

# Manual for the sound card oscilloscope V1.24

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Web-page: [www.zeitnitz.de/Christian/Scope/Scope\\_en.html](http://www.zeitnitz.de/Christian/Scope/Scope_en.html)

**The sound card oscilloscope is a digital oscilloscope with an integrated signal generator, frequency analysis (FFT) and wave file recorder**

## 1 Requirements

- Windows 2000 , XP or Vista
- A PC with a sound card installed.
- 50MB of disk space

## 2 Installation

Unpack the ZIP file in any directory and run setup.exe. The program can be started thereafter through the program menu of the Windows operating system.

## 3 Description

This software can be used for the display and analysis of sound waves. The data can be recorded both directly from the sound card (with a microphone or LINE input), or from a source such as a CD or Mediaplayer. The input to the oscilloscope is defined by the Windows sound mixer (see below). The software obtains its input data for the sound card via the Windows interface. It does not communicate directly with the sound card. Therefore sound card problems should be troubleshot at the operating system level.

The user interface is arranged like a conventional oscilloscope. However, in the program window, additional XY display, frequency analysis, and settings are provided.

### 3.1 Oscilloscope

The software shows the left and right channel of the sound card in the oscilloscope window. The left channel is represented as a green line and the right channel as a red line. In the user interface window there are knobs and input windows for the following three functions: Amplitude, Time, and Trigger.

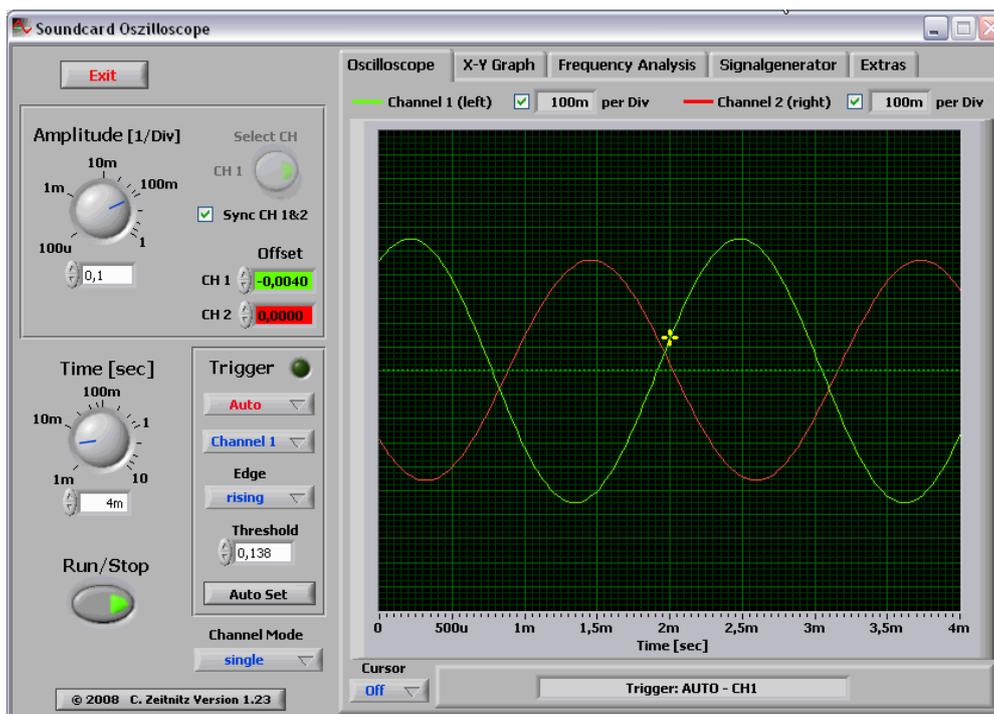


Figure 1: Soundcard oscilloscope

### 3.1.1 Amplitude settings

The amplitude scale of the two channels can be set independently as well as synchronized. The latter is enabled at program start-up and can be disabled by un-checking "Sync CH 1&2" on the front panel. In the case of independent channel control, the active channel has to be selected by the button "Select CH" (see Figure 2).

