CHAPTER

6

Seismic Recording Systems

Günter Asch

6.1 Introduction

During the last ten years, recording devices based on digital technology have completely replaced their old analog predecessors. The latter are costly, require specialized maintenance and consumables, and are incompatible with computer data processing and analysis. They are no longer produced although being still in operation at many older seismological stations and network centers. They are not dealt with in this chapter. Extensive reference to analog photographic and directly visual drum recording and processing of the records is given in the chapters Instruments and Station Operation in Willmore (1979) Manual of Seismological Observatory Practice.

The technological progress in digital signal processing, data storage techniques and highly integrated digital circuits has lead to several instruments being available on the market that all fulfill the basic requirements of a seismic recording instrument and offer several more advanced features as well.

In terms of this chapter, a recording device is an autonomous, self-contained equipment, designed to measure the output signal of a sensor, digitize the signal and record it. In seismological experiments, all three components of ground movement are of interest, whereas in reflection experiments, only the vertical component up to now has been taken into account. Specialized multi-channel recorders with more than 6 channels, preferred in exploration seismics, are not covered here.

Seismic experiments vary from reflection profiles with short recording windows and high sampling rates to the continuous recording of broadband sensors with high resolution at observatories. An instrument well suited for an observatory, may be a bad choice for a wide-angle experiment and vice versa.

This chapter is a short introduction to the principal concept of seismic recording systems and should help the non-technical users to decide which instrument is suitable for their specific requirements. Fig. 6.1 gives an overview of the principal units of a seismic recording system. Each unit will be discussed in detail in the following sections.

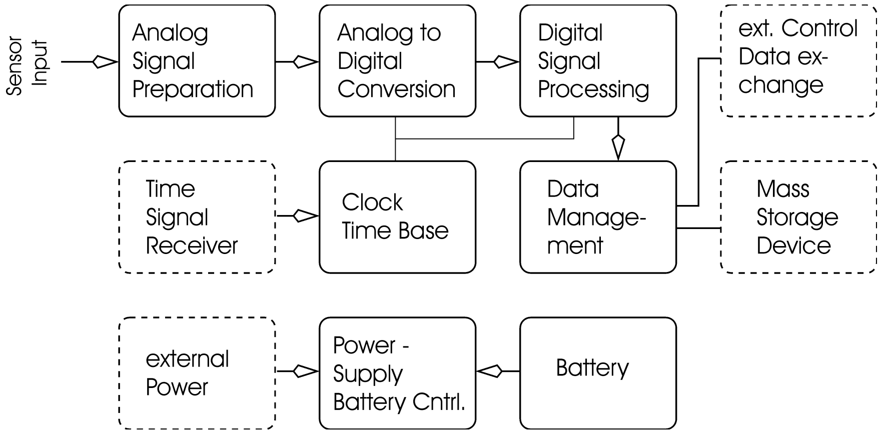

Fig. 6.1 Principal units of a seismic recording device. Dashed boxes relate to optional functions.

6.2 Analog signal preparation

6.2.1 The Analog Signal Preparation section

The seismic sensor is connected to the Analog Signal Preparation (ASP) section. This is not only a problem of correct wiring and polarity, but depends strongly on the type of transducer used. If a passive electrodynamic sensor is used, the impedance of the recording system influences the sensitivity and the frequency of the sensor itself. The sensor's response function (see 5.2.4 to 5.2.9 and Scherbaum, 1996) have to be corrected for the input impedance of the recording device. For this type of sensor, one has to think also about the resistivity and capacity of the cable, used to connect the sensor to the recording device, in case it exceeds several tens of meters. For active sensors, i.e., all broadband sensors, the effect of the input impedance can be neglected because it does not influence the sensor's characteristic, however long cables can introduce noise into the system. In general, short shielded cables, a single common analog ground and high quality connectors help to reduce this type of noise problems. The ASP section is also responsible for protecting the sensitive electronics of the recording system against high input electrostatic voltages.

The next step in ASP is the preamplifier which, together with the Analog to Digital Converter (ADC*), determines the resolution [counts/Volt] of the recording device. The preamplifier has to fulfill several demands, such as linearity with respect to amplitude and phase, low noise, quick recovery from overloading, i.e., no clamping, and in addition, low power consumption. In reality, there is a tradeoff between low noise and low power consumption, and the designer of the preamplifier stage has to find a compromise. In any case, the noise generated by the ASP should be distinctly lower than the least significant bit of the ADC stage. Other requirements are not as critical, but one has to take care in a system with more than one channel, that all the parts in the ASP are identical, so each channel has the same response and sampling is done simultaneously on all channels. The technical approach is explained in section 6.3.4.

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* Terms typed in Arial letters are explained in more detail in 6.7: Glossary of technical terms and links.

6.2.2 Analog filters

Some recorders offer a high-pass filter to remove DC-offset and long-term drifts from the measured signal. These filters are intended to mask temperature and ageing problems, related to electronic components and to the system's specific design. Users should be able to decide whether they wish to activate these filters or not. As an example, the group delay of the optional high-pass filter from a PDAS recorder is given in Fig. 6.2. The high-pass filter is formed by a simple RC-element (15 µF||1 MΩ) with a time constant of about 15 seconds. Signals with periods of about 3 seconds are delayed by approximately 1 sample, assuming a sampling rate of 100 Hz. For longer periods, the situation becomes worse and it is not acceptable to activate this high-pass filter to record signals from broadband sensors. From the scientific point of view, this filter makes no sense; it simply beautifies the signal and substitutes one problem for another.

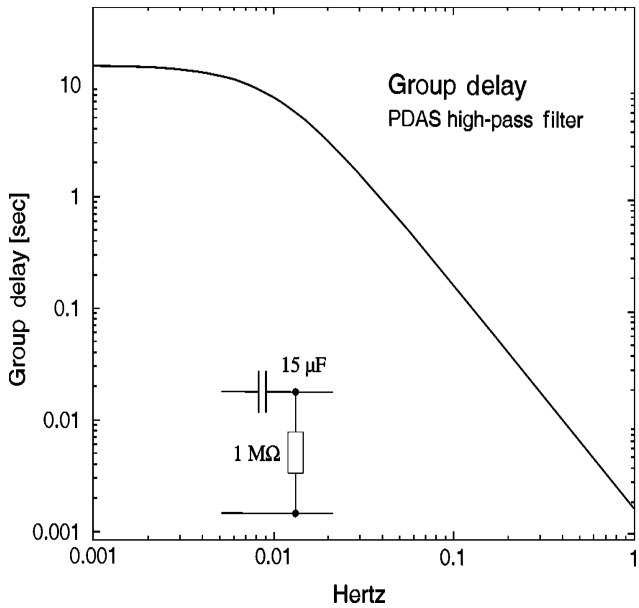

Fig. 6.2 Group delay of the optional PDAS high-pass filter.

Before converting the analog signal into counts, it has to pass a low-pass filter serving as an anti-alias filter. This limits the frequency content of the signal toward higher frequencies. The reason for this fundamental step is described in 6.3.1. In some cases, i.e., continuously integrating converters, this stage can be an integral part of the ADC itself.

6.3 Analog to digital conversion

Conversion of a continuous analog signal into a digital time series is based on quantization with respect to time <u>and</u> amplitude. Therefore it is necessary to understand the influences and limitations introduced by these two different operations. After filtering, the analog signal is sampled and converted into digital values. Since the digital domain consists only of finite length digital words, which have to represent a continuous signal, the conversion step introduces quantization errors.

