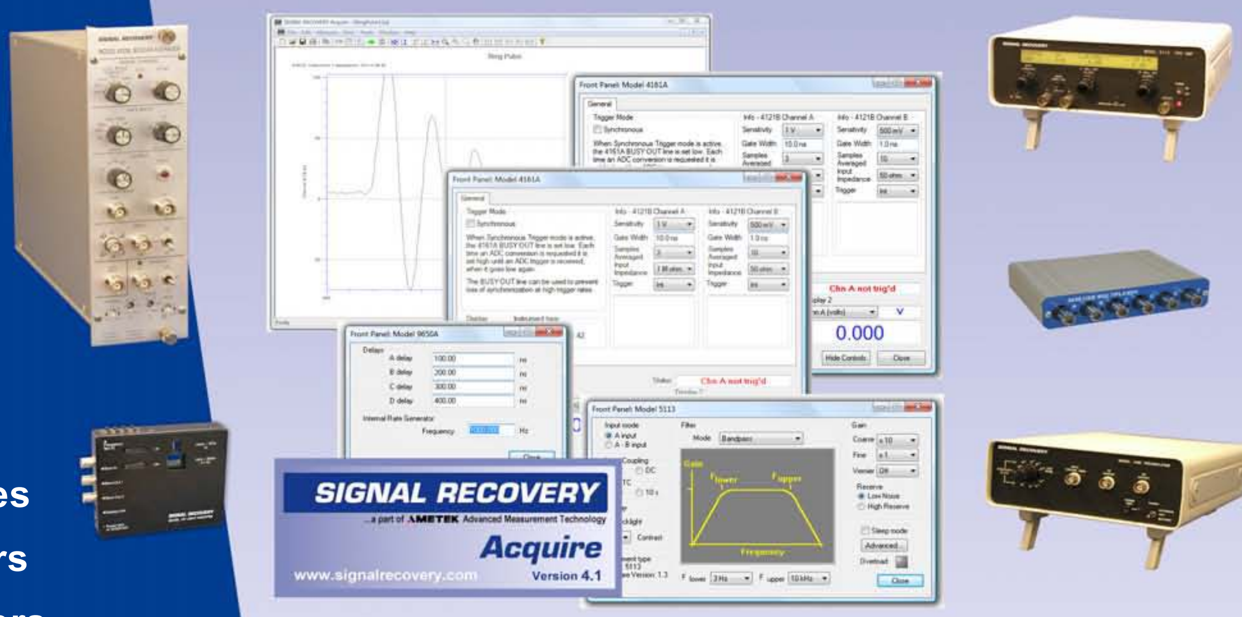
SIGNAL RECOVERY

...part of AMETEK® Advanced Measurement Technology



- Counters
- Software
- Accessories
- Multiplexers
- Preamplifiers
- Light Choppers
- Signal Averagers
- Boxcar Averagers
- Lock-in Amplifiers

Product Catalog
Issue 7

Great products from a company with a great history

ORTEC Brookdeal

PRINCETON APPLIED RESEARCH

Welcome!

Welcome to the latest issue of the catalog for signal recovery instrumentation.

We have retained the same format as previous catalogs, including not only detailed product information but also product selection guides, a model number index, ideas for applications, and our full library of Technical and Applications Notes. As such we are confident it will be a useful addition to your library.

EG&G PRINCETON APPLIED RESEARCH

A Great History...

Our most popular product, the lock-in amplifier, was invented by Princeton University physicist Robert H. Dicke who went on to found Princeton Applied Research (PAR) to develop and market the instrument. Forty years later the PAR brandname is now used only for instruments used in research electrochemistry, but we at ***SIGNAL RECOVERY***, the successor to the original company, continue to design and produce the widest commercially-available range of lock-in amplifiers.

EG&G BROOKDEAL

In those forty year we have undergone several changes of company name, from Princeton Applied Research and Brookdeal Electronics, through EG&G Signal Recovery and PerkinElmer Instruments, to ***SIGNAL RECOVERY***. Now, as part of the $2.1 billion AMETEK® group we have access to all the benefits conferred by a large corporation, while retaining a focussed company structure that allows us to continue to develop new and innovative products. We are proud of our long history and look forward with confidence to continuing meet the needs of you, our customers, for many years to come.

EG&G SIGNAL RECOVERY

Visit us at www.signalrecovery.com

Add our website to your favorites and visit it regularly so that you're always up to date with the products we offer. Download demonstration versions of our software or full LabVIEW drivers, find a copy of the instruction manual for your instrument, or request a quotation. Our online store is open for business so that you can order the products you need - from a simple RS232 null modem cable right through to our most sophisticated lock-in amplifier - whenever you want. And if you have a specific question then send us an e-mail at **info@signalrecovery.com** and we'll get right back to you.

PerkinElmer™ instruments.

Thank you

Thank you for your interest in our products. We look forward to being of service.

SIGNAL RECOVERY

...part of AMETEK® Advanced Measurement Technology

How to use this catalog

- If you know the type of instrument you need, consult the table of contents on page 3 and then turn to the start of the relevant section. When we offer several products of similar type, you will find a product selection guide that should help you narrow down your choice; finally use the detailed product description pages to optimize your selection
- If you know the model number of the instrument you are looking for then the index on page 9 will help identify where it is located, and in the case of obsolete models, suggest the nearest equivalent replacement
- If you are not sure whether any of our products will be of use to you, read the section below and consult the Applications Ideas on pages 7 and 8 before proceeding

Our range of solutions includes:

Amplifiers and Preamplifiers

If the signal you are trying to measure is just too small to be detected by your oscilloscope, voltmeter, spectrum analyzer or ADC card, then one of our preamplifiers may help. They are optimized to add the minimum amount of noise to your signal, and their use can often make possible measurements that could not previously be made.

Lock-in Amplifiers

Lock-in amplifiers use a phase sensitive detector to measure the amplitude of a single frequency of the signal applied to their input, where this frequency is defined by an external or internal reference frequency. Any signals not coherent with the reference are sharply attenuated, so in some ways they can be considered as very narrow bandpass filters with adjustable center frequencies. The output of single-phase instrument is a DC voltage or numeric display that is proportional to the magnitude of the input signal and the cosine of the phase angle between it and the reference; dual phase units, by now far the most common type, give two outputs proportional to the in-phase and quadrature components of the signal and, by derivation, its magnitude and phase with respect to the reference.

Use a lock-in amplifier if:

- The signal can be modulated or is already modulated
- The duty factor is approximately 50%
- The signal is at 2 MHz or below
- You need signal phase information
- Signal waveform information is unimportant

Boxcar Averager

The boxcar averager is a type of sample and hold module with the added ability to integrate the applied signal over a defined gate period, rather than simply sampling it at a fixed point in time. It takes one sample of the input waveform per applied trigger and the built-in output averager allows many such samples to be averaged, thereby improving the measured signal to noise ratio.

Consider a boxcar averager if:

- The signal is repetitive at a rate of up to 80 kHz
- You are interested in determining the amplitude of pulses at this repetition rate where the puse width is in the range 1 ns to 30 ms

Digital Signal Averager

If you need to know the waveform of a complex repetitive signal then a digital signal averager might be useful. In many ways this can be considered simply as a specialized form of digital oscilloscope with the ability to average repeated sweeps, but unlike most 'scopes it allows high repetition rates by using dedicated hardware to carry out the averaging. Whereas a scope might allow the experiment to run at 10 - 20 Hz, the averager can support rates of up to tens of kilohertz, getting results that much faster.

Consider a digital signal averager if:

- You need to recover the waveform of a signal, e.g. in time of flight studies in mass spectrometry
- You need to make time delay measurements, e.g. to detect transit times

Light Choppers

These electromechanical devices are used to modulate beams of light, IR or UV energy to allow their subsequent measurement using lock-in amplifiers. But since the chopping frequency at which they operate can be locked to an external electrical signal, they can also be used in many other light-switching applications.

Counter

The model 3820 Universal Counter is a sophisticated single channel counter/ timer housed in a compact module, powered and controlled from any PC with a USB port. It measures frequency, events and pulse times with high accuracy.

Multiplexer

In cases where you need to select between several different signals for input to a measuring system, the model 3830 USB Multiplexer offers a convenient and compact solution. With six BNC connectors it offers multiple switching options, all controlled via a USB interface.

Software

We offer supporting software for all our instruments that have computer interfaces in the form of LabVIEW drivers, complete applications packages and ActiveX control libraries. These significantly reduce the time needed to assemble complex computer controlled systems, giving you more time to concentrate on your research.

Contents

What's new...

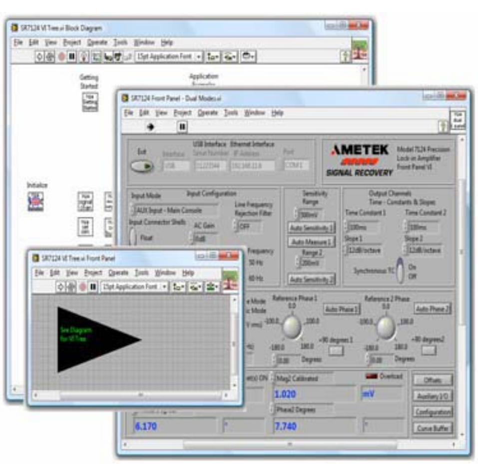
The following products have been introduced since the previous issue of this catalog. Full details can be found on the relevant pages.

Model 7270 Dual Phase DSP Lock-in Amplifier (see page 27)

The model 7270 sets a new standard for general-purpose DSP lock-in amplifiers. It includes the best features of our model 7265 and 7280 instruments, but with even better specifications. The result is a lock-in amplifier of outstanding performance and compact size that is easy to use and suitable for all measurements over a frequency range extending from 1 mHz to 250 kHz.

New Version 4 of Acquire Software (see page 56)

In addition to adding support for our new instruments, the latest version of our Acquire software package allows cursors to be placed on plot traces so that direct reading of signal output values is possible.



Updated LabVIEW Drivers; New Drivers for Models 7124, 7270, and *FASTFLIGHT-2*

Our suite of LabVIEW drivers has again been updated to support operation from LabVIEW version 8.0 or later, and we have added drivers for our latest instruments.

What’s new...

Model 7124 Precision Lock-in Amplifier
(see page 25)

The model 7124 precision lock-in amplifier uses a unique analog fiber optic link to interconnect a remote connection unit (RCU), to which the experiment is connected, and a main instrument console. Using this technique, the model 7124 overcomes one significant limitation of other lock-in amplifiers, which is that the instrument itself can act as a source of digital clock and switching noise that can be coupled back into the experiment via the signal or internal oscillator connectors. The instrument is therefore particularly suited for use in low temperature physics experiments, where the power dissipated by such noise can cause problems.

FASTFLIGHT-2 4 GSa/s Digital Signal Averager
(see page 72)

The *FASTFLIGHT-2* is a high performance digital signal averager in a compact benchtop console, offering digitizing rates of down to 250 ps per point and built-in averaging hardware that reduces the deadtime between succesive data acquisition sweeps to less than 1 µs, Instrument control and data display is via a powerful Windows software package with simple USB interconnection from console to computer.

Computer Interface Cables
(see page 61)

Our model CE0115S (GPIB) and CE0116S (RS232) adaptor cables make it simple to connect our instruments that have these interfaces to computers that are fitted only with USB expansion connectors.

Custom and OEM Solutions

We at ***SIGNAL RECOVERY*** always strive to meet our customers' needs. But sometimes one of our standard products is simply not suitable for the job, so we offer the option of supplying custom instrument solutions.

128-channel DSP Lock-in Amplifier **(see page 35)**

Projects have ranged from several complete 128-channel DSP lock-in amplifiers and operating software supplied to customers in the Far East through to very small tasks to supply accessories to allow our standard products to be used for a specific experiment.

For example, a customer using the 7280 in dual reference mode required a TTL signal at the oscillator frequency. This is not normally possible in this mode, since the REF MON output is at the external reference frequency. Consequently we built a simple comparator circuit into a compact box, powered by the lock-in's preamp power output, which exactly met his needs.

Comparator for use with Model 7280 Lock-in Amplifier

For another customer, we assembled a system to measure the ground (earth) impedance of safety ground connections at electricity supply substations, using two of our model 7265 lock-in amplifiers and a custom power amplifier in ruggedized cases. The system injects an AC current of between 20 Hz and 150 Hz between a remote test probe and the substation ground, while the voltage between the ground and an intermediate probe is recorded at several positions between the remote probe and the ground. One of the lock-in amplifiers measures the test current via a current transformer, and the second lock-in the detected voltage. By measuring at frequencies both below and above the nominal line frequency, it is possible to determine by interpolation what the impedance is at the line frequency, and plot this as a function of distance from the substation.

Earth Impedance Test Set

We are also happy to discuss the use of our products with OEMs (original equipment manufacturers) and can supply instruments in custom liveries and if necessary fitted with specialized firmware and/or hardware. If you are such an OEM and would like to investigate reducing your time-to-market and development costs then please do contact us at:

info@signalrecovery.com

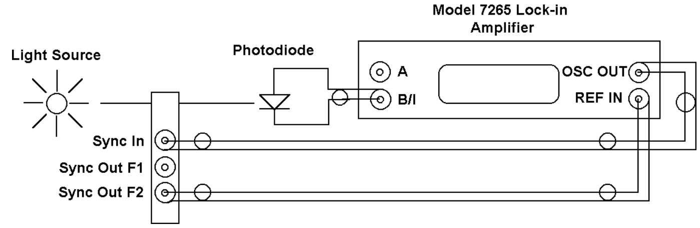
- you've nothing to lose and potentially a great deal to gain.

Application Ideas

Introduction

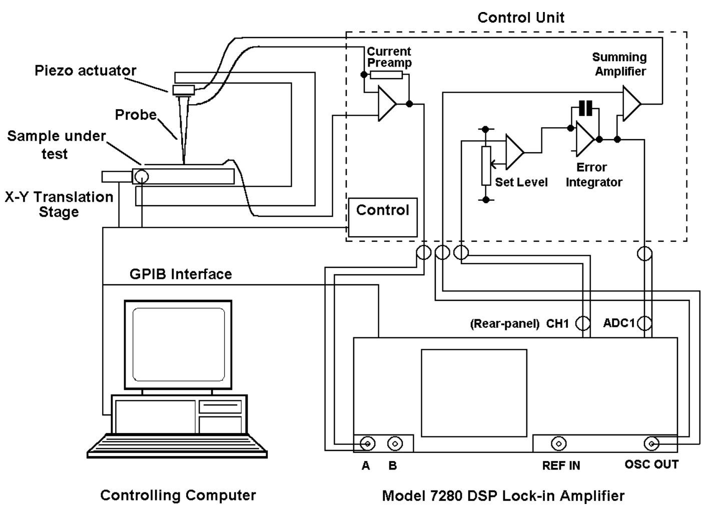
On this and the following page we present a few examples of the required connections for common situations in which our products are used. Further application ideas can be found in the Applications Notes at the end of this catalog (page 121 onwards).

Computerized Frequency Control for Light Choppers



Connect the OSC OUT output of any of our lock-in amplifiers to the SYNC IN input of our light choppers and use it to control the chopper frequency. Because ***SIGNAL RECOVERY*** lock-ins use an independent oscillator, you can even do this when driving the chopper at F but using it to modulate the signal at F/10 simply by coupling the SYNC OUT F/10 output back to the REF IN external reference input, and setting the instrument to external reference mode.

Scanned Probe Microscopy

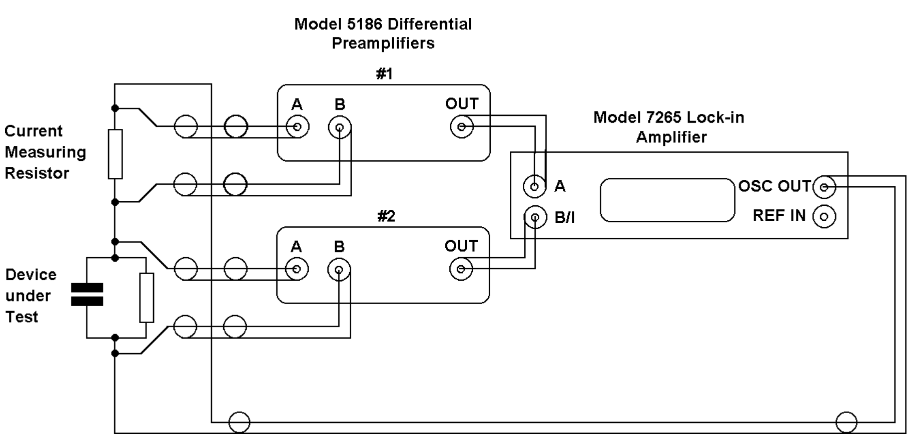


The probe tip vibrates vertically at a frequency set by the lock-in amplifier's oscillator about a mean position set by the output of the error integrator. If the probe gets closer to the sample surface, then the tunnelling current increases and the lock-in amplifier gives a larger output voltage. This in turn causes a change to the error integrator input, resulting in the probe being moved away from the surface. Hence the control loop continuously adjusts the probe position to keep its mean distance above the surface constant.

A controlling computer drives an X-Y translation stage and the mean DC level from the error integrator is digitized and recorded at each measurement point, thereby deriving a map of the sample surface.

Using the model 7280 or 7280BFP allows higher oscillation frequencies and shorter time constants to give faster time per point and hence faster scans than is possible when using traditional 100 kHz lock-ins.

Impedance Measuring System

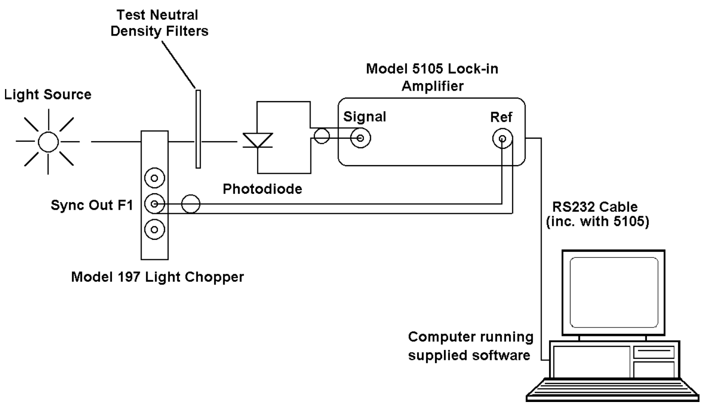


The current flowing through the device under test is measured by detecting the voltage generated across a known current measuring resistor, using a differential amplifier to eliminate ground loops. The voltage generated across the device under test is measured in a similar way.

Using the 7265's switchable A and -B input modes under computer control, both the current through and voltage across the sample can then be sequentially recorded at different frequencies, with the computer's software calculating and storing the corresponding values of complex impedance.

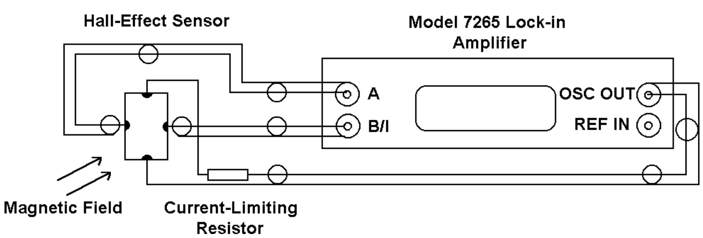
Application Ideas

Lock-in Amplifier Educational Demonstration



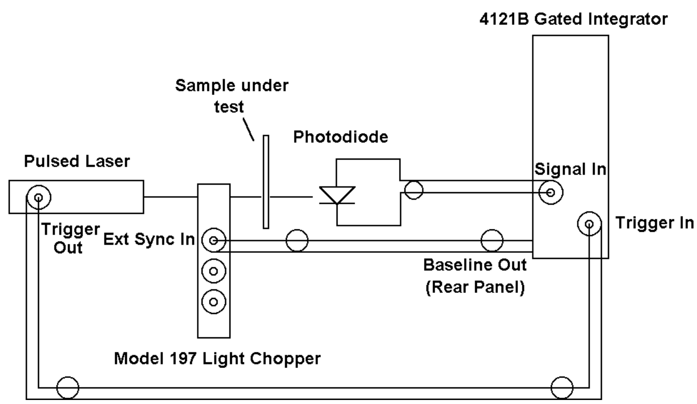
This system can be used to show the advantages of lock-in detection. Use a series of different neutral density filters and record the signal magnitude, plotting magnitude versus optical density. Now repeat but with the chopper removed and the photodiode connected to a microammeter, and note the difficulty in obtaining satisfactory results.

Measuring Hall Effect Voltages



Connect the OSC OUT signal to the Hall-effect device, using a series resistor to limit the current to a suitable value. Set the lock-in to internal reference mode and configure the input as a differential (A-B) stage. With no magnetic field the lock-in magnitude output display should be zero, but if not it may be zeroed using the auto-offset function. Once this is done then the magnitude display is proportional to the magnetic field.

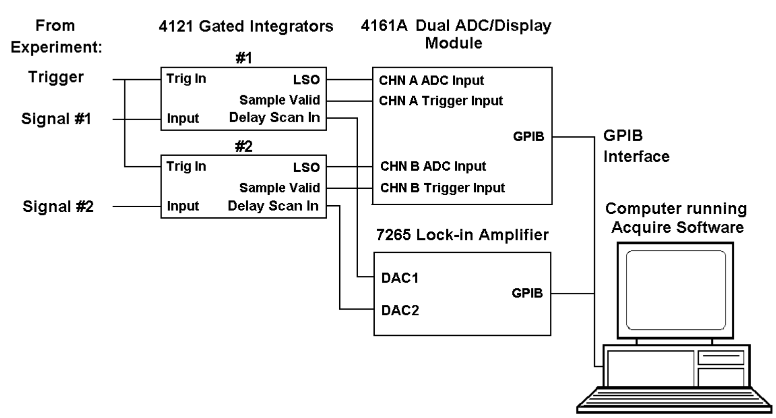
Single Channel Boxcar System with Automatic Baseline Subtraction



Use the Baseline Out output of the 4121B Gated Integrator to drive the EXT SYNC reference input of the 197 or 650-series light choppers for automatic baseline subtraction. With a 1 kHz laser rate the chopper will run at 500 Hz and alternate laser pulses will be prevented from reaching the detector. The averager will then average the difference between the “light” and “dark” signals, compensating for any trigger-coherent pick-up.

Two-Channel Computer Controlled Boxcar Averager System

Apply the system trigger to the trigger inputs of two model 4121B gated integrators, and connect the signals to be measured directly to the integrators’ signal inputs (50 Ω or 1 MΩ). Connect each integrator’s Last Sample Output (LSO) and Sample Valid outputs to the 4161A dual channel ADC. Use the auxiliary DAC outputs on a 7265 lock-in amplifier as programmable voltage sources to adjust the gate delays on the 4121B trigger inputs.



The system is completed with GPIB connections to a controlling computer fitted with a USB-GPIB interface adaptor (part number CE0115S) running the ***SIGNAL RECOVERY*** Acquire software package.

Model Number Index

Preamplifier Selection Guide

Choosing the right ***SIGNAL RECOVERY*** preamplifier is not difficult if you follow this simple guide.
In case of any doubt, simply contact us for further advice

First, identify the frequency or range of frequencies you need to amplify:

- If these exceed 1 MHz then consider the **model 5185**

  If they are below 1 MHz then next decide whether you are amplifying a current signal, for example from a photodiode or ion collector, a voltage signal, for example the voltage generated by a hall-effect sensor, or an optical signal in the range 400 nm to 1650 nm

  - If you want to amplify current signals, consider the **models 181 or 5182**. The former has slightly better performance and can apply a bias voltage to the detector, but the latter includes provision for battery power

    If you want to amplify voltages, then you need to decide whether a single-ended connection to the experiment is acceptable, or whether you need the better common mode rejection offered by a differential input

    If you can work with a single-ended input then use the:
    - **model 5184** if you want the lowest noise but can tolerate an input impedance of 5 MΩ
    - **model 1900** transformer in conjunction with another voltage preamplifier when you are working from very low source impedances and need the lowest possible noise

    If you need true differential input capability, then use the
    - **model 5186**

    - For the ultimate flexibility, consider the **model 5113** that has the ability to work in both single-ended and differential modes, and in addition includes adjustable gain and signal channel filters

Model 5113

Low-Noise Voltage Preamplifier

FEATURES

- **Low-Noise**
- **Single-ended or Differential input modes**
- **DC to 1 MHz frequency response**
- **Optional low-pass, bandpass or high-pass signal channel filtering**
- **“Sleep” mode to eliminate digital noise**
- **Optically-isolated RS232 control interface**
- **Battery or line power**

APPLICATIONS

- **Acoustic research**
- **Radio astronomy**
- **AC bridge measurements**
- **Oscilloscope preamplification**
- **Hall-effect signal amplification**

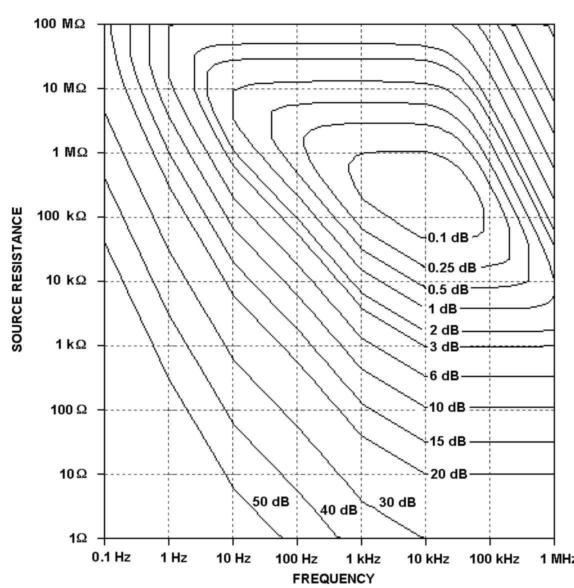
DESCRIPTION

The model 5113 is a high performance, low noise voltage preamplifier with continuously adjustable gain and selectable high, low or bandpass filtering. Its input can be configured for either single-ended or true differential operation with either DC or AC coupling, and its output will deliver up to 1 V pk-pk into a 50 Ω load.

All the principal instrument controls are operated via the three front-panel rotary knobs with a back-lit LCD display to show their present settings. The instrument also includes an optically isolated bi-directional RS232 interface allowing remote operation and interrogation of all controls. Since in some experiments even the very low levels of noise introduced by the internal microprocessor that supports these capabilities may cause problems, the unit includes a "sleep" function whereby every source of digital noise is turned off after a predetermined interval. When in the sleep mode the preamplifier "wakes up" as soon as any control is adjusted and goes back to sleep when adjustment is complete.

The instrument can either be continuously line-powered from the model PS0108 power supply supplied with it, or be run from the internal rechargeable batteries which are charged whenever the power supply is connected. Battery operation often allows troublesome line frequency pick-up to be eliminated, as well as permitting operation away from a source of line power.

If the signal of interest is limited to a single frequency or narrow range of frequencies then the filters allow selective signal amplification, making subsequent signal measurement, for example on an oscilloscope or a lock-in amplifier, easier. The filters can of course be switched out of use to give a flat frequency response.



Noise Figure Contours (Typical)
Gain = x1000, AC Coupling, 10 s coupling time-constant, Flat filter mode

The model 5113 will be of use in applications as diverse as radio astronomy, audiometry, test and measurement, process control and general purpose signal amplification as well as being ideally suited to work with our range of lock-in amplifiers.

Specifications

General

DC or AC coupled voltage amplifier with adjustable gain and a maximum frequency response extending from DC to 1 MHz. Single-ended or differential high-impedance input, and single-ended output, via BNC connectors.

Signal channel high and low pass filters with variable cut-off frequencies and slope may be switched into circuit to give an overall low-pass, high-pass, bandpass or flat response.

Computer control via optically isolated RS232 interface.

Battery powered from internal rechargeable batteries, which recharge when separate line power supply is connected.