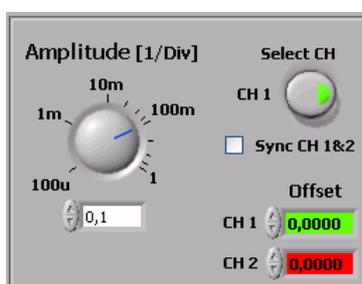


Figure 2: Amplitude settings and channel offsets

The amplitude values are given in units per division of the oscilloscope screen and are displayed for both channels above this screen. The amplitude value corresponds to the digitized sound level divided by 32768. This represents the 16Bit resolution of the data, which are taken with the sound card. Due to the different settings of the volume in Windows the absolute sound level cannot be determined directly! Therefore the presented values are to be understood in arbitrary units. The amplitude setting refers both to the oscilloscope window and to the XY graph. An offset can be assigned to each channel individually via the appropriate input window; thereby the two traces can be separated from each other.

### 3.1.2 Timebase

The **Time** setting refers to the entire represented range and **NOT** to the value per unit as with a normal oscilloscope! The range goes from 1ms to 10,000ms. The larger the range, the smaller is the used scanning rate. This is unavoidable because of the extent of computer cpu use. In the trigger setting "single" the scanning rate is increased again, since computer utilization is less important here.

### 3.1.3 Trigger

The trigger setting modes are "off", "auto", "normal" and "single". These correspond to the standard modes of oscilloscopes. The trigger threshold can be adjusted either in the input window of the trigger selection, or by shifting the **yellow cross** in the oscilloscope window using the mouse. The trigger time can only be adjusted by shifting the cross with the mouse.

In the single SHOT mode of the trigger the RUN/stop switch is deactivated automatically and must be pressed again for a new data-taking run.

The button "Auto set" triggers the program to estimate the optimal time base and trigger level. The main frequency found in the trigger channel is used to obtain the time base. The threshold is taken from the signal amplitude. If the amplitude is too small, the button has no effect. Below approx. 20Hz the result is not reliable due to the limited time window used for the analysis.

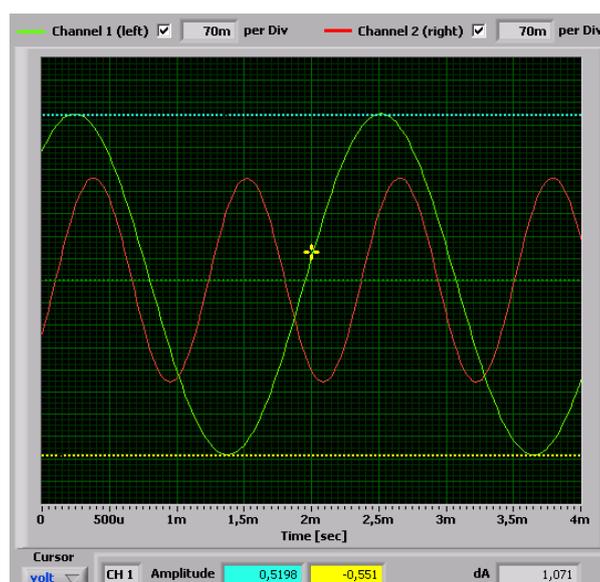
### 3.1.4 Channel Mode

By default, two channels are shown in the oscilloscope window. With the mode selection switch at the bottom of the program window, the sum, difference or product of the channels can be chosen.

### 3.1.5 Data Analysis

On the user interface there is also a run/stop switch, which can be used to interrupt data taking to allow time for analyzing the current window content. The amplitude or Time/frequency can be measured with the help of cursors in the oscilloscope window. The corresponding cursors can be activated through the selector box underneath the window. These can be shifted with the mouse.

In the amplitude mode the values for the two cursors as well as the amplitude difference is displayed.