6.3.1 Sampling theorem

The *sampling theorem* describes the effect of quantization at discrete times on an analog signal. It was first published by Shannon (1949). A simple model is given by a switch, closed periodically for a certain time. At first, this sounds quite simple but in fact, this simple device is a modulator, performing the multiplication of two signals. Fig. 6.3 shows that the output signal is the control signal of the switch, multiplied by the input signal. The control signal is a train of impulses, therefore it is periodic and can be expressed by a Fourier series:

$$P(t) = \sum_{n=-\infty}^{\infty} h \cdot \frac{\tau}{\Delta T} \cdot \frac{\sin(\pi n f_s \tau)}{\pi n f_s \tau} \cdot e^{i2\pi n f_s t}. \tag{6.1}$$

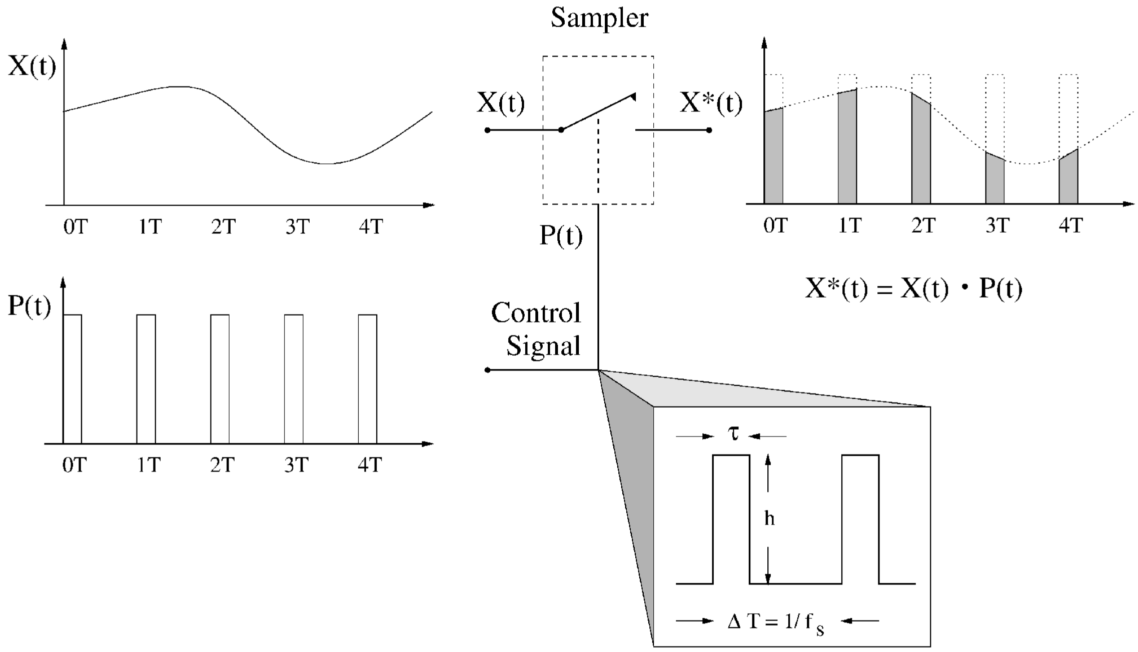

Fig. 6.3 Relation between input-, output- and control signal of a sample device.

The variables h, τ, ΔT, and f_s are explained in Fig. 6.3, and $\sin(\pi n f_s \tau)/\pi n f_s \tau$ is the Fourier transform of the n^{th} boxcar's impulse response. If we look at the sampler as an ideal device, τ approaches 0 and therefore $\lim_{\tau\to 0} \sin(\pi n f_s \tau)/\pi n f_s \tau = 1$, i.e., a so-called Dirac impulse. The energy of a single impulse is given by $A = h\cdot\tau$, so the output signal is described by

$$X^*(t) = X(t) \cdot \underbrace{\frac{A}{\Delta T} \sum_{n=-\infty}^{\infty} e^{i2\pi n f_s t}}_{P(t)} \tag{6.2}$$

Due to the periodic character of the control signal $P(t)$, we get an infinite number of impulse responses, separated from each other by $\Delta T = 1/ f_s$ with f_s as the sampling frequency. This changes the frequency content of the measured signal, and the spectra of the input- and output signal are no longer identical. The Fourier transform of $X^*(t)$

$$X^*(f) = \frac{A}{\Delta T} \int_{-\infty}^{\infty} (X(t) \cdot \sum_{n=-\infty}^{\infty} e^{i2\pi n f_s t}) \cdot e^{-i2\pi f t} dt \tag{6.3}$$

results in

$$X^*(f) = \frac{A}{\Delta T} \sum_{n=-\infty}^{\infty} X(f - n f_s). \tag{6.4}$$

In other words, the spectrum of the input signal is transformed into a sequence of an infinite number of spectra. All these spectra, except the one of the order 0, are called the alias spectra of *X** (*f*).

To illustrate this behavior, a concrete example is given in Fig. 6.4 which demonstrates the situation with the PDAS recording system. Sampling is done with a primary sampling rate of 1 kHz. The anti-alias filters are designed as Butterworth low-pass filter of an order of 6 (36 dB/octave). The corner frequency of this filter is set at 200 Hz. Its transfer function is given by

$$H(j\omega) = \frac{1}{\sqrt{1 + (j\frac{\omega}{\omega_0})^{2n}}} \tag{6.5}$$

with $\omega_0 = 2 \cdot \pi \cdot 200$ Hz and $n = 6$ is the order of the Butterworth filter.

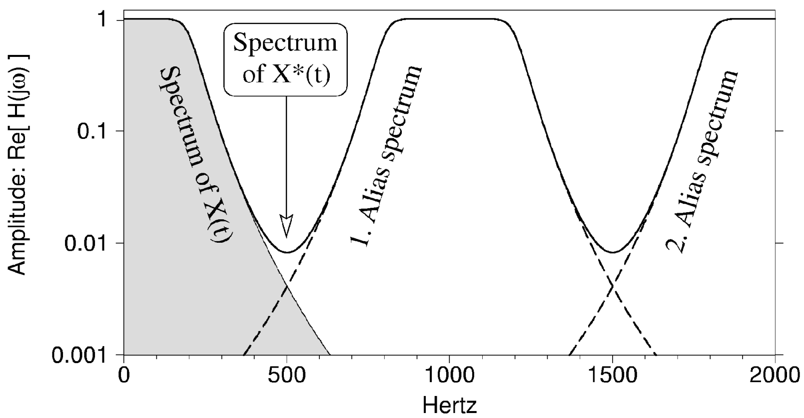

Fig. 6.4 Alias spectra in a PDAS recording system after sampling.

The consequence of aliasing in this example is that all signals with frequencies above the so-called Nyquist frequency $f_N = f_s/2$ are mirrored into the spectrum of *X*(*t*) (Eq. 6.2). A signal with a frequency of 600 Hz can not be distinguished from a 400 Hz signal, and a signal with 1 kHz is seen as DC with an amplitude, depending on its amplitude and the phase difference between the signal and control signal *P*(*t*). That is why we have to assure, by applying an anti-alias low-pass filter to the analog input time series, that its high-frequency amplitudes are drastically reduced and thus do not mirror much energy into the alias spectra. Actually, the output of the sampler *X**(*t*) is the summing curve of all spectra, as shown in Fig. 6.4. The spectra in Fig. 6.4 were calculated on the assumption that the PDAS recording system was fed with a white noise signal.

In field seismology, very little data is collected with a sampling rate of 1 kHz. In local and regional experiments, 100 Hz sampling rate is sufficient, and for teleseismic studies even 20 Hz data are commonly used. Why does the PDAS sample at such a high frequency? One answer is that the first alias spectrum is shifted far away from the frequencies in which we are interested. A second answer is given in the next section.

