima and minima points of a curve with frequency, pitch and pitch deviation from an equally tempered scale, and relative impedance values with the help of the NumDisplay command
- Make all the points of a curve available in text format with the help of the NumDisplay command:
 - Set the detection threshold to the value "0"
 - Copy all the content of the window, and paste it in e.g. An empty excel spreadsheet (see page XXX)
- Select the values shown by the cursor. All current values of the cursor will be inserted automatically into the windows clipboard.
- Open a text editor and paste the current values with Ctrl-V (see picture below and page XXX)

HINT:

A VIAS file is a text file with a header and 5 columns with the following values:

1. Frequency
2. Magnitude
3. Phase
4. Real part
5. Imaginary part

It is possible to read any VIAS measurement with any text editor for further work

The screenshot shows a text editor window titled 'Unbenannt - Editor' containing a table of measurement data. Red arrows point from labels below to specific columns in the table.

Frequency	Pitch and pitch deviation	Value 1st curve	Value 2nd curve	Value 3rd curve	Value 4th curve
302,326Hz	D#4-50 Cent	31,948	19,009	21,750	31,948
338,388Hz	E4+45 Cent	17,232	26,299	33,501	17,232
360,024Hz	F#4-47 Cent	21,840	34,904	10,767	21,840

SETTINGS - LOCK-IN FILTER

To change the frequency detection parameters for the display and lock-in filter:



Tuning

minFreq
10

maxFreq
16000

minLev
-60

minDet
0,6

LogAbsTol
0,3

LogRelTol
0,5

THIS IS IRRELEVANT WHEN DOING IMPULSE MEASUREMENTS!**minFreq** and **maxFreq**

The frequency range in which the signal detection of the lock-in filter will operate is set in these two fields

minLev Specifies the minimum level in dB. Levels below this will not be detected.

minDet: In this example, if more than 60% of the captured periods are consistent, the value will be shown on the graph, otherwise the display of the measured curve will be disabled, until the given number of periods are again consistent. For fast sweeps, choose a lower value.

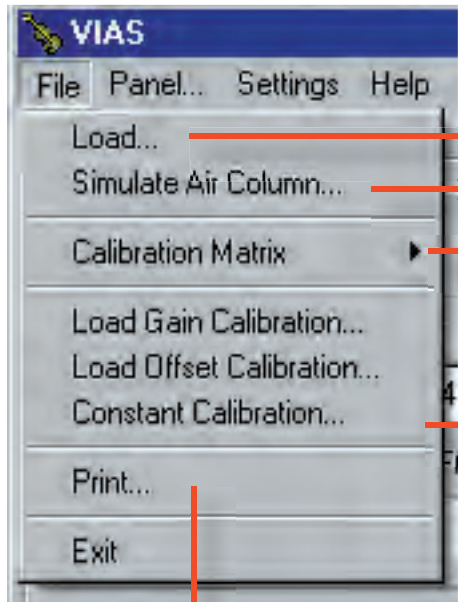
LogAbsTol The entered value must be between 0 and 1. It determines the frequency starting range.

LogRelTol: In a logarithmic sweep, the frequency distance between two consecutive measuring points should not vary arbitrarily. For very fast sweeps, choose a higher value..

Absolute tolerance is related to frequency differences in Hz, whereas relative tolerance is related to frequency differences in percentage.

COMMANDS IN THE MENU BAR

The “File” menu



Load data

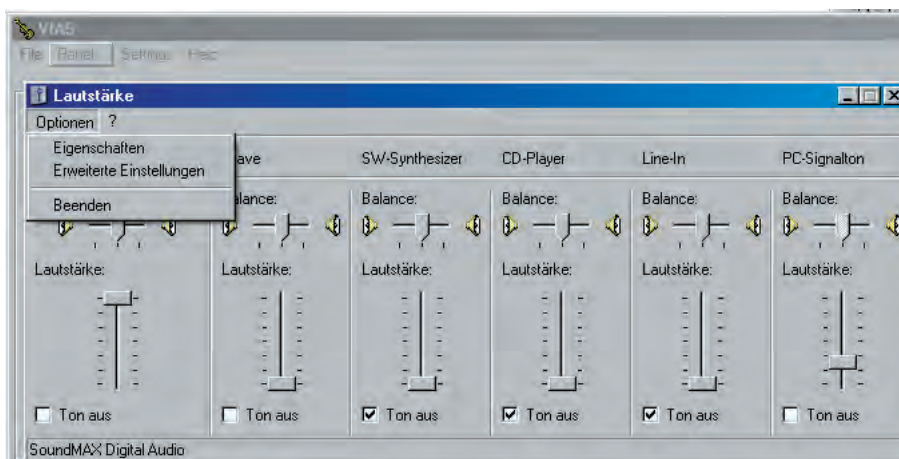
Start the “Physical modeling” tool (page XXX). **Only for BIAS!**

These commands are used to calibrate the BIAS measuring head. The first time that VIAS is run, the BIAS default calibration data will be automatically loaded. **We suggest you keep these data!** Details on how to calibrate are found at the end of this user’s manual.

With help of the VIAS program, it is also possible to obtain calibration data from two high precision (uncalibrated!) measurements of two known volumes, which can be later exported into BIAS.

This procedure requires some experience, and will be described in another section.

Prints the content of the display area with legends.



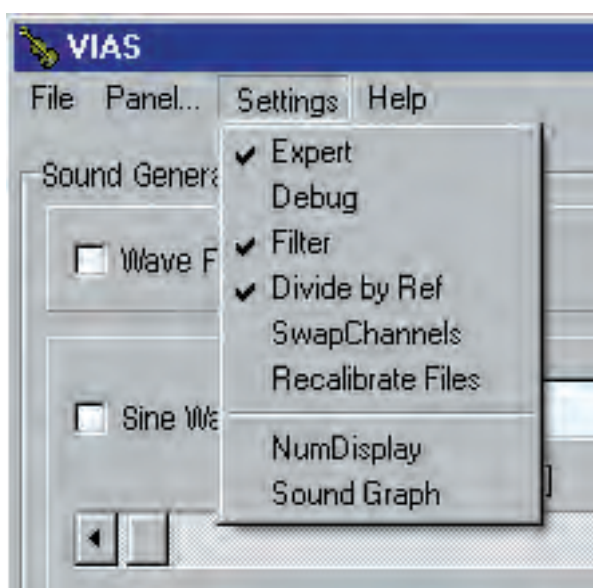
The “Panel...” command

This command opens the Windows sound card window, where you can change the settings of your sound devices.