Inputs

Modes	A or A-B
Coupling	AC or DC
Impedance	
AC coupled	either 10 MΩ or 100 MΩ in parallel with 25 pF and in series with 0.1 µF
DC coupled	either 10 MΩ or 100 MΩ in parallel with 25 pF
Max Input without Damage	
DC coupled	+10 V, -9 V
AC coupled	Coupling capacitors can withstand 100 V. Transients that pass through coupling capacitors must not exceed DC coupled operation limits
Max Input for Linear Operation	
Common mode	1 V peak.
Differential mode	See Table 1

Coarse Gain	Max Peak Input	
	Low Filter Reserve	High Filter Reserve
5 to 25	1 V	1 V
50 to 500	100 mV	1 V
1000 to 5000	10 mV	100 mV
10000 to 50000	10 mV	10 mV

Table 1. Maximum Input as a function of Filter Reserve and Coarse Gain Setting

Common Mode Rejection Ratio, C.M.R.R.	
DC to 1 kHz	>120 dB
1 kHz to 1 MHz	-6 dB/octave
Gain	Coarse gain of ×5 to ×50,000 in 1-2-5 sequence with an accuracy of 1%. Fine gain extends range from ×1 to ×100,000 with an accuracy of 2%. An uncalibrated vernier provides gain adjustment of +20% of coarse gain
Overload Recovery	Front-panel push button or computer command
Voltage Noise	Typically 4 nV/√Hz at 1 kHz referred to input - see also noise contours on page 11

Filters

Type	One high-pass and one low-pass stage
Mode	Low-pass, High-pass, Bandpass, Flat (No filter)
Slope	
Low pass	6 or 12 dB/octave
High pass	6 or 12 dB/octave
Bandpass	6 dB/octave
Frequency Response	
Flat mode	DC to 1 MHz.
Low-pass	-3 dB frequency selectable from 0.03 Hz to 300 kHz in a 1-3-10 sequence (Figure 1)
High-pass	-3 dB frequency selectable from 0.03 Hz to 300 kHz in a 1-3-10 sequence (Figure 2)

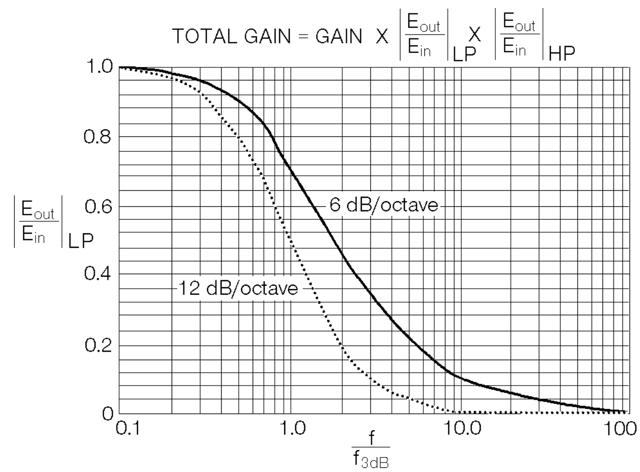


Figure 1, Low-Pass Filter Amplitude vs. Normalized Freq. Response

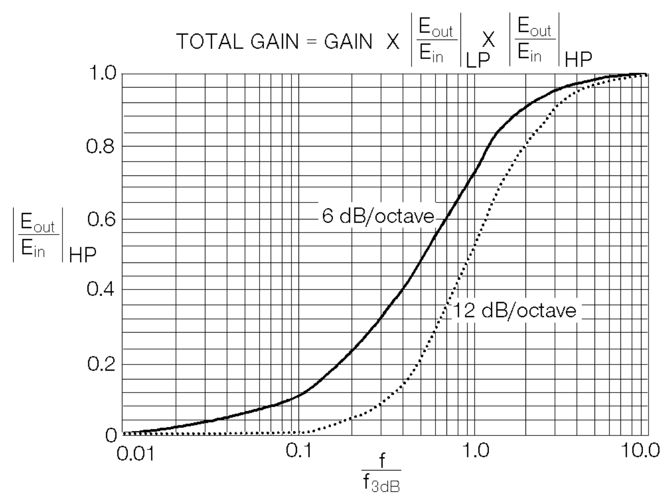


Figure 2, High-Pass Filter Amplitude vs. Normalized Freq. Response

DC Drift	
Referred to Input (DC coupling)	Maximum 10 µV/° C or less than 10 µV per 24 hours at constant ambient temperature
Referred to Output (AC coupling)	
Coarse gain only	75 µV/° C
With Fine Gain	250 µV/° C maximum
DC Input Offset control	Front-panel screwdriver control provides for DC zeroing

Output

Max Output Voltage	2 V pk-pk ahead of 50 Ω
Output Impedance	50 Ω ± 2%

Computer Interface

Type	Opto-isolated RS232
Connector	DB25 25-pin female connector
Baud Rate	300 to 9600 baud
Parameters	No parity, eight data bits and one stop bit

General
Power Requirements
Internal sealed maintenance-free rechargeable lead-acid batteries provide approximately 30 hours operation between charges. An LCD display page provides information on their state of charge

External Power Supply Model PS0108

Input Voltage	110/120/220/240 V AC
Frequency	50-60 Hz
Input Connector	IEC line input; matching power cord supplied
Output Voltage	± 18 V DC nominal, unregulated
Output Connector	DIN 5-pin 180° plug

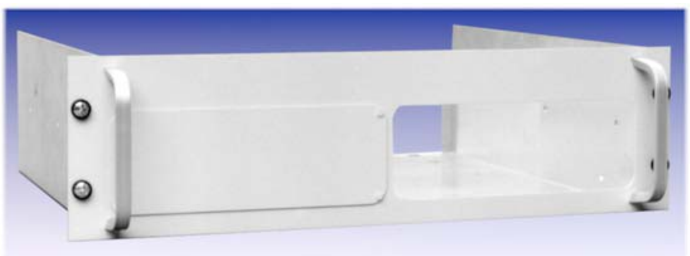
Dimensions

Model 5113		
	Width	8.25" (210 mm)
	Depth	11" (279 mm)
	Height	3.5" (89 mm)
External Power Supply Model PS0108		
	Width	3" (77 mm)
	Depth	5.3" (135 mm)
	Height	2.4" (61 mm)

Weight

Model 5113	8 lbs. (3.7 kg)
External Power Supply	2.2 lbs. (1.0 kg)

Accessories
One or two model 5113's and their associated power supplies may be rack mounted in the model K0304 rack mounting kit.



Model K0304 Rack Mount Kit
for one or two Model 5113 Preamplifiers

The Model 1900 input transformer can increase the 5113's gain by a factor of 100 or 1000 and reduce the noise referred to the input down to a minimum of 0.03 nV/√Hz.

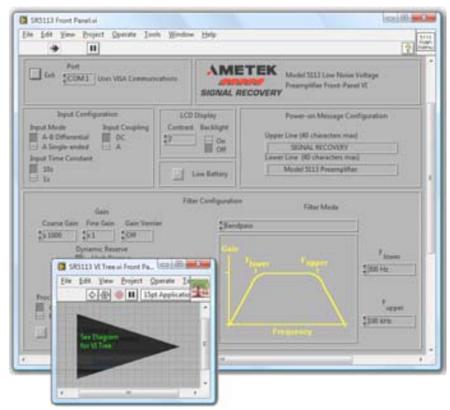
Model 1900 Signal Transformer
(see page 22)

External Line Power Supply
Model PS0108
included with each model 5113

LabVIEW Driver Software

A LabVIEW driver for the model 5113 is available from the **www.signalrecovery.com** website, offering example VIs for all the controls, as well as the usual Getting Started and Utility VIs. It also includes an example soft-front panel built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.

LabVIEW Driver for Model 5113



Why should you choose *SIGNAL RECOVERY* products?

Model 5113 Voltage Preamplifier

SIGNAL RECOVERY **Product Features**	**Benefit to you**
◆ No digital noise when in sleep mode	Digital noise cannot exist when processor is turned off
◆ Unit wakes up as soon as a control setting is change	Easy to change settings
◆ Gain is defined by switches and relays rather than by a cheaper multiplying DAC, as used in competing instruments	Bandwidth remains stable even as gain is changed, so gain changes do not change the shape of the signal being measured as happens in units using a multiplying DAC
◆ RS232 control is bidirectional	Programs can check that settings are correct and can even allow for manual interaction
◆ Excellent LabVIEW driver available	Saves programming time
◆ RS232 Interface is opto-isolated	Removes one potential ground-loop, reducing line frequency pick-up
◆ Rotary knobs allow a wider range of filter settings	Better selection of the wanted signal

Model 518X

Series Preamplifiers

This series of preamplifiers offers convenient bench top instruments with a range of specifications to suit a wide variety of applications. All models in the series can be powered from internally housed (alkaline) batteries (except the model 5185), external low voltage supplies (±15 V or ±18 V) or via the optional line power supply module, model PS0108. Nickel-cadmium rechargeable batteries can also be used, but give reduced operating time and must be recharged in an external charger. In addition, all of the preamplifiers in this range, with the exception of the 5185, can be directly powered from ***SIGNAL RECOVERY*** Lock-in Amplifiers (other than the models 5105 and 5106). Their low noise performance makes them ideally suited for signal recovery applications.

The models 5182, 5184 and 5186 preamplifiers have an output impedance of 450 Ω which when connected to a 50 Ω load creates a convenient 10:1 signal attenuation.

A rack mount kit, model K0304, is available which will accommodate one or two instruments, including their associated power supplies if required (see page 20).

Model 5182

Current Preamplifier

FEATURES

- **Low input impedance**
- **Low noise**
- **Single-ended virtual ground input**
- **Adjustable sensitivity**
- **Bias current monitor (DC) and signal (AC) outputs**
- **DC to 1 MHz frequency response**
- **Battery or external DC power**

APPLICATIONS

- **Photodiode amplification**
- **Photomultiplier amplification**
- **Ion collector amplification**
- **Electron multiplier amplification**

DESCRIPTION

The model 5182 is a current-to-voltage preamplifier of low noise and low input impedance designed to amplify the extremely low currents encountered in such areas as photometry and semiconductor research. It has four standard sensitivity settings but in addition includes a special low-noise mode on the highest gain position for even better low current measurement capability. The unit features two outputs, allowing both the AC and DC components of the input signal to be independently monitored, so that, for example, in a PMT application the bias current can be measured separately from the signal current.

It can be powered from its own internally housed (alkaline) batteries, an external low voltage supply (±15 V or ±18 V) or from the model PS0108 remote line power supply (optional extra). This preamplifier can also be powered from most of our range of lock-in amplifiers.

The model 5182 is ideally suited to amplifying signals from current sources such as electron multipliers, ion collectors, photomultipliers and photodiodes.

Specifications

General

DC coupled current to voltage amplifier with adjustable sensitivity and a maximum frequency response extending from DC to 1 MHz. Single-ended virtual ground input and single-ended DC and AC coupled outputs via BNC connectors.

Battery powered from internal alkaline batteries or external DC power supply.

Inputs

Modes	Single-ended virtual ground
Coupling	DC
Sensitivity	Switch selectable (5 settings)
AC Output	10^{-5}, 10^{-6}, 10^{-7}, 10^{-8}, 10^{-8} low noise A/V
DC Output	10^{-3}, 10^{-4}, 10^{-6}, 10^{-7}, 10^{-8} A/V
Accuracy	±2%
Stability	±300 ppm/° C
Impedance	see Figure 1

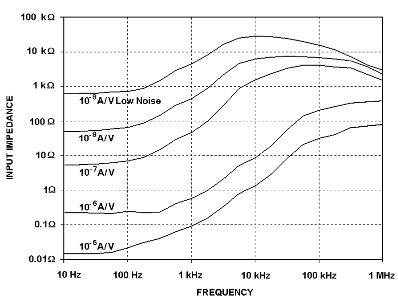


Figure 1, Input Impedance vs. Frequency and Sensitivity

Max input w/o damage	±15 V DC or 10 V rms. AC @ 50 Hz
Noise	see Table 1

Gain A/V	Max DC Input Current	Noise Current at 1 kHz
10^{-5}	9 mA	10 pA/√Hz
10^{-6}	900 µA	5 pA/√Hz
10^{-7}	9 µA	135 fA/√Hz
10^{-8}	900 nA	45 fA/√Hz
10^{-8}, low noise	90 nA	15 fA/√Hz

Table 1, Max DC Current and Noise Current vs. Sensitivity

Frequency Response (AC Output)	lower limit 0.5 Hz upper limit depends on sensitivity setting, see Figure 2
Max DC current at input	see Table 1

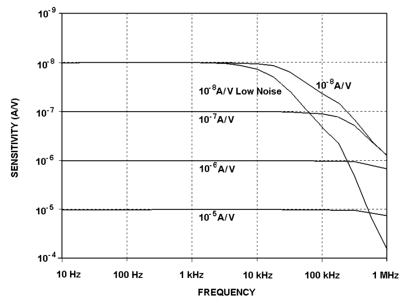


Figure 2, Frequency Response vs. Sensitivity

Outputs

AC Output	
Impedance	450 Ω
Max voltage swing	> 10 V pk-pk
Slew rate	> 22 V/µs
DC Output	
Impedance	10 kΩ
Max voltage swing	> ±9 V
Polarity	Current flowing into the input produces positive output voltage

Power

Internal	Four 9 V alkaline batteries provide approximately 15 hours of use
External	
a)	±15 V or ±18 V DC @ 25 mA
b)	110 V AC or 240 V AC via optional external model PS0108 power supply

Dimensions

(excluding connectors)	8.25" wide x 11" deep x 3.5" high (210 mm wide x 279 mm deep x 89 mm high)
Weight	5.3 lbs. (2.4 kg) excluding optional power supply

Why should you choose *SIGNAL RECOVERY* products?

Models 181, 5182, 5184, and 5186 Preamplifiers

SIGNAL RECOVERY **Product Features**	**Benefit to you**
• Wide choice of units	No need to compromise on specifications. Gives best match to the signal of interest
• Battery, line power (via PS0108) or power from one of our other instruments	Battery operation usually gives the lowest noise
• Can be used not only with our units but also with oscilloscopes, ADC cards and instruments from other suppliers - anywhere that a low-noise, high performance gain stage is required	Preamplifiers are useful general laboratory instruments
• Model 181 can apply an adjustable bias to a detector	Avoid building your own biasing network and eliminate the need to replace batteries

Model 5184

Ultra Low Noise Preamplifier

FEATURES

- **Medium input impedance**
- **Ultra low noise**
- **Pseudo-differential input**
- **Fixed ×1000 gain**
- **0.5 Hz to 1 MHz frequency response**
- **Battery or external DC power**

APPLICATIONS

- **Cryogenic detector amplification**
- **IR detector amplification**
- **Increasing oscilloscope sensitivity**

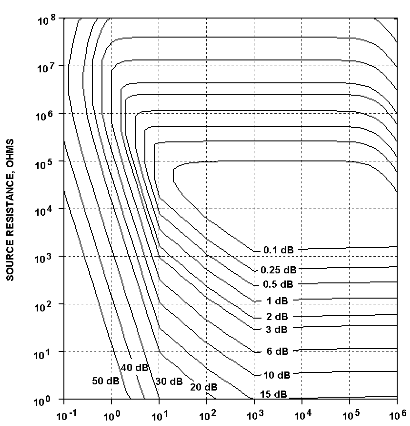


Figure 1, Model 5184 Noise Figure Contours (Typical)

DESCRIPTION

The model 5184 is a medium input impedance, AC-coupled, voltage preamplifier which features an ultra low-noise input stage. It has a frequency response from 0.5 Hz to 1 MHz and a fixed gain of x1000 (60 dB) and incorporates a special pseudo-differential input stage that can be floated to give the ground loop immunity normally associated with true differential inputs but without the associated noise penalty. It can be powered from its own internally housed (alkaline) batteries, an external low voltage supply (±15 V or ±18 V) or from the model PS0108 remote line power supply (optional extra). This preamplifier can also be powered from most of our range of lock-in amplifiers.

The model 5184 is ideal for use with medium impedance cryogenic sources and IR detectors, such as HgCdTe, InSb and InAs.

Specifications

General

AC coupled voltage amplifier with fixed x1000 (60dB) voltage gain and a maximum frequency response extending from 0.5 Hz to 1 MHz. Pseudo-differential input and single-ended output via BNC connectors.

Battery powered from internal alkaline batteries or external DC power supplies.

Inputs

Modes	Asymmetrical differential. Front panel ground terminal provided.
Coupling	AC
Impedance	5 MΩ //50 pF
Frequency Response	0.5 Hz - 1 MHz
C.M.R.R.	> 80 dB (100 Hz to 1 kHz)
Max differential input voltage	10 mV pk-pk
Max common-mode input voltage	300 mV pk-pk
Max signal low potential w.r.t. ground terminal	±600 mV
Max input without damage	±15 V DC or 10 V rms. AC @ 50 Hz
Noise	See Figure 1; typ 800 pV/√Hz @ 1 kHz
Gain	x1000 (60 dB) fixed
Gain Accuracy	±1%
Gain Stability	±800 ppm/° C

Output

Impedance	450 Ω
Max voltage swing	>10 V pk-pk
Slew rate	> 22 V/µs
Polarity	Non-inverting
Distortion	< 0.1% T.H.D.

Power

Internal	Four 9 V alkaline batteries provide approximately 8 hours of use
External	
a)	±15 V or ±18 V DC @ 40 mA
b)	110 V AC or 240 V AC via optional external model PS0108 power supply

Dimensions

(excluding connectors)	8.25" wide x 11" deep x 3.5" high (210 mm wide x 279 mm deep x 89 mm high)
Weight	5.3 lbs. (2.4 kg) excluding power supply

Model 5185

Wideband Preamplifier

FEATURES

- **50 Ω or 1 MΩ input impedance**
- **Low noise**
- **×10 or ×100 gain**
- **DC to > 200 MHz frequency response**
- **DC offset control**
- **Line power**

APPLICATIONS

- **Signal averager preamplification**
- **Boxcar averager preamplification**
- **Increasing sensitivity of oscilloscopes and fast ADC**

DESCRIPTION

The model 5185 is a wideband voltage preamplifier with a frequency response from DC to 200 MHz and switchable gain settings of x10 (20 dB) or x100 (40 dB). It has a selectable input impedance of 50 Ω or 1 MΩ and a DC offset facility.

The 50 Ω frequency response extends from DC to 200 MHz with an equivalent input noise of 10 nV/√Hz at 10 kHz. The 1 MΩ response exceeds 100 MHz, has switch selected AC or DC coupling and an equivalent input noise of 30 nV/√Hz at 10 kHz. A ground switch allows the input signal to be isolated from the output and an adjustable offset facility allows a DC offset on the input signal to be subtracted before it reaches the amplifier output. An overload detector is also provided.

The unit is powered from an external line power supply module, model PS0108, included with each instrument. Signal connections are made via the front-panel BNC connectors.

The model 5185 will prove invaluable for users who need a compact, low cost, high performance wideband preamplifier. It is an ideal accessory for use with oscilloscopes, digitizers, signal averagers and boxcar averager systems.

Specifications

General

DC coupled wideband voltage amplifier with selectable x10 (20dB) or x100 (40dB) voltage gain and a maximum frequency response extending from DC to > 200 MHz. Single-ended input and single-ended output via BNC connectors.

Line powered from model PS0108 power supply included with each unit.

Inputs

Configuration	Single-ended. Front panel ground terminal provided
Coupling	
50 Ω Input	DC only
1 MΩ Input	DC or AC
Impedance	50 Ω or 1 MΩ // 25 pF
Frequency Response	
50 Ω Input	DC to 200 MHz (±1 dB) DC to 250 MHz (+1 to -3 dB)
1 MΩ Input DC	DC to 100 MHz (±1 dB) DC to 125 MHz (+1 to -3 dB)
1 MΩ Input AC	5 Hz to 100 MHz (±1 dB) 5 Hz to 125 MHz (+1 to -3 dB)
Equivalent input noise, rms.	
50 Ω Input	10 nV/√Hz @ 10 kHz
1 MΩ Input	30 nV/√Hz @ 10 kHz
Rise and Fall Times	
50 Ω Input	< 2 ns
1 MΩ Input	< 2.6 ns
Max input voltage	
x10 gain	100 mV pk-pk
x100 gain	10 mV pk-pk
Gain	x10 (20 dB) or x100 (40 dB)
Gain Accuracy	±3% at 10 kHz
Gain Stability	±250 ppm/°C

Output

Impedance	50 Ω
Max voltage swing	>1 V pk-pk
Slew rate	> 2000 V/µs (unloaded)
Polarity	Non-inverting
DC Stability	100 µV//°C (referred to input)
DC Offset Control Range	± 10 mV (referred to input)

Power

a)	±15 V or ±18 V DC @ 300 mA
b)	110 V AC or 240 V AC via external model PS0108 power supply included with unit

Dimensions

(excluding connectors)	8.25" wide x 11" deep x 3.5" high (210 mm wide x 279 mm deep x 89 mm high)
Weight	6.4lbs (2.9 kg) excluding power supply

Model 5186

Differential Voltage Preamplifier

FEATURES

- High input impedance
- Low noise
- True differential input
- Adjustable gain
- 0.5 Hz to 1 MHz frequency response
- Battery or external DC power

APPLICATIONS

- Acoustic research
- Radio astronomy
- AC bridge measurements
- Oscilloscope preamplification
- Hall-effect signal amplification

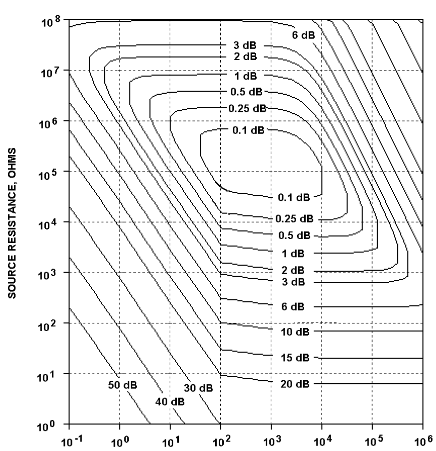


Figure 1, Model 5186 Noise Figure Contours (Typical)

DESCRIPTION

The model 5186 is a high input impedance, low-noise, AC-coupled voltage preamplifier which offers a true differential input. It has a frequency response from 0.5 Hz to 1 MHz and three switched gain settings of ×10, ×100 and ×1000. It is a general purpose preamplifier which has the facility to be connected to grounded sources in a manner which breaks ground loops and since it has a true differential input it can be used to measure floating sources, such as the output from an AC bridge, without imposing an asymmetrical load onto the source. It can be powered from its own internally housed (alkaline) batteries, an external low voltage supply (±15 V or ±18 V) or from the model PS0108 remote line power supply (optional extra). This preamplifier can also be powered from most of our range of lock-in amplifiers.

Specifications

General

AC coupled voltage amplifier with adjustable voltage gain and a maximum frequency response extending from 0.5 Hz to 1 MHz. True differential input and single-ended output via BNC connectors.

Battery powered from internal alkaline batteries or external DC power supplies.

Inputs

Modes	True differential
Coupling	AC
Impedance	100 MΩ // 20 pF
Frequency Response	0.5 Hz to 1 MHz
C.M.R.R.	
x1000 gain	> 110 dB (100 Hz to 1 kHz), degrading by 6 dB/octave above 1 kHz
x10 or x100 gain	> 90 dB (100 Hz to 1 kHz), degrading by 6 dB/octave above 1 kHz
Max common-mode input voltage, x1000 gain	5 V pk-pk
Max input without damage	±15 V DC or 10 V rms. AC @ 50 Hz
Noise	see Figure 1. Typically 4 nV/√Hz @ 1 kHz and x1000 gain; 10 nV/√Hz @ 1 kHz and x10 or x1000 gain

Gain	x10, x100 or x1000 switch selectable
Gain Accuracy	±1%
Gain Stability	±150 ppm/°C

Output

Impedance	450 Ω
Max voltage swing	>10 V pk-pk
Slew rate	> 22 V/μs
Polarity	Non-inverting
Distortion	< 0.01% T.H.D.

Power

Internal	Four 9 V alkaline batteries provide approximately 12 hours of use
External	
a)	±15 V or ±18 V DC @ 27 mA
b)	110 V AC or 240 V AC via optional external model PS0108 power supply

Dimensions

(excluding connectors)	8.25" wide x 11" deep x 3.5" high (210 mm wide x 279 mm deep x 89 mm high)
Weight	5.3 lbs. (2.4 kg) excluding power supply

Model 181

Current Preamplifier

FEATURES

- **Low input impedance**
- **Low noise**
- **Single-ended virtual ground input**
- **Adjustable sensitivity**
- **DC to 200 kHz frequency response**
- **Detector bias control**

APPLICATIONS

- **Photodiode amplification**
- **Photomultiplier amplification**
- **Ion collector amplification**
- **Electron multiplier amplification**

DESCRIPTION

The model 181 is a current-to-voltage preamplifier of low noise and low input impedance designed to amplify the extremely low currents encountered in such areas as photometry and semiconductor research. In photometric applications the low input noise allows the use of photodetectors with dark currents as low as 10^{-14} A/√Hz, while the wide frequency range permits high modulation frequencies to avoid 1/*f* noise and power-line pick-up.

The unit has a high dynamic range, allowing small AC currents to be amplified without overload in the presence of quiescent (DC) detector currents up to ten times the current to voltage converter setting. In semiconductor applications its low input impedance permits the actual bias voltage applied to the device under test to be measured without having to correct for the effects of back bias.

Six switch-selectable sensitivity settings from 10^{-4} A/V to 10^{-9} A/V are available and the instrument has a usable frequency range from DC to 200 kHz. A signal monitor connector is provided on the rear panel and there is an overload indicator light on the front panel.

Bias Control
A bias control (accessible through an opening in the bottom of the unit) allows the application of a detector bias voltage at the input connector in the range 0 V to -5 V, with a nominal source impedance of 10^{-5}/S, where S is the selected sensitivity. For example, if the sensitivity is set to 10^{-7} A/V then the source impedance will be $10^{-5}/10^{-7}$, or 100 Ω. In some cases it may prove convenient to use this bias control to cancel the effect of DC bias accompanying the input signal.

DC Zero Control
A second control, also accessible through an opening in the bottom of the unit, allows the internal electronics to be DC zeroed.

Power
The unit can be powered from an external low voltage, a lock-in amplifier via a suitable power cable, or the models PS0055 or PS0056 remote line power supply modules.

Specifications

General

DC coupled current to voltage amplifier with adjustable sensitivity and a maximum frequency response extending from DC to 200 kHz. Adjustable negative detector bias. Single-ended virtual ground input and single-ended AC coupled output via BNC connectors.

Powered from external DC power supplies.