6.3.2 Oversampling

There are several ways to describe how oversampling increases resolution. The following is adopted from a technical report published by Texas Instruments (1998). The maximum error of an ideal quantizer is ±0.5 LSB (least significant bit). Since the input range of a *n*-bit ADC is divided into 2^n discrete levels, each represented by an *n*-bit binary word, the ADC input range and the word width *n* are a direct measure of the maximum absolute error. In addition, the quantization step can be analyzed in the frequency domain. The number of bits representing the digital value determines the signal-to-noise ratio (SNR). Therefore, by increasing the signal-to-noise ratio, the effective resolution of the conversion will be increased. Assuming the input signal is an ever-changing signal (always true in seismology), the error caused by quantization from an ideal ADC can be viewed as a white noise signal that spreads the energy uniformly across the whole bandwidth from DC to one half of the sampling rate. The RMS amplitude of the quantization error U_{neff} is given by Tietze and Schenk (1990)

$$U_{n_{eff}} = \frac{U_{LSB}}{\sqrt{12}} \tag{6.6}$$

For a full-scale sinewave signal, the RMS amplitude for an *n*-bit ADC is given by:

$$U_{s_{eff}} = \frac{1}{\sqrt{2}} \cdot \frac{1}{2} \cdot 2^n \cdot U_{LSB} \tag{6.7}$$

where the factor 1/2 is caused by the signal having ±maximum amplitude while the converter is from 0 - 2^n. The 1/ √2 comes from conversion to RMS. On the basis of Eqs. (6.6) and (6.7) the signal-to-noise ratio can be calculated

$$\begin{aligned} SNR\,[dB] &= 20\log_{10}\frac{U_{s_{eff}}}{U_{n_{eff}}} \\ &= 20\log_{10} 2^n \cdot \frac{\sqrt{12}}{2\cdot\sqrt{2}} \\ &\approx n \cdot 6\,[dB] + 1.8\,[dB] \end{aligned} \tag{6.8}$$

With an ideal ADC, the quantization noise power $P = U_{neff}^2$ is uniformly distributed across the spectrum between DC and half the sampling rate. This quantization noise power is independent of the sampling rate. If we use higher sampling rates, the noise power is spread over a wider range of frequencies. Therefore the effective noise power density at the band of interest is lower at higher sampling rates. Fig. 6.5 illustrates the effective noise power reduction in the frequency band of interest at a rate of oversampling of k and the resulting sampling rate of $k\,f_s$. One has to use digital low-pass filters to remove all frequencies above f_s /2. The effective resolution is determined by the quality of the digital filter. The remaining noise power beyond f_s /2 is a measure for the quantization noise and therefore responsible for a decrease in the signal-to-noise ratio.

In an oversampled system, the sampling rate is decimated - after or during filtering - by a factor of *k*. In such a case, an ideal low-pass filter and decimator will reduce the quantization noise also by the factor of *k*. Since the signal at the band is not affected by the filter, this leads to an enhancement of the signal-to-noise ratio. The formula for the improved signal-to-noise ratio is:

$$SNR\,[dB] = 20\log_{10}\left(\frac{U_{s_{eff}}}{U_{n_{eff}}}\cdot\sqrt{k}\right) \qquad (6.9)$$

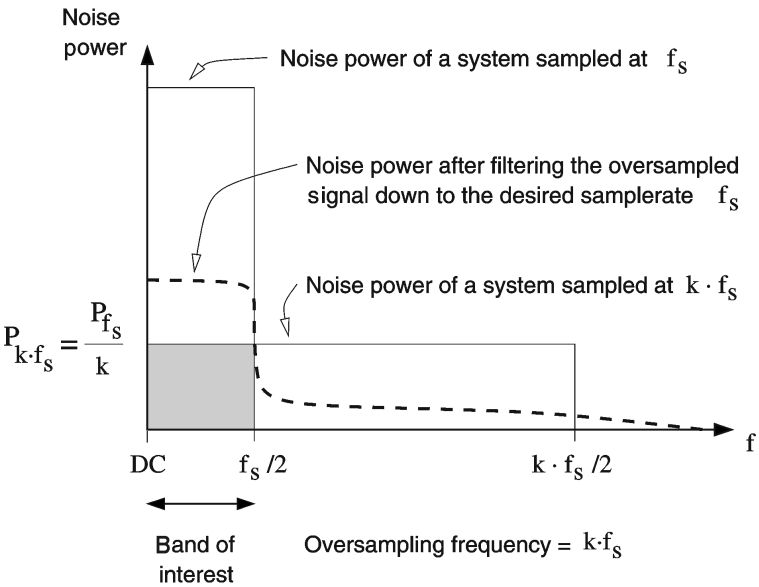

Fig. 6.5 Noise power reduction in an oversampled system.

The increase of the signal-to-noise ratio and the corresponding increase in the number of bits is shown in Fig. 6.6.

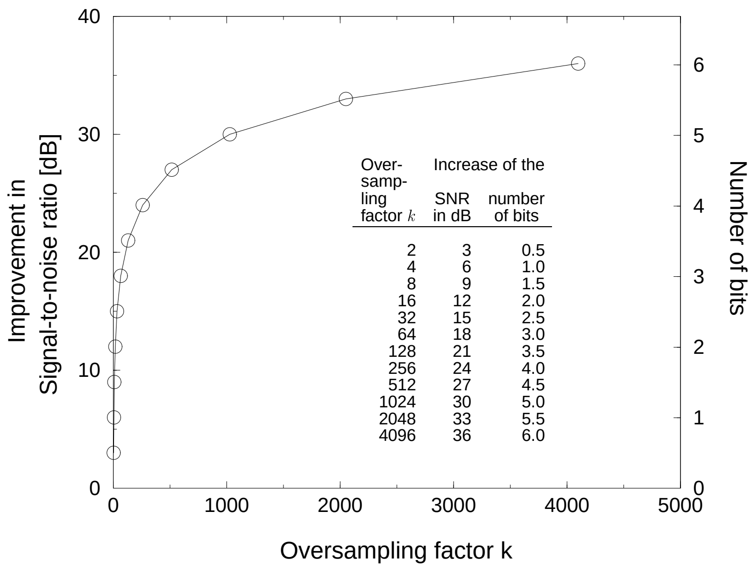

Fig. 6.6 Improvement of the signal-to-noise ratio with oversampling.

6.3.3 Digital filters

The concept of oversampling is strongly related to digital filters. In principal, there are two concepts available, filtering a digital data stream in the time domain. We distinguish between IIR and FIR filters. Both have their specific advantages and drawbacks and are discussed in detail by Buttkus (1991) and Scherbaum (1996). The IIR concept is adopted from analog filter design techniques and utilizes polynomial approximation to the desired transfer function. The name of this type of filter is given by the recursive design of the underlying algorithm. The FIR design is a non-causally approach, applying the time domain representation of the desired transfer function (Laplace transformation of the filter's impulse response; see 5.2.2) to the digital data stream. Both types have their specific benefits and disadvantages. On the one hand IIR filters are fast to compute, on the other hand they are not guaranteed to be stable with increasing order and, as their analog counterparts, introduce a phase shift depending on the frequency of the input signal. FIR filter are easy to implement, always stable and produce no phase shift (if expressed symmetrically). This results in a constant group delay and gives rise to small amplitude precursory artefacts at the output. A compromise getting the best from both approaches is the minimum-phase filter. A brief discussion about this problem and how to deal with is given by Scherbaum (see http://lbutler.geo.uni-potsdam.de/FIR/fir_daaf.htm).

