

Basics of Data Acquisition



About ADInstruments

ADInstruments provides complete and integrated data acquisition and analysis solutions to academic institutions, government organisations and private industry.

At the core of our product line is the world-renowned PowerLab system with LabChart software. Together they offer comprehensive signal processing, data recording, display, and analysis features for a wide variety of research applications. In conjunction with a computer, the systems provide the functionality of a multi-channel, real-time chart recorder, polygraph, XY plotter and digital oscilloscope. You can take advantage of variable sampling rates and remarkable resolution with the benefits of powerful, computer-based data handling and analysis.

Wherever your research leads, our team can support you with the latest technology and powerful but simple tools that give you the power to innovate.



Top 100 universities all use LabChart and PowerLab

In 2010, Quacquarelli Symonds released a list of the Top 100 universities for life science, based on academic citations, peer review, recruiter review, faculty student ratio and international orientation.

PowerLab hardware and LabChart software are used at every one of these institutions, including Harvard, Oxford, and the University of Tokyo.

PowerLab

PowerLab is the original high-performance digital data acquisition device engineered for precise, consistent, reliable data acquisition, giving you reproducible data while meeting the strictest international standards.

ADInstruments provides integrated solutions to advance life science research

LabChart

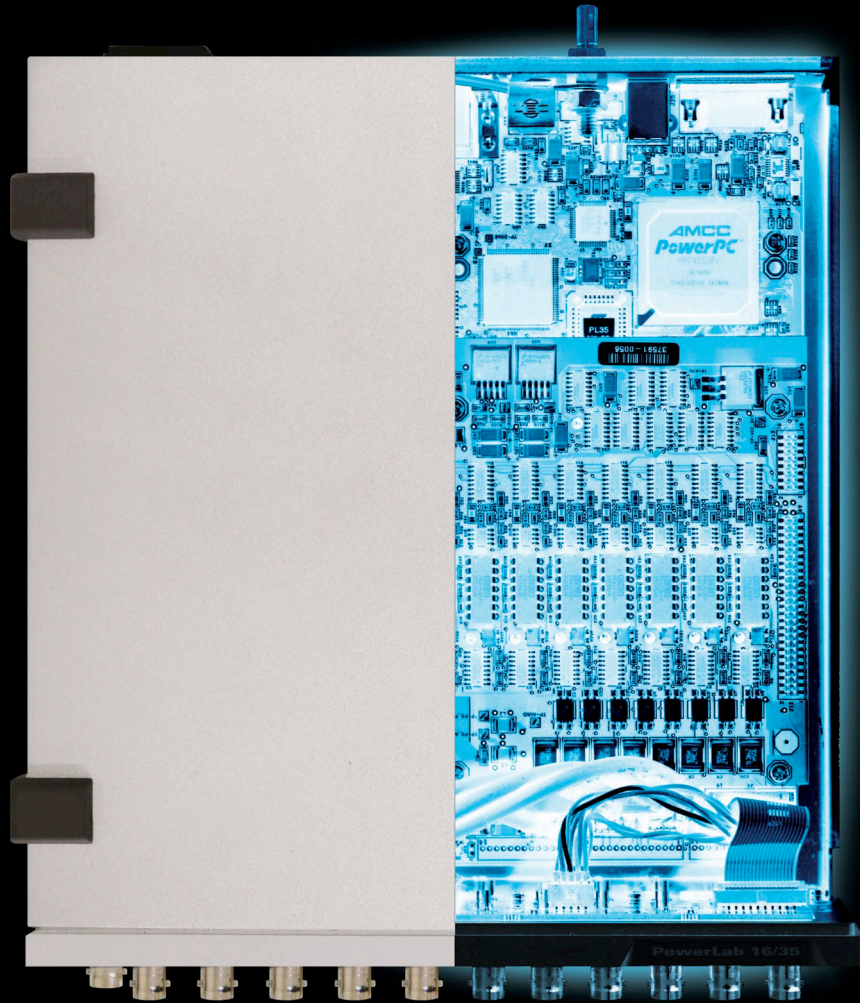
All your analysis in one place

LabChart analysis software is at the centre of all ADInstruments research solutions and acts as a platform to integrate all your data streams into one place.

Designed specifically for life science data, LabChart provides up to 32 channels for data display and either automated or customizable analysis options that are powerful and easy to use.



Much more
than a box



PowerLab

Data with integrity

PowerLab is engineered for precise, consistent, reliable data acquisition, giving you the reproducible data you need while meeting the strictest international safety standards.





Extend your research into new territories

ADInstruments systems provide the flexibility to extend studies into any of these human, animal or in vitro applications. Ask one of our experts to design a system to suit your needs.

Investigating novel or niche research?

With ADInstruments, the power is in your hands. We offer high spec solutions to acquire a range of quality signals, or you can integrate your own custom hardware configurations with PowerLab. With options for both standard analysis features and smart tools for customization in LabChart, you can easily apply advanced calculations as your experiment unfolds.

The ADInstruments advantage

- **Experience you can trust**

ADInstruments systems have been installed in thousands of research institutes, universities, hospitals and commercial laboratories worldwide. With more than 45,000 systems installed worldwide and over 30,000 published scientific research papers featuring our products, you can be assured that the decision to purchase an ADInstruments data acquisition system is the right one.

- **Flexibility**

The flexibility of LabChart allows it to be used in a wide variety of life science applications, maximising the returns of your investment in time and capital.

- **Intuitive and powerful software**

LabChart allows researchers to concentrate on the science. Powerful data extraction and analysis features speed up the research process.

- **Data integrity**

PowerLab systems are calibrated and tested to deliver data you can trust. The software incorporates data integrity features such as multiple block recording with individual settings and calculations stored within a single file, preservation of raw data, and date stamping.

- **GLP and 21 CFR Part 11 compliance**

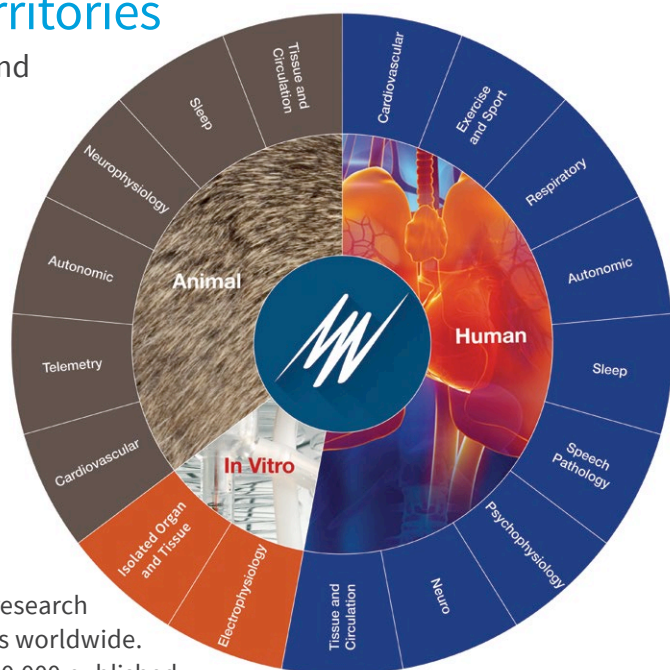
LabChart and our GLP software provide the required user interface, audit trail and signing components for non-repudiation of data under GLP and 21 CFR Part 11 compliance.