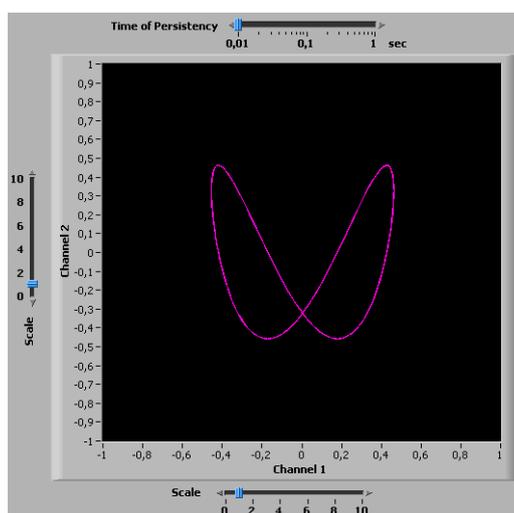


**Figure 3:** Amplitude analysis with the cursors. The shown values correspond to channel 1.

For the time mode the time difference and the appropriate frequency are shown directly. The data can also be examined in more detail by using the zoom. The detail around the position of the perpendicular trigger line is increased. By shifting the trigger line the range can be changed.

## 3.2 X-Y Graph

Here the two channels are displayed against each other. Thereby e.g. Lissajous figures can be produced. For this the frequencies can be adjusted in the signal generator.



**Figure 4:** Lissajous Figure for  $f_1 = 440\text{Hz}$ ,  $f_2 = 880\text{Hz}$  and a phase of  $45^\circ$

The slider above the graph allows to change the time of persistency of the shown data. For a longer time setting increases the time window displayed on the screen. Fast changing signals should better be displayed with a short persistency.

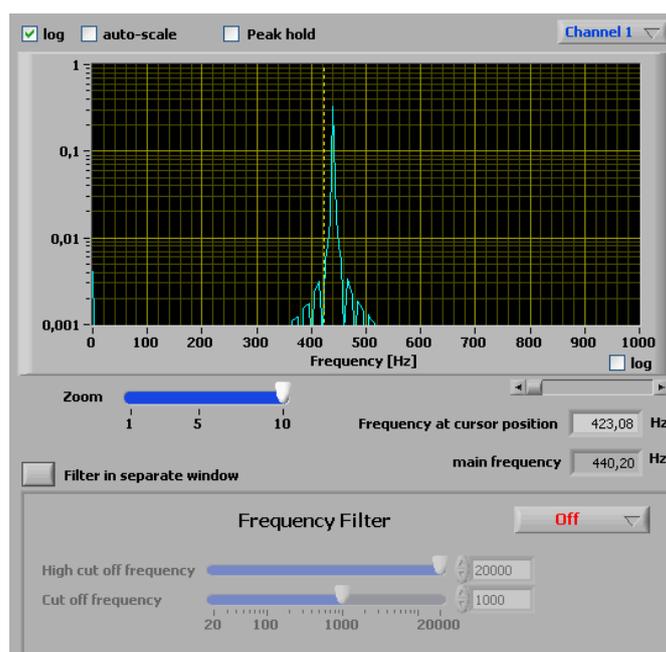
The controllers along the x and y axis permit a scaling of the appropriate channel. The represented range is chosen by adjusting the amplitude knob in the program window.

### 3.3 Frequency Analysis

In the "frequency analysis" window, the display shows the result of the Fourier analysis of the selected channel. The channel can be chosen with the selection button above the grid. By default, the graph shows the amplitude of 0 - 10,000 Hz. The amplitude as well as the frequency can be displayed with a logarithmic scale.

The vertical scale can automatically be adjusted by selecting the auto-scale check-box above the graph. A manual adjustment is possible by double-clicking the maximal or minimal value of the axis and entering a new value. This should be done only if auto-scale is disabled.