All of the digital recorders available use FIR filters to decimate the digital data streams, and most of them are zero-phase filters. At the moment, only Earth Data and Kinemetrics (for the Quanterra records) offer minimum-phase decimation filters for their products.

6.3.4 Analog to Digital Converter (ADC)

The first generation of seismic recorders mainly utilized Successive Approximation Register ADCs (SAR). The concept is based on a DAC combined with a comparator and a shift register in a feedback loop.

It takes n steps to convert one sample with a binary resolution of n-bit. The right-hand part of Fig. 6.7 shows a block diagram of an SAR-ADC. In operation, the system enables the bits of the DAC one at a time, starting with the most significant bit (MSB). As each bit is enabled, the comparator gives an output signifying that the input signal is greater or less in amplitude than the output of the DAC. If the DAC output is greater than the input signal, the bit is reset, i.e., turned off. The system does this with the MSB first, then with the next most significant bit, etc. After n-steps, all the bits of the DAC have been tried, and the conversion cycle is completed. Fig. 6.7 shows the typical components and the signal flow for this type of ADC, as used for example, in the PDAS recorder. The anti-alias filter is followed by a Sample & Hold device. Using one common control signal $P(t)$ for all Sample & Hold devices results in sampling at the same time of all input channels. Thus only one ADC device is necessary for multiple analog input channels, reducing power consumption and cost.

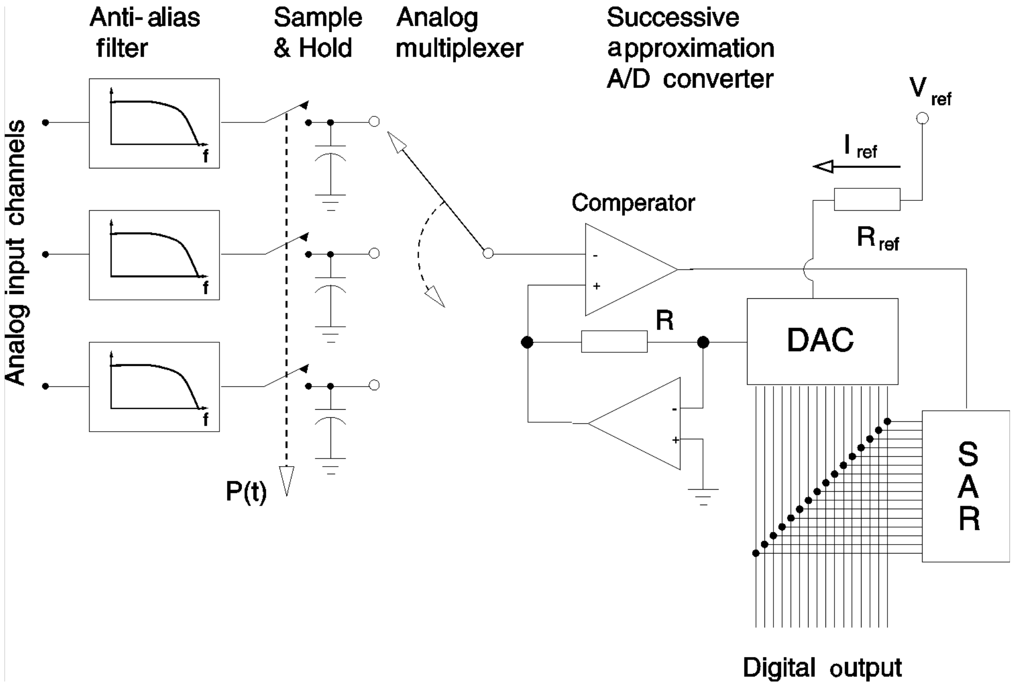

Fig. 6.7 ADC concept used in the first generation of digital recording systems with a 16 bit converter.

This concept, however, has some severe disadvantages. During the conversion cycle, the input voltage must be kept stable below the resolution of the ADC. For this, and the parallel sampling of all input channels, the Sample & Hold unit and the analog multiplexer are introduced. But there is no ideal sampler with $\lim_{\tau\to 0} \sin(\pi n f_s \tau)/\pi n f_s \tau = 1$ as required in Eq. (6.1). Also, the capacitor that holds the input voltage is not an ideal analog storage medium. All this together adds noise to the processed signal. But the most significant problem of this concept is the gap introduced between two Sample & Hold cycles, owing to the input signal being disconnected from the ADC.

Therefore small variations of the input signal are only taken into account if they are significant with respect to the resolution of the ADC stage. The noise reduction achieved by oversampling with this type of converter is proportional to the square root of the oversampling factor, as shown in Eq. (6.9) and Fig. 6.6.

In modern data acquisition systems, there is no clear separation any more of the **Analog to Digital Conversion** and the **Digital Signal Processing** stage, as shown in Fig. 6.1. The ADC itself does a lot of digital signal processing to achieve high resolution, high signal-to-noise ratio, and high dynamics. The used techniques are based on continuous integration and oversampling in combination with carefully designed digital low-pass filters. In most modern seismic recording systems, a Delta-Sigma-Modulator is used as ADC. A simplified block diagram of the signal flow is given in Fig. 6.8. The concept of continuously integrating the analog signal avoids gaps in the sampling process and takes into account variations of the input signal even below the resolution of the ADC. The drawback is that one has to have a separate unit for each input channel, but this is more than compensated for by having abolished the Sample & Hold- and multiplexer devices.

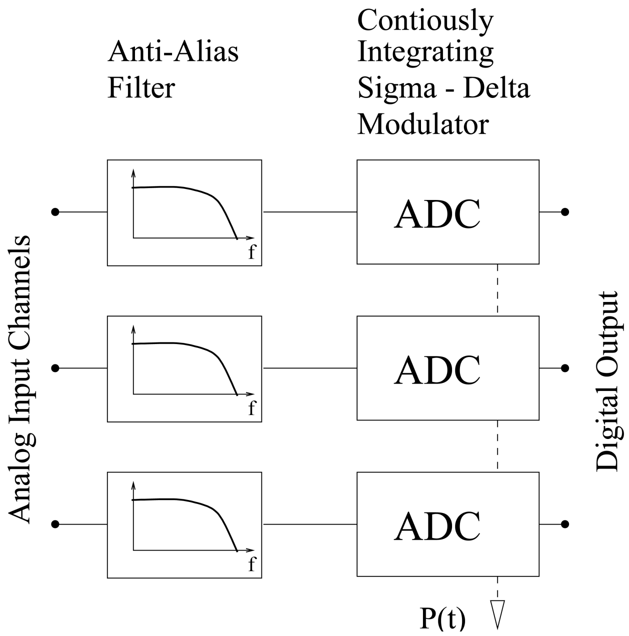

Fig. 6.8 ADC concept used in most modern digital recording systems.

The scheme of a Delta-Sigma-Modulator is given in Fig. 6.9. The negative feedback from the output along the 1-bit DAC and the sum amplifier at the input is performed at a high sampling rate, transferring the quantization noise into the stop band of the digital low-pass filter used for decimation.