- **Quality and reliability**

ADInstruments products are manufactured under a quality system certified by an accredited body as complying with ISO 9001:2008

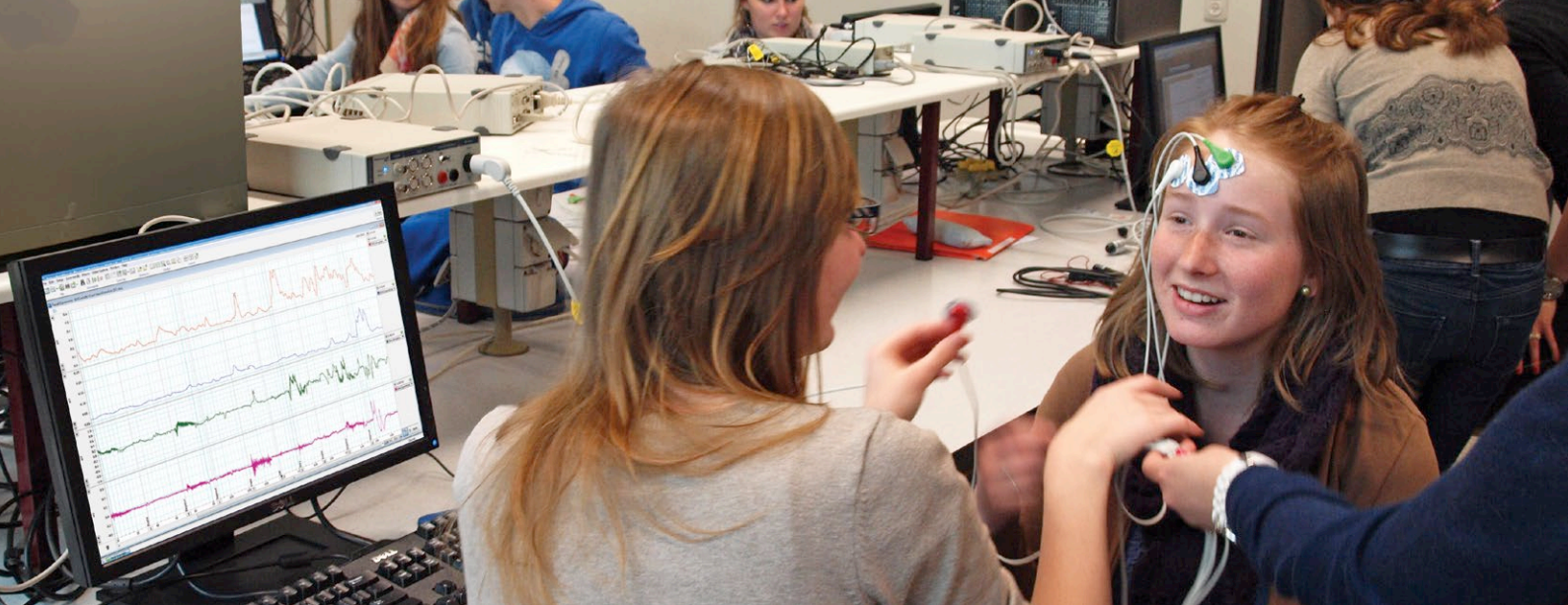
- **Excellence in customer training and support**

Our international graduate and postgraduate support staff are dedicated to making sure our customers are satisfied.



Maximize your potential

If you have a particular research need you would like to discuss, get in touch with our experienced support team. We can work with you to customize an effective solution to record and analyze the specific data that you need.



Background

The purpose of the PowerLab and LabChart data acquisition system is to acquire, store and analyze data. Figure 1 shows a summary of the acquisition. Usually, the raw input signal, which may be from any number of biological or physical sources, is in the form of an analog voltage whose amplitude varies continuously over time. This voltage enters the PowerLab via the input connections and can be modified by amplification and filtering, a process called signal conditioning. After signal conditioning, the analog voltage is sampled at regular intervals. The signal is then converted from analog to digital form before transmission to the attached computer.

LabChart analysis software displays the data directly; it plots the sampled and digitised data points and reconstructs the original waveform by drawing lines between the points. Digital data can be stored on disk for later retrieval. LabChart software can also easily manipulate and analyze the data in a variety of ways.

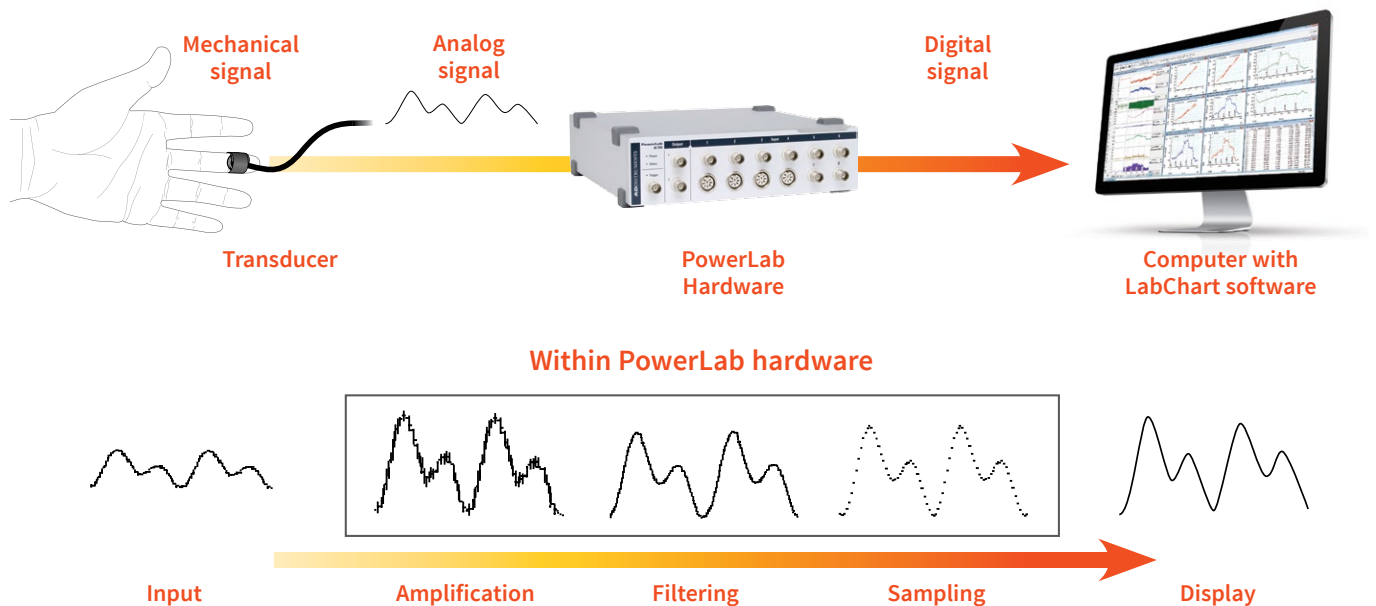


Figure 1 A summary of data acquisition using the PowerLab system.

Most of the parameters that affect acquisition can be set by the user through the software. To make a good recording, the parameters must be appropriate for the signals being recorded.

To acquire good data you need to:

- 1 Select a suitable sampling rate
- 2 Apply the correct input range
- 3 Choose suitable filter settings

In some disciplines you may be able to find tables of suggested sampling rates, ranges, and filter settings, but these should not be applied blindly. You still need to know the science (what you are recording, why you are recording it, and how it relates to real phenomena) and the technique (how best to record, and what limitations or compromises are inherent in the process).

Sampling rate

The first thing to choose is a suitable sampling rate. Samples are taken from the signal at regular time intervals. The appropriate sampling rate depends on the signal to be measured. If the sampling rate is too low, information is irreversibly lost and the original signal will not be represented correctly (Figure 2A). For example, if you sample at 200 Hz, this will display 200 points every second. This would be adequate for a human ECG but you would lose information from a mouse ECG whose heart rate is 5-6 times higher. If the sampling rate is too high, no information is lost, but the excess data increases processing time and may give excessive noise in the signal or result in unnecessarily large disk files.

You can calculate the size of the data file that will be collected based on the sampling rate:

$$\text{Amount of data (bytes)} = 2 \times \text{number of channels} \times \text{sampling rate} \times \text{recording length (in seconds)}$$

You will need this much space on your computer hard disk to record the data, but LabChart will compress the file when saving, so you will not require as much storage space.

Aliasing

Periodic waveform recordings that have been sampled too slowly may be misleading as well as inaccurate. This is known as aliasing where high frequency waveforms may be represented as lower frequency waveforms (Figure 2B).

To prevent aliasing, the sampling rate must be at least twice the rate of the highest expected frequency of the incoming waveform. This sampling rate is known as the 'Nyquist frequency', the minimum rate at which digital sampling can accurately record an analog signal. For example, if a signal has maximum frequency components of 100 Hz, the sampling rate needs to be at least 200 Hz to record it accurately. To provide a safety factor to guard against information loss, it is usual to sample at five to ten times the highest expected frequency rather than the minimum two times.

