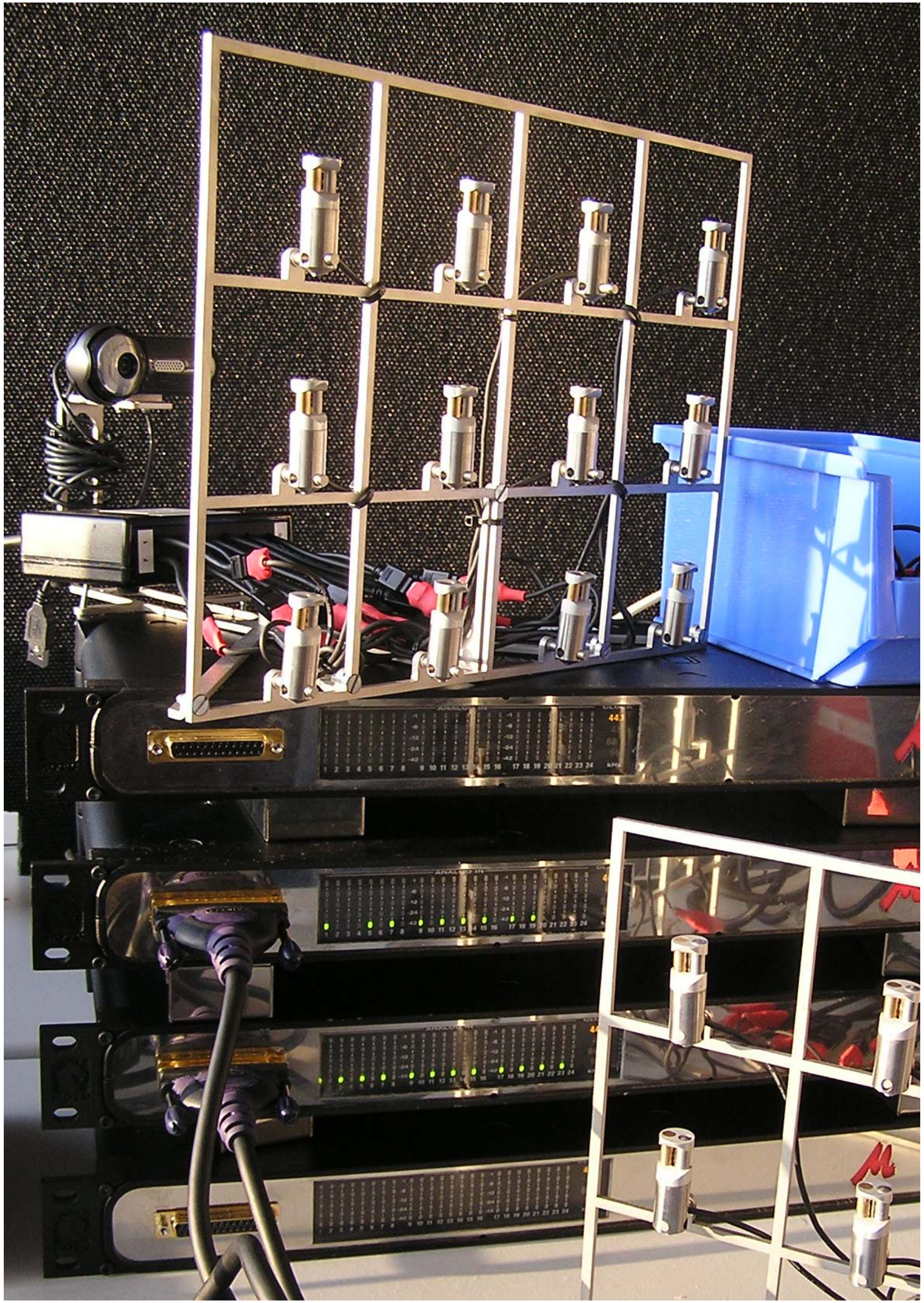

16 Data acquisition

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Fig. 16.1 (previous page): a 96 channel high end sound card (4x24 channel units shown in the back) is modified so that it can be used as data acquisition system for an array of PU probes.

16.1 Summary

In this chapter the use of high quality soundcards for measurement applications is the subject of discussion. Soundcards are A/D converters for audio applications.