Input

Sensitivity — 10^{-4} A/V to 10^{-9} A/V in six ranges

Overload Indicator — Indicates that instantaneous (DC plus peak AC) current has exceeded amplifier capability - see table below

Frequency Response — see table and Figure 1

Gain A/V	Max DC Input Current	Frequency Response DC to
10^{-4}	1 mA	200 kHz
10^{-5}	100 µA	200 kHz
10^{-6}	10 µA	100 kHz
10^{-7}	1 µA	50 kHz
10^{-8}	100 nA	10 kHz
10^{-9}	10 nA	1 kHz

Input Impedance — See Figure 2

Noise Current — See Figure 3

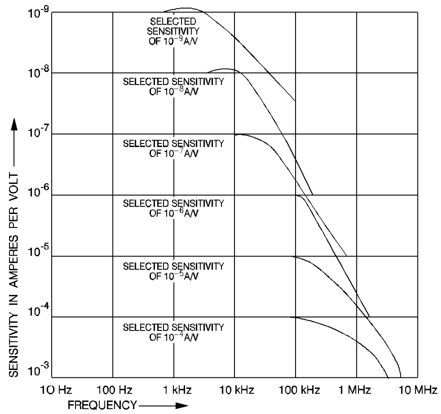


Figure 1, Frequency Response vs. Sensitivity

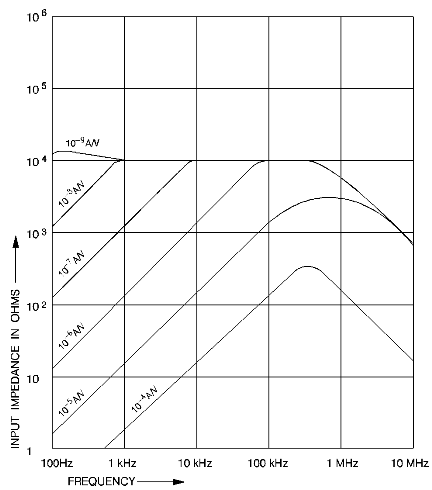


Figure 2, Input Impedance vs. Sensitivity

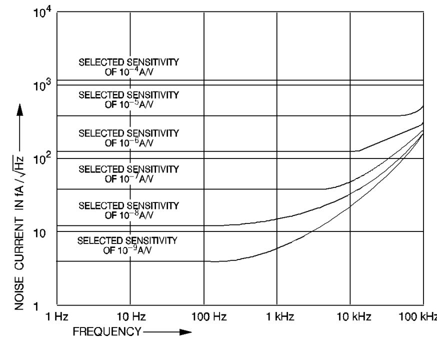


Figure 3, Noise Current vs. Frequency and Sensitivity

Outputs

Monitor Output — 600 Ω rear-panel BNC connector permits monitoring of the input signal

Main Output
- Level — 6.5 V rms maximum
- Impedance — 1 kΩ nominal

Output Attenuator — Provides optional 1:10 attenuation of output voltage

Power — ±15 V or ±24 V at 30 mA

General

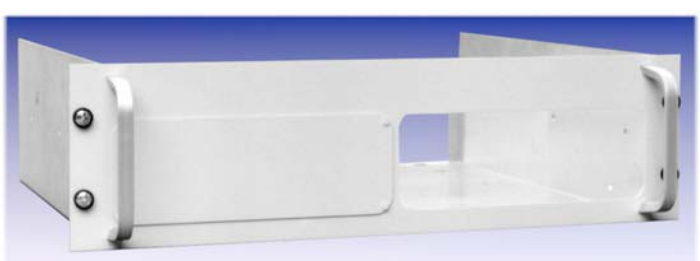
Dimensions (excluding connectors) — 4.5" wide x 6.6"deep x 2.7" high (114 mm wide x 168 mm deep x 69 mm high)

Weight — 1.2lbs (500 g)

Rack Mounting Hardware

Rack Mount Kit
Model K0304

to accommodate 1 or 2 model 5105, 5113, 5182, 5183, 5184, 5185, 5186, 5187, 5188A, or 5188B instruments in a 19” rack



Preamplifier Power Supplies

Remote Line Power Supply for Model 181 (110 V)
Model PS0055
complete with line power input cord

Remote Line Power Supply for Model 181 (220 V)
Model PS0056
complete with line power input cord

Remote Line Power Supply
Model PS0108
for Models
5113, 5182, 5183, 5184, 5185, 5186, 5187, 5188A or 5188B, 110 V or 240 V, complete with line power input cord

Alkaline Batteries (Pack of 20)
Model PS0109
for Models
5182, 5183, 5184
5186, 5187, 5188A,
and 5188B

Preamplifier Power Cables

Power Cable
Model C0145
to power model 181 from models
5102, 5104, 5109, 5110(A), 5205, 5206, 5207, 5208, 5209, 5210, 5302, 7124 7220(BFP), 7225(BFP), 7260, 7265, 7270, 7280, 7280(BFP), or 7310

Power Cable
Model C0218
To power one model
5182, 5183, 5184, 5186, 5187, 5188A or 5188B
from models
5102, 5104, 5109, 5110(A), 5205, 5206, 5207, 5208, 5209, 5210, 5302, 7124, 7220(BFP), 7225(BFP), 7260, 7265, 7270, 7280(BFP), or 7310

BNC Signal Interconnection Cables

Double Shielded BNC Cables

Model C0321
BNC Male - Male Cable, 6” (150 mm) long

Model C0322
BNC Male - Male Cable, 3’ (920 mm) long

Standard BNC Cables

Model 1715-0244
Boxcar Averager cable Kit consisting of:
2 Ea BNC TEE Pieces Male - Female - Female
6 Ea RG-58 BNC Male - Male Cables, 12" (305mm) long
2 Ea RG-58 BNC Male - Male Cables, 24" (610mm) long
4 Ea RG-58 BNC Male - Male Cables, 48" (1220mm) long

Model 1900

Impedance Matching Transformer

FEATURES

- **Selectable ratio**
- **Very low noise**
- **Single-ended inputs and outputs**

APPLICATIONS

- **Matching devices with low source impedances to measuring instruments with medium to high input impedances**
- **High T_c superconductor impedance measurements**
- **Increasing sensitivity of Model 5113 Preamplifier**

DESCRIPTION

The model 1900 low-noise impedance matching transformer is a versatile device for matching sources with impedances in the range from less than 50 mΩ to greater than 500 Ω to measuring instruments with high input impedance and offers two turns ratios of 1:100 or 1:1000. The frequency response depends on both the turns ratio and the source impedance, but under optimum conditions will be essentially flat from below 0.1 Hz to above 2 kHz.

The unit is supplied with a special low-noise double-screened coaxial cable fitted with two BNC plugs to connect its output to the following instrument.

Specifications

General

Precision signal transformer with adjustable turns ratio mounted in a mu-metal case. Signal input and output connections via BNC connectors

Voltage Gain	1:100 or 1:1000 selected by front-panel BNC connector
Frequency Response	See Figure 1

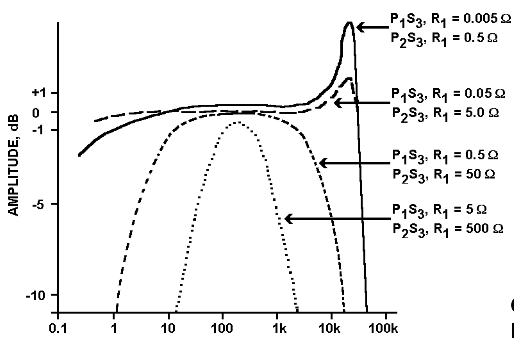


Figure 1, Typical Frequency Response

Noise	See Figure 2
Mounting	Free-standing fully shielded metal case

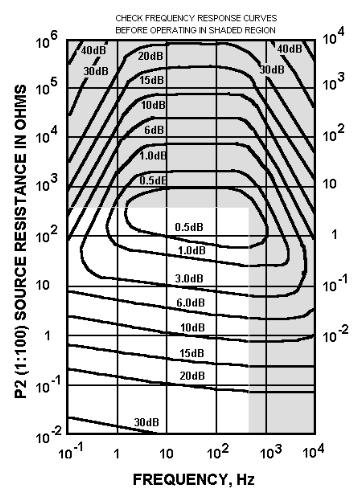


Figure 2, Typical Noise Figure Contours

General

Dimensions (excluding connectors)	3" wide x 5.8" deep x 3" high (76 mm wide x 147 mm deep x 76 mm high)
Shipping Weight	5lbs (2.3 kg)

Lock-in Amplifier Selection Guide

SIGNAL RECOVERY offers the widest range of lock-in amplifiers from a single supplier, so we are almost certain to have a instrument that will meet your needs. Choosing the right model is not difficult if you follow this guide. In case of any doubt, simply contact us for further advice

First, identify the reference frequency range that you need to use. In many experiments there is some latitude in choosing this, but in other cases it is defined by other factors, such as the resonant frequency of a system component or the built in modulation frequency of a laser diode module.

- If you need to use reference frequencies greater than 100 to 150kHz then consider the **models 7270, 7265, 7280** or **7280BFP**. The 7270 and 7265 operate at frequencies up to 250 kHz, but if you also need short output time constants and analog outputs with a fast update rate then the model 7270 is the better unit. For frequencies above 250 kHz, consider the **models 7280** or **7280BFP**
- For multiple-channel detection at frequencies up to 50 kHz, consider the **model 7210.**

  If you will be using reference frequencies in the range 0.5 Hz to 100 - 120 kHz then you need to decide whether you need a simple instrument or something more sophisticated.

  - If you need a simple instrument ideal for basic signal recovery or educational use, don't need an internal oscillator to drive your experiment, and are happy to operate the unit from a PC then consider the **models 5105** or **5106.**

    If you need a traditional instrument complete with full front-panel control and an internal oscillator, you next need to decide if you want an analog or digital (DSP) instrument. Analog units offer true analog outputs and can be the best choice in feedback control loops, whereas digital units have zero output drift and often include additional operating modes.

    - Choose the **models 5209** or **5210** if you want a top quality instrument with an analog demodulator. These units include band-rejection signal channel filters making them especially suited to measuring components of signals at twice the reference frequency in the presence of strong signals at the reference frequency, and continuously adjustable full-scale sensitivity ranges.
    - Choose the **model 7225** or **7225BFP** if you want a unit with DSP demodulation, but don't need the extra features of the model 7270 or 7265.
    - Choose the **model 7270** or **7265** for greater flexibility and to use the extended operating modes provided by these DSP instruments.
    - Choose the **model 7124** for the ultimate performance when you need the minimum possible digital switching noise introduced back into the experiment by the instrument.
- If you need to use reference frequencies in the range 1 mHz to 500 mHz then consider the **models 7225, 7225BFP, 7265** or **7270.**

Model 7124

Precision Lock-in Amplifier

FEATURES

- **Unique analog fiber optic link between the RCU connection module and the main console**
- **No digital clock or switching noise present at the RCU connectors**
- **0.5 Hz to 150 kHz operation**
- **Voltage and current mode inputs**
- **1.0 MHz main ADC sampling rate**
- **10 µs to 100 ks output filter time constants**
- **Precision DDS sinewave oscillator with adjustable amplitude and frequency**
- **Harmonic measurements up to 127 × F**
- **Dual Reference, Tandem, Dual Harmonic and Virtual Reference operating modes**
- **Spectral display mode**

APPLICATIONS

- **Measurement of low impedances in superconductor research**
- **Pump-probe studies**
- **Scanned probe measurements**
- **Atomic force microscopy**

DESCRIPTION

The model 7124 precision lock-in amplifier uses a unique analog fiber optic link to interconnect a remote connection unit (RCU), to which the experiment is connected, and a main instrument console. Using this technique, the model 7124 overcomes one significant limitation of other lock-in amplifiers, which is that the instrument itself can act as a source of digital clock and switching noise that can be coupled back into the experiment via the signal or internal oscillator connectors. Although it is rejected by the lock-in and generally does not impair its performance, the power it dissipates in the sample or device under test can cause serious problems, particularly in low temperature physics experiments.

The fiber link ensures that In normal operation there are no digital clock signals within the RCU, and so it can emit no switching noise. The overall instrument therefore has all the advantages of the latest DSP technology for signal detection, and a powerful processor for easy user operation, as well as the low noise performance that until now has only been available in instruments of all-analog design.

Signal and Reference Connections

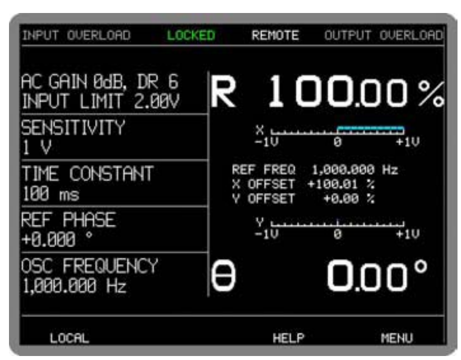
In normal use the 7124's signal and reference connections are made at the RCU. The signal input can be switched to operate in either voltage or current input modes, allowing the best possible connection to be made to the experiment. The RCU also has both general purpose analog and TTL logic reference inputs, as well as the a precision DDS oscillator output that generates a sine wave signal of adjustable frequency and amplitude.

The RCU is connected to the main instrument console via a 16ft (5 m) fiber cable bundle. Complete isolation from all potenttial sources of noise is possibe by ordering the unit with the 7124/99 option that allows the RCU to be powered from external ±24 V DC (battery) supplies.

Main Console

The 7124's main console is a compact, benchtop unit with a color display, and keys for operating the instrument controls, accessing different menus, and easy entry of numeric values. The operating frequency range is from 0.5 Hz to 150 kHz, with a main ADC sampling rate and analog output DAC update rate of 1 MHz, giving excellent performance at even the shortest time constant of 10 µs.

Manual operation is straightforward and based on a similar menu structure to that used on the model 7280, using the color TFT display panel in conjunction with the keys grouped around it and the numeric keypad to adjust the instrument's controls, with the selected outputs being shown

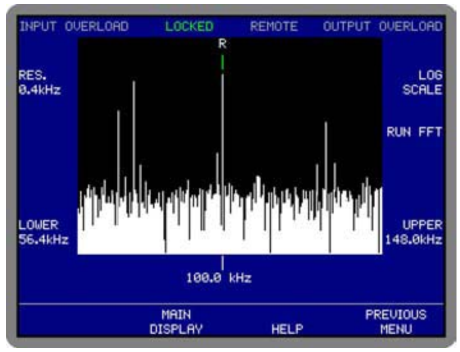


Main Display

both on the display and being available as analog signals from four rear-panel connectors.

The Main Display is used in normal operation and shows four user-selected instrument controls on the left-hand side and four user-selected outputs, output offset status, and the present reference frequency, on the right. The output display selections include digital and bar-graph displays in a variety of formats. Error information, such as input and output overload, and reference unlock indication, is clearly shown along the top edge of the display, while soft keys along the bottom edge are used for selecting controls and to initiate numerical keypad data entry.

Extended Operating Modes
The instrument includes the Dual Reference, Dual Harmonic and Virtual Reference operating modes made



Spectral Display Mode

popular by other ***SIGNAL RECOVERY*** lock-in amplifiers. In addition, the Spectral Display allows the spectrum of the signals present at the input to be calculated and displayed, which can help when choosing the reference frequency

It also includes a new "tandem" demodulation mode which allows an amplitude-modulated signal to be first demodulated at a carrier frequency, with the output from this demodulation being processed by a second demodulator running at a lower frequency.

Main Console Inputs and Outputs
The main console provides a second oscillator output, auxiliary signal channel input, four auxiliary ADC inputs, and four DAC outputs. The DAC outputs can be set to function as analog outputs for the signal measurement (e.g. X, Y, Magnitude, Phase) and/or general purpose programmable analog outputs.

Computer Control
External control of the unit is via USB, RS232 or Ethernet interfaces, using simple mnemonic-type ASCII commands.

Software support is available in the form of a LabVIEW driver supporting all instrument functions, the Acquire™ data acquisition software, and the SRInstComms toolkit.

Specifications

General

Dual-phase DSP lock-in amplifier operating over a reference frequency range of 0.5 Hz to 150 kHz. All-analog front-end generating no digital clock or switching noise with fiber optic connections to main instrument console.

Wide range of extended measuring modes and auxiliary inputs and outputs. User-upgradeable firmware.

Measurement Modes

The instrument can simultaneously show any four of these outputs on the front panel display:

X	In-phase
Y	Quadrature
R	Magnitude
θ	Phase Angle
Noise	
Harmonic	nF, $n \leq 127$

Dual Harmonic
Simultaneously measures the signal at two different harmonics F_1 and F_2 of the reference frequency

Dual Reference
Simultaneously measures the signal at two different reference frequencies, F_1 and F_2 . F_1 can be set to internal or external reference, with and F_2 being the other reference source

Tandem Demodulation
Demodulates the signal using reference frequency F_1, and then passes the resulting X channel output to a second demodulator running at reference frequency F_2

Virtual Reference
Locks to and detects a signal without a reference (100 Hz ≤ F ≤ 150 kHz)

Noise
Measures noise in a given bandwidth centered at the reference frequency F

Spectral Display
Gives a visual indication of the spectral power distribution of the input signal in a user-selected frequency range lying between 1 Hz and 150 kHz. Note that although the display is calibrated in terms of frequency, it is not calibrated for amplitude. Hence it is only intended to assist in choosing the best reference frequency.

Display

320 × 240 pixel (¼ VGA)
colr TFT display giving digital, analog bar-graph and graphical indication of measured signals. Menu system with dynamic key function allocation. On-screen context sensitive help

Signal Channel - Remote Connection Unit

Voltage Input	
Modes	A only, -B only or Differential (A-B)
Full-scale Sensitivity	2 nV to 1 V in a 1-2-5 sequence (e.g. 2 nV, 5 nV, 10 nV, etc.)
Frequency Response	0.5 Hz to 150 kHz (-3dB)
Input Impedance	
FET Input	10 MΩ // 25 pF, AC or DC coupled
Biipolar Input	10 kΩ // 25 pF, input must be DC coupled
Maximum Safe Input	±12.0 V
Voltage Noise	5 nV/√Hz @ 1 kHz
C.M.R.R.	> 100 dB @ 1 kHz
Max. Dynamic Reserve	> 100 dB
Impedance	100 MΩ // 25 pF
Gain Accuracy	±1.0 typ, ±2.0% max.
Distortion	-90 dB THD (60 dB AC gain, 1 kHz)
Current Input	
Mode	Low Noise (10^8 V/A) or Wide Bandwidth (10^6 V/A)
Full-scale Sensitivity	
Low Noise	2 fA to 10 nA in a 1-2-5 sequence
Wide Bandwidth	2 fA to 1 µA in a 1-2-5 sequence
Frequency Response (-3dB)	
Low Noise	0.5 Hz to 500 Hz minimum
Wide Bandwidth	0.5 Hz to 50 kHz minimum
Impedance	
Low Noise	< 2.5 kΩ @ 100 Hz
Wide Bandwidth	< 250 Ω @ 1 kHz
Noise	
Low Noise	13 fA/√Hz @ 500 Hz
Wide Bandwidth	130 fA/√Hz @ 1 kHz
Gain Accuracy	± 2.0% typ, midband
Either Input Mode:	
Max. Dynamic Reserve	> 100 dB
Line Filter	Filter can be set to attenuate 50/60 Hz, 100/120 Hz, or both frequency bands
Grounding	BNC shields can be grounded or floated via 1 kΩ to ground

Signal Monitor

Amplitude	±1 V FS. This is the signal after preamplification and filtering, but before transmission over the optical link
Output Impedance	1 kΩ

Signal Channel - Main Console

Auxiliary Input	
Mode	Single-ended voltage mode input
Impedance	10 MΩ // 25 pF
Maximum Safe Input	±12.0 V
Full-scale Sensitivity	1 V
Signal Monitor	
Amplitude	±1 V FS. This is the signal received from the Remote Connection Unit immediately prior to conversion by the main ADC
Output Impedance	1 kΩ

Reference Input - Remote Connection Unit or Main Console

TTL Input (rear panel)	
Frequency Range	0.5 Hz to 150 kHz
Analog Input (front panel)	
Impedance	1 MΩ // 30 pF
Sinusoidal Input	
Level	1.0 V rms
Frequency Range	0.5 Hz to 150 kHz
Squarewave Input	
Level	250 mV rms
Frequency Range	2 Hz to 150 kHz

Reference Channel

Phase Set Resolution	0.001° increments
Phase Noise at 100 ms TC, 12 dB/octave slope	
Internal Reference	< 0.0001° rms
External Reference	< 0.01° rms @ 1 kHz
Orthogonality	90° ±0.0001°
Acquisition Time	
Internal Reference	Instantaneous acquisition
External Reference	2 cycles + 1 s
Reference Frequency Meter Resolution	4 ppm or 1 mHz, whichever is the greater

Demodulators and Output Processing

Output Zero Stability	
Digital Outputs & Displays	No zero drift on all settings
DAC Analog Outputs	< 100 ppm/°C
Harmonic Rejection	-90 dB
Output Filters	
Time Constant	10 µs to 100 ks in a 1-2-5 sequence
Slope (roll-off)	
TC ≤ 5 ms	6 or 12 dB/octave
TC > 10 ms	6, 12, 18 or 24 dB/octave
Synchronous Filter	Available at F < 20 Hz
Offset	Auto/Manual on X and/or Y: ±300% F.S.
Phase Measurement Resolution	≤ 0.01°
Reference Monitor	TTL signal at current reference frequency, internal or external

Oscillator - General

Frequency	
Range	0.5 Hz to 150 kHz
Setting Resolution	1 mHz
Absolute Accuracy	± 50 ppm
Amplitude	
Range	1 mV to 5 V
Setting Resolution	1 mV
Output Impedance	50 Ω
Sweep	
Frequency	
Output Range	0.5 Hz to 150 kHz
Law	Linear or Logarithmic
Step Rate	1000 Hz maximum
Amplitude	
Output Range	0.000 to 1.000 V rms
Law	Linear
Step Rate	
Main Console	20 Hz maximum
RCU	1 Hz maximum

Oscillator Output - Remote Connection Unit

Amplitude	
Accuracy	±1.0% typ
Stability	100 ppm/°C
Distortion (THD)	-80 dB @ 1 kHz and 100 mV rms

Oscillator Output - Main Console

Amplitude	
Accuracy	±0.2% typ
Stability	50 ppm/°C
Distortion (THD)	-80 dB @ 1 kHz and 100 mV rms

Auxiliary Inputs

ADC 1, 2, 3 and 4	
Maximum Input	±11 V
Resolution	1 mV
Accuracy	±20 mV
Input Impedance	1 MΩ // 30 pF
Sample Rate	250 kHz maximum (one ADC only)
Trigger Mode	Internal, External or burst
Trigger Input	TTL compatible, rising or falling edge

Outputs

Analog Outputs	
DAC1	X, X1, Mag2, User DAC1, Output function
DAC2	Y, Y1, Pha2, User DAC2, Output function
DAC3	X2, Mag, Mag1, User DAC3, Output function
DAC4	Y2, Pha, Pha1, User DAC4, Output function
Output Functions	Noise, Ratio, Log Ratio and User Equations 1 & 2.
Amplitude	
X(1), Y(1), Mag(1), Pha(1)	±2.5 V full-scale; linear to ±300% F.S.
User DACs and Output Functions	±11.0 V full-scale
Impedance	1 kΩ
Update Rate	
X(1/2), Y(1/2), Mag(1/2), Pha(1/2) @ TC < 1 s	1 MHz
User DACs, Output Functions and TC's ≥ 1 s	1 kHz
8-bit Digital Port - RCU	
Mode	8 TTL outputs
Status	Each output line can be set high or low
8-bit Digital Port - Main Console	
Mode	0 to 8 lines can be configured as inputs, with the remainder being outputs
Status	Each output line can be set high or low and the status of each input line read
Power - Low Voltage	±15 V at 100 mA 5-pin 180° DIN connectors on both main console and remote connection unit for powering compatible preamplifiers

Data Storage Buffer

Size	100,000 data points
Max Storage Rate	
Fast Mode	1 MHz (X1, Y1, X2, Y2, ADC1, Demod I/P 1, Demod I/P 2)
Normal Mode	1 kHz

User Settings

Up to 8 complete instrument settings can be saved or recalled from memory as required

Interfaces

USB 2.0, Ethernet, and RS232 on main console allow complete control of instrument settings, and data readout. Four channel 5 m (16ft) fiber optic link between main console and remote connection unit.

General

Power - Main Console & RCU	
Voltage	110/120/220/240
Frequency	50/60 Hz
Power	
Main Console	40 VA max
RCU	15 VA max
Power - RCU with option 7124/99	
Voltage	±24.0 V DC
Current	+300 mA / –170 mA
Dimensions	
Main Console	
Width	15½" (390 mm)
Depth	7¼" (185 mm)
Height	
With feet	7¼" (185 mm)
Without feet	6½" (170 mm)
RCU	
Width	15½" (390 mm)
Depth	7¼" (185 mm)
Height	
With feet	3" (75 mm)
Without feet	2½" (64 mm)
Weight	
Main Console	12.8 lb (5.8 kg)
RCU	7.9 lb (3.6 kg)

Model 7270

DSP Lock-in Amplifier

FEATURES

- **1 mHz to 250 kHz operation**
- **Voltage and current mode inputs**
- **1.0 MHz main ADC sampling rate**
- **10 µs to 100 ks output filter time constants**
- **Precision DDS sinewave oscillator with adjustable amplitude and frequency**
- **Harmonic measurements up to 127 × F**
- **Dual Reference, Tandem, Dual Harmonic and Virtual Reference operating modes**
- **Spectral display mode**

APPLICATIONS

- **Impedance meaasurements**
- **Pump-probe studies**
- **Scanned probe measurements**
- **Atomic force microscopy**

DESCRIPTION

The model 7270 sets a new standard for general-purpose DSP lock-in amplifiers. We've taken advantage of the developments in technology since the first DSP lock-in amplifiers were introduced in the early 1990’s to update the core design, but made sure that we’ve included all the best features of our model 7265 and 7280 instruments. What’s more, the new architecture has allowed us to offer even better specifications in an instrument that is physically much more compact than older designs. The result is a lock-in amplifier of outstanding performance that is easy to use and suitable for virtually all measurements over a frequency range extending from 1 mHz to 250 kHz.

Versatility

In common with other models in our range, the 7270 offers much more than just dual phase lock-in detection at the reference frequency of an applied signal. We’ve included features unique to ***SIGNAL RECOVERY*** instruments such as dual reference and dual harmonic detection, which allow signals at two different frequencies to be measured simultaneously. The spectral display mode shows the power spectral density of the input signal, making it easy to avoid interfering signals when selecting a reference frequency. It is now even possible to perform tandem demodulation. In this mode an amplitude-modulated signal at a (high) "carrier" frequency is first demodulated at that frequency. The resulting in-phase output, at short time constant settings, is a signal at the modulating frequency which is then passed forward for detection by a second set of demodulators running at the same modulating frequency. Such detection techniques, which can be used in pump-probe measurements, have until now required two separate instruments with an analog connection between them.

Fast Data Processing

The main ADC sampling rate and the rate at which the analog signal outputs are updated is 1 MSa/s, giving excellent performance when used at short output filter time constant settings, such as in scanned probe measurements. But we’ve also increased the maximum rate at which data can be stored to the internal curve buffer to 1 µs per point, allowing for the first time direct capture of instrument outputs when using these short time constants. The buffer length has also been increased to 100,000 sets of points, giving recording times of 100 ms at the fastest sampling rates. What’s more, in the fast capture mode

Main Display

the length does not need to be divided by the number of outputs being stored, making it possible, for example, to store the full 100,000 points of X, Y and auxiliary ADC1 values at the same time.

Remote Control

The built-in RS232, USB and Ethernet connections allow full operation from a controlling computer. We offer a comprehensive software package, Acquire, that can operate the instrument via any of these interfaces and makes it easy to set up and run complex experiments, such as measuring a system's frequency response, as well as allowing remote control of every instrument function. Users who wish to do their own programming can use our ActiveX control and toolkit (SRInstComms), or free LabVIEW driver, to simplify the task.

See what you've been missing...

In summary, if you're looking for a general purpose lock-in to work in the range 1 mHz to 250 kHz then you need look no further - you've found it in the ***SIGNAL RECOVERY*** model 7270.

Specifications

General

Dual-phase DSP lock-in amplifier operating over a reference frequency range of 1 mHz to 250 kHz. Wide range of extended measuring modes and auxiliary inputs and outputs. User-upgradeable firmware.

Measurement Modes

The instrument can simultaneously show any four of these outputs on the front panel display:

X	In-phase
Y	Quadrature
R	Magnitude
θ	Phase Angle
Noise	
Harmonic	*n*F, $n \leq 127$

Dual Harmonic
: Simultaneously measures the signal at two different harmonics F_1 and F_2 of the reference frequency

Dual Reference
: Simultaneously measures the signal at two different reference frequencies, F_1 and F_2 . F_1 can be set to internal or external reference, with and F_2 being the other reference source

Tandem Demodulation
: Demodulates the signal using reference frequency F_1, and then passes the resulting X channel output to a second demodulator running at reference frequency F_2

Virtual Reference
: Locks to and detects a signal without a reference (100 Hz ≤ F ≤ 150 kHz)

Noise
: Measures noise in a given bandwidth centered at the reference frequency F

Spectral Display
: Gives a visual indication of the spectral power distribution of the input signal in a user-selected frequency range lying between 1 Hz and 250 kHz. Note that although the display is calibrated in terms of frequency, it is not calibrated for amplitude. Hence it is only intended to assist in choosing the best reference frequency.