COMMANDS IN THE MENU BAR

The “Settings” menu

In this menu, several operations can be turned on/off, so that other measuring settings can be used as well.



The picture shows the standard settings.

Expert enables access to all functions of the program

Debug opens a debug window, where intermediate data is shown

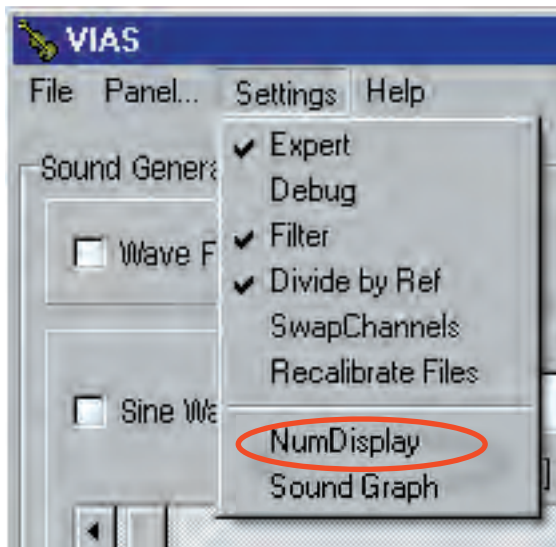
Filter turns on/off the lock-in filter. In the picture, it is turned on. If for any reason your .wav excitation file does not contain purely sinusoidal signals, the lock-in filter should be turned off, as in this case the frequency detection is not possible, and the measuring level display is disabled.

Divide by Ref: To eliminate the influence of different excitation levels on the resulting measurement, the measuring signal will be divided by the shown reference signal (actual level on the speaker membrane in the measuring head).

SwapChannels swaps the reference and measuring signal channels in the sound card. When this command is used together with the division command, the reference signal can be displayed and studied. E.g. to determine if the excitation signal is saturated, turn off the “DividebyRef” command, and turn on the “SwapChannels”. The reference signal will be displayed and plotted instead of the measured signal.

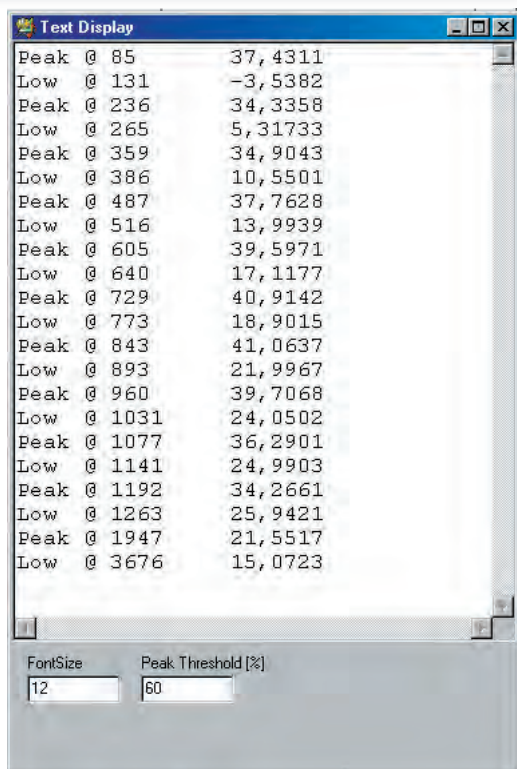
Recalibrate Files: This command loads and already saved (uncalibrated) curve, and calibrates it automatically.

COMMANDS IN THE MENU BAR



NumDisplay shows the frequencies and impedance values of the maxima and minima of a curve, which can be copied and pasted into other applications for further use.

- Turn on NumDisplay
- Choose detection threshold e.g. 50% or 60% (-> maximum or minimum should be at least 60% higher than the surrounding values).
- If the threshold is set to "0" the whole curve is displayed in numerical values. The first column contains the frequency, and the second the impedance.
- Choose the font size, if necessary

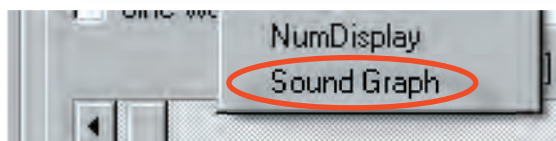


- Load a curve: The result will be shown in a text window (see picture). In the picture, a threshold of 60% was chosen. This is why only the frequency and the impedance of minima and maxima are displayed (Peak = maximum, Low = minimum).

The values can be copied to other applications for further use.

Single rows or the whole contents of the window can be marked and deleted.

- If you load further curves while the NumDisplay window is open, the values for the new curves are appended at the end of the list.