In this chapter it will be shown that soundcards can be used for dedicated measurement applications.

An analyser is a general purpose measurement tool so it can perform absolute measurements (it is a calibrated measurement device). Besides that it has the option to power sensors. A, analyser can be seen as a separate system.

In this context soundcards can only be used to convert analogue signals to the digital domain. They should be used and calibrated in combination with dedicated sensors. The combination of sensor and soundcard can be seen as a complete system.

16.2 Introduction

An analyzer is traditionally used in acoustics and vibrations to collect the analogue signals of microphones, accelerometers etc. and represent them in the frequency domain on a computer.

Apart from the ability to convert analogue signals to the digital domain (and to transform from time domain to frequency domain), an analyzer is used as a measurement tool: from an AC voltage the frequency and RMS value can be obtained accurately and a phase shift between two signals can also be determined very precisely.

On top of that modern analyzers can also power the sensors with a stabilized current source (normally 4mA) and read in additional sensor data like sensitivity, brand serial number etc. (e.g. TEDS).

Each channel can select its own gain to reach an optimal accuracy.

If one also take in mind that the noise and vibrations marked is not very large, it is no wonder that analyzers are expensive pieces of equipment. The prices are in the order of 500-2000 USD per measurement channel in the year 2006 depending on dynamic range (number of bits, usually 24 nowadays), bandwidth (usually 0-20kHz), linearity (0.05dB), harmonic distortion etc. etc.

Per channel an analyzer consists of a low noise precision input preamp that can switch in range, an A/D converter and a DSP to convert the digital signals from the time domain into the frequency domain and do extended (cross spectrum etc.) calculations. A precision input preamp costs around 50-100 USD, an A/D converter 1-15 USD and a DSP is available in the range of 25-50 USD. So for the hardware the average cost per channel are in the order of 150USD per channel. The DSP has to be programmed to operate with the dedicated software of the graphical user interface.

So for the reasons stated above and the need of proper housing and connectors it makes sense that the cost of an analyzer is high.

In the last years, in a complete other market segment, there are interesting developments. In the digital consumer electronics the quality of soundcards is increasing and due to the market volume, the prices are low. For 10USD per channel a high performing A/D converter can be bought.

The question is: can these soundcards be used as an analyzer?

The goal of this chapter is to show that <u>in general</u> a soundcard cannot replace an analyser but for specific cases it can be a perfect alternative.

16.3 The components of analysers

The functionality of a measurement chain with an analyzer is examined and compared to sound card solution. Technically and in general the chain consists of several parts:

1) The sensor

2) Low noise preamplifier of the sensor
3) Cables & connectors
4) Low noise adjustable precision input preamplifier of the analyser
5) Anti aliasing filter (adjustable bandwidth)
6) A/D converter (bitrate)
7) DSP
8) Software & laptop or PC

The several parts are discussed below and compared with a sound card solution.

The sensor

A sensor is a device that has a voltage output in this context. It may be that the sensor has to be (current) powered. The sensor has a dynamic range in the order of 120dB and the highest output is in the order of Volts.

An analyser is a general purpose device so it should have the option to power a sensor and the input voltage range should be large. If a dedicated system is designed, no options are required. This makes the design of the input stage simple.

The low noise preamp of the sensor

A pressure microphone has normally a preamplifier with a unity voltage gain. For single sensor use, a powering module is normally employed that has an adjustable gain. Modern analyzers have powering modules for e.g. microphones build in.

Cables & Connectors

It is a not so rewarding topic but cables and connectors are of crucial importance. For a low number of sensors it is not so difficult to find the right solution. Normally the sensor is connected to the data acquisition system with a shielded cable and connectors. For a low number of sensors the BNC connector is used mostly. The connector of the sensor is normally a BNC (if current power is used), a microdot (accelerometers) or a 7pins LEMO (pressure microphones).

For larger number of channels the BNC connector is not very practical anymore. It is too large and the BNC cables are too thick. The complete system will be a mess (see the opening picture of chapter 8).

If more than, let’s say, 30 channels are required one have to find another solution than one-signal-one-connector. If this is done, the analyzer does not have a standard input anymore and the complete solution has become dedicated in the sense that the connectors will not connect to standard sensors.