You can select the sampling rate in LabChart using a small drop-down menu in the top right corner of the Chart View (Figure 3). The default setting is 1 KHz, but you can adjust it to as low as 1 sample every 10 minutes and as high as 200 KHz (depending on the PowerLab model and the number of channels being used).

In most cases, the highest expected frequency will be known. It may be limited by the transducer used: a bridge transducer to measure mechanical force will not produce high frequencies, for instance. If you are unsure of the frequency range (bandwidth) of your signal, a useful rule is to choose a sampling rate high enough to allow at least 20 samples for any transient peaks or recurring waves in the signal. You can formally determine a signal's highest frequency by sampling the signal at the maximum rate, and looking at the frequency spectrum of the signal using the Spectrum display available under the Window menu item in LabChart.



Figure 2A Undersampling - human finger pulse recorded at 10/s and 200/s. The former sampling rate is too low to depict the signal accurately.

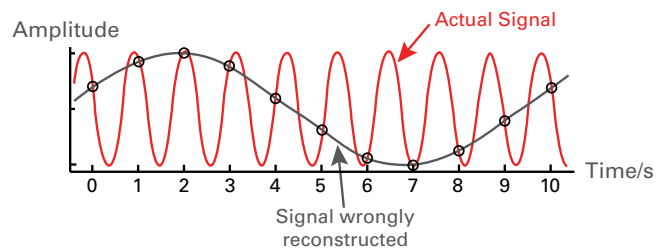


Figure 2B Aliasing - sampling a higher frequency signal at 1 sample per second gives a misleading waveform (real-life effects are more subtle).

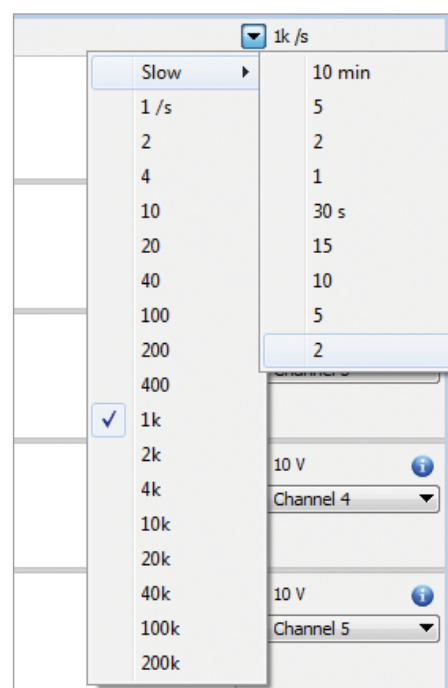


Figure 3 LabChart sampling rate options.

Range and resolution

To accurately record a signal, it is important that the range of the LabChart channel is greater than the signal's amplitude. Many systems use 'gain' when looking at amplified signals. Range is inversely proportional to gain (the amount of amplification), and is a more useful concept than gain since it relates directly to the signal being measured. With a PowerLab system, the range can be set independently for each channel in the right-hand drop down menus.

The resolution of the signal represents the accuracy of the display. PowerLab has a 16-bit resolution which means the range can be divided into a maximum of 2¹⁶ levels (65 536) although LabChart fits 64 000 to the range. So if you are looking at a ± 10 V range the smallest discernible level would be ~ 300 μ V (20 V corresponding to the ± 10 V range divided by 64,000).

If a signal is very small in relation to the range (Figure 4A), then its resolution will be degraded. In extreme cases, the recorded waveform may appear stepped rather than smooth. Even though you could see a ± 380 mV signal on

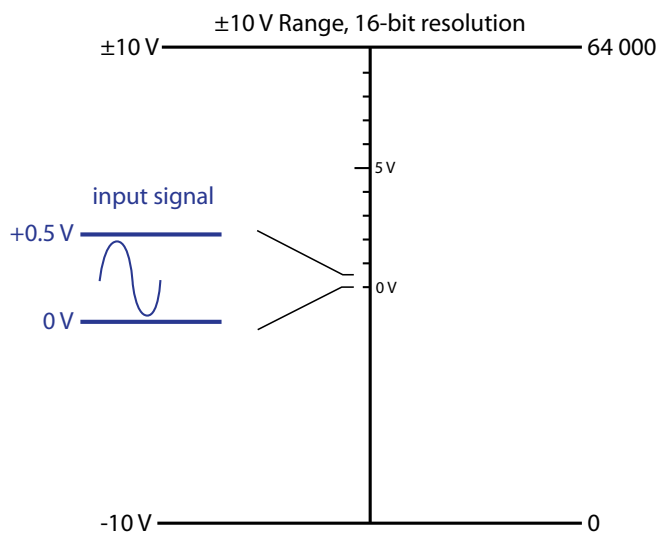


Figure 4A A small signal on a large range will have poor resolution as only a small part of the range is used.

the default 10 V range, it would be preferable to use 500 mV range to measure it at maximum resolution. It would be safer in practice to use a 1 V or 2 V range, though, since unexpectedly large peaks could exceed the 500 mV range if the signal was unpredictable.

If the signal amplitude exceeds the range (Figure 4B), there will be severe information loss. Any signal exceeding the range is 'out of range', a condition indicated where no amplitude can be given. If there is any possibility of this condition occurring, you should set the range to a larger value.

For the best resolution, the maximum amplitude of the signal you are interested in should be reasonably close to the chosen range without exceeding it. That way, the minimum change in voltage discernible in digitisation remains small in relation to the signal being measured. Note that changing a waveform's on screen display (by enlarging it in the Zoom window or by stretching or shrinking its Amplitude axis, for instance) does not affect its resolution, just its appearance.

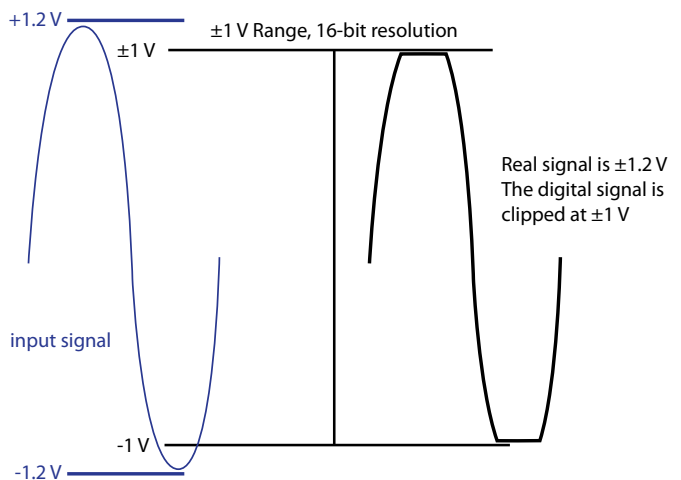


Figure 4B If the signal is larger than the range, the top and bottom is 'clipped' and maximum and minimum value is lost.

Using single-sided or differential inputs

The Single-sided and Differential options in LabChart control signals acquired through the differential inputs (i.e. the pod connectors). These options do not appear where the input for a channel is only single-sided — in this case, the input functions as if the single-sided option were checked permanently.

Single-sided — When this option is selected, only the positive (non-inverting) input on the front of the PowerLab is used, and a positive signal fed into it will be shown as a positive signal on the display. The inverting input is grounded.

Differential — When the Differential option is selected, both positive (non-inverting) and negative (inverting) inputs for that channel are used, and neither is grounded. The signal shown on the display is the difference between the signals at the positive and negative inputs. If both input signals were the same, they would cancel each other out.

Filtering and smoothing

Filters and smoothing are used to get rid of noise from a signal. Noise can be referred to as unwanted signal. It is likely to be a problem at lower range settings, when trying to measure very small signals. Random noise (such as thermal noise) is inherent in all electronic circuits, including those of the PowerLab recording unit. It can be minimised through filtering. Other causes of noise are stray electromagnetic and electrostatic fields including interference (often at the mains frequency of 50 Hz or 60 Hz) from unshielded power lines, switching equipment, fluorescent tubes and computers. Interference can significantly affect a signal, but can be reduced through reasonable care in the arrangement and shielding of equipment and cables.