The command **Sound Graph** starts the sound analysis

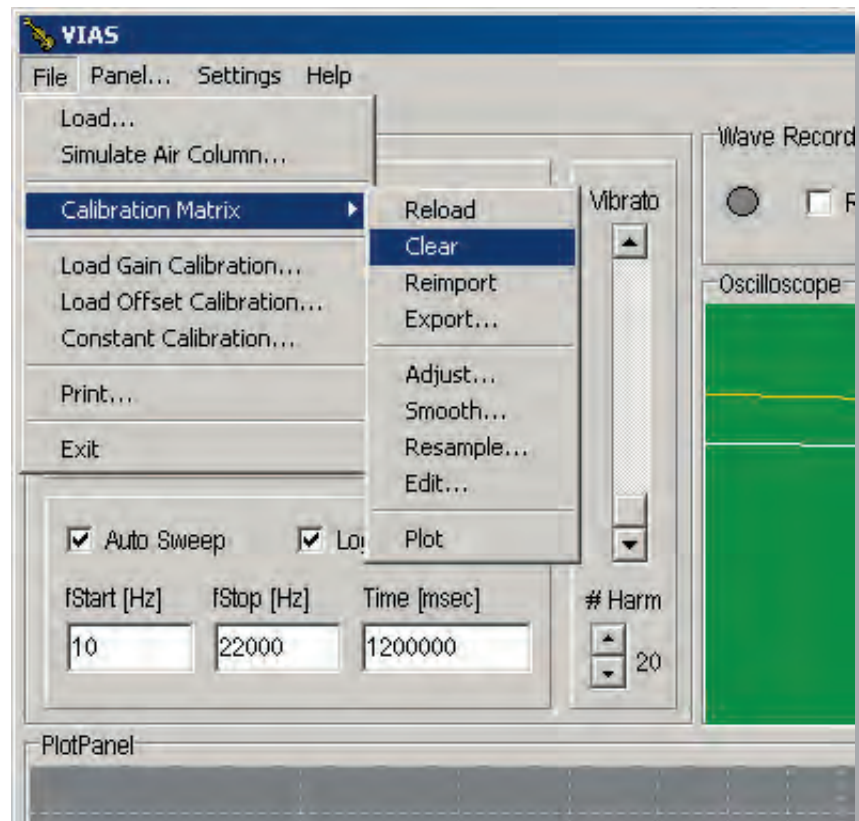
COMMANDS IN THE MENU BAR - CALIBRATION

This section describes how to generate a calibration matrix, to customise the VIAS Program to the acoustical properties of your BIAS measuring head.

In practice, it is usually **not** necessary for the end user to do this, since the VIAS and BIAS software already contains a calibration matrix that corresponds to the BIAS measuring head that was delivered with it. For the sake of documentation and transparency, this procedure is included in this section.

INTRODUCTION

- To obtain the calibration matrix, it is necessary to do an “uncalibrated” measurement! Click on the command “Calibration Matrix” on the “File” menu, and on the submenu, select the command “Clear” (see picture).



MEASURING HEAD SETUP

● To calibrate your system, you will need the following supplied parts:

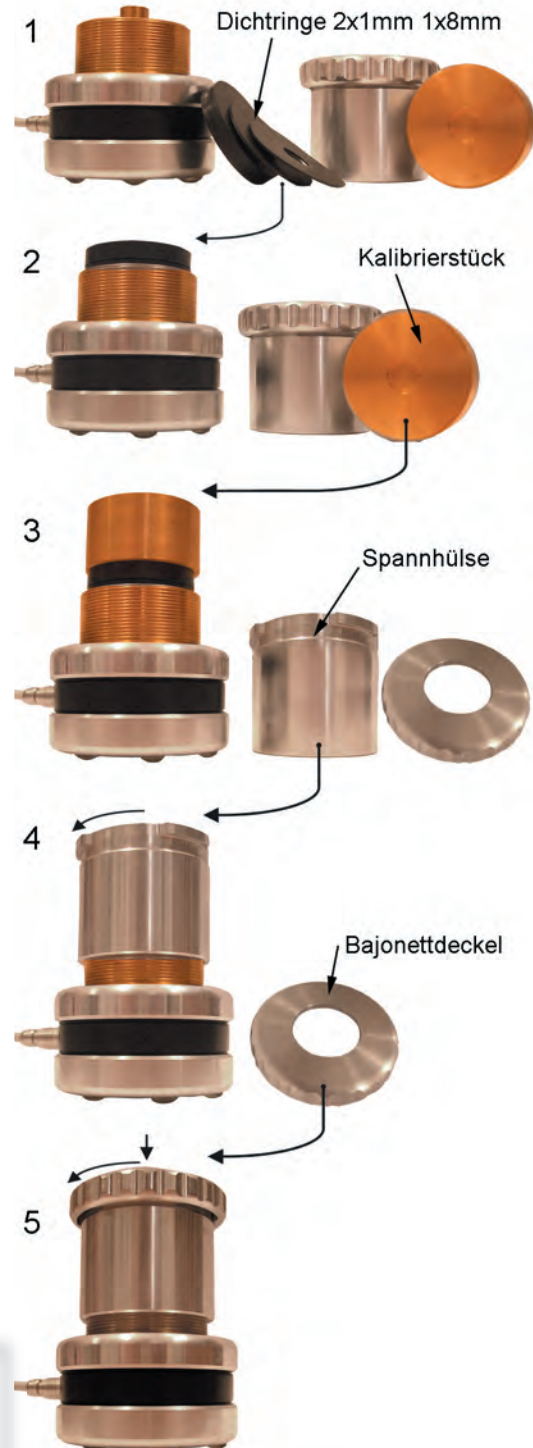
- Measuring head
- Set of sealing rings
- Calibration piece
- Clamping sleeve
- Bayonet cover

● Press the 4 level sealing rings on the measuring head. The attached rings and the top of the measuring head should be aligned in such way that together they form a flat surface

● Place the calibration piece with the 2mm deep volume facing downwards on the measuring head

● Screw the clamping sleeve over the calibration piece on the measuring head

● Place the bayonet cover on the clamping sleeve and screw the cover and the sleeve on the measuring head. The first calibration measurement can now be performed.

**IMPORTANT:**

While placing or changing the calibration piece, **never** leave the clamping sleeve mounted on the measuring head: Always unscrew it first! The air pressure caused by letting the volume fall in/out of the clamping sleeve damages your microphone!!!

CALIBRATION MEASUREMENT

- After having placed the small volume on the measuring head, start a measurement within the desired frequency range (which is set by typing the Start- and Stop- frequency, as it is done in a normal BIAS measurement).

Be patient!