Display

320 × 240 pixel (¼ VGA) color TFT display giving digital, analog bar-graph and graphical indication of measured signals. Menu system with dynamic key function allocation. On-screen context sensitive help

Signal Channel

Voltage Input	
Modes	A only, -B only or Differential (A-B)
Full-scale Sensitivity	2 nV to 1 V in a 1-2-5 sequence (e.g. 2 nV, 5 nV, 10 nV, etc.)
Frequency Response	1 mHz to 250 kHz (-3dB)
Input Impedance	
FET Input	10 MΩ // 25 pF, AC or DC coupled
Biipolar Input	10 kΩ // 25 pF, input must be DC coupled
Maximum Safe Input	±12.0 V
Voltage Noise	5 nV/√Hz @ 1 kHz
C.M.R.R.	> 100 dB @ 1 kHz
Max. Dynamic Reserve	> 100 dB
Impedance	100 MΩ // 25 pF
Gain Accuracy	±0.5% typ, ±1.0% max.
Distortion	-90 dB THD (60 dB AC gain, 1 kHz)
Current Input	
Mode	Low Noise (10^8 V/A) or Wide Bandwidth (10^6 V/A)
Full-scale Sensitivity	
Low Noise	2 fA to 10 nA in a 1-2-5 sequence
Wide Bandwidth	2 fA to 1 µA in a 1-2-5 sequence
Frequency Response (-3dB)	
Low Noise	1 mHz to 500 Hz minimum
Wide Bandwidth	1 mHz to 50 kHz minimum
Impedance	
Low Noise	< 2.5 kΩ @ 100 Hz
Wide Bandwidth	< 250 Ω @ 1 kHz
Noise	
Low Noise	13 fA/√Hz @ 500 Hz
Wide Bandwidth	130 fA/√Hz @ 1 kHz
Gain Accuracy	± 2.0% typ, midband
Either Input Mode:	
Max. Dynamic Reserve	> 100 dB
Line Filter	Filter can be set to attenuate 50/60 Hz, 100/120 Hz, or both frequency bands
Grounding	BNC shields can be grounded or floated via 1 kΩ to ground

Signal Monitor

Amplitude	±1 V F.S.
Output Impedance	1 kΩ

Reference Input

TTL Input (rear panel)	
Frequency Range	1 mHz to 250 kHz
Analog Input (front panel)	
Impedance	1 MΩ // 30 pF
Sinusoidal Input	
Level	1.0 V rms
Frequency Range	0.5 Hz to 250 kHz
Squarewave Input	
Level	250 mV rms
Frequency Range	2 Hz to 250 kHz

Reference Channel

Phase Set Resolution	0.001º increments
Phase Noise at 100 ms TC, 12 dB/octave slope	
Internal Reference	< 0.0001º rms
External Reference	< 0.01º rms @ 1 kHz
Orthogonality	90º ±0.0001º
Acquisition Time	
Internal Reference	Instantaneous
External Reference	2 cycles + 1 s
Reference Frequency Meter Resolution	4 ppm or 1 mHz, whichever is the greater

Demodulators and Output Processing

Output Zero Stability	
Digital Outputs & Displays	No zero drift on all settings
DAC Analog Outputs	< 100 ppm/ºC
Harmonic Rejection	-90 dB

Output Filters

Time Constant	10 µs to 100 ks in a 1-2-5 sequence
Slope (roll-off)	
TC < 5 ms	6 or 12 dB/octave
TC ≥ 5 ms	6, 12, 18 or 24 dB/octave
Synchronous Filter	Available at F < 20 Hz
Offset	Auto/Manual on X and/or Y: ±300% F.S.
Phase Measurement Resolution	≤ 0.01º
Reference Monitor	TTL signal at current reference frequency, internal or external

Oscillator

Frequency	
Range	1 mHz to 250 kHz
Setting Resolution	1 mHz
Absolute Accuracy	± 50 ppm
Amplitude	
Range	1 µV to 5 V
Max Setting Resolution	1 µV
Output Impedance	50 Ω

Sweep	
Frequency	
Output Range	1 mHz to 250 kHz
Law	Linear or Logarithmic
Step Rate	1000 Hz maximum (1 ms/step)
Amplitude	
Output Range	0.000 to 1.000 V rms
Law	Linear
Step Rate	20 Hz maximum (50 ms/step)
Auxiliary Inputs	
ADC 1, 2, 3 and 4	
Maximum Input	±11 V
Resolution	1 mV
Accuracy	±20 mV
Input Impedance	1 MΩ // 30 pF
Sample Rate	200 kHz maximum (one ADC only)
Trigger Mode	Internal, External or burst
Trigger Input	TTL compatible, rising or falling edge
Outputs	
Analog Outputs	
DAC1	X, X1, Mag2, User DAC1, Output function
DAC2	Y, Y1, Pha2, User DAC2, Output function
DAC3	X2, Mag, Mag1, User DAC3, Output function
DAC4	Y2, Pha, Pha1, User DAC4, Output function
Output Functions	Noise, Ratio, Log Ratio and User Equations 1 & 2.
Amplitude	
X(1), Y(1), Mag(1), Pha(1)	±2.5 V full-scale; linear to ±300% F.S.
User DACs and Output Functions	±11.0 V full-scale
Impedance	1 kΩ
Update Rate	
X(1/2), Y(1/2), Mag(1/2), Pha(1/2) @ TC < 1 s	1 MHz
User DACs, Output Functions and TC's ≥ 1 s	1 kHz
8-bit Digital Port	
Mode	0 to 8 lines can be configured as inputs, with the remainder being outputs
Status	Each output line can be set high or low and the status of each input line read
Power - Low Voltage	±15 V at 100 mA 5-pin 180° DIN connectors on rear panel for powering compatible preamplifiers
Data Storage Buffer	
Size	100,000 data points
Max Storage Rate	
Fast Mode	1 MHz (X1, Y1, X2, Y2, ADC1, Demod I/P 1, Demod I/P 2)
Normal Mode	1 kHz

User Settings

Up to 8 complete instrument settings can be saved or recalled from memory as required

Interfaces

USB 2.0, Ethernet, and RS232 allow complete control of instrument settings, and data readout.

General	
Power	
Voltage	110/120/220/240
Frequency	50/60 Hz
Power	40 VA max
Dimensions	
Width	15½" (390 mm)
Depth	7¼" (185 mm)
Height	
With feet	7¼" (185 mm)
Without feet	6½" (170 mm)
Weight	12.8 lb (5.8 kg)

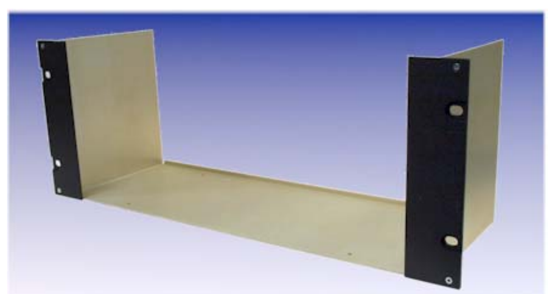
Accesories for use with Models 7124 and 7270

SIGNAL RECOVERY Acquire Software (see page 56)

Users who do not wish to write their own control code but who still want to record the instrument's outputs to a computer file will find the ***SIGNAL RECOVERY*** Acquire Lock-in Amplifier Applications Software, available at a small extra cost, useful. This package, suitable for Windows XP/ Vista, extends the capabilities of the instrument by, for example, adding the ability to record swept frequency measurements. It also supports the internal curve buffer, allowing acquisition rates of up to 1 million points per second independent of the computer's processor speed.

Model K02005

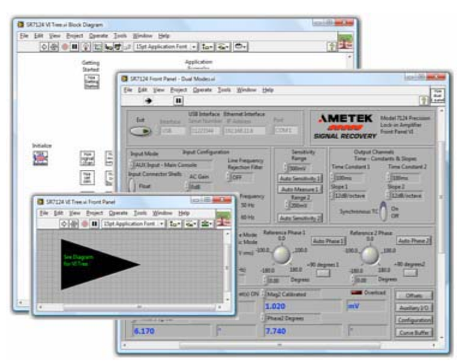
Rack mount to mount one model 7124 (main console) or 7270 in a 19" rack



Model K02004 Rack Mount Kit

LabVIEW Driver Software

LabVIEW drivers for the instruments are available from the **www.signalrecovery.com** website, offering example VIs for all their controls and outputs, as well as the usual Getting Started and Utility VIs. They also include example soft-front panels built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.



Model 7280

Wide Bandwidth DSP Lock-in Amplifier

SIGNAL RECOVERY

FEATURES

- **0.5 Hz to 2 MHz operation**
- **Voltage and current mode inputs**
- **Direct digital demodulation without down-conversion**
- **7.5 MHz main ADC sampling rate**
- **1 µs to 100 ks output time constants**
- **Quartz crystal stabilized internal oscillator**
- **Harmonic measurements to 32F**
- **Dual reference, Dual Harmonic and Virtual Reference modes**
- **Spectral display mode**

APPLICATIONS

- **Scanned probe microscopy**
- **Optical measurements**
- **Audio studies**
- **AC impedance studies**
- **Atomic force microscopy**

DESCRIPTION

The model 7280 DSP Lock-in Amplifier is an exceptionally versatile instrument with outstanding performance. With direct digital demodulation over an operating frequency extending up to 2.0 MHz, output filter time constants down to 1 µs and a main ADC sampling rate of 7.5 MHz it is ideal for recovering fast changing signals. But unlike some other high frequency lock-ins, it also works in the traditional audio frequency band.

In addition to its excellent technical specifications, it is also very easy to use. The front panel is dominated by a large electroluminescent display panel, used both to show the instrument's outputs and for adjusting its controls via a series of menus. Controls are set by a combination of the use of the keys surrounding the display and the numeric keypad, while four cursor-movement keys simplify use of the graphic display menus.

Users of the ***SIGNAL RECOVERY*** models 7260 and 7265 will find switching to the 7280 very easy, since we've designed it with a similar menu structure. The only significant changes are in some of the control menus, where the better resolution of the display allows both the controls and the instrument outputs to be shown simultaneously, for even faster feedback on the effects of control adjustments.

Main Display

Naturally, the instrument includes the extended operating modes like dual reference, dual harmonic and virtual reference made popular by the 7260 and 7265, as well as the spectral display mode used to aid reference frequency selection. It also includes GPIB and RS232 interfaces for remote computer control and a range of auxiliary analog and digital inputs and outputs. Compatible software is available in the form of a LabVIEW driver supporting all instrument functions, and the Acquire lock-in amplifier applications software. The driver and a free demonstration version of the software, DemoAcquire, are available for download from our website at **www.signalrecovery.com**

Auto Functions Menu

In summary, if you need a lock-in capable of working beyond the traditional audio frequency band but still want the drift-free performance that only digital demodulation brings, then look no further - you have found it in the ***SIGNAL RECOVERY*** Model 7280.

Specifications

General

Dual-phase DSP lock-in amplifier operating over a reference frequency range of 0.5 Hz to 2.0 MHz. Direct digital demodulation using a main ADC sampling rate of 7.5 MHz.

Wide range of extended measuring modes and auxiliary inputs and outputs. User-upgradeable firmware.

Measurement Modes

The instrument can simultaneously show any four of these outputs on the front panel display:

X	In-phase
Y	Quadrature
R	Magnitude
θ	Phase Angle
Noise	
Harmonic	*n*F, $n \le 32$

Dual Harmonic
Simultaneously measures the signal at two different harmonics F_1 and F_2 of the reference frequency

Dual Reference
Simultaneously measures the signal at two different reference frequencies, F_1 and F_2 where F_1 is the external and F_2 the internal reference

Frequency Ranges for Dual Harmonic and Dual Reference Modes:

Standard Unit	F_1 and $F_2 \le 20$ kHz
With option -/99	F_1 and $F_2 \le 800$ kHz
With option -/98	F_1 and $F_2 \le 2.0$ MHz

Virtual Reference
Locks to and detects a signal without a reference (100 Hz ≤ F ≤ 2.0 MHz)

Noise
Measures noise in a given bandwidth centered at the reference frequency F

Spectral Display
Gives a visual indication of the spectral power distribution of the input signal in a user-selected frequency range lying between 1 Hz and 2.0 MHz. Note that although the display is calibrated in terms of frequency, it is not calibrated for amplitude. Hence it is only intended to assist in choosing the optimum reference frequency

Display

320 × 240 pixel (¼ VGA) electroluminescent panel giving digital, analog bar-graph and graphical indication of measured signals. Menu system with dynamic key function allocation. On-screen context sensitive help

Signal Channel

Voltage Input	
Modes	A only, -B only or Differential (A-B)
Full-scale Sensitivity	
0.5 Hz ≤ F ≤ 250 kHz	10 nV to 1 V in a 1-2-5 sequence
250 kHz < F ≤ 2.0 MHz	100 nV to 1 V in a 1-2-5 sequence
Max. Dynamic Reserve	> 100 dB
Impedance	100 MΩ // 25 pF
Maximum Safe Input	20 V pk-pk
Voltage Noise	5 nV/√Hz @ 1 kHz
C.M.R.R.	> 100 dB @ 1 kHz
Frequency Response	0.5 Hz to 2.0 MHz
Gain Accuracy	±0.3% typ, ±0.6% max. (full bandwidth)
Distortion	-90 dB THD (60 dB AC gain, 1 kHz)
Line Filter	attenuates 50, 60, 100, 120 Hz
Grounding	BNC shields can be grounded or floated via 1 kΩ to ground
Current Input	
Mode	Low Noise, Normal or Wide Bandwidth
Full-scale Sensitivity	
Low Noise	10 fA to 10 nA in a 1-2-5 sequence
Normal	10 fA to 1 µA in a 1-2-5 sequence
Wide Bandwidth	
F ≤ 250 kHz	1 pA to 100 µA in a 1-2-5 sequence
F > 250 kHz	10 pA to 100 µA in a 1-2-5 sequence
Max. Dynamic Reserve	> 100 dB
Frequency Response (-3 dB)	
Low Noise	≥ 500 Hz
Normal	≥ 50 kHz
Wide Bandwidth	≥ 1 MHz
Impedance	
Low Noise	< 2.5 kΩ @ 100 Hz
Normal	< 250 Ω @ 1 kHz
Wide Bandwidth	< 25 Ω @ 10 kHz
Noise	
Low Noise	13 fA/√Hz @ 500 Hz
Normal	130 fA/√Hz @ 1 kHz
Wide Bandwidth	1.3 pA/√Hz @ 1 kHz
Gain Accuracy	± 0.6% typ, midband
Line Filter	attenuates 50, 60, 100, 120 Hz
Grounding	BNC shield can be grounded or floated via 1 kΩ to ground

Reference Channel

TTL Input (rear panel)	
Frequency Range	0.5 Hz to 2.0 MHz
Analog Input (front panel)	
Impedance	1 MΩ // 30 pF
Sinusoidal Input	
Level	1.0 V rms*
Frequency Range	0.5 Hz to 2.0 MHz
Squarewave Input	
Level	250 mV rms*
Frequency Range	2 Hz to 2 MHz

*Note: Lower levels can be used with the analog input at the expense of increased phase errors

Phase Set Resolution	0.001° increments
Phase Noise at 100 ms TC, 12 dB/octave slope	
Internal Reference	< 0.0001° rms
External Reference	< 0.01° rms @ 1 kHz
Orthogonality	90° ±0.0001°
Acquisition Time	
Internal Reference	instantaneous acquisition
External Reference	2 cycles + 50 ms
Reference Frequency Meter Resolution	1 ppm or 1 mHz, whichever is the greater

Demodulator and Output Processing

Output Zero Stability	
Digital Outputs	No zero drift on all settings
Displays	No zero drift on all settings
Analog Outputs	< 5 ppm/°C
Harmonic Rejection	-90 dB
Output Filters	
X, Y and R outputs only	
Time Constant	1 µs to 1 ms in a 1-2-5 sequence, and 4 ms
Slope (roll-off)	6 and 12 dB/octave
All outputs	
Time Constant	5 ms to 100 ks in a 1-2-5 sequence
Slope	6, 12, 18 and 24 dB/octave
Synchronous Filter	Available for F < 20 Hz
Offset	Auto and Manual on X and/or Y: ±300% full-scale
Absolute Phase Measurement Accuracy	≤ 0.01°

Oscillator

Frequency	
Range	0.5 Hz to 2.0 MHz
Setting Resolution	1 mHz
Absolute Accuracy	± 50 ppm
Distortion (THD)	-80 dB @ 1 kHz and 100 mV rms
Amplitude (rms)	
Range	1 mV to 1 V
Setting Resolution	1 mV
Accuracy	±0.2%
Stability	50 ppm/°C
Output Impedance	50 Ω
Sweep	
Amplitude Sweep	
Output Range	0.000 to 1.000 V rms
Law	Linear
Step Rate	20 Hz maximum (50 ms/step)
Frequency Sweep	
Output Range	0.5 Hz to 2.0 MHz
Law	Linear or Logarithmic
Step Rate	20 Hz maximum (50 ms/step)

Auxiliary Inputs

ADC 1, 2, 3 and 4	
Maximum Input	±10 V
Resolution	1 mV
Accuracy	±20 mV
Input Impedance	1 MΩ // 30 pF
Sample Rate	
ADC 1 only	40 kHz max.
ADC 1 and 2	17.8 kHz max.
Trigger Mode	Internal, External or burst
Trigger Input	TTL compatible

Model 7280 Specifications

Outputs

Main Analog (CH1 and CH2) Outputs

Function	X, Y, R, θ, Noise, Ratio, Log Ratio and User Equations 1 & 2.
Amplitude	±2.5 V full-scale; linear to ±300% full-scale
Impedance	1 kΩ
Update Rate:	
X, Y or R @ TC ≤4 ms	7.5 MHz
All outputs @ TC ≥5 ms	1 kHz

Signal Monitor

Amplitude	±1 V FS
Impedance	1 kΩ

Auxiliary D/A Output 1 and 2

Maximum Output	±10 V
Resolution	1 mV
Accuracy	±10 mV
Output Impedance	1 kΩ

8-bit Digital Port

0 to 8 lines can be configured as inputs, with the remainder being outputs. Each output line can be set high or low and each input line read to allow interaction with external equipment. Extra line acts as trigger input

Reference Output

Waveform	0 to 3 V rectangular wave
Impedance	TTL-compatible

Power - Low Voltage	±15 V at 100 mA rear panel 5-pin 180° DIN connector for powering ***SIGNAL RECOVERY*** preamplifiers

Data Storage Buffer

Size	32k × 16-bit data points, may be organized as 1×32k, 2×16k, 3×10.6k, 4×8k, etc.
Max Storage Rate	
From LIA	up to 1000 16-bit values per second
From ADC1	up to 40,000 16-bit values per second

User Settings

Up to 8 complete instrument settings can be saved or recalled at will from non-volatile memory

Interfaces

RS232 and GPIB (IEEE-488). A second RS232 port is provided to allow "daisy-chain" connection and control of up to 16 units from a single RS232 computer port

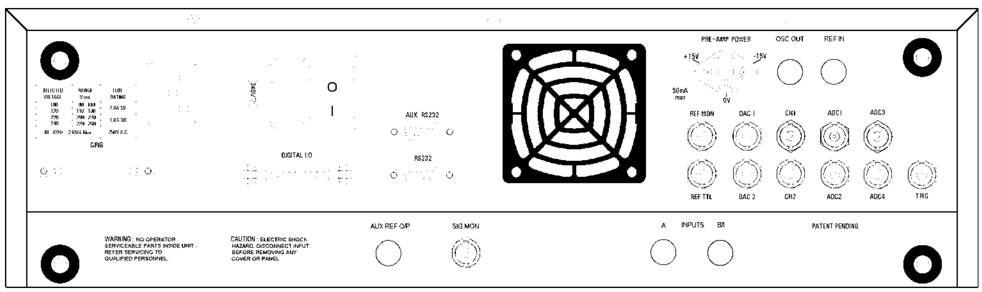
General

Power Requirements

Voltage	110/120/220/240 VAC
Frequency	50/60 Hz
Power	200 VA max

Dimensions

Width	17¼" (435 mm)
Depth	19" (485 mm)
Height	
With feet	6" (150 mm)
Without feet	5¼" (130mm)
Weight	25.4 lb (11.5 kg)



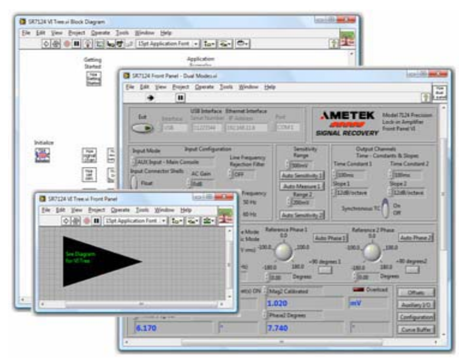
Model 7280 Rear Panel

SIGNAL RECOVERY Acquire Software (see page 56)

Users who do not wish to write their own control code but who still want to record the instrument's outputs to a computer file will find the ***SIGNAL RECOVERY*** Acquire Lock-in Amplifier Applications Software, available at a small extra cost, useful. This 32-bit package, suitable for Windows XP/Vista, extends the capabilities of the instrument by, for example, adding the ability to record swept frequency measurements. It also supports the internal curve buffer, allowing acquisition rates of up to 1000 points per second independent of the computer's processor speed.

LabVIEW Driver Software

A LabVIEW driver for the instrument is available from the **www.signalrecovery.com** website, offering example VIs for all its controls and outputs, as well as the usual Getting Started and Utility VIs. It also includes example soft-front panels built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.



Ordering Information

Each model 7280 is supplied complete with a comprehensive instruction manual. Users may download the instrument's LabVIEW driver software and a free demonstration copy, DemoAcquire, of the ***SIGNAL RECOVERY*** lock-in amplifier applications software package, from the **www.signalrecovery.com** website.

Optional Accessories

Model 7280/99	Extended frequency range (800 kHz) for Dual Reference and Dual Harmonic Modes
Model 7280/98	Extended frequency range (2.0 MHz) for Dual Reference and Dual Harmonic Modes
Acquire™	32-bit lock-in amplifier applications software for use with Windows XP/Vista operating systems
Model K02004	Rack mount to mount one model 7280 in a 19" rack

Model 7280BFP

Wide Bandwidth DSP Lock-in Amplifier with blank front-panel

FEATURES

- **0.5 Hz to 2 MHz operation**
- **Voltage and current mode inputs**
- **Direct digital demodulation without down-conversion**
- **7.5 MHz main ADC sampling rate**
- **1 µs to 100 ks output time constants**
- **Quartz crystal stabilized internal oscillator**
- **Harmonic measurements to 32F**
- **Dual reference, Dual Harmonic and Virtual Reference modes**

APPLICATIONS

- **OEM's**
- **Systems needing multiple detection channels**

DESCRIPTION

The model 7280BFP has exactly the same specifications as the standard 7280, but can only be operated from a remote computer. As such, it is ideal for use in complex systems both by end users and manufacturers where front-panel operation is not required.

All signal connections are made via the rear-panel BNC connectors giving an uncluttered appearance, especially when the unit is mounted in an equipment rack.

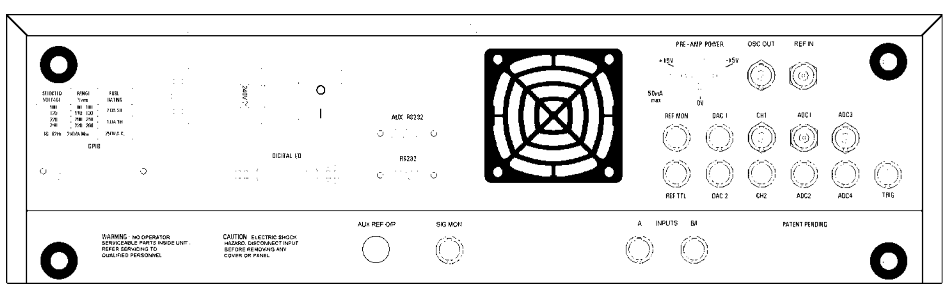
Control of the unit is via either the RS232 or GPIB interfaces, using simple mnemonic-type ASCII commands. A second RS232 port allows up to sixteen 7280BFP or compatible instruments to be operated from a single RS232 computer port by connecting them in a "daisy-chain" configuration.

Compatible software is available in the form of a LabVIEW driver supporting all instrument functions, the Acquire lock-in amplifier applications software and the SRInstComms ActiveX control and software toolkit. The driver and a demonstration version of the applications software, DemoAcquire, are available for download from our website at **www.signalrecovery.com**

Instruments are supplied complete with a C01003 RS232 cable and a simple Windows software package that allows the unit to be configured and a basic functional check to be preformed. This software also allows the user to install the 7280/99 and 7280/98 options.

Specifications

Please see the specifications for the model 7280 on pages 30-32



Model 7280BFP Rear Panel Layout

Why should you choose ***SIGNAL RECOVERY*** products?

Models 7280 and 7280BFP Wide Bandwidth DSP Lock-in Amplifiers

SIGNAL RECOVERY Product Features	Benefit to you
• They are the only commercially available 2 MHz genuine DSP lock-in amplifiers	Allows use in systems requiring short output time constants without problems caused by an insufficient number of samples per signal cycle
• Analog outputs updated at 7.5 MHz for use with time constants down to 1 µs	Ideal for scanned probe microscopy feedback control loops
• Spectral Display (Model 7280 only)	See in the frequency domain where interfering signals are and choose a quiet region for your reference frequency
• Dual Reference	Measure two signals at two different frequencies simultaneously, without the expense involved in buying two instruments
• Dual Harmonic	Measure two signals at two different harmonics simultaneously, without the expense involved in buying two instruments
• Curve Buffer Graphical Display	Strip chart mode display is invaluable for monitoring during manual adjustment of experiments
• Virtual Reference	Recover signals even without a reference
• Large high resolution electroluminescent display (Model 7280 only)	Excellent viewing angle for good visibility even across a crowded laboratory
• Easy to set controls with keypad and cursor movement keys (Model 7280 only)	Enter the exact setting you need without having to fiddle with a sensitive rotary knob. Move the cursors on the graphical display with ease
• User upgradeable firmware	Benefit from future firmware upgrades without having to send the instrument to a service facility
• 2-input multiplexing using A and -B inputs - even under computer control	Measure two signals sequentially under computer control using the same lock-in without having to switch connections
• 8 User Settings Memory (Model 7280 only)	Several users can share an instrument but keep their own personalized settings
• Internal Oscillator can be used independently of rest of instrument	Set OSC OUT to a different frequency to the reference e.g. Use it to control a ***SIGNAL RECOVERY*** chopper at *f* and then connect the lock-in's reference input to the chopper's *f*/10 SYNC output
• Auxiliary Digital Input and Output port	Eliminate the need for separate digital I/O cards when building complex computer controlled experiments
• Excellent LabVIEW driver	Saves programming time
• Compatible with Acquire software	Eliminates the need to develop programs
• Compatible with SRInstComms	Control the instrument from any ActiveX enabled programming language, such as Visual Basic, VBA (Excel, Word, Access) and VBScript (Internet Explorer)

Model 7210

Multichannel DSP Lock-in Amplifier

FEATURES

- **Up to 32 DSP dual phase lock-in amplifier channels operating in parallel**
- **Common reference frequency**
- **Independent per-channel control of sensitivity, reference phase and time constant**
- **Units may be interconnected to increase available channels**
- **Voltage or Current mode signal channel inputs**
- **Complete with software**

APPLICATIONS

- **Spectroscopy**
- **Magnetic measurements**
- **Superconductivity tests**
- **Impedance measurements**
- **Pump-probe experiments**

DESCRIPTION

The ***SIGNAL RECOVERY*** Model 7210 represents a significant advance in the application of DSP technology in the design of a lock-in amplifier. Until now, instruments have been restricted to a single signal channel, allowing only one, or at most two, signals to be measured at any one time. The model 7210, with its use of the latest technology, allows up to 32 signals to be measured simultaneously. What is more, units can be linked together to give more detection channels. For example, four units give 128 channels, while sixteen would give 512 channels.

The instrument can effectively operate as 32 parallel dual-phase lock-in amplifiers, running at the same external reference frequency, measuring 32 signals and generating 32 pairs of X and Y outputs. It can also operate in a *tandem* mode (see page 37) in which it generates a second reference signal which is an integer division of the external reference. This second reference is applied to the external experiment in such a way as to amplitude modulate the signal at the first reference frequency.

The amplitude modulation is detected by the first set of demodulators, which run at the external frequency, and then further demodulated by a second set of demodulators running at the generated reference frequency, to give a second set of X and Y outputs per channel. This detection method would previously have required two lock-in amplifiers connected in series, so in this mode the 32-channels of the 7210 are equivalent to 64 dual phase lock-in amplifiers. To date, no other lock-in amplifier matches this capability.

Specifications

General

Dual-phase 32-channel DSP lock-in amplifier operating over a reference frequency range of 20 Hz to 50.5 kHz. External Reference mode only. Independent control of sensitivity, AC Gain, reference phase and time constant on each channel.

Tandem and 2F detection modes. User-upgradeable firmware.

Measurement Modes

Single-frequency 32 channel dual-phase lock-in amplifier, running with an external reference frequency in the range 20 Hz to 50.5 kHz. Outputs in this mode are X1 and Y1 (in-phase and quadrature components) for each channel

Tandem-operation 32 channel dual-phase lock-in amplifier, running with a first, external reference frequency (the *carrier* frequency) in the range 20 Hz to 50.5 kHz and generating the second reference frequency by integer division of the first. The range of the second frequency is 0.001 Hz to 100 Hz. Outputs in this mode are X1 and Y1 of the carrier frequency and X2 and Y2 of the amplitude modulation of the carrier frequency by the second reference frequency.