There are two types of filter available within PowerLab and LabChart: analog/hardware filters or digital/software filters.

Analog/hardware filters

Analog/hardware filters are used to filter the incoming, continuous signal before it is sampled by the analog to digital converter (ADC). These filters are built into ADInstruments front-ends (Bio Amps, Bridge Amps etc) and some PowerLab units. ADInstruments front-ends initially amplify the signal to a level suitable for filtering. The analog filters are then used to remove unwanted frequencies, before further amplification is performed before digitisation. Filtering the signal prior to full amplification is essential for biopotential measurements to improve the signal-to-noise ratio.

Digital/software filters

LabChart's digital or software filters filter the data after it has been sampled and recorded by the PowerLab. Digital filters are used during or after data acquisition and are advantageous because:

- It is possible to design digital filters that are impractical to make in analog form
- They are stable over time and provide consistent, reproducible signal filtering
- In LabChart, they can be applied post data acquisition while the raw data is retained

A disadvantage of post-acquisition digital filtering is that unless analog/hardware filters have also been used prior to digitisation, any noise or baseline offset will also be amplified. This will have a negative effect on the signal's resolution.

Basic filtering terms

To understand the basics of filtering, it is first necessary to learn some important terms used to define filter characteristics. While these terms apply to all types of filters, for simplicity the following examples will only refer to low-pass filters.

- **Cut-off frequency (f_c)** — Also referred to as the corner frequency, this is the frequency or frequencies that define(s) the limits of the filter range(s). It is the desirable cut-off point for the filter.
- **Stop band** — The range of frequencies that is filtered out.
- **Pass band** — The range of frequencies that is let through.
- **Transition band** — The range of frequencies between the pass band and the stop band where the gain of the filter varies with frequency.

Low-pass filters

A low-pass filter allows signal frequencies below the low cut-off frequency to pass and blocks frequencies above the cut-off frequency. It is commonly used to help reduce environmental noise and provide a smoother signal.

A simple way to understand how a filter works is to plot signal frequency against signal gain (Figure 5A). When a signal is unfiltered, it is recorded at a gain of 1, that is, the full signal is being recorded. All low-pass filters have a frequency (f_a) above which the gain is very small and the signal is virtually non-existent.

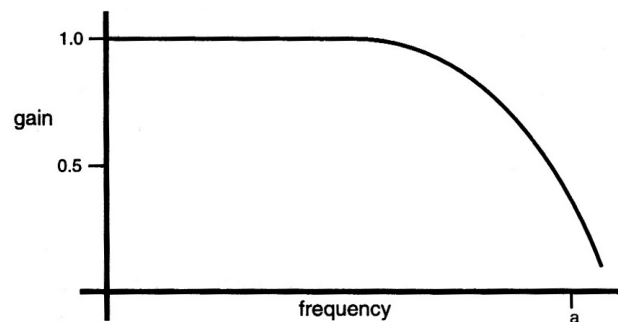


Figure 5A Gain versus frequency (S.S. Young, 2001).

Ideally, a perfect low-pass filter would have a gain of 1 for signals having frequencies that are meant to pass through the filter, and a gain of zero for frequencies that are meant to be blocked (like the ideal brick wall filter in Figure 5B). However, filters are imperfect, and the gain of a low-pass filter never quite falls to zero. The frequency at which the gain starts to decrease by a reasonable amount is the cut-off (corner) frequency (f_c). This reduction in signal gain after the cut-off frequency is often referred to as signal attenuation and is commonly presented in decibel (dB) units.

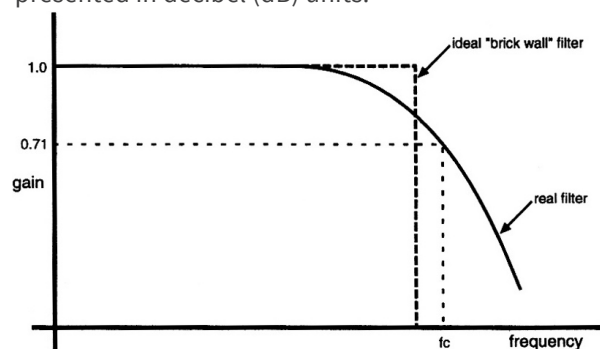


Figure 5B Effect of low-pass filtering on signal gain (S.S. Young, 2001).

Note: Decibels are not units of measurement in the conventional sense but represent a log ratio, thereby describing how much bigger or smaller one thing is compared to another.

All signal frequencies below the cut-off frequency are referred to as the pass band (Figure 6). All signal frequencies above the cut-off frequency are referred to as the stop band. The region between the pass and stop bands is referred to as the transition band or transition width. This width (in Hz) depends on how sharply the filter response drops from the pass band to the stop band. Related to this is the roll-off rate, which, for low-pass filters is the rate at which the signal gain decreases when the signal is above the cut-off frequency. The narrower the transition band, the steeper the roll-off.

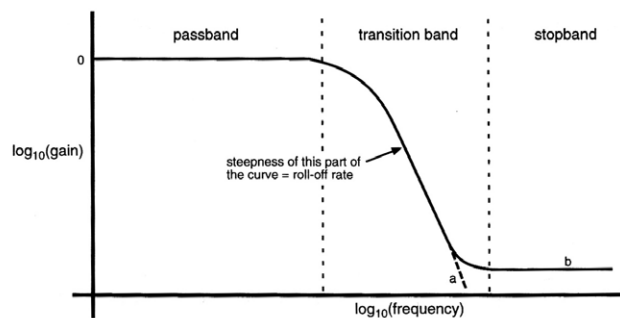


Figure 6 Filter bands (S.S. Young, 2001).

Other filter types

High-pass filters

A high-pass filter allows frequencies higher than the cut-off frequency to pass and removes any steady direct current (DC) component or slow fluctuations from the signal. Such filters are often used to stabilise the baseline of a signal (i.e. minimise baseline drift in an ECG signal). A useful comparison of the effects of a low-pass filter in comparison to a high-pass filter is presented in Figure 7.

Mains filters

Mains interference (50/60 Hz) from power lines or electrical equipment is not static and may vary during the day, with more variation in some countries than others. An adaptive mains filter tracks and removes the mains noise (including the harmonics of the fundamental) with minimal distortion to the recorded signal. Digital mains filters are included in LabChart for use with PowerLab/10, /15, /20, /25, /26, /30 and /35 series models.

Notch filters

A notch filter removes a particular frequency from a signal and has a frequency response that falls to zero over a narrow range of frequencies. For example, a 50 Hz notch may block signals from 49.5 – 50.5 Hz. Notch filters are available in all ADInstruments Bio Amps.

Narrow band-pass filters

Narrow band-pass filters are used to remove all signal frequencies except for a particular band, say to record 8 – 12 Hz activity in EEG recordings. Frequencies either side of this band are blocked.

Band-pass filters

A band-pass filter may be used to pass a larger range of frequencies say 0 – 100 Hz in EEG activity. Frequencies either side of this band are blocked.

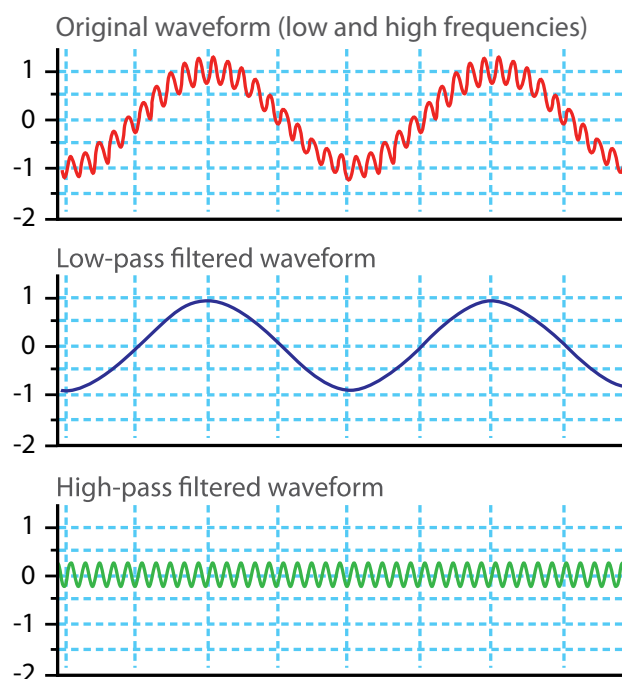


Figure 7 Comparison of the effects of low and high pass filters on a signal waveform.