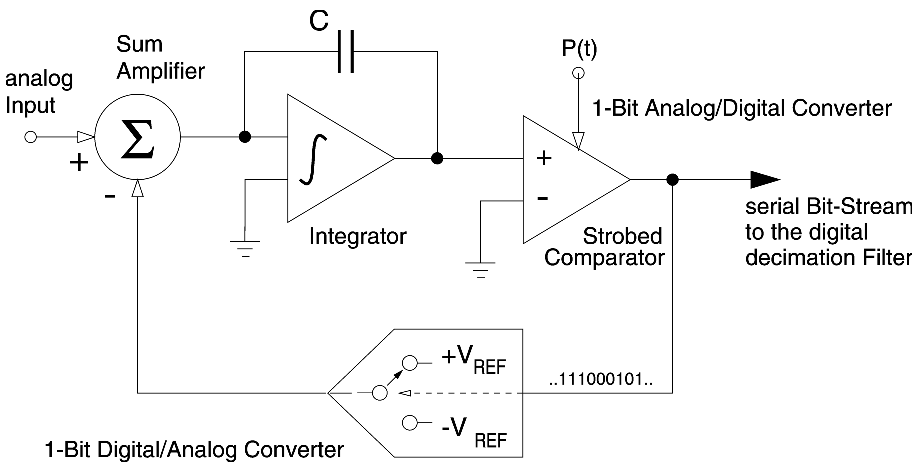

Fig. 6.9 First order Delta-Sigma Modulator.

The principal elements in the block diagram of the Delta-Sigma-ADC shown in Fig. 6.9 are :

- continuously sampling integrator;
- 1-Bit A/D converter (strobed comparator);
- 1-Bit D/A converter (feedback);
- sum amplifier;
- digital low-pass filter (not shown in Fig. 6.9);

In operation, the sampled analog signal is fed to the sum amplifier, along with the output of the 1-bit DAC. The integrated difference signal is fed to the strobed comparator, whose output samples the difference signal at a frequency (the actual sampling frequency) many times that of the analog signal frequency. The output of the comparator provides the digital input for the 1-bit DAC, so the system functions as a negative feedback loop which minimizes the difference signal by tracking the input. The integrator is continously fed with the differential signal and there are no gaps in the analog input signal, as introduced by sample and hold devices. The digital information representing the analog input voltage is coded in the polarities of the pulse train appearing at the output of the comparator. It can be retrieved as a parallel binary data word applying a digital filter operator. In general Delta-Sigma ADCs are described by the order of the integrator, which is in fact an analog low-pass filter. Fig. 6.9 shows a simple first order integrator. In real world systems, an order of four is used. This reflects a compromise between oversampling factor and stability of the modulator limited only by the performance of the analog part of the Delta-Sigma-Modulator.

6.3.5 Noise test

The overall noise level of a recording system can be tested by shortening the input and recording for an adequate period of time. The variation of the measured values reflects the overall noise of the recording device. This test also helps to check temperature influences and find problems in the instruments design. Loading one channel with a defined signal and cross correlating with the others, gives a detailed insight into the system's real performance.

Fig. 6.10 gives an example measured with a PDAS recorder. Channel 1 was fed with a $50mV_{pp}$ sinusoidal signal with a period of one second. Channel 2 was shortened by a 2.5kΩ resistor to simulate the output impedance of a standard passive geophone. All channels were recorded with 100 Hz sampling rate. The lower left section a) of Fig. 6.10 shows the measured signals on both channels with the left ordinate giving the µVolts of the input signal on channel 1 and the right ordinate the µVolts of the signal recorded on channel 2.

The two amplitude-spectra in Fig. 6.10 b) show a zoomed section between 0.5 to 5 Hz from a 2^{18}-point FFT, calculated from a 45 minute measured record. The signals at 2, 3, 4 and 5 Hz are overtones of the signal from the signal generator. They are the result of the non-ideal sinusoidal shape of the signal generator's output, used in this test. The mean level of the spectrum of channel 1 is about one decade above the spectrum of channel 2. This shift mainly reflects the difference in resolution in the gain-ranged mode, that we used in this example. In the record of channel 1, the resolution is 30 nV/count whereas channel 2 is resolved with 4 nV/count. But one can clearly see a peak at 1 Hz in the spectrum of channel 2. Its amplitude is about 6 decades below that of the input signal on channel 1. The cross correlation between the two channels over 500 samples is shown in Fig. 6.10 c). The peak-to-peak amplitude is 0.082 µVolts. From this, the cross coupling can be calculated as $20 \times \log(82nV/50mV) = -115$ dB for this specific instrument. This is a fairly good separation of the input channels for field installations if one keeps in mind that 3-component signals measured (not only) in seismology are highly correlated.

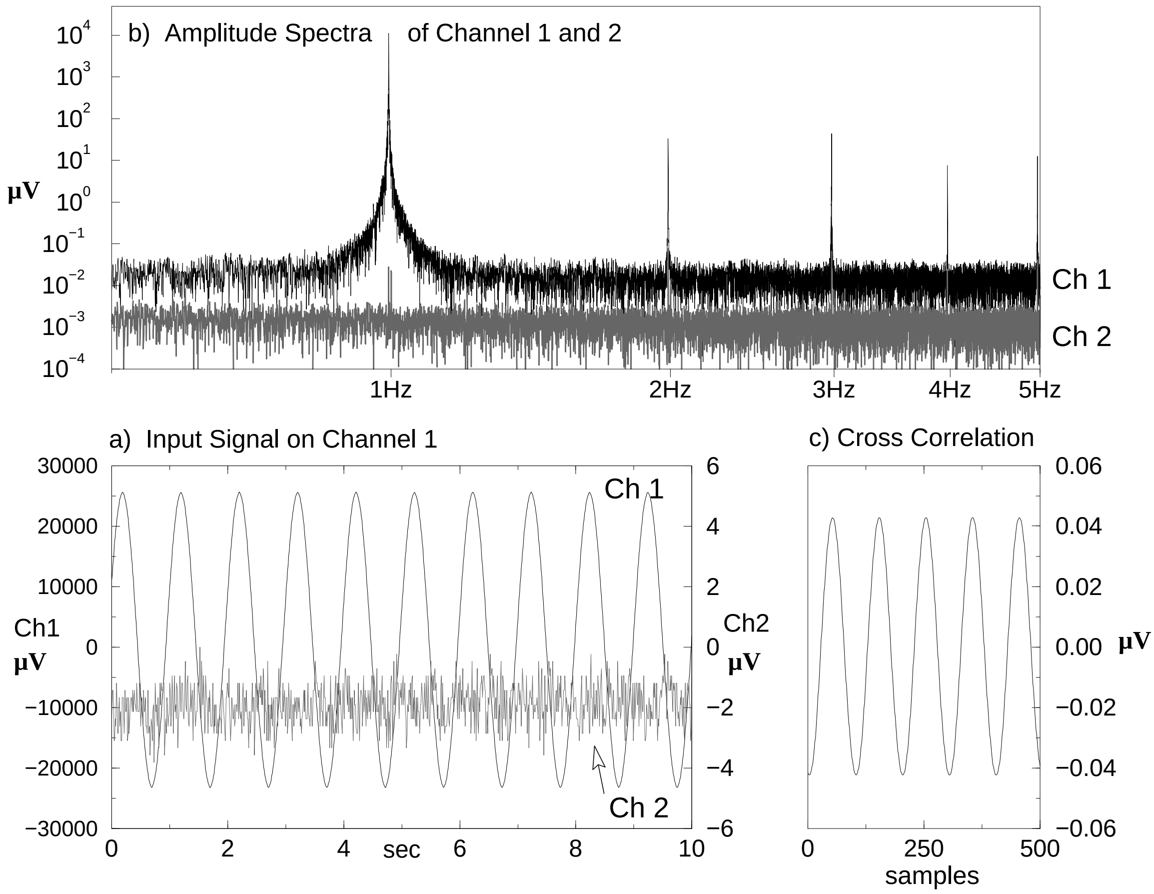

Fig. 6.10 Signal with 50 mV_{pp} amplitude and 1 Hz of channel 1 of a PDAS recorder and the cross coupled signal on channel 2. Shown are the a) measured signals, b) their amplitude-spectra and c) their cross correlation.