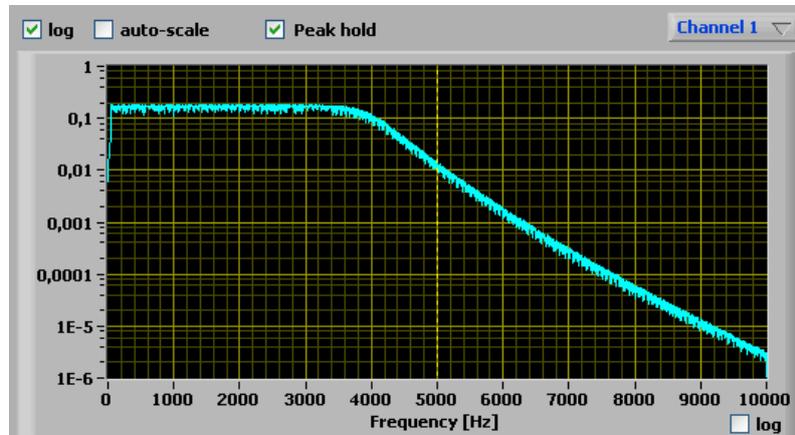
Below the graph is a roll bar and a zoom sliding control; they permit the indicated range to be changed. These should be only used if data taking has been stopped with the run/stop button. The zoom shot slider shows details of the frequency analysis: use the mouse to set the perpendicular yellow line to the frequency of interest and drive the zoom shot slider up to the desired detail.



**Figure 5:** Frequency analysis of a 440Hz signal

The two output values underneath the sliders show the frequency at the cursor position and the value of the strongest frequency found from a harmonious analysis of the data. Note that the Fourier analysis is always based on data with the full sampling rate of 44.1kHz. Therefore, the time controller automatically jumps to a pre-defined value when this window is active.

Selecting "peak hold" allows to store the maximal amplitude values of the Fourier analysis. This allows to display the transfer function, when using the white noise generator.



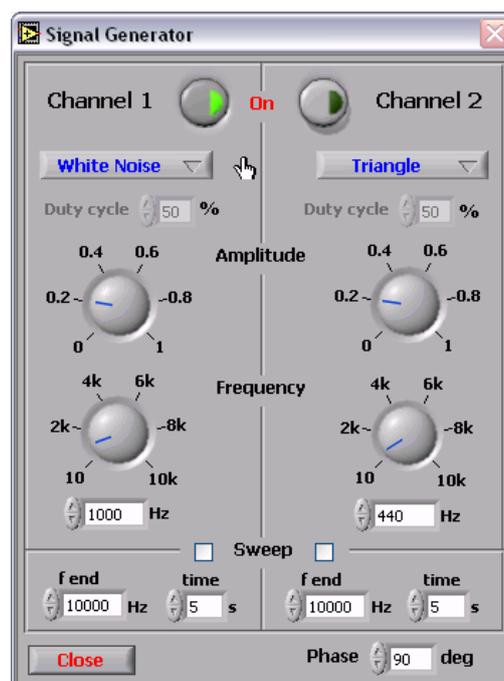
**Figure 6:** Transfer function utilizing the peak hold function with the white noise generator

Under the frequency analysis an adjustable frequency-selective filter (Besselfilter 10th order) is also provided. Three kinds of filter can be selected: Low-pass, high-pass and band-pass filter. The critical frequencies can be adjusted with the sliding controls accordingly.

Above the frequency-selective filter is a button to open filter control in a separate window. This function allows one to observe the effect of the filter directly in the oscilloscope window. Double-clicking on the button or closing the window re-establishes the original settings.