Band-stop filters

A band-stop filter blocks a certain range of frequencies and allows frequencies either side of this range to be passed. For example, you may wish to block Beta [β 1: 16 – 32 Hz] activity from an EEG recording but record all other frequencies between 0 – 15 Hz and 33 – 100 Hz.

Anti-aliasing filters

The anti-aliasing filter is a sampling rate-dependant low-pass filter. Aliasing can be caused by under sampling, so for this filter to be effective it is important that the correct sampling rate has been applied. In LabChart, the cut-off frequency of the anti-aliasing filter is set to half the sampling frequency but disabled at sampling rates lower than 100 Hz. The anti-aliasing filter is a digital filter available in the 15 and 26 series PowerLabs.

Smoothing

Another means of removing unwanted high frequencies, noise or clutter from a waveform, is to use 'smoothing'. In LabChart, it works both offline and online. Choosing the Smoothing... command from any Channel Function drop-down menu displays the Smoothing dialogue for that channel.

Four smoothing methods are available: a moving average with a Triangular Window, Savitzky-Golay, Median Filter and Averaging.

- **Triangular (Bartlett) window smoothing** works by taking the sample point together with a variable number of points on each side in the moving average window, weighting the values (most at the middle, which decreases to zero going out towards the window edges) and averaging them to give the smoothed value at the sample point (Figure 8).

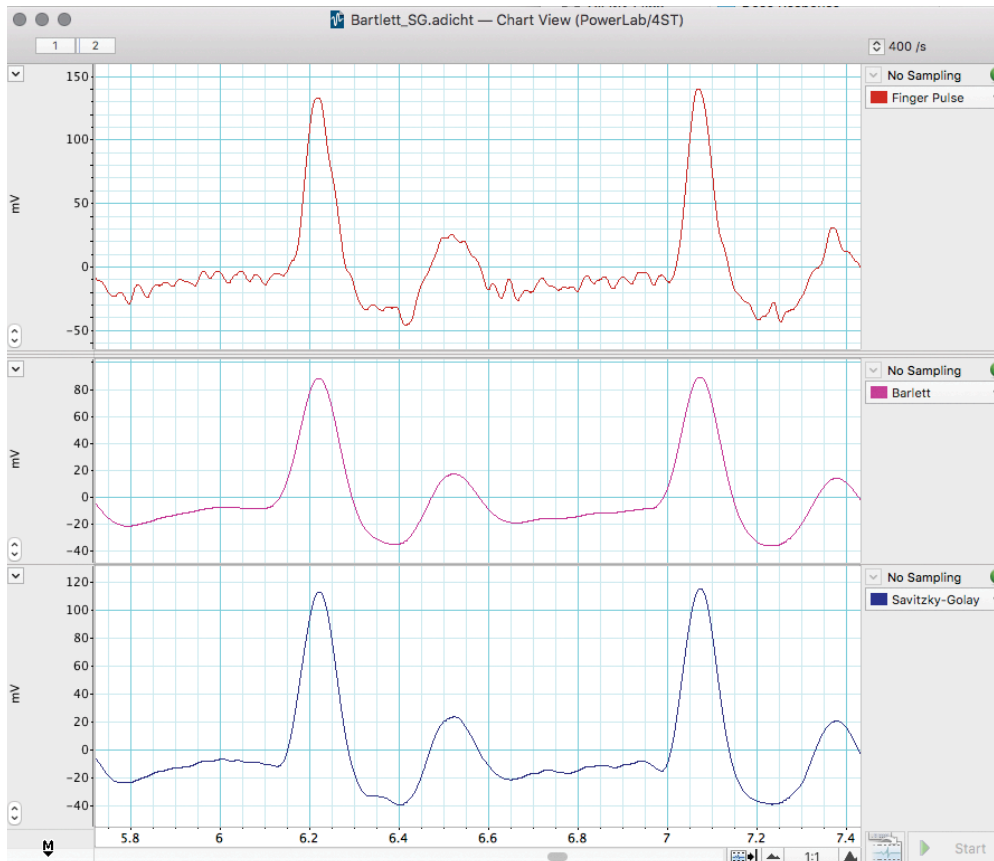


Figure 8 Example used in LabChart Training courses, examining the differing effects of the Bartlett and Savitzky-Golay smoothing functions.

- **Savitzky-Golay smoothing** works by fitting a polynomial in a window around each sample point, using least squares fitting. You can specify the order of the polynomial from two to six.
- **Median filter smoothing** sorts the data values in the window around each sample point and returns the middle value.
- **Averaging (decimation) smoothing** replaces all the data values in the window with a true ("boxcar") averaged value. This compresses the data and effectively results in a change to the sampling rate.

With all the smoothing methods, the number of points used to calculate each smoothed value is set with Window width. The window width is always an odd number (3–2000001 for triangular smoothing, 3–999 for Savitzky-Golay smoothing, 3–255 for the median filter and 2–2000001 for averaging). The window width should be chosen with caution. A window width that is large in comparison with the time-span of changes in the signal slope will bias the calculation of the smoothed value, whereas a too-small width will not effectively remove noise.

Suggested settings for selected life science applications

Signal	Range	Sampling rate	Filters/Other information
ECG	10 -20 mV	<ul style="list-style-type: none"> • Mouse 4 kHz • Rat 2 – 4 kHz • Rabbit 1 – 2 kHz • Guinea Pig 1 – 2 kHz • Dog 400 Hz – 1 kHz • Pig 400 Hz – 1 kHz • Human 400 Hz – 1 kHz Should be 15-30 sample points in the QRS Complex	ECG signals contain limited information above 100 Hz. <ul style="list-style-type: none"> • High pass: 0.3 Hz to minimise iso-electric (baseline) drift • Low pass: $\leq 25 - 50$ % sampling rate (typically 200 – 1 kHz low-pass cut-off) • Notch: Do NOT use as may distort ECG signal. • Mains: Will suppress electrical interference without distorting signal.
EEG/ECOG	200 – 500 μ V	400 – 1000 Hz	EEG signals contain limited information above 50 – 100 Hz (Humans, Rats, Mice and Sheep). <ul style="list-style-type: none"> • High-pass: 0.1 or 0.3 • Low-pass: 100 – 200 Hz cut-off • Notch: No (unless the user is aware of affect on signal) • Mains: Yes Frequencies of particular interest: <ul style="list-style-type: none"> • Delta: 0.5 – < 4 Hz • Theta: 4 – < 8 Hz • Alpha: 8 – < 12 Hz • Spindles: 12 – 14 Hz • Sigma: 12 – < 16 Hz • Beta: (β1) 16 – 32 Hz • Beta: (β2) 30 – 60 Hz
EMG	Variable between species and muscle types	2 – 4 kHz	EMG signals contain variable frequencies; however, most common frequency bands recorded are 0.3 Hz – 1 or 2 kHz. <ul style="list-style-type: none"> • Notch: No (unless the user is aware of effect on signal) • Mains: Yes Removing ECG artifacts, particularly in intrathoracic recordings (i.e. esophageal EMG) often requires difficult template matching (not available in LabChart). If you need to remove low frequency artifacts > 2 Hz (i.e. respiratory or ECG artifacts) prior to recording/ digitizing the signal, then use the high-pass filters available in the Bio Amps.
Blood Pressure		For accurate reproduction of dirotic notches and BP harmonics, sampling speeds of 50 to 100 times the heart rate (in Hz) are desirable. <ul style="list-style-type: none"> • Humans: 400 Hz • Guinea Pigs/Rats: 2 kHz • Mice: 2 kHz 	Typical heart rates: <ul style="list-style-type: none"> • Humans: 80 – 200 bpm (max 4 Hz) • Guinea Pigs/Rats: 200 – 400 bpm (max 7 Hz) • Mice: 500 – 700 bpm (max 12 Hz) As a minimum, the low-pass filter frequency should be 10 times the heart rate (in Hz) and half the sampling rate. Recommended low-pass cut-off frequency: <ul style="list-style-type: none"> • Humans: 100 – 200 Hz • Guinea Pigs/Rats: 100 Hz – 1 kHz • Mice: 100 Hz – 1 kHz

Data display

Apart from biopotential signals such as ECG and EEG (where you are interested in the electrical signal as a voltage), few biological signals will have much meaning as a raw voltage signal and may require further modification to display the parameters of interest.