Signal Channel

The signal input specifications depend on the type of signal board fitted, of which three are available:

7210/99 Signal Board - Voltage Mode Inputs

Voltage

Mode	Virtual Ground, floating
Connector	BNC

Impedance Shell to Ground
1 kΩ or 0 Ω - set by internal pin jumpers
Input Impedance 10 MΩ
Input Voltage Noise < 10 nV/√Hz at 1 kHz
Max Safe Input ± 12.0 V

Frequency Response over which following four specifications apply: 20 Hz to 51 kHz
Gain Accuracy Overall ± 0.5%
Gain Match between Channels ± 1.0%
Phase Accuracy Overall ± 2°
Phase Match between Channels ± 1°

Full-scale sensitivity 100 µV to 1 V rms in a 1-3-10 sequence (9 settings)

7210/98 Signal Board - Wide Bandwidth Current Mode Inputs
Current Input
Mode Virtual Ground, floating
Connector BNC
Impedance Shell to Ground
1 kΩ or 0 Ω - set by internal pin jumpers
Input Impedance ≤ 1 kΩ at 1 kHz to virtual ground
Input Current Noise < 150 fA/√Hz at 1 kHz
Max Safe Input ± 12.0 V

Frequency Response over which following four specifications apply: 20 Hz to 51 kHz
Gain Accuracy Overall ± 0.5%
Gain Match between Channels ± 1.0%
Phase Accuracy Overall ± 2°
Phase Match between Channels ± 1°

Full-scale sensitivity 100 pA to 1 µA rms in a 1-3-10 sequence (9 settings)

7210/97 Signal Board - Low Noise Current Mode Inputs
Current Input
Mode Virtual Ground, floating
Connector BNC
Impedance to Ground
1 kΩ or 0 Ω - set by internal pin jumpers
Input Impedance ≤ 1 kΩ at 1 kHz to virtual ground
Input Current Noise < 50 fA/√Hz at 1 kHz
Max Safe Input ± 12.0 V

Frequency Response over which following four specifications apply: 20 Hz to 5 kHz
Gain Accuracy Overall ± 0.5%
Gain Match between Channels ± 1.0%
Phase Accuracy Overall ± 2°
Phase Match between Channels ± 1°
Full-scale sensitivity 10 pA to 100 nA rms in a 1-3-10 sequence (9 settings)

Reference Channel
External Reference Input
Impedance 1 MΩ//35 pF
Level 250 mV to 2.5 V rms
Connector BNC
Frequency Range, f1
20 Hz to 50.5 kHz
Lock Acquisition Time
2 seconds max
Reference Phase Shifter (each channel)
Set Resolution 10 m°
Orthogonality 90° ± 0.001°

External Reference Frequency Meter
Resolution 1 Hz
Reference Output (Tandem frequency)
Frequency, f2 f1/n, where n, an integer, is calculated by the instrument to give a frequency as close as possible to a user-specified value in the range 0.001 Hz to 100 Hz
Amplitude > 3 V pk-pk square-wave
Impedance < 200 Ω
Connector BNC
Harmonic Detection f and 2f (2f in single -frequency operation only, 2f < 50.5 kHz)
Tandem Reference Frequency Meter
Resolution 0.001 Hz

Demodulator
Main ADC's, each channel
Type 12 bit
Sampling Rate 208 kHz < f_s< 250 kHz, synchronous to external reference (f1) frequency

Single-Frequency Operation
Time Constants 4 ms to 1 ks in 1-3-10 sequence (12 steps)
Slope 12 dB/octave
Type Synchronous digital FIR filters
Harmonic Rejection > 90 dB
Dynamic Reserve > 80 dB

Tandem-Frequency Operation
Applying to f1 outputs:-
Time Constants 4 ms to 1 ks in 1-3-10 sequence (12 steps)
Slope 12 dB/octave
Type Synchronous digital FIR filters

Applying to f2 outputs:-
Time Constants 30 ms to 1 ks in 1-3-10 sequence (11 steps)
Slope 12 dB/octave
Type Synchronous digital FIR filters
Harmonic Rejection > 90 dB
Dynamic Reserve > 80 dB

Data Outputs
The outputs available from the instrument are:-

Single Reference Mode:
X1 & Y1

Tandem Mode:
X1, Y1, X2 and Y2

All outputs are for each of 32 channels. Outputs can be read directly on receipt of a command, or stored on receipt of a GPIB trigger or the GET command for later readout. The output values can be read using commands generating binary or ASCII responses.

Interconnections
Instruments can be interconnected to provide more than 32 detection channels. Interconnections are via RG45 multipole connectors. Each instrument has a rear-panel switch to select whether the connectors function as outputs, in which case the unit is the "master", or inputs, when the unit is a "slave".

Indicators
Front-panel LEDs indicate the following conditions:-

Power On - a single LED which is lit when line power is applied and the unit switched on.

Communications Activity - indicates when command is being received and response is waiting to be read or being transmitted.

Master/Slave - when lit indicates that the instrument is set to function as a "master" and that its synchronizing signal connectors are configured as outputs

Internal Oscillator - reserved for future expansion

Reference Unlock - lights when no suitable reference is applied

Signal Channel Overload - a single LED warning of input or output overload in any one of the 32 channels. It is possible to identify via a computer status command which channel(s) is affected and the type of overload condition

General
Computer Interfaces
Type GPIB (IEEE-488) and RS232
Connectors Standard GPIB Centronics connector, 9-pin female RS232
Comms Settings Set by rear-panel DIP switches
Command Set ASCII commands for all instrument controls and data readout. Binary dump commands for data readout
Power Requirements
Voltage 100/120/220/240 V AC
Frequency 50 - 60 Hz
Power 200VA max
Dimensions
Width 446 mm
Height 3U (133.5 mm)
Depth 435 mm
Weight 12.5 kg

Single and Tandem Reference modes explained

Single Reference Mode

This is the conventional mode of operation common to all lock-in amplifiers. The instrument measures the amplitude of the components of the signal at its inputs that are in-phase and in quadrature (i.e. 90° out of phase) with an internally-generated sinusoidal demodulator signal. This demodulator signal is in turn phase locked to the applied external reference signal.

These two components are conventionally known as the X and Y channel outputs. All signal channels are measured with respect to the same external reference signal, so with a 32 channel instrument there are 64 output values.

Tandem Reference Mode

If an amplitude-modulated sinusoidal carrier signal is applied to a conventional lock-in amplifier operated at the carrier frequency and with its reference phase adjusted to yield zero Y channel output, then the X output signal will be the modulating signal. This only applies if the output time constant is sufficiently short to allow the modulation to pass.

If this X output signal is applied to a second lock-in amplifier, but this time running at the modulating frequency, then the second lock-in can directly measure the amplitude of the modulation.

Historically, this type of experiment would have required two instruments, with a physical cable coupling the X output of one to the input of the second. However, the 7210 includes this capability as a standard feature.

In order to allow the second lock-in amplifier's demodulator to run synchronously with the first, it is desirable for its reference frequency to be the result of an integer division of the first reference frequency. This condition is best satisfied by ensuring that the second reference frequency be internally generated by the instrument and made available via a connector so that it can be used as the source of modulation for the signal.

Consequently the 7210 is fitted with two reference connectors; **REF 1 IN** is used to apply the external reference frequency at which the first demodulation stage operates, and the second, **REF 2 OUT**, outputs a TTL reference waveform at the frequency of the second stage. The user can specify the divisor used to generate the second reference from the first.

It will be appreciated that in tandem mode there are four outputs per signal channel, an X and Y pair from the first stage and an X and Y pair from the second. To avoid confusion, the outputs from the first stage, even when the unit is operating in single reference mode, are referred to as X1 and Y1 and those from the second as X2 and Y2.

It can also be seen that in Tandem mode an instrument with 32 channels generates 128 output values.

Ordering Information

In view of the specialized nature of this product, the model 7210 is **currently available to special order only**, with instruments being individually configured to meet customer requirements.

The basic model 7210 will support up to eight signal boards, each with four signal channels. Three types of board are available:

7210/99 Signal Board - Voltage Mode Inputs (20 Hz - 51 kHz)

7210/98 Signal Board - Wide Bandwidth Current Mode Inputs (20 Hz - 51kHz)

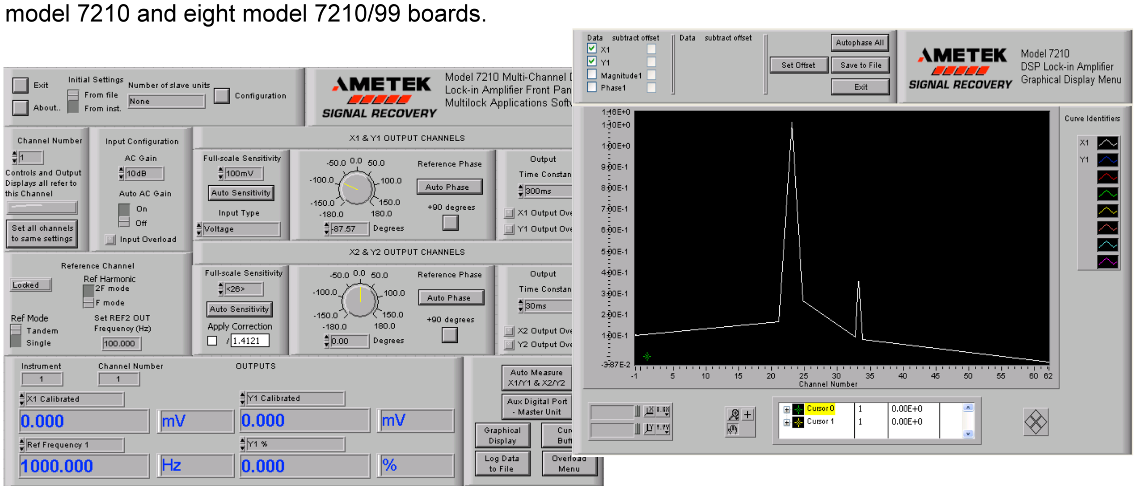
7210/97 Signal Board - Low Noise Current Mode Inputs (20 Hz - 5 kHz)

For example, a 32 channel unit with voltage mode inputs would require one model 7210 and eight model 7210/99 boards.

When multiple instruments are ordered together, 1 meter long interconnecting GPIB and reference link cables are supplied. Each instrument is of course supplied complete with a comprehensive instruction manual containing full programming information.

Software

A free LabVIEW driver that allows full instrument control is available for this instrument, which can be used *as issued* to control up to four instruments or incorporated into user programs. A fully compiled version of this program, MULTILOCK, is also available which also allows data to be saved directly to disk for later analysis using third-party software.



Multilock Applications Software

Model 7265

DSP Lock-in Amplifier

FEATURES

- **0.001 Hz to 250 kHz operation**
- **Voltage and current mode inputs**
- **Direct digital demodulation without down-conversion**
- **10 µs to 100 ks output time constants**
- **Quartz crystal stabilized internal oscillator**
- **Synchronous oscillator output for input offset reduction**
- **Harmonic measurements to 65,536F**
- **Dual reference, Dual Harmonic and Virtual Reference modes**
- **Spectral display mode**
- **Built-in experiments**

APPLICATIONS

- **Scanned probe microscopy**
- **Optical measurements**
- **Audio studies**
- **AC impedance studies**
- **Atomic force microscopy**

DESCRIPTION

The ***SIGNAL RECOVERY*** model 7265 uses the latest digital signal processing (DSP) technology to extend the operating capabilities of the lock-in amplifier to provide the researcher with a very versatile unit suitable both for measurement and control of experiments. At the same time due consideration has been given to the needs of those users wishing only to make a simple measurement quickly and easily.

Operating over a frequency range of 1 mHz to 250 kHz, the model 7265 offers full-scale voltage sensitivities down to 2 nV and current sensitivities to 2 fA. The instrument has a choice of operating modes, signal recovery or vector voltmeter, for optimum measurement accuracy under different conditions, and the use of DSP techniques ensures exceptional performance.

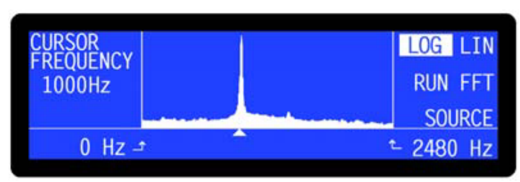
The instrument performs all of the normal measurements of a dual phase lock-in amplifier, measuring the in-phase and quadrature components, vector magnitude, phase angle and noise of the input signal.

Several novel modes of operation are also include to give greater levels of versatility than ever before, for example:

◆ **Virtual Reference™**
Under suitable conditions, this mode allows measurements to be made in the absence of a reference signal

◆ **Dual Reference**
In this mode the instrument can make simultaneous measurements on two signals at different reference frequencies, which is ideal, for example, for use in source compensated optical experiments

◆ **Spectral Display**
This allows the spectrum of the signals present at the input to be calculated and displayed, which can help when choosing the reference frequency



Spectral Display

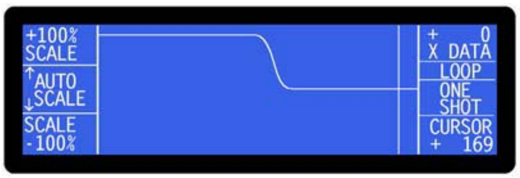
◆ **Transient Recorder**
In this mode, the auxiliary ADC inputs can be used as a 40 kSa/s (25 µs/point) transient recorder, with the captured transient being displayed graphically

◆ **Frequency Response**
This built-in experiment allows the internal oscillator frequency to be swept between preset frequencies, while simultaneously measuring the input signal magnitude and phase. The mode is ideal for determining the frequency and phase response of external networks

◆ Harmonic Analysis

Most lock-in amplifiers will measure signals at the applied reference frequency or its second harmonic. In the 7265, operation is possible at harmonics up the 65,536th, and in Dual Harmonic mode, simultaneous measurements can be made on two harmonics

Three auxiliary ADC inputs, one of which is a special integrating converter, four DAC outputs and eight output logic lines are provided. These can be used to record the magnitude of external signals associated with the experiment, such as temperature or pressure, or to generate voltages to control or switch other equipment. Information from the ADCs together with the lock-in amplifier's output data can be stored in the 32k point buffer memory, and even displayed graphically on screen.



Graphical Display

Pop-up Keypad to set Controls

The model 7265 is extremely easy to use. All instrument controls are adjusted using soft-touch, front panel push-buttons, with the present settings and measured outputs being displayed on the centrally located, cold fluorescent backlit dot-matrix LCD. A particularly convenient feature is the pop-up keypad which is used when setting controls that need adjusting to a large number of significant figures.

Control selection and adjustment is aided by the logical structure of on-screen menus and sub-menus, supported by a series of context-sensitive help screens. A number of built-in automatic functions are also provided to simplify instrument operation.

External control of the unit is via either the RS232 or GPIB interfaces, using simple mnemonic-type ASCII commands. A second RS232 port allows up to sixteen 7265 or compatible instruments to be operated from a single RS232 computer port by connecting them in a “daisy-chain” configuration.

Compatible software is available in the form of a LabVIEW driver supporting all instrument functions, and the Acquire lock-in amplifier applications software. The driver and a free demonstration version of the software, DemoAcquire, are available for download from our website at **www.signalrecovery.com**

Specifications

General

Dual-phase DSP lock-in amplifier operating over a reference frequency range of 0.001 Hz to 250 kHz.

Wide range of extended measuring modes and auxiliary inputs and outputs.

User-upgradeable firmware.

Measurement Modes

The instrument can simultaneously show any four of these outputs on the front panel display:

X	In-phase
Y	Quadrature
R	Magnitude
θ	Phase Angle
Noise	

Harmonic — nF, $n \leq 65{,}536$

Dual Harmonic
Simultaneously measures the signal at two different harmonics F_1 and F_2 of the reference frequency

Dual Reference
Simultaneously measures the signal at two different reference frequencies, F_1 and F_2 where F_1 is the external and F_2 the internal reference

Frequency Range for Dual Harmonic and Dual Reference Modes: F_1 and F_2 $\leq$ 20 kHz

Virtual Reference
Locks to and detects a signal without a reference (100 Hz $\leq$ F $\leq$ 250 kHz)

Noise
Measures noise in a given bandwidth centered at the reference frequency F

Spectral Display
Gives a visual indication of the spectral power distribution of the input signal in a user-selected frequency range lying between 1 Hz and 60 kHz. Note that although the display is calibrated in terms of frequency, it is not calibrated for amplitude. Hence it is only intended to assist in choosing the optimum reference frequency

Display

240 × 64 pixel cold fluorescent backlit LCD panel giving digital, analog bar-graph and graphical indication of measured signals. Menu system with dynamic key function allocation. On-screen context sensitive help

Signal Channel

Voltage Input

Modes	A only, -B only or Differential (A-B)
Full-scale Sensitivity	2 nV to 1 V in a 1-2-5 sequence
Max. Dynamic Reserve	> 100 dB
Impedance	
FET Input	10 MΩ // 30 pF
Bipolar Input	10 kΩ // 30 pF
Maximum Safe Input	20 V pk-pk
Voltage Noise	
FET Input	5 nV/√Hz @ 1 kHz
Bipolar Input	2 nV/√Hz @ 1 kHz
C.M.R.R.	> 100 dB @ 1 kHz
Frequency Response	0.001 Hz to 250 kHz
Gain Accuracy	±0.2% typ
Distortion	-90 dB THD (60 dB AC gain, 1 kHz)
Line Filter	attenuates 50, 60, 100, 120 Hz
Grounding	BNC shields can be grounded or floated via 1 kΩ to ground

Current Input

Mode	Low Noise or Wide Bandwidth
Full-scale Sensitivity	
Low Noise	2 fA to 10 nA in a 1-2-5 sequence
Wide Bandwidth	2 fA to 1 µA in a 1-2-5 sequence
Max. Dynamic Reserve	> 100 dB
Frequency Response (-3 dB)	
Low Noise	≥ 500 Hz
Wide Bandwidth	≥ 50 kHz
Impedance	
Low Noise	< 2.5 kΩ @ 100 Hz
Wide Bandwidth	< 250 Ω @ 1 kHz
Noise	
Low Noise	13 fA/√Hz @ 500 Hz
Wide Bandwidth	1.3 pA/√Hz @ 1 kHz
Gain Accuracy	± 0.6% typ, midband
Line Filter	attenuates 50, 60, 100, 120 Hz
Grounding	BNC shield can be grounded or floated via 1 kΩ to ground

Model 7265 Specifications (continued)

Reference Channel

TTL Input (rear panel)	
Frequency Range	0.001 Hz to 250 kHz
Analog Input (front panel)	
Impedance	1 MΩ // 30 pF
Sinusoidal Input	
Level	1.0 V rms*
Frequency Range	0.3 Hz to 250 kHz
Squarewave Input	
Level	250 mV rms*
Frequency Range	2 Hz to 250 kHz

*Note: Lower levels can be used with the analog input at the expense of increased phase errors

Phase Set Resolution	0.001° increments
Phase Noise at 100 ms TC, 12 dB/octave slope	
Internal Reference	< 0.0001° rms
External Reference	< 0.01° rms @ 1 kHz
Orthogonality	90° ±0.0001°
Acquisition Time	
Internal Reference	instantaneous acquisition
External Reference	2 cycles + 50 ms
Reference Frequency Meter Resolution	1 ppm or 1 mHz, whichever is the greater

Demodulator and Output Processing

Output Zero Stability	
Digital Outputs	No zero drift on all settings
Displays	No zero drift on all settings
Analog Outputs	< 5 ppm/°C
Harmonic Rejection	-90 dB
Output Filters	
X, Y and R outputs only	
Time Constant	10 μs to 640 μs in a binary sequence
Slope (roll-off)	6 dB/octave
All outputs	
Time Constant	5 ms to 100 ks in a 1-2-5 sequence
Slope	6, 12, 18 and 24 dB/octave
Synchronous Filter	Available for F < 20 Hz
Offset	Auto and Manual on X and/or Y: ±300% full-scale
Absolute Phase Measurement Accuracy	≤ 0.01°

Oscillator

Frequency	
Range	0.001 Hz to 250 kHz
Setting Resolution	1 mHz
Absolute Accuracy	± 50 ppm
Distortion (THD)	-80 dB @ 1 kHz and 100 mV rms
Amplitude (rms)	
Range	1 μV to 5 V rms
Setting Resolution	
1 μV to 4 mV	1 μV
4 mV to 500 mV	125 μV
500 mV to 2 V	500 μV
2 V to 5 V	1.25 mV
Accuracy	
> 1 mV	±0.3%, *F* ≤ 60 kHz, ±0.5%, *F* > 60 kHz
100 μV - 1 mV	±1%, *F* ≤ 60 kHz ±3%, *F* > 60 kHz
Stability	50 ppm/°C
Output Impedance	50 Ω
Sweep	
Amplitude Sweep	
Output Range	0.000 to 5.000 V rms
Law	Linear
Step Rate	20 Hz maximum (50 ms/step)
Frequency Sweep	
Output Range	0.001 Hz to 250 kHz
Law	Linear or Logarithmic
Step Rate	20 Hz maximum (50 ms/step)

Auxiliary Inputs

ADC 1 & 2	
Maximum Input	±10 V
Resolution	1 mV
Accuracy	±20 mV
Input Impedance	1 MΩ // 30 pF
Sample Rate	
ADC 1 only	40 kHz max.
ADC 1 and 2	17.8 kHz max.
Trigger Mode	Internal, External or burst
Trigger Input	TTL compatible
ADC 3	
Maximum Input	±10 V
Resolution	12 to 20 bit, depending on sampling time
Input Impedance	1 MΩ // 30 pF
Sampling Time	10 ms to 2 s, variable

Outputs

Fast Outputs	
Function	X and Y or X and Mag
Amplitude	±2.5 V full-scale; linear to ±300% full-scale
Impedance	1 kΩ
Update Rate	166 kHz
Main Analog (CH1 and CH2) Outputs	
Function	X, Y, R, θ, Noise, Ratio, Log Ratio and User Equations 1 & 2.
Amplitude	±10.0 V full-scale; linear to ±120% full-scale
Impedance	1 kΩ
Update Rate	200 Hz
Signal Monitor	
Amplitude	±10 V FS
Impedance	1 kΩ
Auxiliary D/A Outputs 1, 2, 3 and 4	
Maximum Output	±10 V
Resolution	1 mV
Accuracy	±10 mV
Output Impedance	1 kΩ
8-bit Digital Output Port	
8 TTL-compatible lines that can be independently set high or low to activate external equipment	
Reference Output	
Waveform	0 to 5 V rectangular wave
Impedance	TTL-compatible
Power - Low Voltage	±15 V at 100 mA rear panel 5-pin 180° DIN connector for powering ***SIGNAL RECOVERY*** preamplifiers

Data Storage Buffer

Size	32k × 16-bit data points, may be organized as 1×32k, 2×16k, 3×10.6k, 4×8k, etc.
Max Storage Rate	
From LIA	up to 1000 16-bit values per second
From ADC1	up to 40,000 16-bit values per second

User Settings

Up to 8 complete instrument settings can be saved or recalled from non-volatile memory

Interfaces

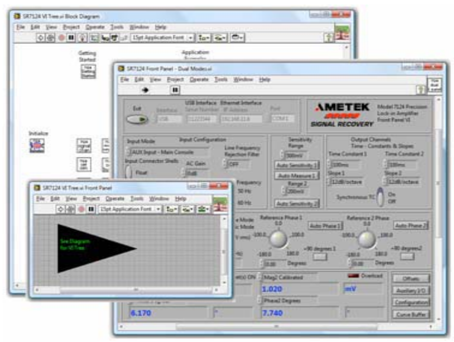
RS232 and GPIB (IEEE-488). A second RS232 port is provided to allow “daisy-chain” connection and control of up to 16 compatible instruments from a single RS232 computer port

General

Power Requirements	
Voltage	110/120/220/240 VAC
Frequency	50/60 Hz
Power	40 VA max
Dimensions	
Width	13¼" (350 mm)
Depth	16½" (415 mm)
Height	
With feet	4¼" (105 mm)
Without feet	3½" (90mm)
Weight	18 lb (8.1 kg)

LabVIEW Driver Software

A LabVIEW driver for the instrument is available from the **www.signalrecovery.com** website, offering example VIs for all its controls and outputs, as well as the usual Getting Started and Utility VIs. It also includes example soft-front panels built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.

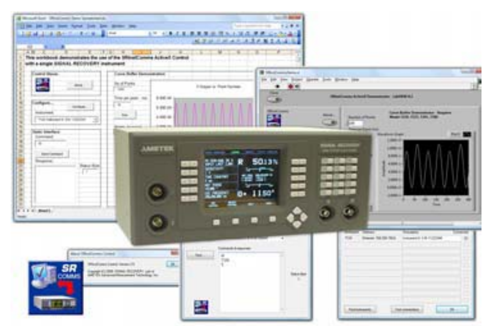


SIGNAL RECOVERY Acquire Software (see page 56)

Users who do not wish to write their own control code but who still want to record the instrument's outputs to a computer file will find the ***SIGNAL RECOVERY*** Instruments Acquire Lock-in Amplifier Applications Software, available at a small extra cost, useful. This 32-bit package, suitable for Windows XP/Vista, extends the capabilities of the instrument by, for example, adding the ability to record swept frequency measurements. It also supports the internal curve buffer, allowing acquisition rates of up to 1000 points per second independent of the computer's processor speed.

SRInstComms Software (see page 59)

Control up to ten ***SIGNAL RECOVERY*** instruments directly from Visual Basic, Visual C++, LabVIEW, Visual Basic for Applications (included in Word, Excel, Outlook, Access and other Microsoft products) and VBScript (supported by Internet Explorer 3 and later) without having to worry about low-level communications routines. The SRInstComms control handles all the communications between your software and the instrument(s) via the RS232 and/or GPIB interfaces, leaving you free to develop the code to run your experiment.

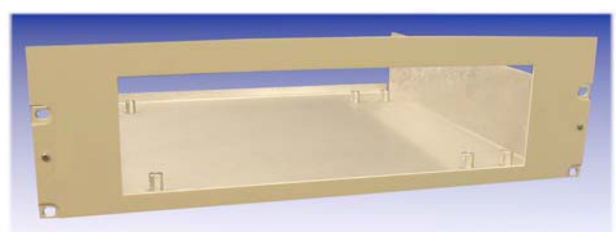


Ordering Information

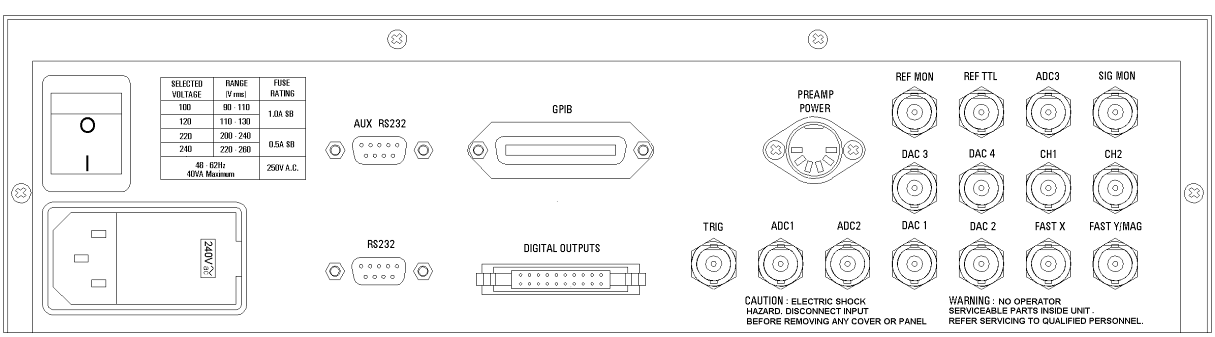
Each model 7265 is supplied complete with a comprehensive instruction manual. Users may download the instrument's LabVIEW driver software and a free demonstration copy, DemoAcquire, of the ***SIGNAL RECOVERY*** lock-in amplifier applications software package, from the **www.signalrecovery.com** website.