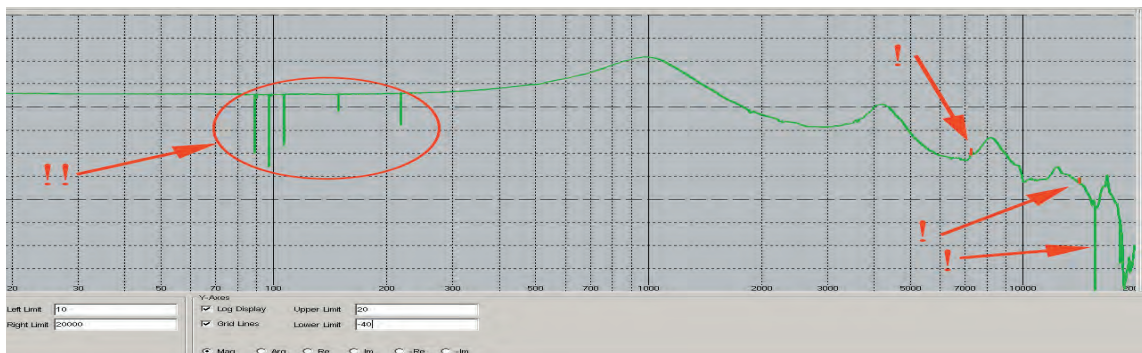
- A calibration measurement in the frequency range between 10 Hz and 12 kHz should never last less than 10 minutes (600,000 seconds)!
- Make sure that the **AGC** box is checked and that the volume scrolling bar is set at its maximum before starting the measurement.

Press the “Start” button, and watch the oscilloscope window. Both oscillations often saturate (look like a square signal). Wait until the AGC function has compensated, and the oscillations shown in the oscilloscope window are again sinusoidal (round).

- Shortly after, press the “Stop” button and discard the measurement.
- Start the measurement again. This time, no saturation should occur.

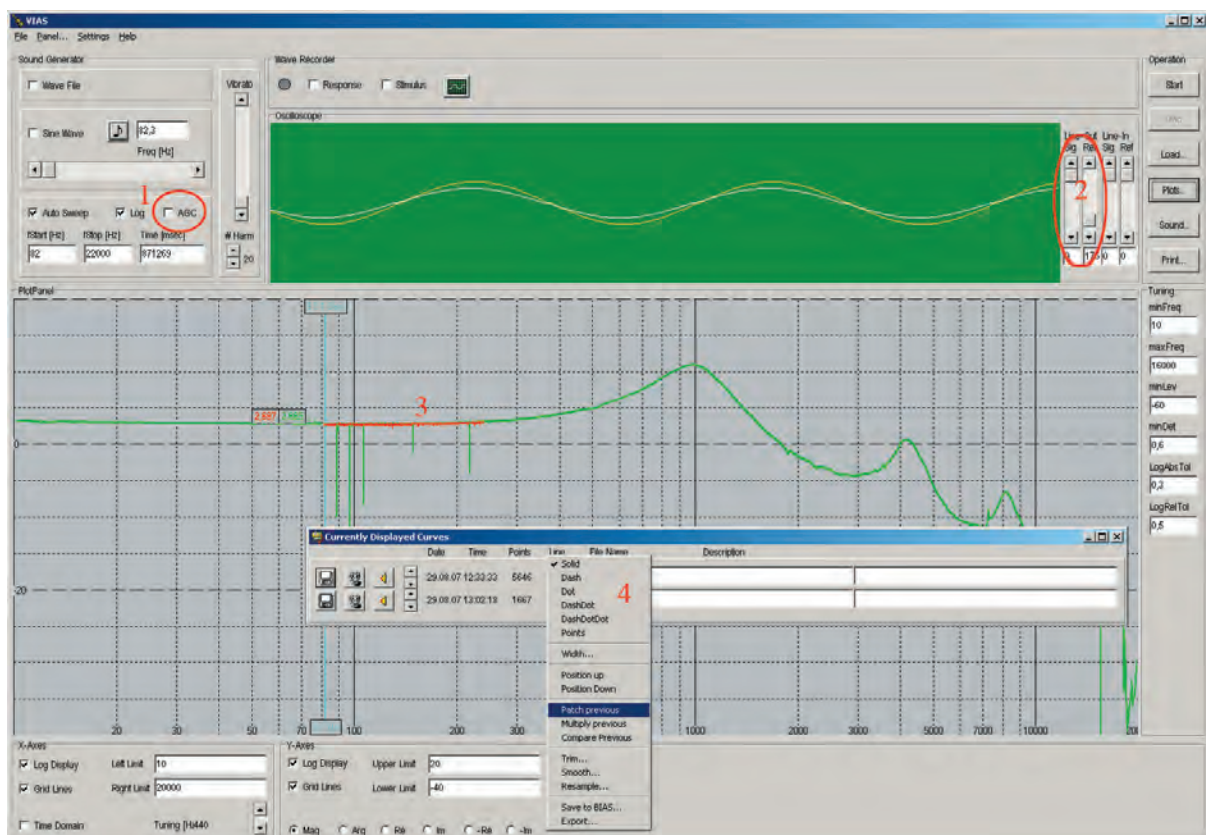
Once the measurement is completed, a plot similar to that shown in the picture will appear in the “Plot” window.

The AGC level adjustment sometimes leads to flaws in the measurement: e.g. at high frequencies it is not possible for the AGC to adjust the level of the excitation signal fast enough (see the range shown in red in the picture). It will be necessary to remeasure these points manually. Saturation in the measured signal will be automatically shown in **RED** in the “Plot” window.