If one is building a dedicated solution, a soundcard becomes a viable option.

Low noise adjustable precision input preamplifier of the analyser

The input preamp is a high quality amplifier that has an excellent linearity, low self noise, high dynamic range and the gain is adjustable in precise steps. For those reasons those preamps are expensive.

Nowadays the A/D converter is available in 24 bits resolution so the need for an adjustable (multistep) gain is diminished. The dynamic range of a 24 bit A/D converter is 145dB. It may be useful to have a 40dB attenuation option for the preamp to get a more general purpose input.

If sensors and their preamplifiers are optimised to be connected to a specific A/D interface, an input preamp may be excluded. This is because the dynamic range of the sensors is normally in the range of 120dB. However to be on the save side, a 40dB attenuation is recommended to increase the dynamic range so that the lowest signal is converted with enough bits.

The input preamplifier can be omitted if the preamplifier of the sensors is 40dB adjustable. This statement sounds like just shifting the problem but in practice it shows that making the preamplifier of a sensor adjustable is far more simple to realise than to include a high quality adjustable preamplifier in an analyser.

An analyser has usually the option that the gain of each channel can be set separately. This increases the overall quality of the measurements because each channel then uses the maximal possible bits. A soundcard usually one gain setting for all channels.

The anti-aliasing filter

An A/D converter transforms time data in the frequency domain. If the analogue signal has frequency components higher than half the sampling frequency of the A/D converter, something goes wrong: the high frequency signals are displayed in the low frequency band.

The only remedy to avoid this problem is to filter the analogue signal. Signals above half the sample frequency have to be blocked.

Such filters are called ‘anti aliasing filters’. It is impossible to make an analogue filter that passes all signals below a frequency and that blocks all signals above that frequency.

If the A/D converter has an adjustable sample frequency to adjust the bitrate, the anti aliasing filter must also have an adjustable bandwidth.

Simple data acquisition cards do not even have an anti aliasing filter so if the analogue input signal has higher frequency components, the representation in the frequency domain goes completely wrong.

If the sample frequency is chosen much higher than that the required bandwidth would suggest, e.g. 96kHz sample frequency for a bandwidth of 20kHz, then the anti aliasing is not very difficult anymore.

If a dedicated sensor is used (e.g. the PU probe), the frequency spectrum of the sensor is known. The preamp can be designed so that it contains an anti aliasing filter.

The A/D converter

The A/D converters that are used in analyzers are the same or similar to the ones that are used in high performance soundcards. The cost of a high quality 24 bits A/D converter chip is in the order 10USD. The cost of a quality soundcard is in the order of 10USD per channel. The cost of a multi-channel high quality soundcard (so also high quality connectors and housing) is in the order of 1200USD per 24channels (50USD per channel).

The DSP

The DSP is e.g. able to convert the steam of time data into frequency data. This processing is important if the data has to be represented in real time, i.e. if the measurements have to be shown directly on a display.

A few years ago the computers where not so fast so it was impossible to process the data and display it with a very short delay so that the visualization seemed to be real time. Nowadays (year 2007) a normal PC can process two channels easily but also the frequency representation of more than 10 channels can be processed and displayed in real time.

Time data of 96 channels with 24bits resolution and 92kHz sample frequency can be read in and stored on hard disk and post processed in seconds.

With dedicated (e.g. C++ programmed) software it is possible to process 96 channels with 24 bits of resolution in real time on a regular PC.

The need for a DSP still exists if more than 100 channels have to be processed in real time. An acoustic camera based on holography methods that displays the results in real time is an example that requires many DSP’s.

If the visualization has not to be real time but if the pictures may be shown with a delay of a few seconds, the DSP is not required anymore.

The graphical user interface

The data that is generated by a sound card or an analyzer has to be visualized and input setting etc. must be manipulated. This is done by a program that is called: a ‘graphical user interface’ or GUI.

The more dedicated and the more complex the program is, the more labour intensive it gets to write such program.

A GUI to be able to do two channel general measurements with a soundcard is available for less than 100 USD. A program that visualises the NAH (see chapter 8: 'Holography') of a 256 channel array may cost over 100.000USD. The costs is high because it is complex matter and there are not much customers.