Calibration

Most transducers require calibrating before use. The calibration process converts the voltage signal into units of the quantity being measured by the transducer, such as units of pressure or temperature.

Most transducers and amplifiers are linear. This means the voltage output relative to the quantity measured is a straight line. These can be calibrated in LabChart with a 2-point 'units conversion', where the two points cover the range of interest. For example, if recording mammalian body temperature, the temperature probe could be calibrated with 35°C and 45°C. In this case, the voltage output would be measured with the probe in a solution at each temperature. These readings would be then converted to the temperature in °C (Figure 9). If the transducer or amplifier is non-linear then a multi-point calibration must be carried out which requires an additional LabChart extension.

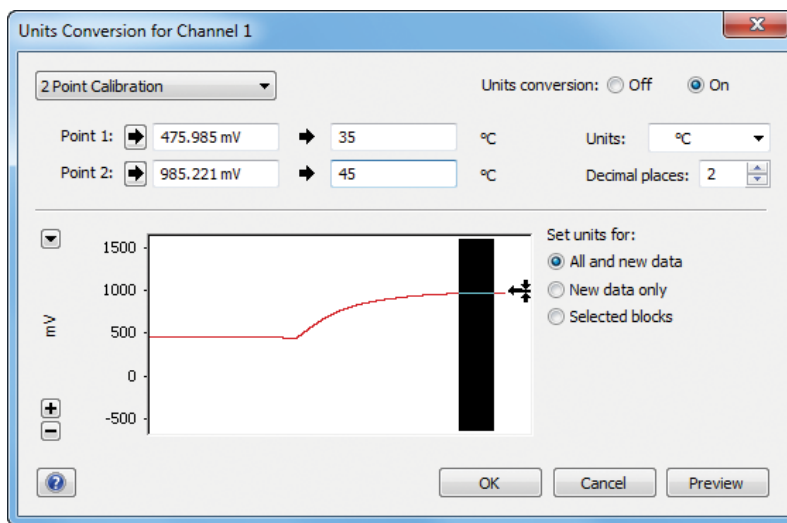


Figure 9 Two points conversion of a voltage signal into temperature.

Data calculations

Even with the data being displayed in the correct units, you may wish to display other features of the signal which may have more meaning for your study such as a heart rate, the change in pressure over time (dp/dt) or the area under the signal. Each LabChart Channel has a Channel Function drop-down menu with additional features (Figure 10).

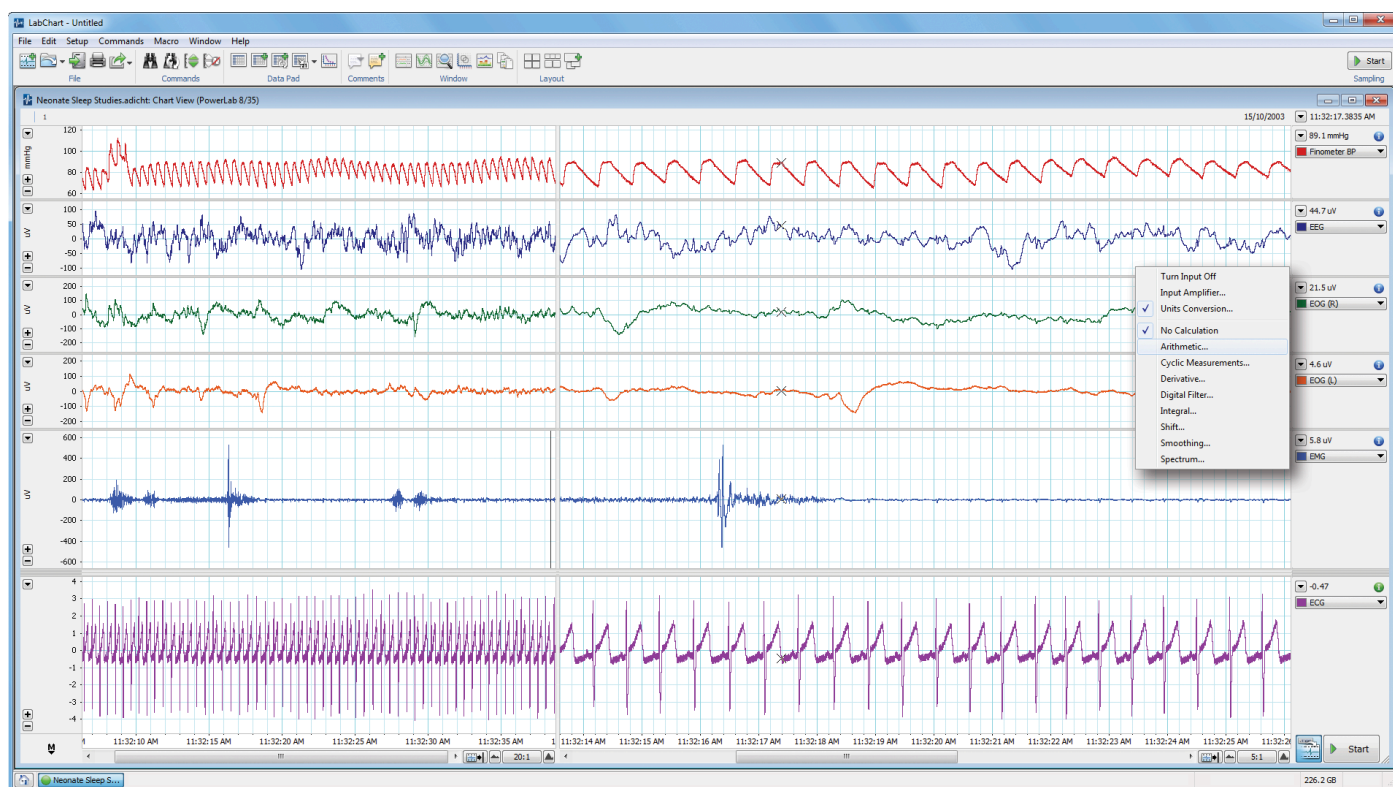


Figure 10 LabChart Channel Function drop-down menu.

Cyclic measurements

Cyclic Measurements analyse periodic waveforms, either online or offline. LabChart preprocesses the signal, detects cycles in the waveform, and then uses those detected cycles to perform various calculations on the cycles (cyclic minimum, maximum, mean, rate, period frequency etc). Choose **Cyclic Measurements...** from the Channel Function drop-down menu. The input data is selected and the cyclic function chosen. LabChart has a series of pre-set detection waveforms to allow the cycles to be determined and then the selected measurement can be displayed over the raw data or on a separate channel (Figure 11).