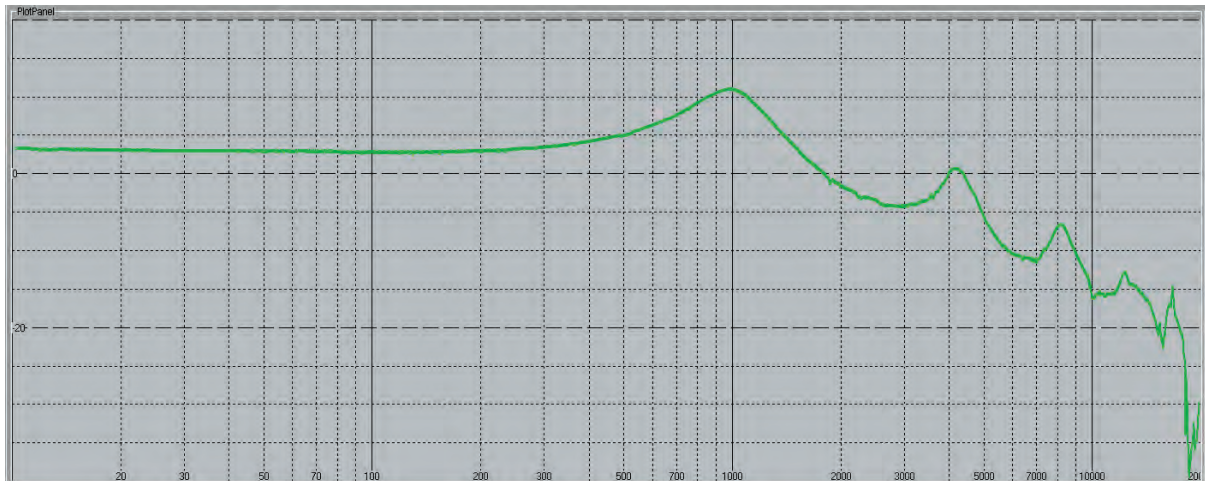
MANUAL CORRECTION OF MEASURING ERRORS

- Choose a different colour for the current curve
- Disable the AGC function (1). Find out with the aid of the cursor, the start and end frequencies of the range with errors, and start a new measurement within this new frequency range.
- Monitor the oscilloscope window closely: In case of saturation, stop and discard the measurement.
- Adjust the sound level with the scrolling bar “Line-Out-Ref”, until a good (not saturated) signal is displayed in the oscilloscope window (2).
- Stop the measurement as soon as the new curve has gone through the interval to be corrected. Measuring errors that lie close together (as in the range marked red on the left of the picture) can be corrected in one measurement (3).
- Use the “Patch previous” function to replace the old measurement with the new error free measurement (4).



MANUAL CORRECTION OF MEASURING ERRORS

After having corrected all the intervals with errors, the new error free measured curve should look like the one shown in the picture.

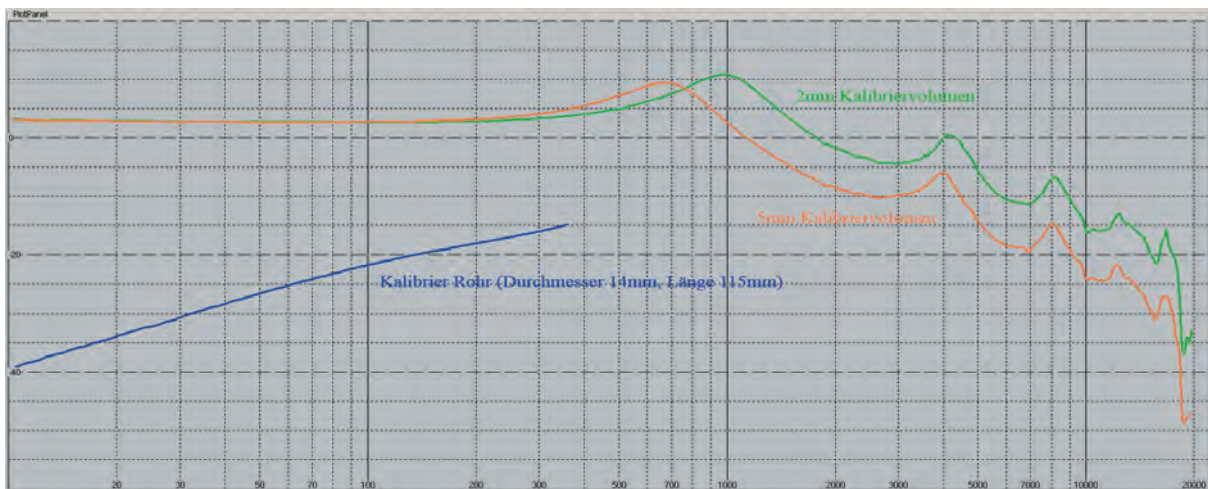


- Save this measurement as .via file. It is recommended to include in the file name the volume (V2 for the 2mm volume or V5 for the 5 mm one), as well as the frequency range of the measurement (e.g. “V2_10_20k.via”)
- After having successfully saved the file, it is recommended to smooth and resample the curve by executing the command “**Resample**”, using the standard settings (both values at 25 cents).
- Save this last measurement with a new name, e.g. under “V2_10_20k_res.via”
- Change the calibration piece (**CAUTION: Unscrew the clamping sleeve first!**) and repeat the measurement with the new calibration volume.

CALIBRATION IN THE LOW FREQUENCY RANGE

- To increase the quality of the measurement at low frequencies, it is possible to do a third calibration measurement using a tube of approximately 15 mm length.
- This tube should have a constant inner diameter of at least 14 mm and a smooth surface. Make sure that the coupling between the tube and the measuring head is air tight!
- It is important that the first resonance of the tube lies well over 400 Hz! You can determine in advance if this is the case by e.g. doing a simulation
- Set the frequencies “fstart” to 10 Hz and “fstop” to 400 Hz.
- The measurement will take approximately 12 seconds to complete. Proceed as when measuring with the calibration volume (calibration piece).