The advantage of using a soundcard is that these devices are widely used and low cost general purpose software can be found.

If a DSP is used, the communication between the GUI and the DSP is different for each DSP and each supplier. So it is impossible to use a program of one supplier with the analyser of another supplier. One could say that the analyser is a hardware dongle for the software.

If software is developed for a soundcard, all soundcards can be used. The hardware and the software can be seen as separate entities. This has the advantage that software and hardware can be acquired from different parties that each have other specialities.

If soundcards are used, the software can e.g. be written in MATLAB (or e.g. Adobe Audition or LABVIEW). In this way the development of software is relatively simple keeping the development time short and the development cost low.

The housing, connectors, powering etc.

It seems not important to mention but the connectors and housing of an analyser may be very important. In most cases the cost of a measurement is high and the results are analysed at a later time. Having bad connectors is something one can do without. Also setup time may be expensive. In such case sensor powering by the analyser is time and thus money saving. Analyses are available that have internal battery power and have internal hard disk data storage. These properties make analysers more suited in some situations.

16.4 Properties of soundcards versus analyzers

A soundcard is compared with an analyzer. In this test a Siglab analyzer is used as reference. The Siglab analyser is a 16-bit high quality data acquisition system that is operated by a Matlab based GUI. The sound card is a 10 Euro stereo sound blaster.

The Creative Sound Blaster is a low cost stereo 16-bit soundcard system that is operated by USB. AtSpec software (www.taquis.com) is used for the calculations and as GUI. The total cost of this solution is less than 50 Euro.

Signal to noise ratio

The signal to noise ratio is shown by measuring a 1kHz pure sine wave. The signals are processed with 8192 points record length FFT, 10 times averaging and a Hanning window.

As can be seen in Fig. 16.2, the signal to noise ratio of the 2 channel soundcard is comparable but slightly less good.

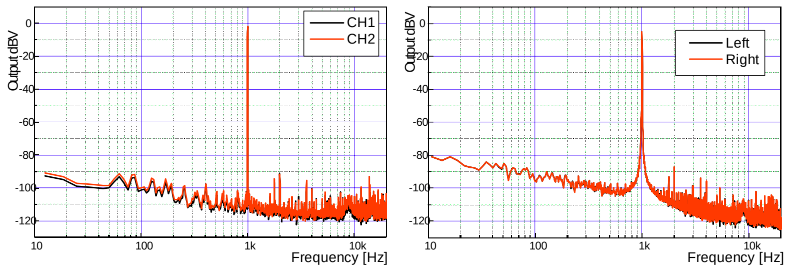

Fig. 16.2: A 1kHz sine wave is measured with (left) a Siglab analyser and (right) a low cost two stereo sound card.

Phase accuracy between 2 channels

To check the phase accuracy 0.5VRMS white noise (this is full scale for the soundcard) was put on both channels and the transfer function was measured. It was measured with a Hanning window, 30times averaging and 8192 points record length FFT. If both channels would measure exactly the same, the phase response would be zero degrees. It shows that the phase accuracy is very good and that the sound card performs slightly better.

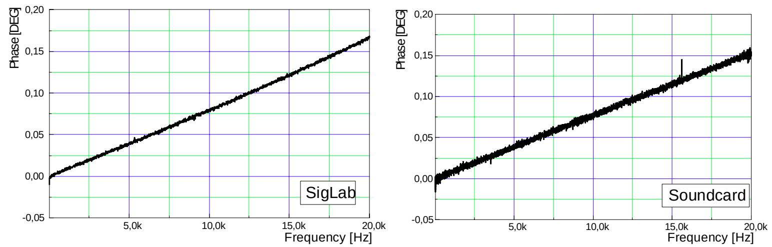

Fig. 16.3: Transfer function of two channels is measured to check the phase accuracy.

Amplitude accuracy between 2 channels

To check the amplitude accuracy 0.5VRMS white noise (this is full scale for the soundcard) was put on both channels and the transfer function was measured. It was measured with a Hanning window, 30times averaging and 8192 points record length FFT. If both channels would measure exactly the same, the phase response would be zero degrees. It shows that the amplitude accuracy is very good and that the analyser performs slightly better.