Optional Accessories

Model K02003 Rack mount to mount one model 7265 in a 19" rack



Model K02003 Rack Mount Kit



Model 7265 Rear Panel Layout

Why should you choose *SIGNAL RECOVERY* products?

Model 7265 DSP Lock-in Amplifier

SIGNAL RECOVERY **Product Features**	**Benefit to you**
◆ Physically compact	Saves valuable space in crowded laboratories
◆ Spectral Display	See in the frequency domain where interfering signals are and choose a quiet region for your reference frequency
◆ Dual Reference	Measure two signals at two different frequencies simultaneously, without the expense involved in buying two instruments
◆ Dual Harmonic	Measure two signals at two different harmonics simultaneously, without the expense involved in buying two instruments
◆ Curve Buffer Graphical Display	Strip chart mode display is good for monitoring during manual adjustment of experiments
◆ Virtual Reference	Recover signals even without a reference
◆ Easy to set controls - pop-up keypad	Enter the exact setting you need without having to fiddle with a sensitive rotary knob
◆ Experiments - frequency response	Perform complete swept-frequency response measurement and display the results graphically without having to write any program
◆ Transient Recorder	Capture the waveform of any signal at up to 40 kSa/s
◆ User upgradeable firmware	Benefit from future firmware upgrades without having to send the instrument to a service facility
◆ Synchronous Oscillator output	Allows input offset removal (see Applications Note AN1001 on page 123)
◆ 2-input multiplexing using A and -B inputs - even under computer control	Measure two signals sequentially under computer control using the same lock-in without having to switch connections
◆ 8 User Settings Memory	Several users can share an instrument but keep their own personalized settings
◆ Internal Oscillator can be used independently of rest of instrument	Set OSC OUT to a different frequency to the reference e.g. Use it to control a ***SIGNAL RECOVERY*** chopper at *f* and then connect the lock-in's reference input to the chopper's *f*/10 SYNC output
◆ Excellent LabVIEW driver	Saves programming time
◆ Compatible with Acquire software	Eliminates the need to develop programs
◆ Compatible with SRInstComms	Control the instrument from any ActiveX enabled programming language, such as Visual Basic, VBA (Excel, Word, Access) and VBScript (Internet Explorer)

Model 7225

DSP Lock-in Amplifier

FEATURES

- **0.001 Hz to 120 kHz operation**
- **Voltage and current mode inputs**
- **Direct digital demodulation without down-conversion**
- **10 µs to 100 ks output time constants**
- **Quartz crystal stabilized internal oscillator**
- **Synchronous oscillator output for input offset reduction**
- **Harmonic measurements to 32*f***

APPLICATIONS

- **Chopped light measurements**
- **AC bridge measurements**
- **Audio studies**
- **AC impedance studies**
- **Vibration studies**
- **Thermal wave detection**

DESCRIPTION

The ***SIGNAL RECOVERY*** model 7225 offers a cost-effective solution to the researcher needing the performance provided by DSP demodulation but not requiring the additional features or higher operating frequencies of the models 7280 and 7265.

The instrument performs all of the normal measurements of a dual phase lock-in amplifier, measuring the in-phase and quadrature components, vector magnitude, phase angle and noise of the input signal.

Two auxiliary ADC inputs, four DAC outputs and eight output logic lines are provided. These can be used to record the magnitude of external signals associated with the experiment, such as temperature or pressure, or to generate voltages to control or switch other equipment. Information from the ADCs together with the lock-in amplifier's output data can be stored in the 32k point buffer memory prior to transfer back to a controlling computer.

The model 7225 is extremely easy to use. All instrument controls are adjusted via the left-hand display panel and its associated keys, while the right hand panel shows the two selected instrument outputs. Auto functions need only two keypresses to activate and in many cases eliminate the need for manual control adjustment.

External control of the unit is via either the RS232 or GPIB interfaces, using simple mnemonic-type ASCII commands. A second RS232 port allows up to sixteen 7225 or compatible instruments to be operated from a single RS232 computer port by connecting them in a “daisy-chain” configuration.

Compatible software is available in the form of a LabVIEW driver supporting all instrument functions, the Acquire lock-in amplifier applications software and the SRInstComms ActiveX control and software toolkit. The driver and a demonstration version of the applications software, DemoAcquire, are available for download from our website at **www.signalrecovery.com**

Specifications

General

Dual-phase DSP lock-in amplifier operating over a reference frequency range of 0.001 Hz to 120 kHz. Wide range of auxiliary inputs and outputs and user-upgradeable firmware.

Measurement Modes

The instrument can simultaneously show any two of these outputs on the front panel display:

X	In-phase
Y	Quadrature
R	Magnitude
θ	Phase Angle
Noise	

Harmonic: *n*F, $n \leq 32F$

Noise

Measures noise in a given bandwidth centered at the reference frequency F

Displays

Two 2-line 16 character backlit LCD panels giving digital indication of measured signals

Signal Channel

Voltage Input

Modes	A only, -B only or Differential (A-B)
Full-scale Sensitivity	2 nV to 1 V in a 1-2-5 sequence
Max. Dynamic Reserve	> 100 dB
Impedance	
FET Input	10 MΩ // 30 pF
Bipolar Input	10 kΩ // 30 pF
Maximum Safe Input	20 V pk-pk
Voltage Noise	
FET Input	5 nV/√Hz @ 1 kHz
Bipolar Input	2 nV/√Hz @ 1 kHz

Model 7225 Specifications

Voltage Input (continued)

C.M.R.R.	> 100 dB @ 1 kHz
Frequency Response	0.001 Hz to 120 kHz
Gain Accuracy	±0.2% typ
Distortion	-90 dB THD (60 dB AC gain, 1 kHz)
Line Filter	attenuates 50, 60, 100, 120 Hz
Grounding	BNC shields can be grounded or floated via 1 kΩ to ground

Current Input

Mode	Low Noise or Wide Bandwidth
Full-scale Sensitivity	
Low Noise	2 fA to 10 nA in a 1-2-5 sequence
Wide Bandwidth	2 fA to 1 µA in a 1-2-5 sequence
Max. Dynamic Reserve	> 100 dB
Frequency Response (-3dB)	
Low Noise	≥ 500 Hz
Wide Bandwidth	≥ 50 kHz
Impedance	
Low Noise	< 2.5 kΩ @ 100 Hz
Wide Bandwidth	< 250 Ω @ 1 kHz
Noise	
Low Noise	13 fA/√Hz @ 500 Hz
Wide Bandwidth	1.3 pA/√Hz @ 1 kHz
Gain Accuracy	± 0.6% typ, midband
Line Filter	attenuates 50, 60, 100, 120 Hz
Grounding	BNC shield can be grounded or floated via 1 kΩ to ground

Reference Channel

TTL Input (rear panel)	
Frequency Range	0.001 Hz to 120 kHz
Analog Input (front panel)	
Impedance	1 MΩ // 30 pF
Sinusoidal Input	
Level	1.0 V rms*
Frequency Range	0.3 Hz to 120 kHz
Squarewave Input	
Level	250 mV rms*
Frequency Range	2 Hz to 120 kHz

*Note: Lower levels can be used with the analog input at the expense of increased phase errors

Phase Set Resolution	0.001° increments
Phase Noise at 100 ms TC, 12 dB/octave slope	
Internal Reference	< 0.0001° rms
External Reference	< 0.01° rms @ 1 kHz
Orthogonality	90° ±0.0001°
Acquisition Time	
Internal Reference	instantaneous acquisition
External Reference	2 cycles + 50 ms
Reference Frequency Meter Resolution	1 ppm or 1 mHz, whichever is the greater

Demodulator and Output Processing

Output Zero Stability	
Digital Outputs	No zero drift on all settings
Displays	No zero drift on all settings
Analog Outputs	< 5 ppm/°C
Harmonic Rejection	-90 dB
Output Filters	
X and Y outputs only:	
Time Constant	10 µs to 640 µs in a binary sequence
Slope (roll-off)	6 dB/octave
All outputs	
Time Constant	5 ms to 100 ks in a 1-2-5 sequence
Slope	6, 12, 18 and 24 dB/octave
Synchronous Filter	Available for F < 20 Hz
Offset	Auto and Manual on X and/or Y: ±300% full-scale
Absolute Phase Measurement Accuracy	≤ 0.01°

Oscillator

Frequency	
Range	0.001 Hz to 120 kHz
Setting Resolution	1 mHz
Absolute Accuracy	± 50 ppm
Distortion (THD)	-80 dB @ 1 kHz and 100 mV rms
Amplitude (rms)	
Range	1 mV to 5 V rms
Setting Resolution	
1 mV to 500 mV	1 mV
500 mV to 2 V	4 mV
2 V to 5 V	10 mV
Accuracy	±0.3%, *F* ≤ 60 kHz, ±0.5%, *F* > 60 kHz
Stability	50 ppm/ºC
Output Impedance	50 Ω
Sweep (computer control only)	
Amplitude Sweep	
Output Range	0.000 to 5.000 V rms
Law	Linear
Step Rate	20 Hz maximum (50 ms/step)
Frequency Sweep	
Output Range	0.001 Hz to 120 kHz
Law	Linear or Logarithmic
Step Rate	20 Hz maximum (50 ms/step)

Auxiliary Inputs

ADC 1 & 2	
Maximum Input	±10 V
Resolution	1 mV
Accuracy	±20 mV
Input Impedance	1 MΩ // 30 pF
Sample Rate	
ADC 1 only	40 kHz max.
ADC 1 and 2	17.8 kHz max.
Trigger Mode	Internal, External or burst
Trigger Input	TTL compatible

Outputs

Fast Outputs	
Function	X and Y
Amplitude	±2.5 V full-scale; linear to ±300% f.s.
Impedance	1 kΩ
Update Rate	166 kHz
Main Analog (CH1 and CH2) Outputs	
Function	X, Y, R, θ, Noise, Ratio, Log Ratio and User Equations 1 & 2.
Amplitude	±10.0 V full-scale; linear to ±120% full-scale
Impedance	1 kΩ
Update Rate	200 Hz
Signal Monitor	
Amplitude	±10 V FS
Impedance	1 kΩ
Auxiliary D/A Outputs 1, 2, 3 and 4	
Maximum Output	±10 V
Resolution	1 mV
Accuracy	±10 mV
Output Impedance	1 kΩ
8-bit Digital Output Port	8 TTL-compatible lines that can be independently set high or low to activate external equipment
Reference Output	
Waveform	0 to 5 V rectangular wave
Impedance	TTL-compatible
Power - Low Voltage	±15 V at 100 mA rear panel 5-pin 180° DIN connector for powering ***SIGNAL RECOVERY*** preamplifiers

Data Storage Buffer

Size	32k × 16-bit data points, may be organized as 1×32k, 2×16k, 3×10.6k, 4×8k, etc.
Max Storage Rate	
From LIA	up to 1000 16-bit values per second
From ADC1	up to 40,000 16-bit values per second

Interfaces

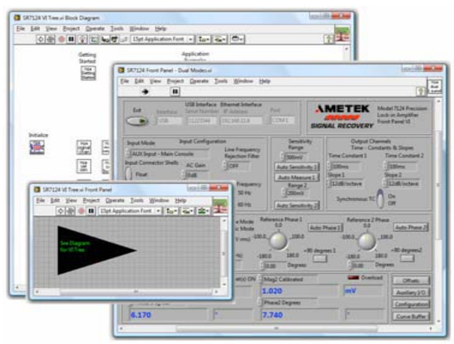
RS232 and GPIB (IEEE-488). A second RS232 port is provided to allow “daisy-chain” connection and control of up to 16 units from a single RS232 computer port

General

Power Requirements	
Voltage	110/120/220/240 VAC
Frequency	50/60 Hz
Power	40 VA max
Dimensions	
Width	17" (432 mm)
Depth	16½" (415 mm)
Height	
With feet	3" (74 mm)
Without feet	2¼" (60 mm)
Weight	16 lb (7.4 kg)

LabVIEW® Driver Software

A LabVIEW driver for the instrument is available from the **www.signalrecovery.com** website, offering example VIs for all its controls and outputs, as well as the usual Getting Started and Utility VIs. It also includes example soft-front panels built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.

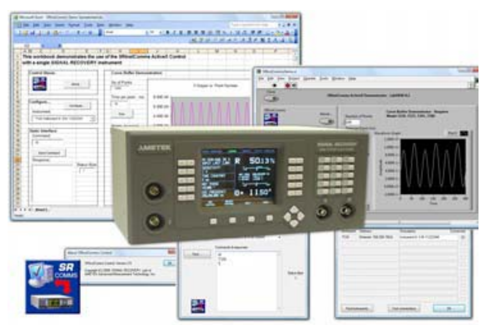


SIGNAL RECOVERY Acquire Software (see page 56)

Those users who do not wish to write their own control code but who still want to record the instrument's outputs to a computer file will find the ***SIGNAL RECOVERY*** Acquire Lock-in Amplifier Applications Software, available at a small extra cost, useful. This 32-bit package, suitable for Windows XP/Vista, extends the capabilities of the instrument by, for example, adding the ability to record swept frequency measurements. It also supports the internal curve buffer, allowing acquisition rates of up to 1000 points per second independent of the computer's processor speed.

SRInstComms Software (see page 59)

Control up to ten ***SIGNAL RECOVERY*** instruments directly from Visual Basic, Visual C++, LabVIEW, Visual Basic for Applications (included in Word, Excel, Outlook, Access and other Microsoft products) and VBScript (supported by Internet Explorer 3 and later) without having to worry about low-level communications routines. The SRInstComms control handles all the communications between your software and the instrument(s) via the RS232 and/or GPIB interfaces, leaving you free to develop the code to run your experiment.

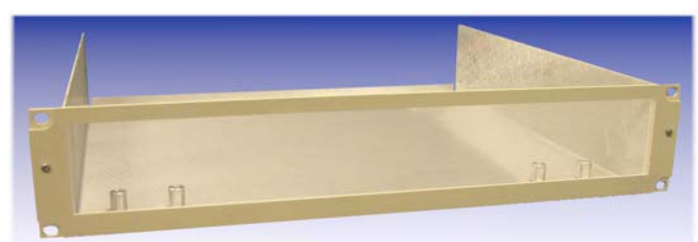


Ordering Information

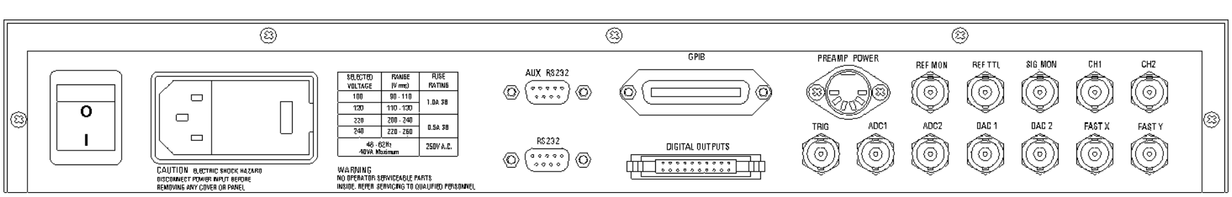
Each model 7225 is supplied complete with a comprehensive instruction manual. Users may download the instrument's LabVIEW driver software and a free demonstration copy, DemoAcquire, of the ***SIGNAL RECOVERY*** lock-in amplifier applications software package, from the **www.signalrecovery.com** website.

Optional Accessories

Model K02002 Rack mount to mount one model 7225 or 7225BFP in a 19" rack



Model K02002 Rack Mount Kit



Model 7225 Rear Panel Layout

Why should you choose *SIGNAL RECOVERY* products?

Model 7225 DSP Lock-in Amplifier

SIGNAL RECOVERY **Product Features**	**Benefit to you**
• Matches all the important specs of its closest competitor, but typically at a lower cost	Saves you money
• Physically compact	Saves valuable space in crowded laboratories
• Easy to set controls	Auto functions allow most signals to be measured with only two key presses
• User upgradeable firmware	Benefit from future firmware upgrades without having to send the instrument to a service facility
• Synchronous Oscillator output	Allows input offset removal (see Applications Note AN1001 on page 123)
• Internal Oscillator can be used independently of rest of instrument	Set OSC OUT to a different frequency to the reference e.g. Use it to control a ***SIGNAL RECOVERY*** chopper at *f* and then connect the lock-in's reference input to the chopper's *f*/10 SYNC output
• Transient Recorder (computer control)	Capture the waveform of any signal at up to 40 kSa/s
• Excellent LabVIEW driver	Saves programming time
• Compatible with Acquire software	Eliminates the need to develop programs
• Compatible with SRInstComms	Control the instrument from any ActiveX enabled programming language, such as Visual Basic, VBA (Excel, Word, Access) and VBScript (Internet Explorer)

Model 7225BFP DSP Lock-in Amplifier

SIGNAL RECOVERY **Product Features**	**Benefit to you**
• Cost-effective for quantity orders	Ideal for OEM users needing an instrument operating under computer control only
• Suitable for customized firmware or hardware	We are happy to quote for custom versions with special hardware and/or firmware. Capabilities similar to 7225 but in principle can include features from 7265 as well. Can be supplied finished in your company's color scheme
• Synchronous Oscillator output	Allows input offset removal (see Applications AN1001 on page 123)
• Internal Oscillator can be used independently of rest of instrument	Set OSC OUT to a different frequency to the reference e.g. Use it to control a ***SIGNAL RECOVERY*** chopper at *f* and then connect the lock-in's reference input to the chopper's *f*/10 SYNC output
• Transient Recorder (computer control)	Capture the waveform of any signal at up to 40 kSa/s
• Excellent LabVIEW driver	Saves programming time
• Compatible with SRInstComms	Control the instrument from any ActiveX enabled programming language, such as Visual Basic, VBA (Excel, Word, Access) and VBScript (Internet Explorer)

Model 7225BFP

DSP Lock-in Amplifier with blank front-panel

FEATURES

- **0.001 Hz to 120 kHz operation**
- **Voltage and current mode inputs**
- **Direct digital demodulation without down-conversion**
- **10 µs to 100 ks output time constants**
- **Quartz crystal stabilized internal oscillator**
- **Synchronous oscillator output for input offset reduction**
- **Harmonic measurements to 32F**

APPLICATIONS

- **OEM's**
- **Systems needing multiple detection channels**

DESCRIPTION

The ***SIGNAL RECOVERY*** model 7225BFP has exactly the same specifications as the standard 7225, but can only be operated from a remote computer. As such, it is ideal for use in complex systems both by end users and manufacturers where front-panel operation is not required.

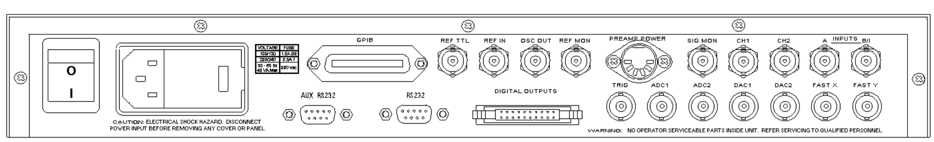
All signal connections are made via the rear-panel BNC connectors giving an uncluttered appearance, especially when the unit is mounted in an equipment rack.

Control of the unit is via either the RS232 or GPIB interfaces, using simple mnemonic-type ASCII commands. A second RS232 port allows up to sixteen 7225BFP or compatible instruments to be operated from a single RS232 computer port by connecting them in a "daisy-chain" configuration.

Compatible software is available in the form of a LabVIEW driver supporting all instrument functions, the Acquire lock-in amplifier applications software and the SRInstComms ActiveX control and software toolkit. The driver and a demonstration version of the applications software, DemoAcquire, are available for download from our website at **www.signalrecovery.com**

Specifications

Please see the specifications for the model 7225 on pages 43-45



Model 7225BFP Rear Panel Layout

Models 5209 and 5210

Single and Dual-Phase Analog Lock-in Amplifiers

FEATURES

- **0.5 Hz to 120 kHz operation**
- **Voltage and current mode inputs**
- **Continuous full-scale sensitivity control**
- **Sinewave or squarewave demodulation**
- **Powerful fourth-order signal channel Bandpass, Low Pass or Notch filter**
- **Up to 130 dB dynamic reserve**
- **Synchronous 15-bit ADC for lower output jitter**

APPLICATIONS

- **Auger spectroscopy**
- **Feedback control loops**
- **Replicating existing experimental setups**
- **Direct optical transmission/ reflection measurements**

DESCRIPTION

Over the past few years, the ***SIGNAL RECOVERY*** models 5209 (single-phase) and 5210 (dual-phase) have become the benchmark lock-in amplifiers against which others are judged. They are widely referenced in technical publications describing a diverse range of research applications including optical, electrochemical, electronic, mechanical and fundamental physical studies.

Although more recently the introduction of instruments using digital signal processing has brought advances in phase sensitive detection techniques, instruments using analog demodulators are still the first choice for many experiments. These include those requiring a true analog output, for example in some feedback control loops, or where the instrument is used to recover the envelope modulation of a carrier frequency. Of course, they are also chosen for compatibility with previous experimental setups.

Voltage or current inputs...
The instruments include a current preamplifier with two transimpedance settings and so can directly measure signals from current sources such as photodiodes. With an input impedance of down to typically only 25 Ω, the resulting voltage generated across the source by the signal current is minimized for the very best performance.

Continuous full-scale sensitivity control...
As with all lock-ins the models 5209 and 5210 have a range of calibrated full-scale sensitivity settings. However, unlike other units they also have a sensitivity vernier control, allowing the full-scale sensitivity to be set to any value between the calibrated values. Suppose you are performing an optical transmission experiment and you want to measure transmission in terms of a percentage relative to that of a “reference” sample. All you need to do is put the reference sample in the optical path and press the auto vernier control on the lock-in. It will then adjust the sensitivity so that the display reads 100%. Now replace the reference sample with the test sample and read the percentage transmission directly.

Unique Walsh Function Demodulators...
The simplest method of implementing the phase sensitive detector at the heart of an analog lock-in is with a reversing switch driven at the reference frequency, giving excellent linearity, dynamic range and stability. This is known as a “squarewave” demodulator since the instrument responds to signals not only at the reference frequency but also at its odd harmonics. It offers much better performance than can be achieved by using a true analog multiplier, which requires the synthesis of a very pure reference sinusoid and is very nonlinear when handling large levels of interfering signal.

Squarewave demodulation is ideal for many applications, such as experiments using chopped light beams where the signal being detected is a square-wave, since the odd

harmonics contain useful information. However in other cases the requirement is for “sinewave” or “fundamental” response where only signals at the reference frequency are measured.
In theory, a squarewave can be modified to a sinewave response by inserting a low-pass or bandpass filter in the signal channel ahead of the demodulator. However this requires a highly selective filter in order to reject signals at the third harmonic without at the same time causing significant phase and magnitude errors for signals at the reference frequency.

The ***SIGNAL RECOVERY*** models 5209 and 5210 use a modified form of switching demodulator, known as the Walsh demodulator, which multiplies the applied signal by a stepped approximation to the reference sinusoidal waveform. This gives a demodulator that does not respond to signals at the third and fifth harmonics, although it does respond to higher harmonics. A fourth-order signal channel filter is therefore included to reject these harmonics, giving an overall sinewave response. The advantages of the switching demodulator are thereby retained without the phase and magnitude errors associated with the use of highly selective filters.

The instruments can be switched to operate in either sinewave or squarewave mode, giving you the choice of the optimum detection method for your experiment. Only ***SIGNAL RECOVERY*** gives you this flexibility.

Choice of signal channel filter modes…
In the usual sinewave response mode, the filter is set to the bandpass or low-pass modes. But what if you are trying to measure a signal at twice the reference frequency in the presence of a strong signal at the reference frequency? In this case, the filter can be set to a notch (band-stop) mode and tuned to the reference frequency, leaving the signal at 2F unattenuated and easy to measure.

In addition to the main signal channel filter, a line-frequency rejection filter operating at 50/60 Hz and/or 100/120 Hz is also included, for elimination of troublesome line pickup.

High dynamic reserve...
The combination of the Walsh demodulator(s) and the signal channel filter gives the instruments a dynamic reserve of up to 130 dB - implying that you can, for example, measure a signal of 1 µV in the presence of an interfering signal of more than 1 V. No other analog lock-in amplifiers can deliver this performance.

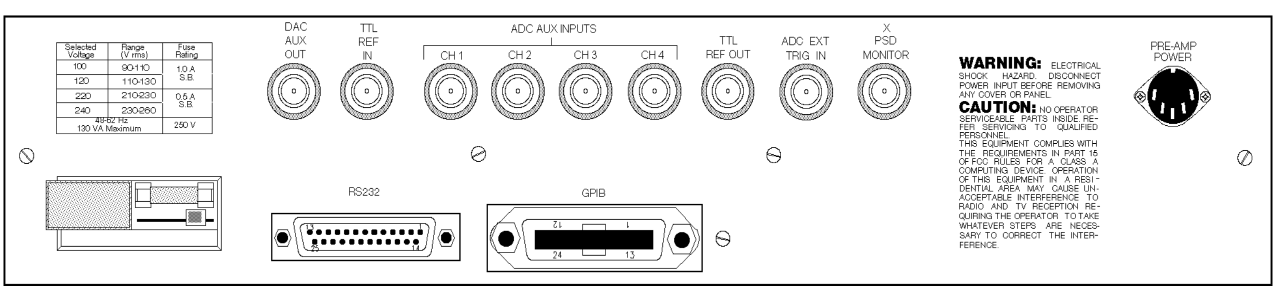
Output filters...
The output low-pass filters offer time constants in the range 1 ms to 3 ks, with all settings available at slopes of both 6 and 12 dB/octave. In addition, the instruments include a rear-panel connector giving the signal at the output of the in-phase (X-channel) demodulator with a time constant of typically only 100 µs, for use in those applications such as tandem demodulation where the largest output bandwidth is required.

Synchronous ADC trigger...
The analog outputs from the demodulator(s), after filtering by the output low-pass filter(s), need to be digitized by an analog to digital converter (ADC) for display or for transfer to the controlling computer. If this conversion is carried out asynchronously then the resulting values can display significant jitter. This is because the demodulator output contains not only the required DC level, but also signals at twice the reference frequency. When the output is sampled for conversion, this 2F signal means that some samples will be smaller and some larger than the mean.
Of course, the 2F component can be reduced to any arbitrarily small value by increasing the time constant, but this reduces the response time to changes in input signal, slowing down data throughput. The ***SIGNAL RECOVERY*** models 5209 and 5210 therefore offer a unique reference synchronous ADC trigger mode, which guarantees that the output is sampled at the same point in time relative to the reference waveform and thereby removes this source of error.

Internal oscillator...
With the models 5209 and 5210 there is no need to buy a separate oscillator to use as an excitation source for your experiment, since both instruments include one capable of generating a low distortion sinewave output signal over a frequency range of 0.5 Hz to 120 kHz. Although in most lock-ins the frequency of the internal oscillator can be adjusted, in the models 5209 and 5210 the amplitude can also be controlled over the range 1 mV to 2 V rms.

Manual or computer control...
In manual operation the backlit control setting indicators, the two digital displays and the analog panel meter make the instruments very easy to use, with the settings of all the important controls being instantly visible. Six auto functions further simplify control adjustment, while red overload and reference unlock LEDs warn of conditions which will result in measurement errors. All the front panel indicators can be turned off for use in blackout conditions.

The instruments include GPIB (IEEE-488) and RS232 computer interfaces, allowing virtually all the controls to be operated, and all the outputs that can be displayed to be read, via simple ASCII mnemonic-type commands. The communications interface parameters, such as baud rate and GPIB address are set by front-panel controls, with no difficult DIP switches to adjust.



Model 5210 Rear Panel Layout

Specifications

General

Single-phase (model 5209) and dual-phase (model 5210) analog lock-in amplifiers operating over a reference frequency range of 0.5 Hz to 120 kHz. Wide range of auxiliary inputs and outputs.