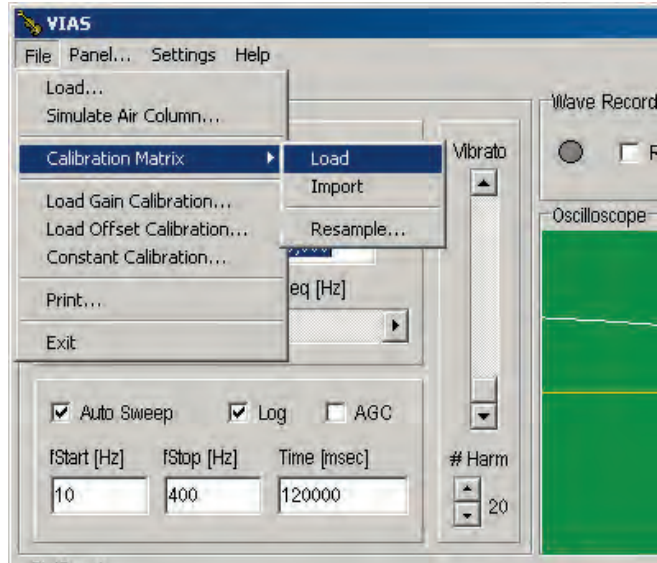
The following picture shows the plot window with the three calculated calibration curves



GENERATE THE CALIBRATION MATRIX

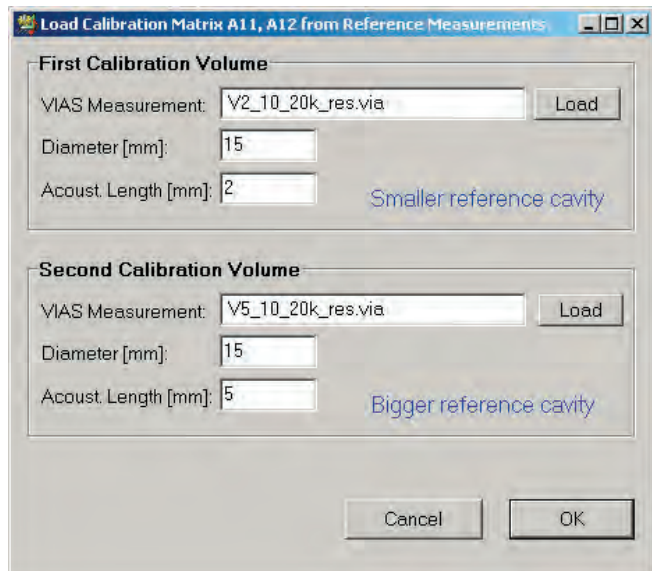
In the next step, the three measurements will be loaded into the so called “Calibration Matrix”. Calibrated measurements can be performed only after the calibration matrix has been generated.

- In the “File” menu, click on “Calibration Matrix” - “Load”



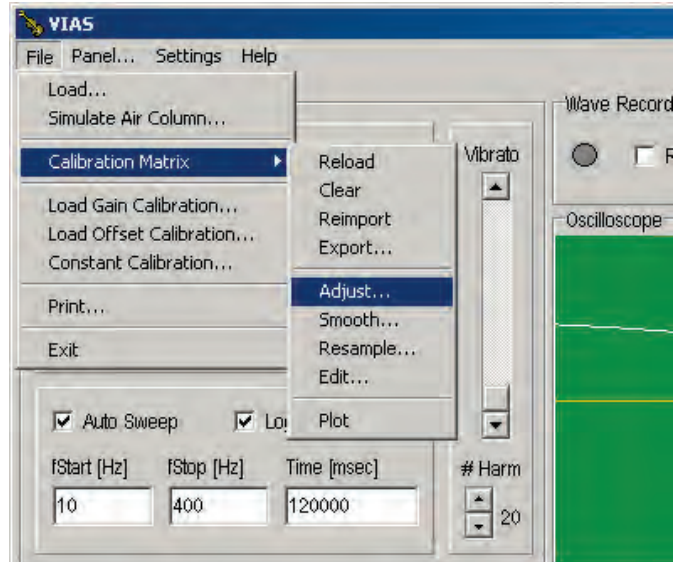
- Load the two curves measured with the calibration volume, the top one with the small volume, and the bottom one with the big volume.

If you have used a calibration piece with other dimensions, you must type in the corresponding dimensions in the fields “Diameter” and “Acoustic length”

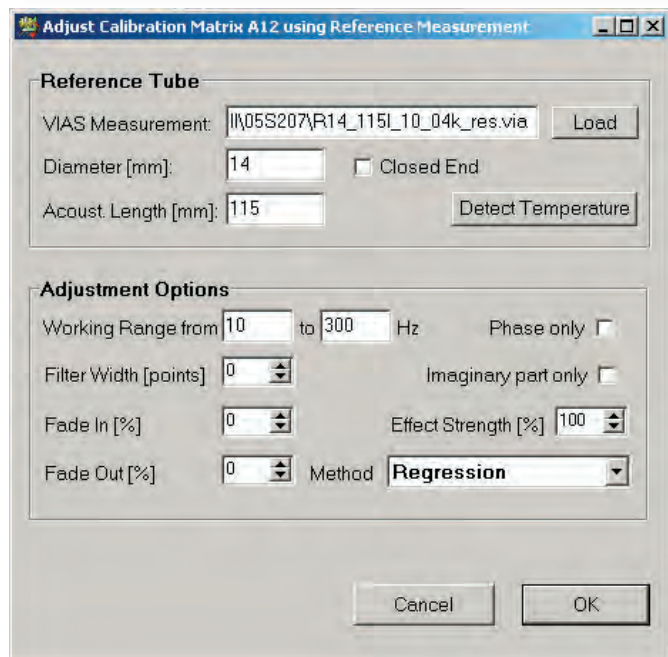


GENERATE THE CALIBRATION MATRIX

- In the “File” menu, click on “Calibration Matrix” - “Adjust”