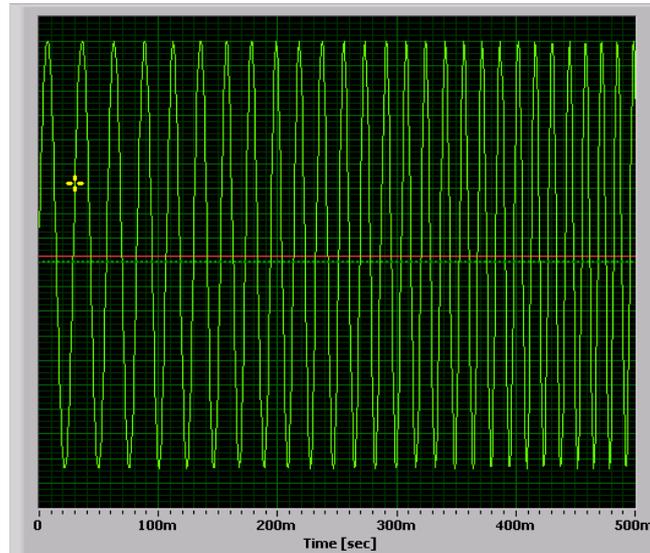
### 3.4 Signal Generator

A 2-channel signal generator is integrated into the program. The generator can be released from the program window by pushing the button above the panel. A second click on the button will embed the generator again.



**Figure 7:** Signal Generator

The generator outputs sine, rectangle, triangle, and saw tooth waves with variable amplitude and frequency. A white noise generator is included as well. The phase of the signal can be adjusted. The "Sweep mode" allows to sweep the frequency from the main frequency to f(end) continuously within the specified time window.

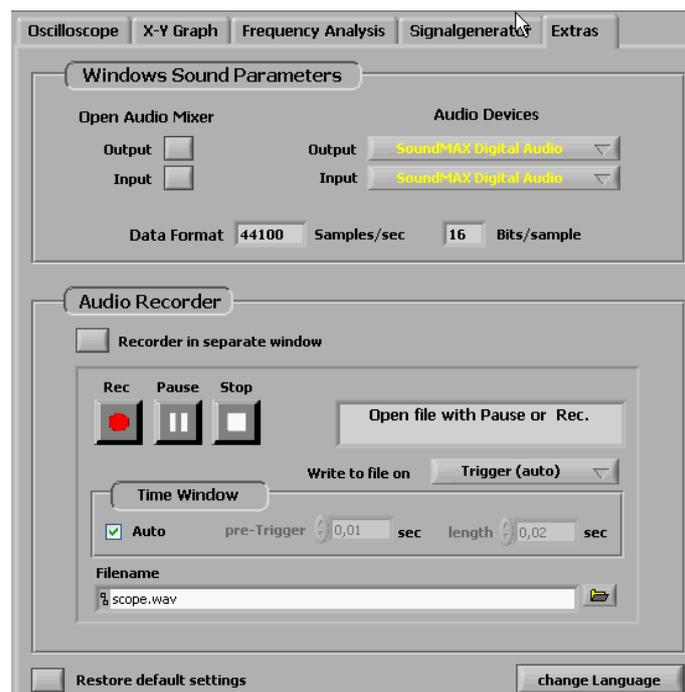


**Figure 8:** Automatic frequency sweep

Upon opening the signal generator, both channels are deactivated and must be switched on by a button at the bottom of the window. The frequency can be changed in steps of 0.5Hz. The generator signal can be sent directly to the sound card. This must be activated in the sound mixer of the Windows operating system (usually designated as "Wave Out"). If in addition the recording of the "Wave" source is activated, the signals are visible in the oscilloscope and can be displayed (e.g. to produce Lissajous figures).

### 3.5 Extras

In this window, there are some settings for the Windows audio devices. On the right side are the audio devices for sound input and output. If several sound systems are present, the equipment used can be selected here.



**Figure 9:** Extras Tab contains the windows sound settings and the sound recorder

On the left side are buttons to start the Windows audio mixers operating. Note that each push of a button opens a small mixer window! In the mixers, the inputs and outputs can be configured. At the bottom of the settings window is a button to reset the program settings. This includes ALL settings; any changes made by the user thus far will be lost!

The language of the program can be with the corresponding button. The change of the language will be applied at the next startup of the program



Figure 10: Language selection window

**For experts only:** The standard settings for the soundcard are 44.1kHz with 16Bit resolution per sample. Higher sampling rates and sample resolutions can be set in the initialization file *scope.ini* located in the installation path of the program. The corresponding parameters are "SamplingRate" and "Bits", which are commented in the original file. Most current soundcards (even onboard versions) support up to 100kHz and 16Bit. If the soundcard does not support the sampling rate and/or bit resolution, an error message will be shown at program startup.