Measurement Modes

The model 5209 can show one of these outputs on the front panel display:

X	In-phase
Noise	
Ratio	X/ADC1
Log Ratio	Log_{10}(X/ADC1)

The model 5210 can also simultaneously show one of these outputs on the front panel display:

Y	Quadrature
R	Magnitude
θ	Phase Angle

Harmonic: F or 2F

Noise
Measures noise in a given bandwidth centered at the reference frequency F

Displays

Two 3½-digit LCD displays and analog panel meter

Signal Channel

Voltage Input

Modes	A only or Differential (A-B)
Full-scale Sensitivity	100 nV to 3 V in a 1-3-10 sequence and vernier adjustment
Max. Dynamic Reserve	> 130 dB
Impedance	100 MΩ // 25 pF
Maximum Safe Input	30 V pk-pk
Voltage Noise	5 nV/√Hz @ 1 kHz
C.M.R.R.	> 100 dB @ 1 kHz
Frequency Response	0.001 Hz to 120 kHz
Gain Accuracy	1% typical in Flat mode, 2% typical in tracking Bandpass mode
Gain Stability	200 ppm/°C typical
Distortion	-90 dB THD (60 dB AC gain, 1 kHz)
Grounding	BNC shields can be grounded or floated via 1 kΩ to ground

Current Input

Mode	10^{-6} A/V or 10^{-8} A/V
Full-scale Sensitivity	
10^{-6} A/V	100 fA to 3 µA in a 1-3-10 sequence and vernier adjustment
10^{-8} A/V	1 pA to 300 µA in a 1-3-10 sequence and vernier adjustment
Max. Dynamic Reserve	> 130 dB
Impedance	
10^{-6} A/V	< 250 Ω at 1 kHz
10^{-8} A/V	< 2.5 kΩ at 100 Hz
Maximum Input	15 mA continuous, 1 A momentary without damage. 10 µA AC pk-pk without saturation on 10^{-6} A/V; 100 nA AC pk-pk without saturation on 10^{-8} A/V
Noise	
10^{-6} A/V	130 fA/√Hz at 1 kHz
10^{-8} A/V	13 fA/√Hz at 500 Hz
Frequency Response	
10^{-6} A/V	-3 dB at 60 kHz
10^{-8} A/V	-3 dB at 330 Hz
Gain Accuracy	1% typical in Flat mode, 2% typical in tracking Bandpass mode
Gain Stability	200 ppm/°C typical
Grounding	BNC shield can be grounded or floated via 1 kΩ to ground

Signal Channel Filters

Line Frequency Rejection Filter	
Center frequency, F (factory set)	50/100 or 60/120 Hz
Mode	Off, F, 2F, F & 2F
Main Signal Channel Filter	
Mode	Fourth-order Low-pass, Bandpass, Notch or Flat (Disabled)
Frequency	Auto or Manual tuning
Signal Monitor	Front-panel BNC connector allows viewing of signal immediately ahead of the demodulator(s)

Reference Channel

TTL Input (rear panel)	
Frequency Range	0.5 Hz to 120 kHz
Analog Input (front panel)	
Impedance	1 MΩ // 30 pF
Sinusoidal Input	
Level	1.0 V rms*
Frequency Range	0.5 Hz to 120 kHz
Squarewave Input	
Level	250 mV rms*
Frequency Range	2 Hz to 120 kHz

*Note: Lower levels can be used with the analog input at the expense of increased phase errors

Phase Set Resolution	0.005° increments
Phase Set Accuracy	± 1°
Phase Noise	0.005° rms @ 1 kHz, 100 ms, 12 dB TC
Phase Drift	< 0.05°/°C
Orthogonality (model 5210 only)	± 0.5° above 5 Hz, degrading to ± 5° at 0.5 Hz
Acquisition Time	100 ms + 2 cycles max
Lock Indicator	LED warns of frequency/phase unlock

Demodulator and Output Processing

Mode: Sinewave (Walsh demodulator + BP/LP filter) or Squarewave

Zero stability/Dynamic Reserve

Mode	Dynamic Reserve Filter On	Filter Off	Zero Stability
High DR	130 dB	60 dB	500 ppm/°C
Normal	110 dB	40 dB	50 ppm/°C
High Stability	90 dB	20 dB	5 ppm/°C

Harmonic Rejection	> 80 dB with Low-pass, and > 60 dB with Bandpass main signal filter
Output Filters	
Time Constant	1 ms - 3 ks (1-3-10 sequence)
Roll -off	6 dB/oct or 12 dB/oct for all TC settings
Offset	Auto and Manual on X and/or Y: ±100% full-scale (±1000% full-scale with Expand on)

Oscillator

Frequency Range	0.5 Hz - 120 kHz
Amplitude Range	0 - 2 V rms (front panel or computer); 5 V rms fixed (computer only)
Amplitude Resolution	
0 - 500 mV	1 mV
500 mV - 2 V	4 mV
Distortion (THD)	0.5%
Output	sinewave from 900 Ω source

Auxiliary Inputs

ADC 1, 2, 3 and 4

Maximum Input	±15 V
Resolution	1 mV
Accuracy	±20 mV
Input Impedance	1 MΩ // 30 pF
Sample Rate	100 Hz
Trigger Mode	Internal, External or ref synchronous
Trigger Input	TTL compatible

Outputs

Demodulator Monitor	100 µs TC @ 6 dB/octave (5210: X output only)
Main Analog (CH1 and CH2) Outputs	
5209:	One ±10 V FS
5210:	Two ±10 V FS (X, Y or R, θ)
Resolution	1 mV
Impedance	1 kΩ
Update Rate	100 Hz
Expand	Expands X output by factor of 10
Auxiliary D/A Outputs	
5210	One output, ±15 V
5209	Two outputs, ±15 V
Resolution	1 mV

Accuracy	±10 mV
Output Impedance	1 kΩ
Reference Output	
Waveform	0 to 5 V rectangular wave
Impedance	TTL-compatible
Power - Low Voltage	±15 V at 45 mA rear panel 5-pin 180° DIN connector for powering ***SIGNAL RECOVERY*** preamplifiers

Interfaces

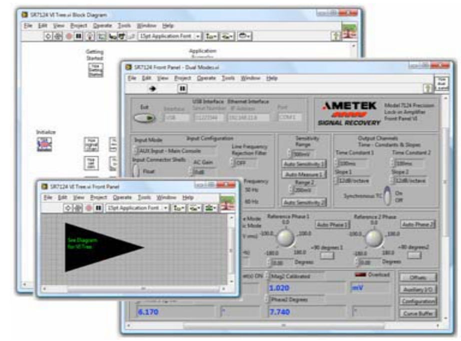
RS232 and GPIB (IEEE-488). All instrument controls except A, A-B, 10^{-6}A/V, 10^{-8}A/V and FLOAT/GND can be operated and all outputs that can be displayed can be read

General

Power Requirements	
Voltage	110/120/220/240 VAC
Frequency	50/60 Hz
Power	130 VA max
Dimensions	
Width	17¼" (440 mm)
Depth	19½" (500 mm)
Height	3½" (90 mm)
Weight	16.8 lb (7.6 kg)
Temperature Range	0 - 50°C
Rack Mounting	Hardware included

LabVIEW Driver Software

A LabVIEW driver for these instruments is available from the **www.signalrecovery.com** website, offering example VIs for all their controls and outputs, as well as the usual Getting Started and Utility VIs. It also includes example soft-front panels built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.



SIGNAL RECOVERY Acquire Software (see page 56)

Those users who do not wish to write their own control code but who still want to record the instrument's outputs to a computer file will find the ***SIGNAL RECOVERY*** Acquire Lock-in Amplifier Applications Software, available at a small extra cost, useful. This 32-bit package, suitable for Windows XP/ Vista, extends the capabilities of the instrument by, for example, adding the ability to record swept frequency measurements.

Ordering Information

Each model 5209 and 5210 is supplied complete with a comprehensive instruction manual. Users may download the instrument's LabVIEW driver software and a free demonstration copy, DemoAcquire, of the ***SIGNAL RECOVERY*** lock-in amplifier applications software package, from the **www.signalrecovery.com** website.

Why should you choose ***SIGNAL RECOVERY*** products?

Models 5209 and 5210 Analog Lock-in Amplifiers

SIGNAL RECOVERY Product Features	Benefit to you
• The benchmark analog lock-ins	It is likely that someone else has already successfully used one of these instruments in the same way as you intend
• Continuous full-scale sensitivity control	Set up your "100%" signal level and then press Auto Vernier to set the output display to 100%. Read % transmission values directly, saving calculation time
• Analog signal channel filtering	Exceptional dynamic reserve - up to 130 dB - means that these instruments can measure signals buried in noise when others can't
• Choice of filter modes	Notch filter is especially useful when measuring a signal at 2*f* in the presence of a strong signal at *f*
• Internal Oscillator can be used independently of rest of instrument.	Set OSC OUT to a different frequency to the reference e.g. Use it to control a ***SIGNAL RECOVERY*** chopper at *f* and then connect the lock-in's reference input to the chopper's *f*/10 SYNC output
• Excellent LabVIEW driver	Saves programming time
• Compatible with Acquire software	Eliminates the need to develop programs

Model 5105

Dual-Phase Analog Lock-in Amplifier Module

FEATURES

- **5 Hz to 20 kHz operation (or single "spot" frequency up to 100 kHz)**
- **Voltage mode input**
- **Squarewave demodulation**
- **Adjustable low-pass and high-pass signal channel filters**
- **Up to 80 dB dynamic reserve**
- **Complete with software**

APPLICATIONS

- **Chopped light measurements**
- **Multiple instrument systems**
- **Teaching the principles of phase-sensitive detection**

DESCRIPTION

The model 5105 is a compact dual-phase lock-in amplifier ideal for those signal recovery applications not demanding the performance offered by more sophisticated instruments in the ***SIGNAL RECOVERY*** range. It does not incorporate controls for manual operation but instead is operated entirely via an RS232 interface using simple ASCII character string commands. This approach allows the unit to be located closer to the signal source than is the case with PC card based instruments, thereby improving performance.

The instrument uses two switching type (squarewave) demodulators to measure the magnitude of the input signal in-phase (X) and in quadrature (Y) with the applied reference signal, and outputs both analog and digital representations of these values. The analog outputs are provided at front panel BNC connectors while the digital values, and in addition the resulting signal vector magnitude and phase angle, are available as responses to RS232 commands.

The signal channel includes high and low-pass filters which can be set to "bracket" the signal of interest thereby further improving the noise rejection, while the reference channel will operate from an external TTL or analog reference waveform.

Included with each instrument is a copy of **5105Acquire,** a simple but versatile software package supporting up to ten instruments for Windows PC, giving access to all the instrument's controls and outputs. In addition, LabVIEW drivers are available for users wishing to use that environment to develop their own control software.

Supplied complete with a separate line power supply and 9-pin RS232 cable, the model 5105 is ready to use "out of the box". Its low cost and high performance allows phase sensitive signal recovery techniques to be used in many new applications.

Specifications

General

Dual-phase analog lock-in amplifier operating over a reference frequency range of 5 Hz to 20 kHz, but also available calibrated for use at one user-specified spot frequency in the range 20 kHz to 100 kHz

Measurement Modes

The instrument can simultaneously measure these outputs:

X	In-phase
Y	Quadrature
R	Magnitude
θ	Phase Angle
Harmonic	F only

Signal Channel

Modes	Pseudo-differential
Grounding	BNC shield can be grounded or floated via 1 kΩ to ground using internal jumper
Full-scale Sensitivity	10 µV to 1 V in a 1-3.16-10 sequence (10 dB steps)
Max. Dynamic Reserve	> 80 dB
Impedance	10 MΩ // 30 pF
Maximum Safe Input	20 V pk-pk
Voltage Noise	< 30 nV/√Hz @ 1 kHz
C.M.R.R.	> 40 dB @ 1 kHz
Frequency Response	5 Hz to 100 kHz

Model 5105 Specifications

Input (continued)

Gain Accuracy — ± 2% typical for digital outputs; ± 6% typical for analog outputs
Gain Stability — 200 ppm/°C typical

Signal Channel Filters

High-pass Signal Channel Filter
- -3 dB frequency — 1 Hz, 10 Hz, 100 Hz or 1 kHz

Low-pass Signal Channel Filter
- -3 dB frequency — 50 Hz, 500 Hz, 5 kHz or 50 kHz

Frequency Accuracy — ± 10%

Reference Channel

Mode — TTL or Analog
Frequency Range — 5 Hz to 20 kHz
- Analog Impedance — 1 MΩ // 30 pF

Phase Set Resolution — 0.1° increments
Phase Set Accuracy — ± 1°
Phase Noise — ≤ 0.015° rms @ 1 kHz, 100 ms, 12 dB TC; ≤ 0.007° rms @ 10 kHz, 100 ms, 12 dB TC
Phase Drift — < 0.05°/°C
Orthogonality — ± 1°
Acquisition Time — 1 s + 2 cycles max

Demodulator and Output Processing

Mode — Squarewave switching demodulator + HP/LP filters

Zero stability/Dynamic Reserve

Mode	Dynamic Reserve (Filters Off)	Zero Stability
High DR	46 dB	500 ppm/°C
Normal	26 dB	100 ppm/°C
High Stability	6 dB	40 ppm/°C

Output Filters
- Time Constants
  - Analog and Digital Outputs
    - Fast Mode — 300 µs, 1 ms, 3 ms or 10 ms (316 µV to 1 V FS sensitivity)
    - Normal Mode — 30 ms, 100 ms, 300 ms or 1 s
  - Digital Outputs only — 3 s and 10 s
- Accuracy — ±10%
- Slope — 6 dB/octave or 12 dB/octave

Offsets — ±20% full-scale, X and/or Y

Outputs

Main Analog (X and Y) Outputs
- Amplitude — ±1 V FS
- Impedance — 1 kΩ

Signal Monitor — 10 V pk-pk maximum
Reference Output
- Waveform — 0 to 5 V rectangular wave
- Impedance — TTL-compatible

Interface

Type
RS232 via 9-pin D type plug, configured as a DTE device. Two ports are provided allowing up to sixteen model 5105 or compatible instruments to be controlled from a single computer port

Parameters (fixed)
4800 baud, no parity, 8 data bits and 1 stop bit

Addressing
Rear panel rotary switch assigns a unique address to each instrument

Controls
Sensitivity, High and Low-Pass Filter settings, Dynamic Reserve, Reference Phase, Time Constant and Slope can all be set and read via RS232 command

Auto Functions
Auto-Phase and Auto-Offset

Data Transfer Rate
6 - 8 readings per second typical

Outputs
- X and Y — Max count = ±1200 (±1000 = FS)
- Magnitude — Max count = 1200 (1000 = FS)
- Signal Phase — Max count = ±1800, corresponding to ±180°
- Ref Frequency — Response in millihertz

General

Software & RS232 Cable

5105Acquire, a full applications package for IBM PC or 100% compatible computer and supporting up to ten instruments, is supplied with each unit. This package allows access to all instrument controls and displays two selected instrument outputs.

In addition, a LabVIEW driver suitable for version 4.01 and later of LabVIEW is available by download from our website at **www.signalrecovery.com**

The instrument is also compatible with the full ***SIGNAL RECOVERY*** Acquire Lock-in Amplifier Applications software. A free demonstration version can be downloaded from the above website.

2 meter null-modem cable suitable for connecting the instrument to a 9-pin D-type RS232 plug on a PC computer is also included

Power Requirements — +18 V DC unregulated @ 300 mA; -18 V DC unregulated @ 80 mA
A separate power supply (model PS0108) suitable for 110 V 60 Hz or 230 V 50 Hz operation is supplied with each instrument

Dimensions
- Width — 8¼" (209 mm)
- Depth — 10½" (266 mm)
- Height — 3½" (85 mm)

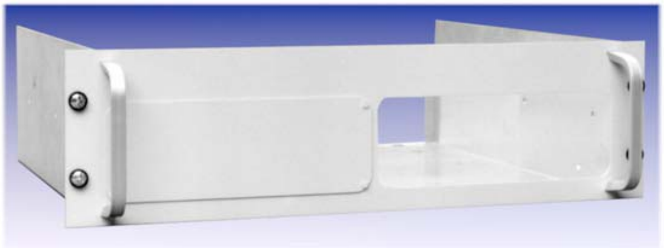
Weight — 5 lb (2.3 kg)

Remote Line Power Supply
Model PS0108

included with each instrument

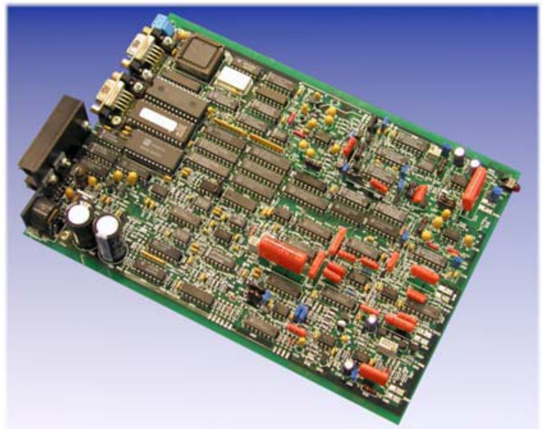
Rack Mount Kit
Model K0304

Allows 1 or 2 model 5105 lock-in amplifiers to be mounted in a standard 19" rack.



Model 5106

Dual-Phase Analog Lock-in Amplifier PCB Assembly



FEATURES

- **5 Hz to 20 kHz operation (or single "spot" frequency up to 100 kHz)**
- **Voltage mode input**
- **Squarewave demodulation**
- **Adjustable low-pass and high-pass signal channel filters**
- **Up to 80 dB dynamic reserve**
- **Complete with software**

APPLICATIONS

- **Chopped light measurements**
- **Multiple instrument systems**
- **Teaching the principles of phase-sensitive detection**
- **OEM's**

DESCRIPTION

The model 5106 is the tested printed circuit board assembly as used in the model 5105 (see page 52) and therefore has the same specifications as that instrument. It is especially suitable for OEM or multiple instrument use where the user is able to provide an appropriate enclosure and the necessary unregulated DC power supply.

Signal, reference and analog output connections are made either by connector or soldering to Berg pins mounted at 0.1" centers on the board. Power should be supplied to the on-board 5-pin 180º DIN connector from a remote source. The ***SIGNAL RECOVERY*** model PS0108 is available as an optional extra for this purpose.

Included with each unit is a copy of **5105Acquire,** a simple but versatile software package supporting up to ten instruments for an IBM PC or compatible computer, giving access to all the instrument's controls and outputs. In addition, LabVIEW drivers are available for users wishing to use that environment to develop their own control software.

Supplied complete with a 9-pin RS232 null modem cable, the card is ready to use once power is applied. Its especially low cost and high performance mean that the use of phase sensitive signal recovery techniques becomes cost-effective in even more situations than ever before.

Specifications

Specifications are the same as for the model 5105 (page 52-53), except as follows:-

General

Power Requirements	+18 V DC unregulated @ 300 mA -18 V DC unregulated @ 80 mA	Dimensions	
		Width	6½" (167 mm)
		Depth	9¼" (233 mm)
		Height	1½" (40 mm)
		Weight	14oz (400 g)

Optional Remote Line Power Supply
Model PS0108
suitable for use with model 5106

Why should you choose ***SIGNAL RECOVERY*** products?

Model 5105 Dual Phase Analog Lock-in Amplifier Module

SIGNAL RECOVERY Product Features	Benefit to you
◆ Low cost module	Saves you money
◆ Ideal for teaching applications	Students learn the advantages of lock-in detection and, having done this can move on to develop their own simple data acquisition and analysis program to control the instrument
◆ Genuine analog outputs	When used as part of feedback loop, the experiment can be designed to be unconditionally stable
◆ Switching-type demodulator	Response matches square wave signals generated by chopped light experiments, giving outputs nearly a fifth bigger for the same signal than with sinusoidal responding instruments
◆ Daisy Chain RS232	Multiple instruments can be operated from a single RS232 port, avoiding the expense of a GPIB card and cables
◆ Excellent LabVIEW driver	Saves programming time
◆ Complete with operating software and compatible with the full Acquire package	Eliminates the need to develop programs
◆ Compatible with SRInstComms	Control the instrument from any ActiveX enabled programming language, such as Visual Basic, VBA (Excel, Word, Access) and VBScript (Internel Explorer)

Model 5106 Dual Phase Analog Lock-in Amplifier PCB Assembly

SIGNAL RECOVERY Product Features	Benefit to you
◆ Lowest cost ***SIGNAL RECOVERY*** lock-in	Ideal for incorporating into larger systems and for OEM use
◆ Genuine analog outputs	When used as part of feedback loop, the experiment can be designed to be unconditionally stable
◆ Switching-type demodulator	Response matches square wave signals generated by chopped light experiments, giving outputs nearly a fifth bigger for the same signal than with sinusoidal responding instruments
◆ Daisy Chain RS232	Multiple instruments can be operated from a single RS232 port, avoiding the expense of a GPIB card and cables
◆ Excellent LabVIEW driver	Saves programming time
◆ Complete with operating software and compatible with the full Acquire package	Eliminates the need to develop programs
◆ Compatible with SRInstComms	Control the instrument from any ActiveX enabled programming language, such as Visual Basic, VBA (Excel, Word, Access) and VBScript (Internel Explorer)

Acquire

Data Acquisition Software

New Version

FEATURES

- Operates with all current ***SIGNAL RECOVERY*** Lock-in Amplifiers, Boxcar Averagers, and the Model 5113 Preamplifier
- Suitable for Windows XP/ Vista
- Remote Front Panel mode
- Experiment Recording mode - take data versus time, frequency or auxiliary ADC values
- Input and output triggers
- Method and Data storage
- ASCII text export utility
- GPIB or RS232 operation
- Free demonstration version available

APPLICATIONS

- Record outputs versus time
- Frequency response measurements
- Transient recording
- Remote control of instruments

DESCRIPTION

Acquire is a comprehensive data acquisition package designed to operate most current and many former ***SIGNAL RECOVERY*** instruments from a personal computer. It is suitable for use with all our lock-in amplifiers, boxcar averager, and 5113 preamplifier, and operates via Ethernet, USB, RS232, or GPIB (IEEE-488) interfaces. For most users, the software eliminates the need for them to write control software, so that they can concentrate on the task of taking data. It will also prove invaluable for others who simply want to operate an instrument from a remote location or who wish to integrate their instrument with other computer controlled systems. Up to ten instruments can be controlled at the same time.

The package provides two principal modes of operation. First, in remote front panel mode virtually all of the functions of the connected instrument(s) can be controlled from the computer via a series of simple dialogs. The software is instrument sensitive and adjusts the content of these dialogs automatically to reflect the measurement capabilities and functions available in the connected unit. The data outputs to be displayed can be chosen from the range available and these are then clearly shown on-screen.

The second mode, experiment recording, allows selected instrument outputs to be recorded as a function of time, with the additional option of sweeping certain outputs (e.g. oscillator frequency, auxiliary DAC voltage, digital filter frequency, digital delay and/or digital port setting) as the experiment proceeds. When used with a lock-in amplifier, any auxiliary ADC inputs can be configured as trigger inputs, allowing data to be logged as function of external trigger events.

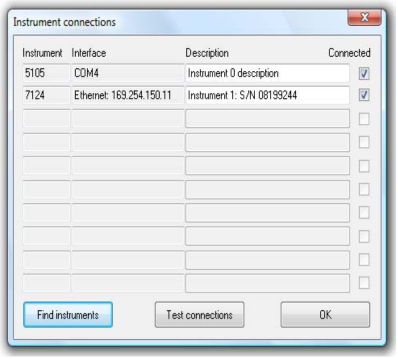
As data is acquired, it is displayed on screen and can be printed, as well as being saved for later use. Displayed plots can use a variety of line formats, while four curve cursors allow direct readout of measured values. However, with the very wide range of applications in which SIGNAL RECOVERY instruments can be used, it is not possible to anticipate every possible format in which the acquired data will be displayed. Hence many users take advantage of the export function to save the data to disk for display and/or further manipulation using other software.

A comprehensive help system is built in and free support is available to registered users.

Specifications

Compatible Instruments

Acquire will operate the ***SIGNAL RECOVERY*** Models 4161A, 5105, 5106, 5113, 5110(A), 5209, 5210, 7124, 7220, 7260, 7225, 7225BFP, 7265, 7270, 7280, 7280BFP, 7310, and 9650A. Up to ten instruments can be operated simultaneously.



Instrument Connections Dialog

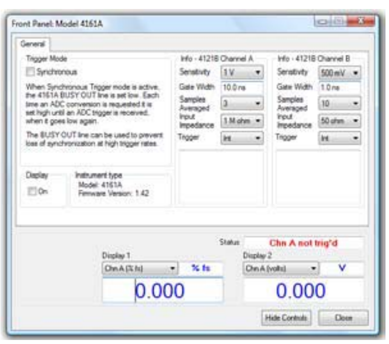
Capabilities

Instrument Connection

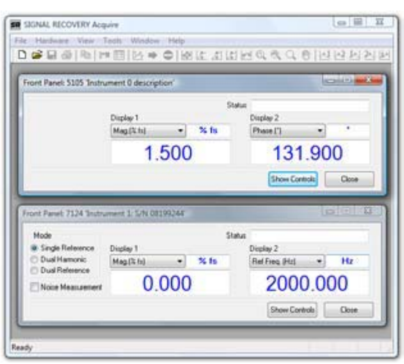
The package automatically detects compatible instruments connected via Ethernet, USB, RS232 or GPIB interfaces and displays a connections dialog where the instruments can be allocated meaningful names.

Remote Front Panel

All functions of the connected instrument(s) may be controlled remotely, with selectable on-screen display of outputs from those available. The display updates regularly, depending on speed of computer but typically at 2 - 3 Hz. The control panel can be shown in two sizes, one with tabs for the instrument controls and the second with just the output meter displays.



Remote Front Panel - Controls and Outputs Displayed



Remote Front Panel - Outputs for two instruments displayed

Front panel operation of the connected instrument(s) is inhibited while the software is running to prevent unauthorized interference with settings.

Define Experiment

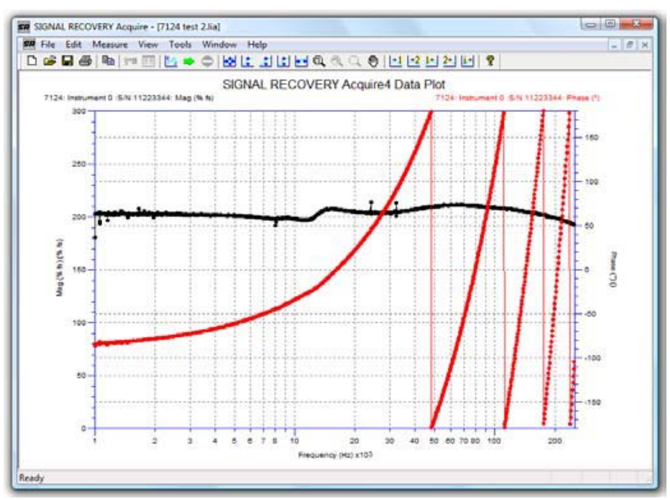
Users can define an experiment in which Y-axis data will be recorded as a function of an X-axis variable. The X-axis may be chosen as follows:-

- *Models 5105, 5106, 9650A, 5113*
  Time only - Data acquisition may only be initiated from the software.
- *Model 7310*
  Time, digital filter frequency, digital output port value, and trigger events.
- *All others*
  Time, oscillator frequency, oscillator amplitude, auxiliary DAC output voltage, digital output port value, and trigger events. Data acquisition can initiated directly from the software or on receipt of a trigger, and can then either free-run or be on the basis of one point per trigger.

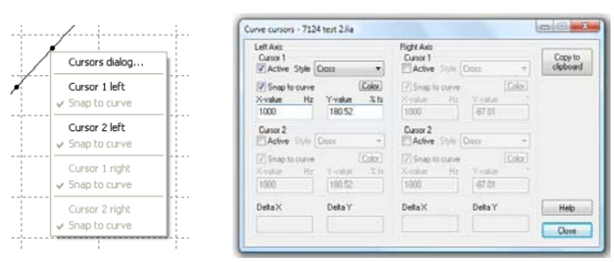
The Y-axis data to be recorded is selected from the outputs provided by the instrument(s). Hence, for example, dual phase lock-ins may record X, Y, Magnitude and Phase outputs; the 7310 Noise Rejecting Voltmeter can record output voltage, maximum and minimum outputs; the 4161A can record Channel 1 and Channel 2 voltage. Between one and eight outputs can be recorded in a given experiment.

File Storage and Data Display

Acquired data may be stored and recalled from disk, and displayed on user-adjustable axes. The line format used on plots can be selected, and four curve cursors allow direct readout of data point values.



Typical Data Plot



Curve cursors for easy readout of data values

Data plots may be manipulated for optimum display prior to printing, and can be copied to the clipboard for subsequent pasting into other applications.

Data can also be exported to ASCII text files suitable for import to third party software to allow further analysis.