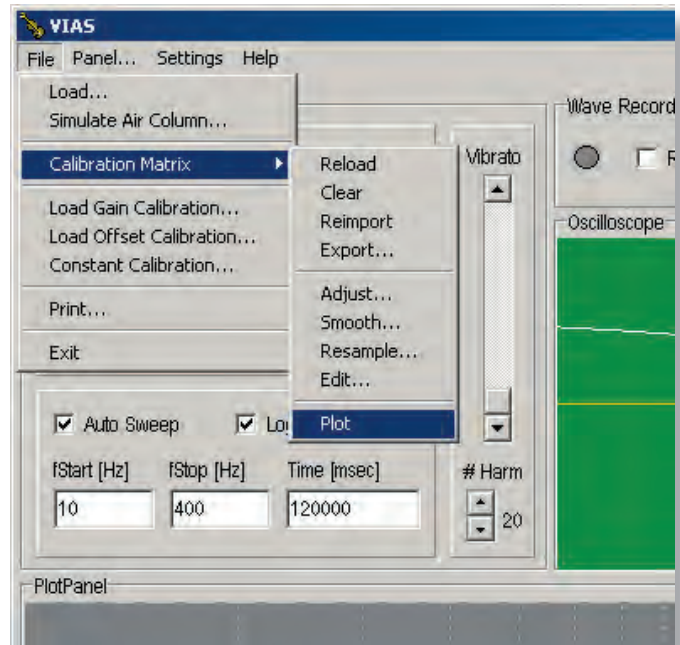


- Load the additional measurement with the tube, and type in the fields “Diameter” and “Acoustic Length” the values that correspond to the inner dimensions of the tube used for the measurement.
- Set the “Working Range” to 10 to 300 Hz
- Set the “Filter Width [points]” to 0 (see picture)
- Press the “OK” button

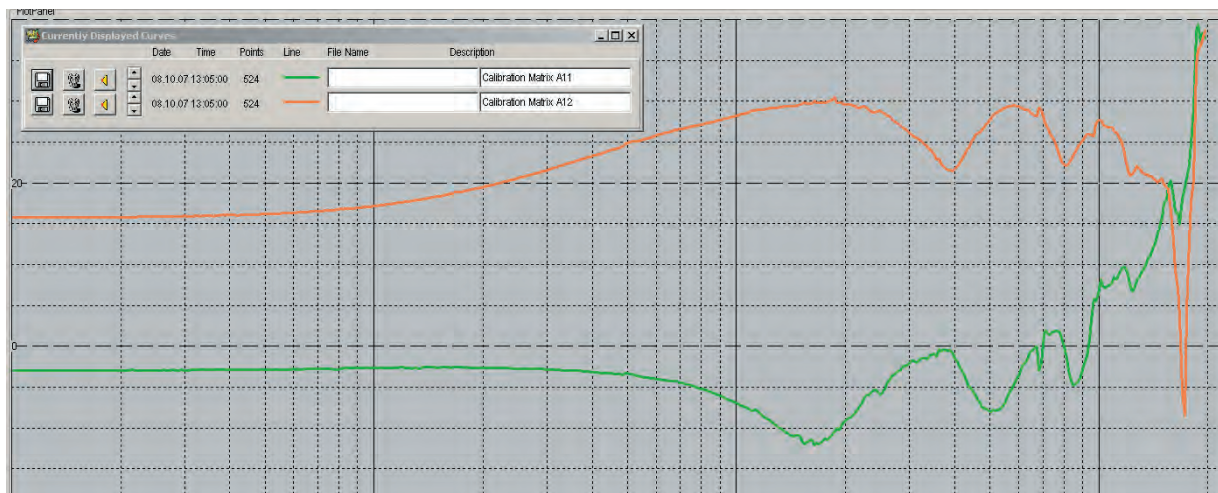


GENERATE THE CALIBRATION MATRIX

- Verify that the “Calibration Matrix” is correct, by looking at the plot generated by clicking on “File” - “Calibration Matrix” - “Plot”



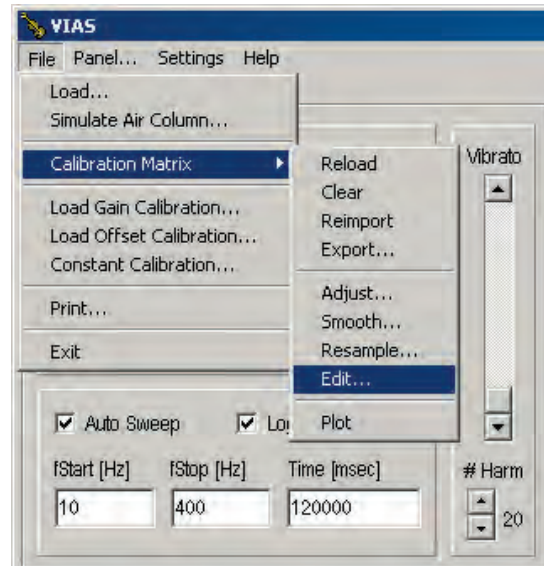
The picture shows the plot of a typical “Calibration Matrix” with a frequency range from 10 Hz to 20 kHz



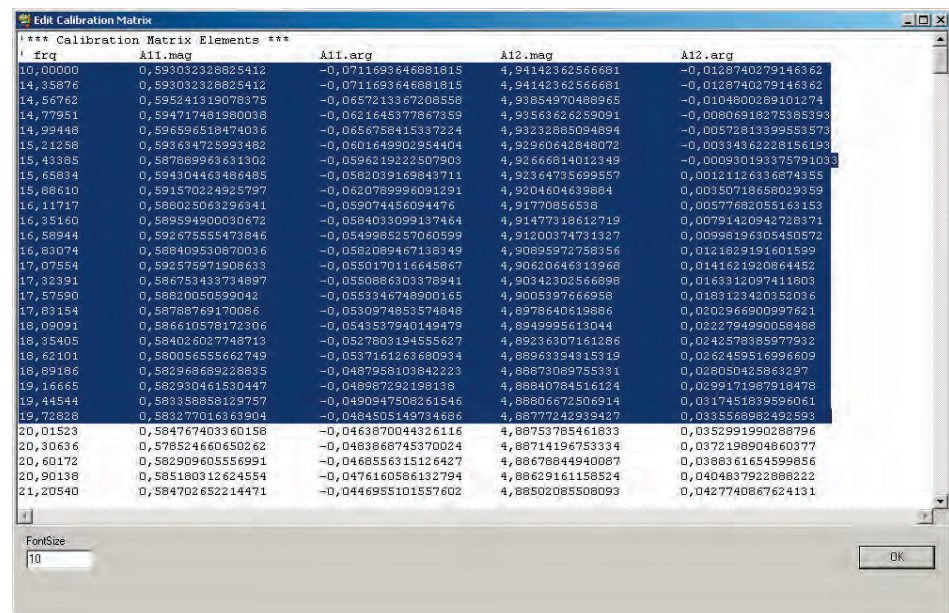
EDIT THE CALIBRATION MATRIX

In case the frequency range between 10 Hz and 30 Hz appears to have significant errors or fluctuations, you can smooth the curve directly by editing the calibration matrix. This step is optional, and not always necessary.