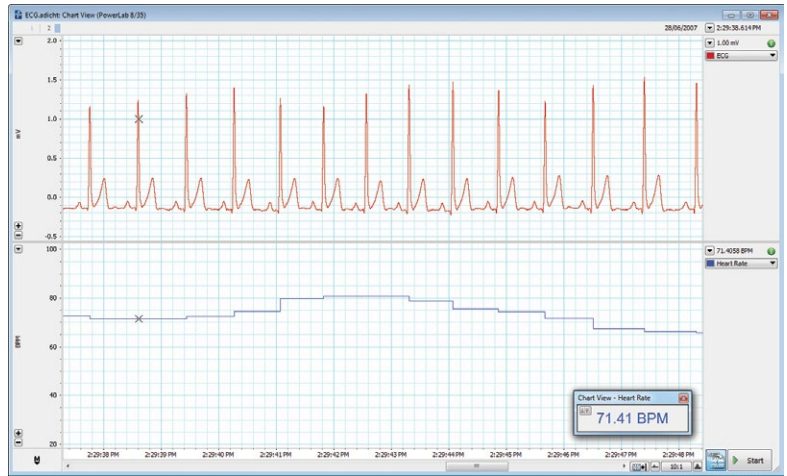
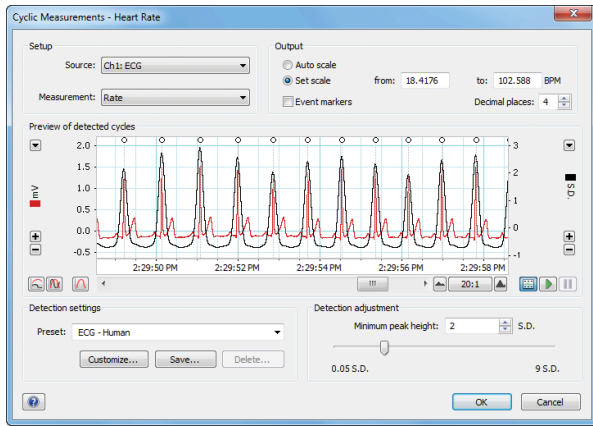


Figure 11 Cyclic Measurements set-up (left) and output display.

Arithmetic

The Arithmetic calculation allows you to arithmetically combine waveform data from different channels as well as applying an algorithm to data inputs. It works online (during recording) as well as offline.

Choose **Arithmetic...** from the Channel Function drop-down menu to open the Arithmetic dialogue (Figure 12). The channel in which results will be displayed is indicated in the dialogue title.

Expression entry box

Source data channel

Manually set the Amplitude Axis scale, or ask LabChart to do it automatically.

Enter units for the calculated channel.

- Math
- Trig
- Logs
- Noise
- Sign()
- Abs()
- Sqrt()
- Pi
- SampleTime
- Round()
- Int()
- Mod()
- Threshold()
- Window()
- Differentiate()
- Integrals
- Shift()
- Smooth()
- SmoothSec()
- MedianFilter()
- Filters

- +
-
- *
- /
- ^
- Log10()
- Ln()
- Exp()
- Integrate()
- IntegrateAbs()
- IntegratePos()
- IntegrateNeg()

- Sin()
- Cos()
- Tan()
- ASin()
- ACos()
- ATan()
- ATan2()
- RadToDeg()
- DegToRad()
- Random()
- Gaussian()
- Lowpass()
- Highpass()
- Notch()
- NarrowBandpass()
- Bandpass()
- Bandstop()
- RCLowpass()
- NanRemover()

Figure 12 Arithmetic set-up.

Integral

The integral of a waveform is equal to the area underneath it. The Integral calculation can calculate integrals with respect to time both online and offline.

A typical application where this is useful is in respiratory work where air movement is normally quantified by measuring flow with a pneumotachograph. The flow signal is then integrated to give the volume.

Choose **Integral...** from the Channel Function drop-down menu of the channel you wish to display the calculation result. In the Integral dialogue (Figure 13) for that channel, you choose the source channel, the type of integral, the resetting mode, the limits of the integral and the scale to be used. The accumulative integration can be displayed (no reset) or it can be reset by time, cycle or event.

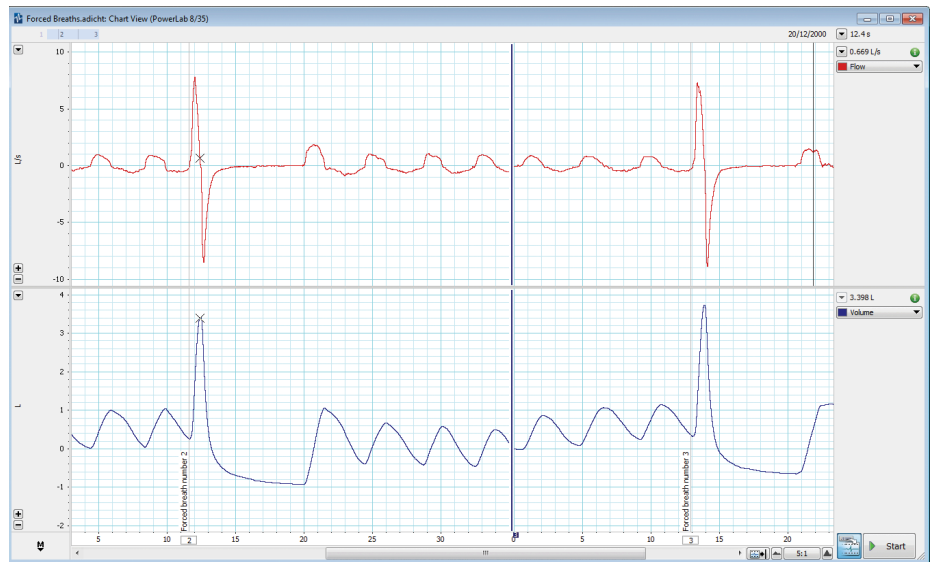
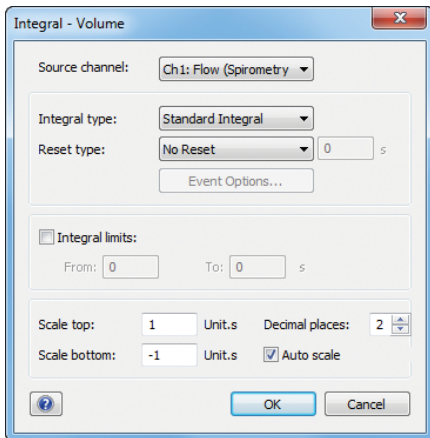


Figure 13 Integral set-up. The Flow of Channel 1 is integrated to give Volume on Channel 2.

Derivative

It is also possible to find the first or second order derivative of a signal waveform with LabChart. A typical application where this may be of use is when measuring ventricular pressure and you are interested in the rate of the pressure change (dP/dt). You can determine this using the first order derivative of the pressure signal (Figure 14). Choosing the **Derivative...** command from a Channel Function drop-down menu displays the Derivative dialogue for that channel. As with most channel calculations, you specify a source channel in the Source channel drop-down list.

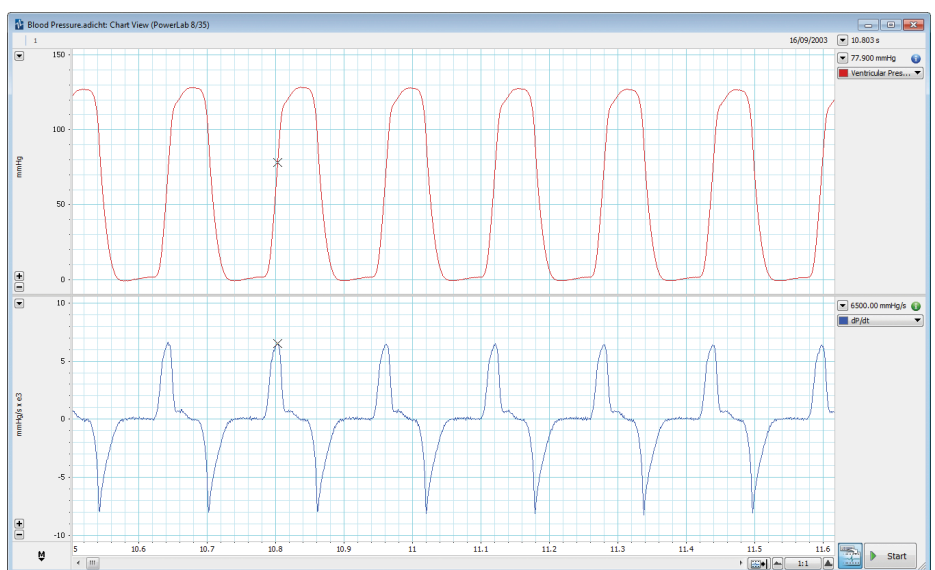
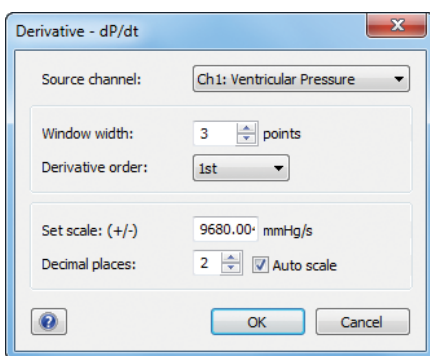


Figure 14 Derivative set-up. The dP/dt (mmHg/s) of the pressure waveform is displayed.