**Addition information:** be aware, that high sampling rate/bit resolution can lead to a significant CPU load. For 100kSample with 16Bit resolution the load is more than four times larger than under standard conditions. So monitor the CPU load, when increasing the settings !

### 3.5.1 Signal Sources for the Oscilloscope

The following inputs are usually available:

- **Line-In** Port on the PC
- **Microphone** Port on the PC, or internal (e.g. Laptop) – often only mono
- **Wave Out** internal sound, e.g. MP3 player, Media-Player; signal generator
- **CD Player** Music directly from a CD

The equipment to appear on the oscilloscope must be selected from the inputs mentioned above. With some sound cards, several sources can be selected at the same time. The volume of the equipment can also be adjusted here. This has a direct effect on the amplitude of the oscilloscope!

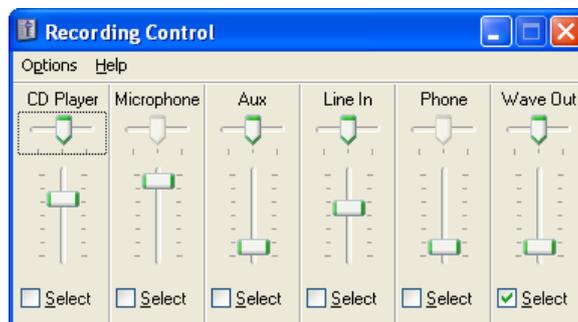


Figure 11: Selection of inputs in Windows Audio Mixer

### 3.5.2 Signal Output via Sound Card

In order to define which sound is sent to the sound card output, the appropriate equipment must be selected in the Windows Audio Mixer. Frequently several sources are merged at the same time here.

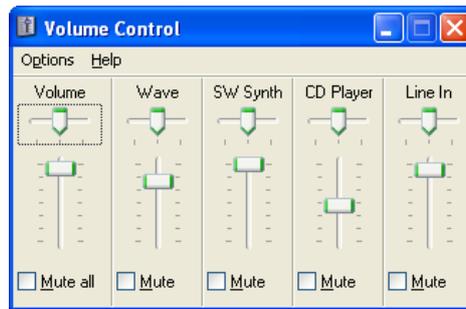


Figure 12: Selectable outputs

**Important:**

It can sometimes occur that an input or an output is not listed in the window. In this case it must be activated under:

→ Options → Properties

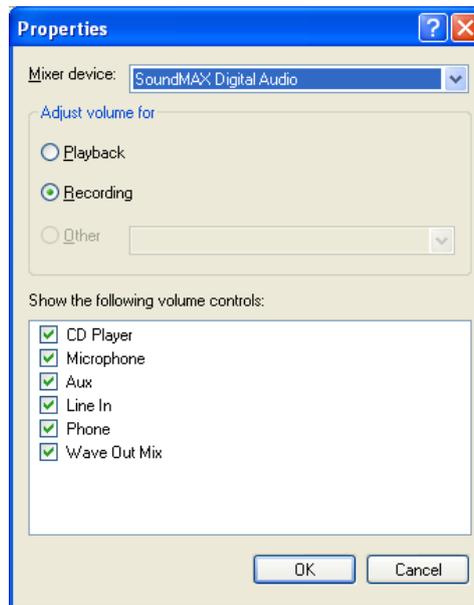


Figure 13: Properties of audio input devices

### 3.5.3 Audio Recorder

The Audio Recorder allows to save data to a Wave file. The name of the output file has to be selected before the Pause or Record button is pressed.