- In the “File” menu, click on “Calibration Matrix” – “Edit...”. The “Edit Calibration Matrix” window will open (see following picture)



- Change the font-size to “10”

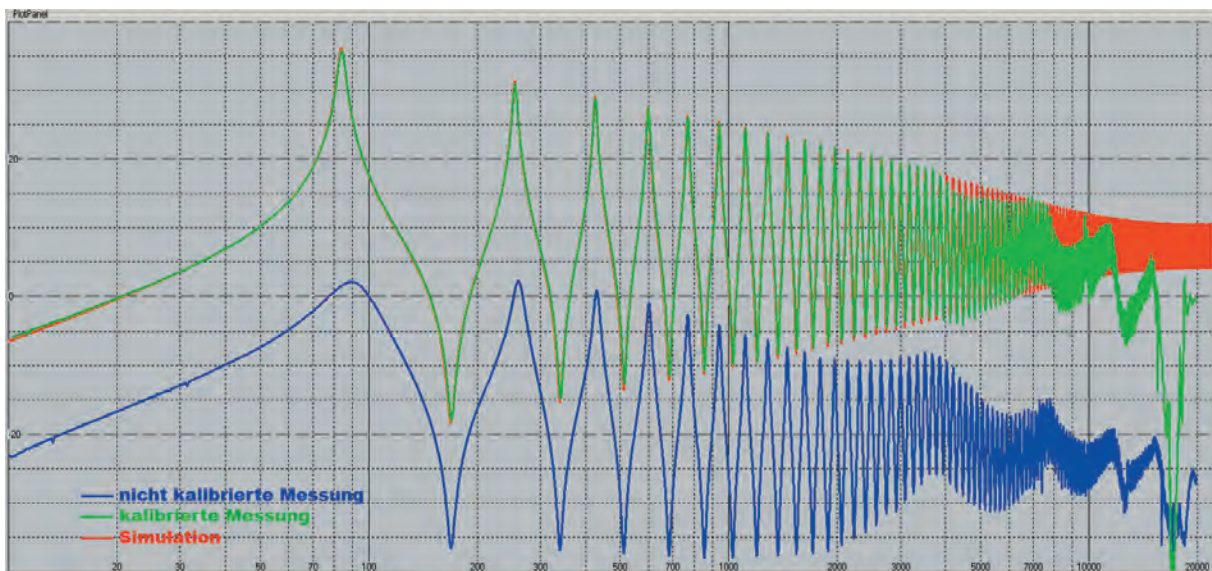


- Mark the range including the first data row (as shown in the picture) up to the frequency where no more changes should be made (usually this range is between 10 Hz and 30 Hz). The first column of the matrix corresponds to the frequency.
- Delete the marked range by pressing the “Del” key on your keyboard.

EDIT THE CALIBRATION MATRIX

- Mark the first data row and copy the data to the clipboard ("Ctrl + C")
- Paste the contents of the clipboard by pressing "Ctrl + V" **as first data row**
- Mark with the mouse pointer the first **frequency value** and change that value to 10.00000
- Click "OK"

To verify the modification, it is recommended to plot the calibration matrix again.



The picture shows a comparison between a non calibrated measurement, a calibrated measurement and a simulation.

Short description of measuring methods

BIAS input impedance measurement

Excitation with a sinusoidal sweep (10 Hz to 4096 Hz). Duration: 2 seconds. The reference signal (from the microphone placed over the loudspeaker membrane in the measuring head) and the measured signal (from the microphone inside the mouthpiece) are temporarily recorded in .wav format, and the measured signal is divided through the reference signal.

The spectrum (FFT) of this standardised signal (measured signal divided through reference signal) is calculated, resulting in the impedance curve. Transformation from time to frequency domain.

Advantages: Provides resulting impedance curve very quickly.

Disadvantages: Suboptimal use of the energy bandwidth, no flexibility in terms of frequency range.

VIAS input impedance measurement

Excitation with a slow sweep (frequency range: arbitrary, supplied by the user). Duration: Arbitrary, depending on frequency range to be measured. The measured signal (from the accelerometer) is filtered through an octave band lock-in filter, and then divided over the reference (excitation) signal.

The RMS level is calculated from this quotient, and displayed as impedance curve.

Advantages: Allows sufficient time to build standing waves, better use of the energy from the excitation signal, flexible frequency range, higher accuracy.

Disadvantages: Long measuring time

VIAS impulse hammer admittance measurement:

Excitation with an impulse. Reference signal (from the force sensor built in the impulse hammer) and measured signal (from the accelerometer) are stored temporarily, and the measured signal is divided through the reference signal.

The spectrum (FFT) of this quotient is calculated and displayed as admittance curve.

Advantages: Very short measuring time, unproblematic and flexible measurement setup, reproducible results when averaging.

Disadvantages: Small amplitudes from around 4 kHz. If results over 4 kHz are required, more measurements are necessary.

VIAS admittance measurement with shaker

Excitation with a slow sweep (frequency range: arbitrary, supplied by the user). Duration: arbitrary, depending on frequency range to be measured. The measured signal (from the accelerometer) is filtered through an octave band lock-in filter, and then divided over the reference (excitation) signal.

The RMS level is calculated from this quotient, and displayed as admittance curve.

Advantages: Generation of higher oscillation amplitudes, flexible frequency range, high accuracy.

Disadvantages: Long measuring time, only possible if the mass of the excitation mechanism is negligible