Summary

The bare essentials of data acquisition and display

- **Define the variables to be measured.** You will need to know the normal range and allow for any potential effects of your intervention.
- **Convert the physiological effect to a voltage.** A transducer and amplifier will be required to convert the variable into a voltage signal.
- **Choose the sampling rate.** The general rule is to sample at least twice the highest expected frequency of the incoming signal.
- **Set the range.** This allows you to record your signal with good resolution, but be careful not to set it too low or data could be irreversibly lost.
- **Apply appropriate filters.** Use filters to reduce noise or unwanted frequencies either with a hardware or digital filter. Using the latter in LabChart allows you to maintain your raw data.
- **Calibrate your transducer.** Change the voltage signal into meaningful units.
- **Set up channels for calculated data.** Channel calculations can display meaningful parameters of your raw input signal.

Advanced data acquisition and analysis - LabChart Modules

LabChart Modules are application-specific acquisition and analysis LabChart Add-Ons software.

Module	Purpose
• Blood Pressure Module	The Blood Pressure Module detects, analyzes, displays and reports a set of cardiovascular parameters from arterial or ventricular pressure signals. It can be used online or offline.
• Cardiac Output Module	The Cardiac Output Module allows for cardiac output calculations to be derived from a LabChart recording of a thermodilution curve.
• DMT Normalization Module	The DMT Normalization Module calculates the optimal pretension conditions for microvessels prior to commencing experiments using DMT wire myographs.
• Dose Response Module	The Dose Response Module generates dose response curves and calculated values such as EC50 and Hill slopes within LabChart. It can be used in online or offline modes.
• ECG Analysis Module	The ECG Analysis Module detects and examines ECG components online or offline providing statistical and graphical analysis. It features ECG averaging and is suitable for human and animal recordings.
• HRV Module	The HRV Module analyzes variability in ECG or arterial pulse recordings. A number of heart rate variability parameters, graphs and a report can be generated online or offline.
• Metabolic Module	The Metabolic Module enables real-time acquisition and online or offline analysis of human metabolic parameters such as RER, VCO ₂ , VO ₂ and VE.
• Peak Analysis Module	The Peak Analysis Module provides automatic detection and analysis of multiple signal peaks in acquired waveforms. The module can be used in online or offline modes.
• PV Loop Module	The PV Loop Module provides a variety of views, plots and tables for the analysis of left ventricular pressure and volume data in small and large mammals. The module can be used in online or offline modes.
• Spike Histogram Module	The Spike Histogram Module allows the detection, discrimination and analysis of extracellular neural spike activity online or offline.
• Video Capture Module	The Video Capture Module is used to record and synchronize a video movie with a LabChart data file. This allows simultaneous playback and correlation between data and video recorded events.

Reference

S.S Young – 'Computerised Data Acquisition and Analysis for the Life Sciences – a Hands-On Guide'. 2001.

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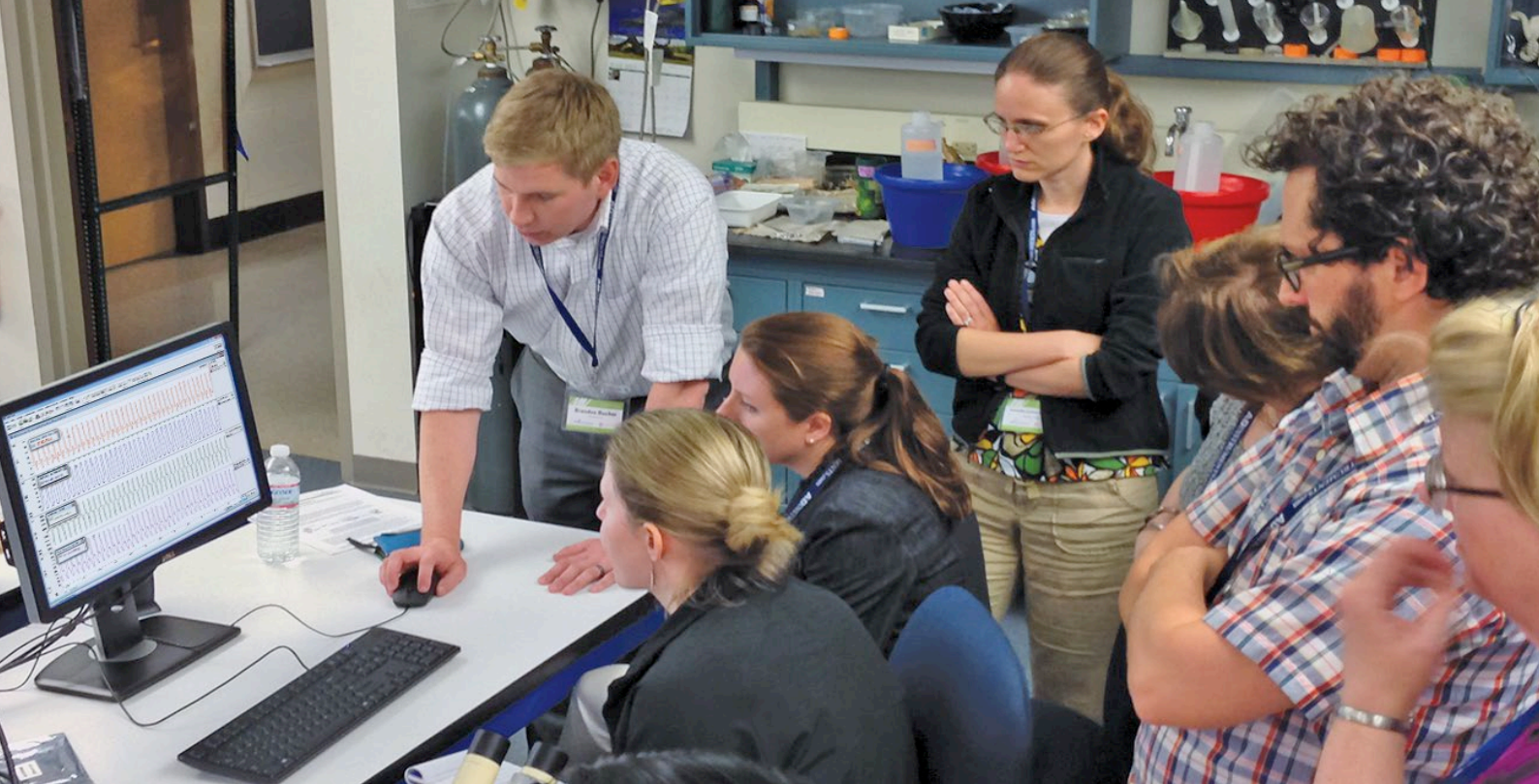
All your analysis
in one place



LabChart

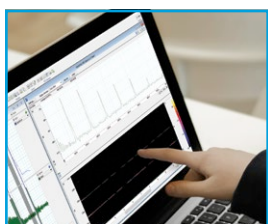
Enabling discovery

LabChart data analysis software creates a framework for all of your recording devices to work together, allowing you to acquire signals from multiple sources simultaneously and apply advanced calculations as your experiment unfolds. Even better, LabChart tracks every action you take and never modifies your raw data, ensuring the integrity of your results so you can focus on the true insights of your research.



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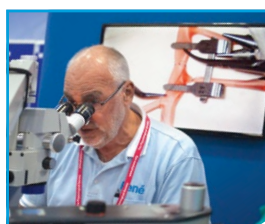
Software Training



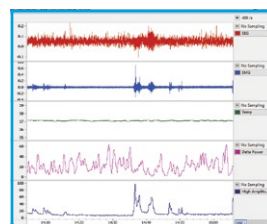
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