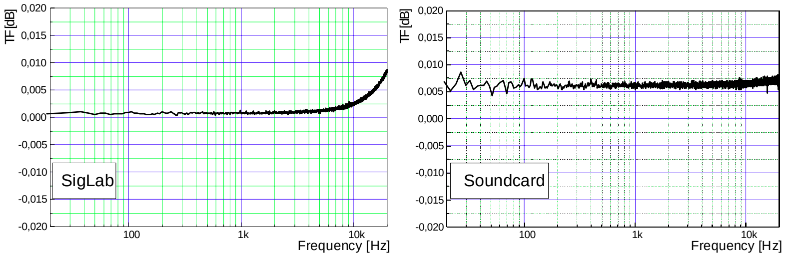

Fig. 16.4: Transfer function of two channels is measured to check the amplitude accuracy.

Effect of lower signals

The measurement of the previous paragraph is repeated but the input signal is lowered compared to the previous signal to 0.05Vrms and 0.005Vrms white noise. It was again put on both channels and the transfer function was measured with a Hanning window, 30times averaging and 8192 points record length FFT.

As can be seen, the lower input signal affects the result. The 0.5Vrms input signal was close to the full scale input, the 0.05Vrms is 20dB less and 0.005Vrms is 40dB less. The lower signal means that 8 bits less bits are used.

If the signal is measured with less bits used, the results becomes noisier. If the 16 bit resolution soundcard is used 40dB below full scale, the noise in the transfer function becomes less than 0.1dB. This is still acceptable but it is not recommended to lower the signal more.

The analyser is not affected by this because the preamplifier gain is increased by 40dB.

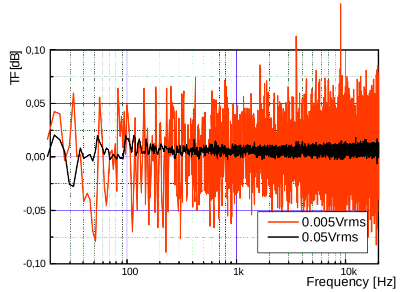

Fig. 16.5: Transfer function measured at 0.05V and 0.005V: if the input signal is too low the accuracy is low. (0.5V is full scale)

Synchronisation

Delay of two sine waves

16.5 Calibration of a soundcard

The calibration of a soundcard can be done for two purposes: 1) the absolute calibration, i.e. the digital number as result of input voltage. Or: 2) the relative calibration. That is the similarity between channels.

Relative calibration

If the sound probes are connected to a sound card only a relative calibration is needed for the

16.6 Conclusion soundcards versus analyzers

In general a soundcard cannot be used as an analyser. This is because the overall quality (linearity, phase accuracy, stability) of an analyser is higher than a soundcard. The question is however if this high quality is needed. If the sensors (and the preamplifiers of the sensors) that are connected to the analyser have less dynamic range and if the harmonic distortion is higher than there is no need for an analyser.

Nowadays sensors and analysers are seen as two separate systems. If the sensor and soundcard are seen as a single system, the complete system (sensor + soundcard) can be calibrated and ripples in the frequency response are not of any importance. The only thing that is important is that the transfer function is constant in time, the dynamic range of the soundcard is higher than the dynamic range of the sensors and that the harmonic distortion is much lower than the harmonic distortion of the sensor.

It is difficult to answer the question if in general a soundcard can be used as an analyzer. Under some restrictions it is possible:

1) The sensors have lower dynamic range than the A/D conversion.
2) The preamplifiers of the sensors have a 40dB extra gain switch.
3) The maximal output level of the sensor plus preamplifier equals the full scale input of the A/D converter.
4) Less than 100 channels real time monitoring of simple algorithms (e.g. the direct method, see chapter 8).
5) A few seconds time is acceptable for 96 channels (24bit, 94kHz sample rate) if algorithms are more complex.
6) The phase and amplitude stability is within limits.
7) The complete system (sensor plus soundcard) is calibrated. Or the sound card is calibrated with a dual channel analyser.
8) Data acquisition is not used as absolute measurement system (it must be calibrated before use).
9) All channels have a comparable input level.

If these requirements are met it is possible to use a soundcard based data acquisition system.