6.3.6 The Crystal chip set

Crystal's CS5321 analog modulator and the CS5322 digital filter function together as a high resolution ADC. Therefore they are widely used in geophysical applications. The Reftek and Güralp recording systems, as well as Nanometrics's Orion are based on this chip set. The CS5322/CS5321 combination performs sampling, A/D conversion and anti-alias filtering. The CS5321 utilizes a fourth order Delta-Sigma-Modulator architecture to produce highly accurate conversions at low power dissipation (< 100 mW). It provides an oversampled serial bit stream at 128 kBit/second (f_{in} = 512 kHz, 4th order oversampling architecture) to the CS5322 FIR decimation filter. From the manufacturer's data sheet, one can compile the characteristics of the CS5322/CS5321 modulator and filter combination as shown in Fig. 6.11.

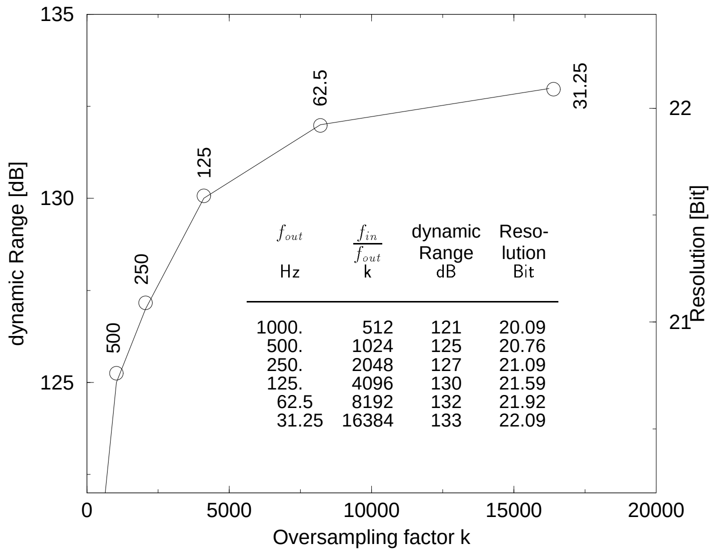

Fig. 6.11 Dynamic range of the CS5322/CS5321 chip set (f_{in} = 512 kHz).

The analog input is resolved between 20 and 22 bits, depending on the oversampling ratio k. The resolution in Fig. 6.11 is limited by the modulator, not by the filter. The CS5322 provides the digital decimation filter for the CS5321 modulator output. It is not a general purpose DSP but consists of a multi-stage FIR-filter. The decimation factor by which the oversampling frequency is reduced, is selectable from 64× to 4096×. The data at the output of the digital filter is represented in a 24-bit serial format.

The -3 dB bandwidth of each decimation rate is approximately 82% of the Nyquist frequency. The filter achieves a minimum of 130 dB signal reduction at the Nyquist frequency for all filter selections. Tab. 6.1 gives an overview of the relation of the user selectable output sampling rate f_{out} with a Reftek data logger and the associated -3 dB bandwidth of the output spectra. Due to the zero-phase FIR design of the digital decimation filter CS5322, phase shift does not depend on frequency. This results in a constant factor for the group delay.

Tab.6.1 Selectable sampling rates, -3 dB bandwidth, and associated group delay in a Reftek system.

f_{out} Hz	f_{in} kHz	$\frac{f_{in}}{f_{out}}$ k	-3 dB Point Hz	Group Delay sec
1000	512.0	512	412.2	0.029
500	512.0	1024	206.0	0.058
250	512.0	2048	103.0	0.116
125	512.0	4096	51.5	0.232
200	409.6	2048	82.4	0.145
100	409.6	4096	41.2	0.290
50	409.6	8192	20.6	0.580
25	409.6	16384	10.3	1.160
40	327.68	8192	16.5	0.725
20	327.68	16384	8.2	1.450
10	81.92	8192	4.1	2.900
5	81.92	16384	2.1	5.800

The CS5322 is realized as a cascade of three symmetrical FIR filters, which produce by design a constant delay of the output signal of 29/ f_{out} (see the manufacturer's data sheet for details of the design). The resulting group delay is given in the last column of Tab. 6.1. The software of the recording system has to shift the time stamp of a sample at the output of the filter stage by this amount of time, which the DSP needs to calculate the filtered output sample. But one has to take into account that the filter settles to full accuracy only after all filter stages have been completely recalculated, i.e., after two times the group delay, according to 57 output words. A transient signal like a step appears at the output of the DSP 29 words later. But to settle to full accuracy it will take another 28 steps. This example demonstrates that the field of application for Delta-Sigma converters is to track slowly changing signals, but not to record transient signals like spikes and steps.

6.3.7 The Quanterra family

The main impact in modern broadband observation was the availability of the Wielandt/Streckeisen (1972) sensor, the presentation of the doctoral thesis of Steim (1986), and the cooperation of Wielandt and Steim (1986). Steim introduced a 24-bit digitizer (US Patent 4866442) which even today gives outstanding performance. It is the de-facto standard in leading broadband experiments such as IRIS GSN, TERRAScope and GEOFON, just to name a few. Steim's digitizer is a variant of the Delta-Sigma-Modulator described in section 6.3.4, but is built up from discrete electronic parts and circuits. The main difference with respect to Fig. 6.9 is the use of a 16-bit ADC instead of the strobed comparator. The digital decimation filter is fed with a 16 bit data stream at 20 kHz. The feedback loop within the Delta-Sigma-Modulator is realized as a 1-bit stream, similar to the design shown in Fig. 6.9.

The main drawback of the original Quanterra data loggers is their rather high electric power consumption. By no means is it a portable instrument; it is designed for the installation in an observatory. These days Kinemetrics offers an array of Quanterra products for different kinds of application, including portables.

6.4 Time base

The usual thing to say about timing on all recording devices is that they allow synchronizing the internal time base against an external clock signal. With the availability of relatively cheap GPS receivers, the timing problem has been solved in a global sense. This type of time signal receivers provides a stable clock reference in the order of µseconds along with time, date and geographic position information, which allow synchronized data acquisition all over the world.

Moreover, the way a recording device synchronizes the internal data streams against the external clock signal differs, depending on the design goals. In active seismic field experiments, predefined time windows are recorded. The time slots vary from some tens of seconds in reflection - up to some tens of minutes in wide angle experiments. Also, in recent seismological experiments a triggered recording mode was mainly used due to limited storage capacities and computer facilities. In this mode, recording windows of several tens of minutes to hours are produced. Synchronizing a clock at the beginning of a recording window and keeping it running for several minutes up to hours will normally not produce significant drift errors. The situation becomes more complicated, if one has to synchronize a continuous stream of data against an external clock signal. If the basic design is not done carefully, life can become a nightmare for people evaluating the data.

The standard procedure sounds quite simple. An internal clock signal is compared against an external time signal, normally the 1 pps (pulse per second) signal derived from a GPS receiver's output. From the difference in time, a control signal is derived and the internal clock is adjusted accordingly. Normally this is done by a phase locked loop in combination with a voltage controlled oscillator. All systems distinguish between two operation modes. If the difference is bigger than a selectable amount of time, the adjustment is done instantly. If everything works perfectly, the so-called jam set mode only occurs during the initializing phase of the system. While switching on the recorder, it updates the absolute time and date by reading the GPS input, and synchronizes the internal time base. During normal operation the time base advance and retard control provides the capability to adjust the internal oscillator without disturbing data acquisition. Adjustments in the order of µseconds per millisecond of the time base are allowed to smoothly synchronize the internal timing clocks with an external source or to simply adjust the internal clocks for known drift rates.