Three different modes are available to store data:

1. **Trigger (auto)** Save automatically the currently triggered data
2. **Trigger (manual)** Manually save the last triggered data to the file
3. **Rec. Button** Start the writing to the file with the record button (independent of the trigger)

Independent of the mode only a limited chunk size is written to the output file. The length is defined by the corresponding selectors in the Recorders window. The length is by default defined by the oscilloscopes window, but can be set by the user to a different value (uncheck the Automatic box). In all cases the writing will stop, when Pause or Stop is pressed.

Be aware, that the selected file will be overwritten WITHOUT any warning! Since the current file will be closed after the Stop button has been pressed, define a new output file BEFORE pressing Pause or Record! The resulting Wave file will contain 100 samples of silence between the recorded data chunks. Cue points at the beginning of the Wave file mark the start of each written chunk.

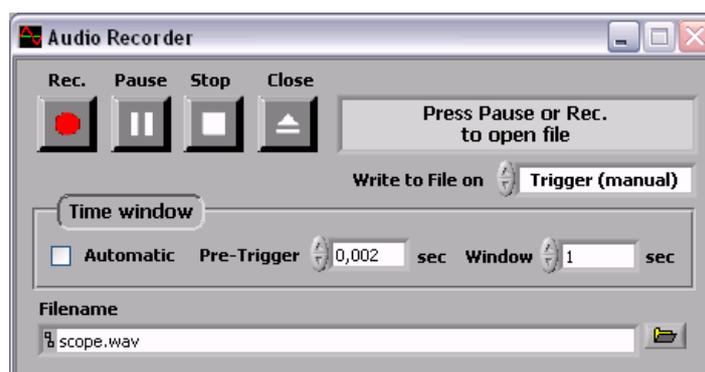


Figure 14: Audio Recorder window

## 4 Conclusion

I hope you will have a lot of fun with this program. If something goes wrong and you discovered a bug, please send mail to [Christian@Zeitnitz.de](mailto:Christian@Zeitnitz.de).

If you use the program for a project at and university or school I would like to know about it.

This program can be used and passed on for use within the school and private sector freely. For planned commercial use please contact [Christian@zeitnitz.de](mailto:Christian@zeitnitz.de).

## 5 Trouble Shooting

Certainly this program might still have some errors, however some standard problems are caused by the sound card and/or Windows.

### An error message is shown when starting the program

The error message, that the installed LabView Run-Time engine (version 7.1) is incompatible with the required one (version 7.1.1), is caused by a previously installed LabView Run-Time engine. In order to get the program running, uninstall the Run-Time engine (Control panel → Add/Remove Program → National Instruments ) prior to a re-installing this software.

### No soundcard is found

Check in the hardware manager, that Windows actually has a soundcard correctly installed. Some soundcards recognize, if speakers or a microphone is installed. This is the default behaviour under Windows Vista. In this case you have to check, that at least one input/output device is enabled in the sound settings (green check mark). If no enabled output device is existing, the program will complain about it and terminate immediately.

### The oscilloscope shows no signal and the display is frozen

Unfortunately it sometimes happens that communication with Windows breaks down. Here only terminating and re-starting the program helps!

### No signal on the oscilloscope

If the signal generator is used and a channel is also SWITCHED ON, the user must select "Wave Out" for the audio mixer equipment .

### No sound audible

In order that a signal on the speaker is audible, the appropriate equipment must not be deactivated. In this case check the audio mixer and enable the appropriate device. When using the signal generator, "Wave" must be selected.

### Strange jumps in the signal

A large signal can overdrive the input. The maximum possible value should actually be sent to the output. With some sound cards, however, this leads to an overflow and instead of a large *positive* value, a large

*negative* value is sent, which leads to a complete distortion of the signal. If such jumps are observed, the input signal should be attenuated

### **Program reacts very slowly**

The CPU load on a slow computer (less than 1GHz) can go up to 100%, especially in the frequency analysis mode. The program will only react slowly. A solution is to reduce the amount data to be processed by changing the sampling rate in the file *scope.ini*. For this uncomment the line with the key word `SamplingRate` and put in a value of 22050. This corresponds to a reduction of the amount of data by a factor of 4.