There are important differences in the technical realization of the clock control, resulting in quite different behaviors of the data acquisition systems. Selecting the start time of a recording window for a PDAS, an Orion, or a Güralp, results in digital data streams, starting exactly at the full second of the selected time. A Reftek or a Quanterra will start somewhere in the vicinity of the desired start time. The difference in design is quite small, but of great consequence. In an experiment with more than one instrument of these two types, each recorder samples at the same sampling rate, but at different times. A sample, related to midnight, is randomly distributed over two days on different recorders. In the case of a Reftek or a Quanterra, the ADC/DSP timing signal is derived from an internal oscillator, only providing the clock signal for the ADC/DSP unit and not synchronized by the external clock. Synchronizing is done at the output of the digital decimation filter. The PDAS, Orion, and Güralp recorders are so-called mono-oscillator systems. All timing signals are derived from a master oscillator, providing a 1 pps signal, which controls the timing of the whole recording system. This internal 1 pps signal is adjusted to zero phase with respect to the synchronizing external 1 pps clock signal and forces synchronizing of the ADC, i.e., at the input of the decimation filter. With this timing concept, all instruments in an array or on a profile, take their samples at the same time, starting at full seconds. This simplifies data acquisition of continuous data streams to a great extent.

6.5 Data management

6.5.1 Storage media

In field experiments with continuously recording instruments, one collects about 30-50 MBytes of data per day at a 3-component station with 100 Hz sampling rate. During the last years, hard disks have overcome other mass storage technologies in field recording. They have became reliable, are quite cheap, robust if switched off, their power consumption is moderate, and access is much faster than on other mass storage media. The big advantage of hard disks against other magnetic- and opto-magnetic media is the fact that they are encapsulated and insensitive in rough and dusty environments. For connecting the hard disk to the recorder, many systems support SCSI. This bus system is an accepted and well defined industry standard, available on most platforms. Up to 7 devices can be connected in parallel on a 8-bit SCSI bus and the technical specification even allows hot pluggable devices, i.e., the operator can change the hard-disc without interrupting continuous recording. With an Orion or a Güralp system, it is possible to change the hard disk without interrupting the recording process. On a PDAS this will not only result in a corrupted file system, but one also has a good chance to kill the SCSI controller chip. The only problem with SCSI hard-discs is that they are developing more and more to the high performance server market. In this market segment, low-power (and low price) is not a basic design goal. All the hard-discs fulfilling the requirements for seismic recording systems, were designed for laptops, where IDE is the standard system interface. But to swap IDE devices in a running computer system for reading or writing, ranges from tricky to impossible. So we are still waiting for the manufacturer of seismological recording systems providing us with a hot pluggable mass storage device based on low-power IDE drives and equipped with a TCP/IP connection, which would allow access to a practically unlimited number of storage media at the same time, with all the benefits of a network connection.

6.5.2 Data formats, compression and metadata

Even a hard disk, designed for random access, can be used as a sequential block device, just like a tape drive. The data stream is subdivided into blocks of a fixed length, including the header which defines the type of data and holds the date and time stamp of the first sample. Besides the raw data streams, state of health and status information are generated. Additional channels containing information about internal voltages, temperatures, and information related to the synchronization against the external clock are recorded.

Each subsequent block of data is written to the hard disk. No file or directory structure is necessary and it is quite easy to implement compression algorithms or ring buffer structures. Steim (1986) introduced a widely used algorithm to compress integer time series without loss of information. Blocks of data are organized in frames with a fixed length. Only the difference with respect to the first sample in the frame is stored. An associated table, occupying a 2-bit/sample within the frame, holds the information about the significant word length of the stored differential value (8-, 16- or 32-bit). The compressional rate depends on the difference between two subsequent samples, or in terms of seismology, on the noise situation of the measured signal. In general the compression rate achieved with this so-called STEIM1 algorithm is in-between 1/3 and 2/3. Steim extended this scheme introducing 4 bit differences in the STEIM2 algorithm and gaining a compression rate up to 30% against STEIM1. The Quanterra, the Orion, and the Reftek compress their data with the STEIM1 algorithm, whereas Güralp has its own compression method, but there is no principal difference.

If the block header holds two additional pointers to its predecessor and successor, this is called a tagged file system. In fact, it allows only sequential access to the data of a specific stream. Some systems like the Orion reserve a certain amount of disk space for each raw data stream. This saves the two pointers and in addition, allows an easy implementation of ring buffers. After a ring buffer is filled, the oldest data are subsequently overwritten. This user selectable behavior utilizes continuous data acquisition with a fixed length of time history. If an event in which one is interested occurs, the length of the ring buffer gives the time in which one has to access the station and download the data. Sampling 3 channels at a rate of 100 Hz, a 4 GByte hard disc will provide recording capacity for about 80 days.

To read the data into a computer system, one has only to be able to access the physical raw data device and to know the logical structure of the data frames. The big disadvantage of the frame structure with compressed data is that there is no possibility to address an item directly as recorded at a specific time. One has to read all subsequent block headers from the beginning until finally accessing the data. This storage and compression concept in seismic recording systems is optimized with respect to disk space. This is the best solution for acquiring but a bad choice for processing the data. So the first step after reading the data into a computer system, is to convert and organize it within a file system. There are several different operating systems around, and each operates with several different versions or types of file systems. If a recording system relies on a specific file system, this can become a problem for the user, who has to organize the data on a different operating system.

File systems introduce an additional abstract layer to the data model and also have their specific limitations, starting with the number of characters in a file name, up to the maximum length of a single file. The PDAS is the only recorder mentioned here writing MS-DOS™ data files directly and organizing the data streams in a hierarchical directory structure. The Orion also utilizes the same file system. If the host system is able to mount an MS-DOS™ file system, the ring buffers are visible as ordinary files.

The conclusion of all the different data formats on different recording systems is that each one needs its special treatment. The only common level at the moment is the SCSI hardware, but even this will change in future. All systems have their own way to store the data logically. Even if Steim's compression algorithm is used to write the raw data, the block header structures may differ. But this is not the real annoyance. There is no common agreement on what type of additional information is or should be recorded by the system.

Each manufacturer has his own ideas, how to synchronize against the external clock signal and how to correct the drift of the time base, and of course, how to report this. So, from here on, no general recipe for converting raw data to the user's file system can be given. Also, no recipe is available on how to treat timing errors from different data loggers (there are systems around, even reporting *unknown error*).

The user's file system normally depends on the software being run to process the data. This is the scientific part of the game and may vary from project to project. To buy an instrument from a manufacturer who is supporting only one specific operating system and withholding information about the low-level data structures, may be a bad investment.

Unfortunately, the approach to unify the low level data formats for seismic recording systems, the SUDS-2 format initialized by Ward (1992), failed because of the complexity of the abstract data model and the non-agreement on a common platform for Unix and PC-based systems. Thus, the problem handling the metadata is still unsolved. All the data describing the instrumentation and the site location related to the stored waveform are called metadata. For archiving and exchange of waveform data, there are several data formats in use, mainly, the different variants of SEGY and GCF (pure ASCII code), which are independent of the hardware platform and even email-proof on 7-Bit mail servers. But the most complete and widely accepted standard is the SEED format and it's MSEED variant. SEED is the only standard holding the most important metadata and the waveform data within one single file, a so-called SEED volume. MSEED is a subset of the SEED definition, holding the waveform-data only. Thanks to the excellent software library QLIB2, MSEED has become an easy-to-use and platform-independent data format.

The software packages provided by the manufacturers are mainly reduced to set-up, quick lock, and quality check functions and are normally not intended for scientific evaluation of the data. The software should also provide converters to more common formats mentioned above. Here the GEOFON, the ORFEUS, and the PASSCAL home pages are ideal starting points to find all kinds of useful software and information.

6.6 Conclusions and final remarks

There is no "best" recording system available. It always depends on several aspects, which instrument to choose - and the weights are different from project to project. The requirements for a temporary installed, portable network are completely different from a permanent setup in an observatory, connected to the power supply system and the Internet. Beyond this, Delta-Sigma-Modulators with 130-140 dB and GPS-timing are state of the art, Web-hosted communication via Internet is coming up, and prices, hopefully, down. Within this chapter several products from various manufacturers are named. They are only used as examples to show specific differences in the technical realization of seismic recording systems. In fact, there are more products available but this chapter has no intention to give an overview of the market. If one particular product is not mentioned here, it is not an opinion in terms of its quality.

6.7 Glossary of technical terms and links

ADC : **A**nalog to **D**igital **C**onverter. A device that converts data from analog to digital form.

CS5321/CS5322 : A 24-bit word-length variable bandwidth ADC. The CS5321 is a Sigma-Delta-Modulator which functions together with the CS5322 digital filter as a high resolution ADC. The CS5322/CS5321 combination performs sampling, AD conversion, and anti-alias filtering. The circuits are manufactured by *Crystal* (http://www.crystal.com/).

DAC : **D**igital to **A**nalog **C**onverter. A device which takes a digital value and outputs a voltage which is proportional to the input value.

DSP : **D**igital **S**ignal **P**rocessor. The *big four* programmable DSP chip manufacturers are Texas Instruments, with the TMS320 series of chips; Motorola, with the DSP56000, DSP56100, DSP56300, DSP56600, and DSP96000 series; Lucent Technologies (formerly AT&T), with the DSP1600 and DSP3200 series; and Analog Devices, with the ADSP2100 and ADSP21000 series. For further information have a look at www.bdti.com/pocket/dsp_guide.htm.

Earth Data : Earth Data Limited specializes in the research, design and manufacture of data acquisition, telemetry and 24-Bit portable recording systems (www.kenda.co.uk/edata).

FIR : **F**inite **I**mpulse **R**esponse filter, also named acausal- or, in their symmetrical realization, zero-phase-filters. For a brief description of digital filters (see Scherbaum,1996; and http://lbutler.geo.uni-potsdam.de/service.htm).

Gain-ranged Mode: A Gain Ranging Amplifier is scaling the input signal to fit the ADC's digitizing window based on signal level. This method increases the dynamic range, but not the resolution of a recording system. The resolution is limited to something less than the number of bits available from the ADC, due to noise introduced by the uncertainties of the different gain steps. Gain-ranged mode was used in older 16-bit ADC based systems to achieve dynamic ranges up to 130 dB.

GEOFON : The GEOFON program of the GeoForschungsZentrum Potsdam presently operates, together with its partner organizations, 40 permanent and a varying number of longterm temporary broadband seismological stations. 30 permanent stations are located in Europe and the Mediterranean. Most of these stations are equipped with Streckeisen STS-2 very broad band seismometers and adequate Quanterra Q380 or Q4120 dataloggers (see www.gfz-potsdam de/geofon/index.html).

GPS : **G**lobal **P**ositioning **S**ystem. There are a lot of good sites to start from, for example www.skydiversdepot.com/gps2.htm.

Güralp : Güralp Systems Ltd (GSL). Manufacturer of seismometers and data acquisition systems (www.guralp.demon.co.uk).

GSN : The IRIS Global Seismographic Network. The goal of the GSN is to deploy 128 permanent seismic recording stations uniformly over the Earth's surface. These stations continuously record seismic data from very broad band seismometers at 20 samples per second. It is also the goal of the GSN to record data with a dynamic range of 140 db (24 bit digitizers). (www.iris.washington.edu/GSN/index1.htm)

IDE : **I**ntegrated **D**rive **E**lectronics. Originally called IDE, the ATA interface was invented by Compaq around 1986. Standardized by the ANSI group X3T10 (who named it *Advanced Technology Attachment* (ATA)). Ratification in November 1990.

IIR : **I**nfinite **I**mpulse **R**esponse filter, also named causal-filter. All analog filters are of this type. For brief discription of digital filters see Scherbaum (1996).

IRIS : **I**ncorporated **R**esearch **I**nstitutions for Seismology. 1200 New York Ave. NW, Suite 800, Washington, DC 20005 (www.iris.edu).

Kinemetrics : Kinemetrics Inc. Manufacturer of seismic sensors and data acquisition systems. (see www.kinemetrics.com).

Lennartz : Lennartz electronic GmbH. Manufacturer of seismic sensors and data acquisition systems (see www.lennartz-electronic.de).

MARSlite : The MARS recorders are an array of portable data loggers developed and manufactured by Lennartz electronic GmbH. Recent products are the MARSlite and M24 data loggers.

Nanometrics : Manufacturer of data acquisition systems. Nanometrics also offers complete network solutions based on satellite communication (see www.nanometrics.ca).

ORFEUS : **O**bservatories and **R**esearch **F**acilities for **E**uropean **S**eismology. ORFEUS is the European non-profit organization that aims at coordinating and promoting digital, broad-band seismology in Europe (see orfeus.knmi.nl).

ORION : Digital recording system, manufactured by Nanometrics.

PASSCAL : **P**rogram for the **A**rray **S**eismic **S**tudies of the **C**ontinental **L**ithosphere. PASSCAL Instrument Center, New Mexico Tech, 100 East Road, Socorro, NM 87801, U.S.A (see www.passcal.nmt.edu/passcal/resources.htm).

PDAS : Portable Data Acquisition System. Solid first generation digital recording system manufactured by Geotech/Teledyne, 3401 Shiloh Road, Garland, Texas 75041 (see www.geoinstr.com). This data-logger is no logger available on the market, but still widely used.

QLIB2 : Software library available as source code (C/Fortan) from Doug Neuhauser (doug@seismo.berkeley.edu) to read and write MSEED data files on different hardware platforms. Jewel found in the PASSCAL software distribution (Ver. 1.9) under contrib/mseed/qlib2. Unfortunately the PASSCAL routines do not profit from this library, because they used their own.

Quanterra : Advanced broad band remote data acquisition system. Developed by Steim (1986), it is the defacto standard in global broadband seismology. These days Kinemetrics is selling an array of Quanterra products with different specifications.

Reftek : Refraction Technology, Inc. 2626 Lombardy Lane, Suite 105, Dallas, TX 75220, U.S.A. (see www.reftek.com).

SAR-ADC : Succesive Approximation Analog to Digital Converter.

SCSI : Small Computer System Interface. Bidirectional, parallel interface to connect up to 7 (15 with wide SCSI) external devices to a computer. ANSI-X-3T9.2 *American National Standards Institute* (see www.ieee.org/index.html).

TCP/IP : Transmission Control Protocol over Internet Protocol. The de facto standard Ethernet protocols. TCP/IP was developed by DARPA for internet working and encompasses both network layer and transport layer protocols. While TCP and refer to the entire DoD protocol suite based upon these, including telnet, FTP, UDP and RDP.

TERRAscope : is a very broadband seismographic network in Southern California. Each station consists of a Wielandt/Streckeisen seismograph, a strong-motion sensor, and a barograph. The data is recorded on-site using the 24-bit Quanterra dataloggign system and collected by the Caltech Seismo Lab. (www.gps.caltech.edu/terrascope/TerraInfo.html).

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Recommended overview readings (see References under Miscellaneous in Volume 2)

Buttkus (1991)
Havskov and Alguacil (2004)
Scherbaum (1997)
Scherbaum (2001)
Shannon (1949)
Ward (1992).
Willmore (1979)