Points	Time s	7265: Instrument 0 description: Osc Frequency Hz			7265: Instrument 0 description: Mag.(% fs) % fs	7265: Instrument 0 description: Phase (°) °
0	0	1.00E+01			8.91E+01	-1.11E+00
1	0.28	1.19E+01			9.92E+01	-9.83E+00
2	0.5	1.38E+01			9.87E+01	-1.24E+01
3	0.72	1.57E+01			9.79E+01	-1.49E+01
4	1.05	1.76E+01			9.69E+01	-1.87E+01
5	1.27	1.95E+01			9.63E+01	-2.01E+01
6	1.54	2.14E+01			9.52E+01	-2.35E+01
7	1.82	2.33E+01			9.40E+01	-2.64E+01

Export Data as ASCII Text Files

Ordering Information

Acquire includes the software supplied on CD and a 83-page instruction manual. It is also possible to download the full program from the **www.signalrecovery.com** website. When installed, this runs in a demonstration mode, known as DemoAcquire, but can be converted to the full program by purchasing an Activation Code.

Acquire is licenced for use on a single computer; for multiple or redistribution licenses please contact us first.

Optional Accessories

Model CE0114S	National Instruments PCI-GPIB Interface Board
Model CE0115S	National Instruments USB-GPIB Interface Cable
Model CE0116S	USB-RS232 Serial Adaptor
Model SC0073	2m GPIB cable
Model SC0067	4m GPIB cable
Model SC0066	1m GPIB cable
Model C01001	9F - 9F Null Modem RS232 cable (for models 5105 and 5106)
Model C01002	9F - 25M Null Modem RS232 cable (for models 5109, 5110, 5209 and 5210)
Model C01003	9F - 9M Null Modem RS232 cable (for models 7124, 7220, 7260, 7225, 7225BFP, 7265, 7270, 7280 and 7280BFP)
Model K02001	25F - 9M RS232 adapter.
Model 796190	6' (2 meter) long USB type A to type B cable for connecting model 7124 and 7270 to the USB port on a computer

Free Demonstration Version

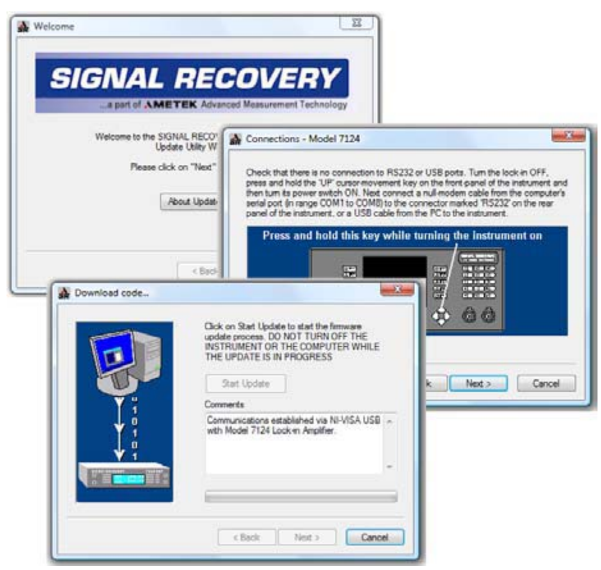
We offer a version of the program, DemoAcquire, which allows you try out the software and decide whether or not the full version will meet your needs. You can download it and the instruction manual from our website at **www.signalrecovery.com**

Firmware Updates

for DSP Lock-in Amplifiers

The operating firware in all ***SIGNAL RECOVERY*** DSP lock-in amplifierscan be updated to the latest version by downloading Update Packs from the **www.signalrecovery.com** website. Each pack contains the firmware, release notes, installation program, and full instructions making it a very simple task to keep your instrument completely up to date. All that is required in addition is a Windows PC and a suitable RS232 or USB cable.

All models can be updated via the RS232 interface; the models 7124 and 7270 can in addition be updated via USB.

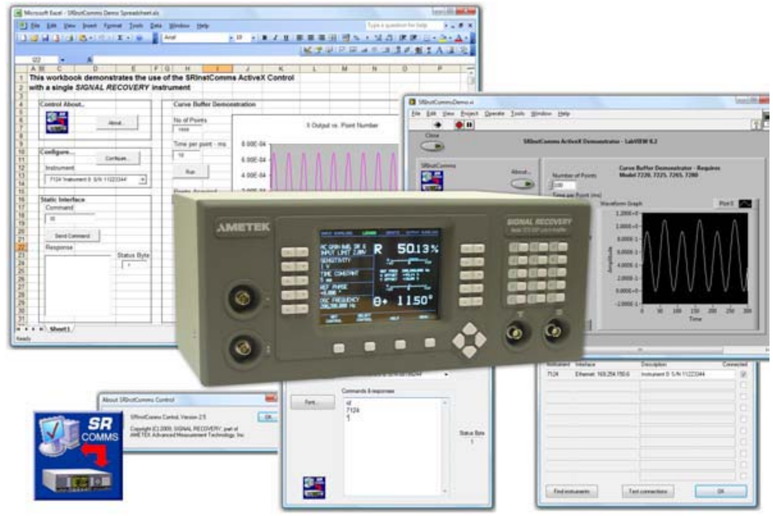


Firmware Update Utility

SRInstComms

ActiveX Control for *SIGNAL RECOVERY* instruments

New Version



FEATURES

- Easy PC control for compatible instruments
- Operates from one to ten instruments simultaneously
- Uses GPIB, RS232, USB, or Ethernet interface
- Automatic detection of instruments
- Supplied with example programs in Visual Basic, Visual C++, Visual C#, LabVIEW, Excel and HTML.
- Full printed and on-screen documentation

APPLICATIONS

- Direct instrument control and output charting from an Excel spreadsheet
- Experiments using multiple instruments
- Remote control systems
- Data acquisition direct to an Access database
- Web-based test systems

DESCRIPTION

SRInstComms is an ActiveX control that allows users of ***SIGNAL RECOVERY*** instruments to control them from PC's running Windows XP/Vista. Unlike the Acquire applications software which we also offer (see page 56), it is not a complete package but rather a component conforming to recognized industry standards that allows instruments to be controlled by user-developed programs. The only requirement is that these programs must be written in a language that supports such controls, which in practice is virtually all modern languages capable of developing Windows applications.

The control takes care of all communication between the user-developed program and the instrument, performing the necessary handshaking and decoding status signals over the selected interface, which can be GPIB, RS232, Ethernet or USB, depending on the type of interface fitted to the instrument being controlled. With the exception of speed, the interface type is essentially transparent to the user, making programs portable between systems with different interfaces.

It includes an automatic search routine which will find any compatible instruments that are connected to the computer. In most cases, this eliminates the need to adjust the communications settings controls on the instrument. The complete "profile" of connected instruments, together with any user-entered descriptive comments, is then securely saved in the system registry. Subsequent data transmissions to and from the instrument use this information to give the fastest possible communication.

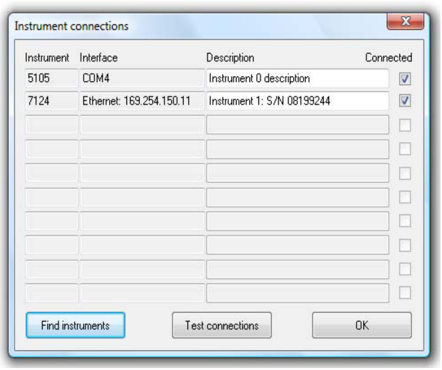
Up to ten compatible instruments can be controlled independently or simultaneously, so that for example in a system measuring impedance one lock-in amplifier can measure the sample current while a second measures the voltage. Both instruments can be read via the control and the output readings combined to determine the impedance.

The package includes a full printed instruction manual, as well as on-screen help so that programming information is always easily available. In addition, sample applications in Visual Basic, Visual C++, Visual C#, LabVIEW, Excel and VBScript (HTML web page) are supplied. The VB, Visual C++ and Visual C# examples include a working executable as well as a full project workspace with all the corresponding source files. Similarly the LabVIEW, Excel and VBScript demonstration programs are complete with all source code information so that they can be easily edited by the user.

User Interface

The control offers a dialog box that programs can activate to allow users to check and if necessary update details of the instruments connected to the system. This box can also be used to initiate a search for instruments via the "Find instruments" button.

The "Test connections" button checks whether an instrument recorded as being of a certain type and connected to a given port is actually present.

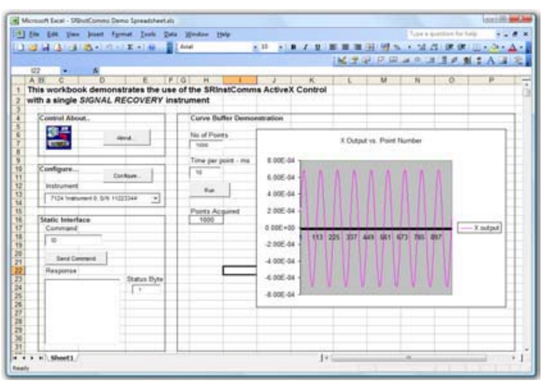


User Interface Dialog

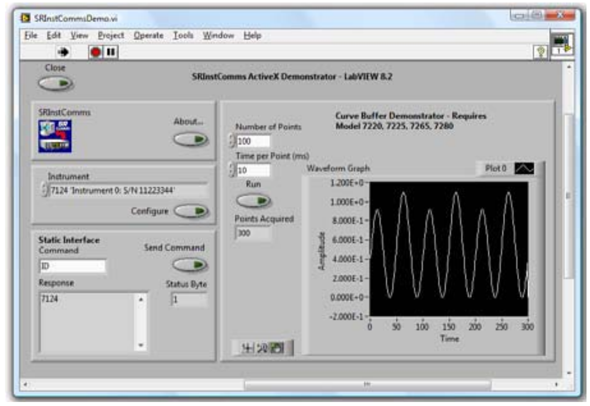
Sample Programs

The supplied example programs offer a quick way to start developing a program. Unlike some software toolkits, this ensures that you have access to code that is known to work with a wide range of hardware.

The Excel spreadsheet and LabVIEW VI demonstrate perfectly the power of the control, allowing data to be taken directly from the instrument and plotted to a graphical display.



Demonstration Excel Spreadsheet



Demonstration LabVIEW VI

Technical Requirements

In addition to a compatible operating system and suitable programming software, the control requires at least one free USB or RS232 port on the computer, or a connection to an Ethernet network, or a spare PCI/PCMCIA slot to accommodate a National Instruments GPIB Interface card, typically a PCI-GPIB or PCMCIA-GPIB. **Other manufacturer's cards or cables will not work.**

Ordering Information

The SRInstComms software pack includes the software supplied on CD and a 57 page instruction manual. The package is licenced for use on a single computer; for multiple or redistribution licenses please contact us first.

Optional Accessories

Model CE0114S	National Instruments PCI-GPIB Interface Board
Model CE0115S	National Instruments USB-GPIB Interface Cable
Model CE0116S	USB-RS232 Serial Adaptor
Model SC0073	2m GPIB cable
Model SC0067	4m GPIB cable
Model SC0066	1m GPIB cable
Model C01001	9F - 9F Null Modem RS232 cable (for models 5105 and 5106)
Model C01002	9F - 25M Null Modem RS232 cable (for models 5109, 5110, 5209 and 5210)
Model C01003	9F - 9M Null Modem RS232 cable (for models 7124, 7220, 7260, 7225, 7225BFP, 7265, 7270, 7280 and 7280BFP)
Model K02001	25F - 9M RS232 adapter.
Model 796190	6' (2 meter) long USB type A to type B cable for connecting model 7124 and 7270 to the USB port on a computer

Computer Cables, RS232 Adapter and GPIB Interface Card

RS232 Cables

Model C01001
9-pin female to 9-pin female, modem eliminator, for use with models 5105 and 5106.

Model C01002
9-pin female to 25-pin male, modem eliminator, for use with models 5109, 5110, 5209, 5210, and 5302.

Model C01003
9-pin female to 9-pin male, modem eliminator, for use with models 7124, 7220(BFP), 7225(BFP), 7260, 7265, 7270, 7280(BFP), 7210, and 7310.

RS232 Adapter
Model K02001
25-pin female to 9-pin male adapter allowing the use of any of the above RS232 cables with 25-pin RS232 computer ports. The required cable must be ordered separately.

USB Cable
Model 796190
6' (2 meter) long USB type A to type B cable for connecting models 7124 and 7270 to the USB port on a computer

USB-RS232 Adaptor Cable
Model CE0116S
USB to 9-pin RS232C Serial Interface Cable. Requires model C01001, C01002 or C01003 null modem serial cable in order to connect to ***SIGNAL RECOVERY*** products. Complete with driver software suitable for XP/Vista.

GPIB (IEEE-488) Cables

Model SC0067
Shielded GPIB Cable - 4 meter

Model SC0073
Shielded GPIB Cable - 2 meter

Model SC0066
Shielded GPIB Cable - 1 meter

GPIB (IEEE-488) Interface Card
Model CE0114S
National Instruments GPIB Interface Card type PCI-GPIB

GPIB (IEEE-488) Interface Cable
Model CE0115S
National Instruments GPIB Interface Cable type USB-GPIB

Light Chopper Selection Guide

Follow this simple guide to choose the right ***SIGNAL RECOVERY*** light chopper for your application. In case of any doubt, simply contact us for further advice

First, decide if space considerations in your experiment require the use of a small diameter blade. If so:

- If the required chopping frequency is in the range 30 Hz to 290 Hz, choose the **model 652-1**
- If the required chopping frequency is in the range 60 Hz to 1.1 kHz, choose the **model 652-2**

If you can use a larger blade, decide whether you are doing optical mixing experiments in which you want to apply two chopped light beams onto your sample and detect a resulting signal at the sum of their frequencies:-

- If you are doing optical mixing experiments, or want to use the dual reference modes of the model 7265 or 7280 lock-in amplifiers for optical source compensation work, consider the **model 198A**

If you are not doing mixing experiments, the final decision is whether you want a self-contained chopper or a chopper with a remote head.

- If you want a unit with a separate head, choose the **model 651-1**
- If you want a self-contained unit, choose the **model 197**

Why should you choose *SIGNAL RECOVERY* products?

Light Choppers models 197, 198A and 650-series

SIGNAL RECOVERY **Product Features**	**Benefit to you**
• Crystal controlled chopper frequency	Very stable and repeatable - ideal for long term measurements
• Sync input	Allows locking to an external frequency, e.g. to allow computerized frequency control connect to the oscillator output of one of our computer interfaced lock-in amplifiers.
• 652-1 and 652-2 have very compact chopper heads	Ideal for use where space is limited, e.g. in laser housings
• Blades are protected	Virtually eliminates the possibility of accidental damage. Preventing blades getting bent also significantly increases the motor bearing life
• Dual aperture blades on the models 197 and 651-1	When used with the 7265/7280 dual reference modes allows dual beam path ratiometric measurements using a single lock-in amplifier

650-Series Light Choppers

Model 651-1

FEATURES

- **Remote chopper heads**
- **Quartz crystal frequency accuracy and stability**
- **Internal or external frequency reference**
- **Sync outputs**
- **Choice of 3 models**
- **Fully enclosed housings for safety and low noise**

APPLICATIONS

- **Optical absorption, reflection and transmission measurements**
- **Dual-beam ratiometric measurements**
- **Automatic background subtraction in boxcar averager experiments**

DESCRIPTION

The 650 series of light choppers features a control box with separate chopping heads for remote operation. There are three different models within the range, the 651-1 using a standard dual port chopping head and 652-1 and 652-2 employing a single port micro head for use where space is a prime consideration.

All the models in the range offer precision frequency control via digital push-buttons or by the application of an AC signal to the sync input on the control unit. When used in conjunction with a ***SIGNAL RECOVERY*** lock-in amplifier, computer control of the chopping frequency can be achieved by use of the lock-in amplifier's oscillator as a drive signal for the chopper and controlling the oscillator frequency from the computer. In all cases, frequency accuracy and stability are excellent. A LED indicator on the front panel gives constant indication of frequency lock. This indicator can, however, be extinguished for those measurements that need to be executed in total blackout conditions.

One special feature of the 650 series choppers is the ability to add extension leads (model 653A) to increase the remote distance between the chopping head and the control unit. Up to two additional leads can be employed, each 2 m long, producing a maximum separation distance between head and controller of 5.5 m (18 ft).

Both the standard head and the micro heads are fully enclosed designs which serve to reduce errors resulting from external air motion and to minimize the risk of accidental damage to the chopping blades.

Quartz Crystal Frequency Accuracy and Stability

The models 651-1, 652-1 and 652-2, in common with all ***SIGNAL RECOVERY*** light choppers, uses a quartz crystal oscillator as their primary frequency standard. The oscillator signal is divided down to yield the required chopper frequency, and then the motor speed is continuously adjusted to phase lock the actual chopper frequency to this required value. The result are choppers with an output frequency as stable as any other modern frequency source.

External Frequency Control

Like many other choppers, the frequency can be controlled externally. However, unlike other units the control is via an applied TTL reference rather than an analog voltage. This means that the modulation frequency generated is exactly that required, and in the case of the dual-aperture model 651-1 allows it to be used in conjunction with the dual reference modes offered by our model 7270, 7124, 7265 and 7280 lock-in amplifiers to implement two-channel source compensation experiments. Figure 1 on page 64 shows the model 198A used in this mode but is equally valid when this is replaced by the 651-1. The application is discussed in detail in Applications Note AN1005 on page 147.

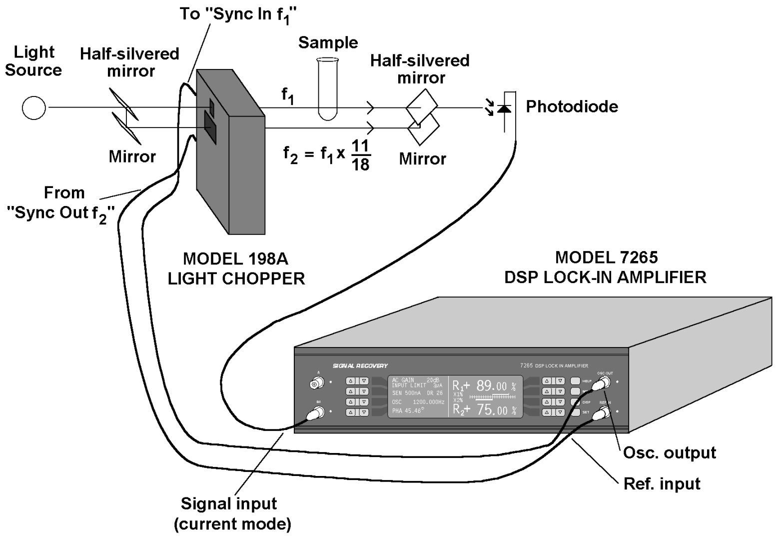
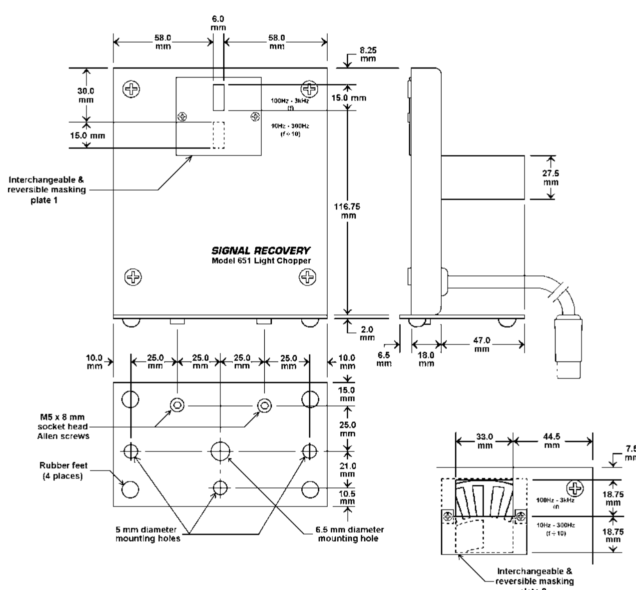


Figure 1, Using the Model 198A (651-1 may be substituted) with a Model 7265 to implement a dual-channel source compensation experiment

Model 651-1
Dual Aperture Light Chopper

The model 651-1 uses the standard head. This is a dual port design allowing two optical beams to be modulated simultaneously at different frequencies, f_1 and f_2. Two reference (sync) outputs are made available at the control unit, corresponding to f_1 and f_2, for use with signal recovery processing electronics, such as a lock-in amplifier.

Two small blanking plates are provided with the standard head and one of these allows either the top port or the bottom port to be blanked off to reduce the effect of any stray light passing through the unused path. The second plate provides some measure of beam restriction when using large diameter beams on the higher frequency (and therefore smaller aperture) port. The standard head is supplied complete with a flat base plate to allow it to be used in a free standing mode on a bench type surface. A special feature of this support allows the head to be used in the horizontal plane for modulating vertical beams. The base plate can be removed and standard optical mounting posts employed instead if preferred.



Model 651-1 Mechanical Dimensions

Specifications - Model 651-1

General

Dual-aperture remote head chopper with internal or external reference frequency. Two sync outputs.

Frequency	10 Hz to 3200 Hz
outer sector	100 Hz to 3200 Hz
inner sector	10 Hz to 320 Hz
Control	
manual	Digital push-button
external	Application of 0.5 V to 10 V pk-pk sine or squarewave, 100 Hz to 3200 Hz to EXT SYNC BNC connector
Internal Frequency	
accuracy	±30 ppm at 25 °C
stability	±50 ppm/ °C (range 10 °C to 30 °C)

Jitter (measured pk-pk and presented as a % of a full cycle)

outer sector	
100 to 200 Hz	blade only: 0.2%; blade + electronics: < 7%
200 to 400 Hz	blade only: 0.2%; blade + electronics: < 4%
400 to 2500 Hz	blade only: 0.2%; blade + electronics: < 1.5%
2500 to 3200 Hz	blade only: 0.2%; blade + electronics: < 2%
inner sector	
10 to 20 Hz	blade only: 0.2%; blade + electronics: < 0.7%
20 to 40 Hz	blade only: 0.2%; blade + electronics: < 0.4%
40 to 250 Hz	blade only: 0.2%; blade + electronics: < 0.2%
250 to 320 Hz	blade only: 0.2%; blade + electronics: < 0.3%
Lock indication	green LED when locked - can be extinguished
Settling Time	<40 s nominal

Outputs

Sync Out 1	10 V pk-pk squarewave at outer sector chopping frequency, 100 - 3200 Hz
Sync Out 2	10 V pk-pk squarewave at inner sector chopping frequency, 10 - 320 Hz
Connectors	BNC
Impedance	10 kΩ. Note that although the output voltage is 10 V pk-pk, the high output

impedance means that the outputs can be directly connected to the external reference input of any ***SIGNAL RECOVERY*** lock-in amplifier without causing problems.

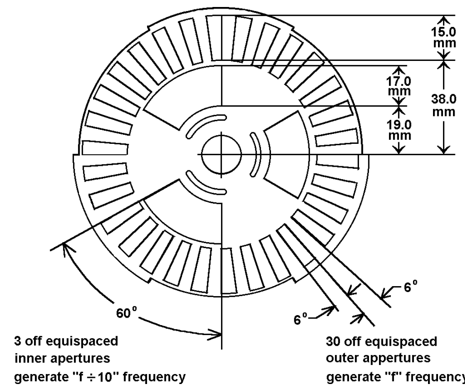
General

Power Requirements	110 V AC, 50/60 Hz or 220/240 V AC, 50/60 Hz supply. State which voltage is required when ordering

Dimensions	
Controller	
Width	6½" (168 mm)
Height	3¼" (79 mm)
Depth	9½" (236 mm)
Chopper head, overall, inc. base and feet	
Width	4¾" (122 mm)
Height	6" (150 mm)
Depth	2¾" (72 mm)

Options

Model 653A	2 m (6'6") extension cable

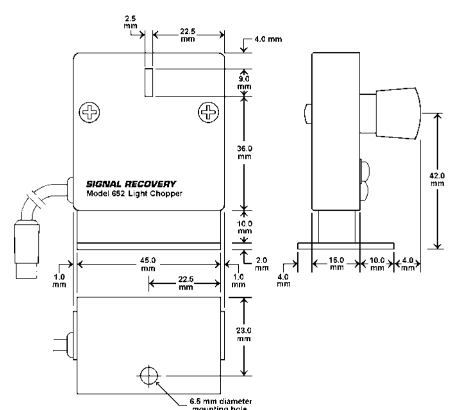


Model 651-1 Blade Dimensions

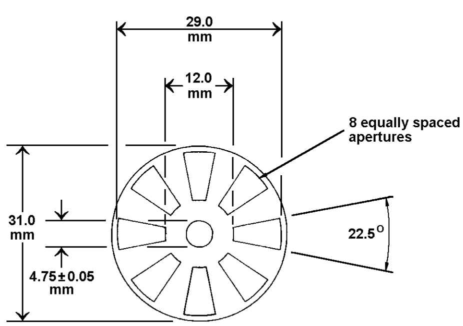
Models 652-1 and 652-2 Micro Head Light Choppers

The models 652-1 and 652-2 utilize a micro head. This is a single port modulator for use where space is limited. The former uses a 2-slot chopping disc and the latter an 8-slot disc to achieve a different frequency range. Micro heads can be employed as free standing modulators on a bench type surface, or can be supported on standard mounting hardware.

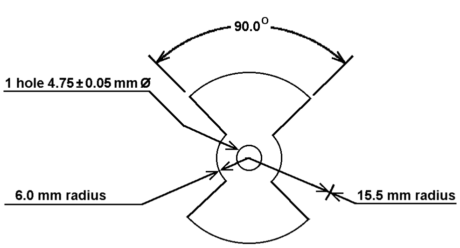
Model 652-1



Models 652-1 and 652-1 Mechanical Dimensions



Model 652-2 Blade Dimensions



Model 652-1 Blade Dimensions

Specifications - Models 652-1 and 652-2

General

Single aperture remote head chopper with internal or external reference frequency. Sync output.

Frequency	
652-1	2 slots, 30 Hz to 290 Hz
652-2	8 slots, 60 Hz to 1100 Hz

Control	
manual	Digital push-button
external	Application of 0.5 V to 10 V pk-pk sine or squarewave within the chopper's range to EXT SYNC BNC connector
Internal Frequency	
accuracy	±30 ppm at 25 °C
stability	±50 ppm/ °C (range 10 °C to 30 °C)
Jitter (measured pk-pk and presented as a % of a full cycle)	
652-1	
30 to 50 Hz	blade only: 0.2%; blade + electronics: < 5%
50 to 290 Hz	blade only: 0.2%; blade + electronics: < 2%
652-2	
60 to 100 Hz	blade only: 0.2%; blade + electronics: < 6%
100 to 290 Hz	blade only: 0.2%; blade + electronics: < 2.4%
Lock indication	green LED when locked - can be extinguished

Specifications

Models 651-1 and 651-2 (continued)

Settling Time	<40 s nominal

Outputs

Sync Out 1	10 V pk-pk squarewave at chopping frequency
Connector	BNC
Impedance	10 kΩ. Note that although the output voltage is 10 V pk-pk, the high output impedance means that the outputs can be directly connected to the external reference input of any ***SIGNAL RECOVERY*** lock-in amplifier without causing problems.

General

Power Requirements	110 V AC, 50/60 Hz or 220/240 V AC, 50/60 Hz supply. State which voltage is required when ordering.
Dimensions	
Controller	
Width	6½" (168 mm)
Height	3¼" (79 mm)
Depth	9½" (236 mm)
Chopper head, overall, inc. base and feet	
Width	2" (48 mm)
Height	2½" (62 mm)
Depth	1½" (35 mm)

Options

Model 653A	2 m (6'6") extension cable

Model 197

Precision Light Chopper

FEATURES

- **Self contained chopper head**
- **Quartz crystal frequency accuracy and stability**
- **Internal or external frequency reference**
- **Sync outputs**
- **Fully enclosed housings for safety and low noise**

APPLICATIONS

- **Optical absorption, reflection and transmission measurements**
- **Dual-beam ratiometric measurements**
- **Automatic background subtraction in boxcar averager experiments**

DESCRIPTION

The model 197 is a compact, high performance chopper, offering features and benefits that are ideal for use in modern photometric systems.

The unit is self contained, comprising a dual aperture chopper blade, motor and the necessary driving electronics. Each aperture provides an independent reference output allowing simultaneous dual frequency operation (10:1 ratio) for dual-path experiments. Frequency control is by a precision internal oscillator set by a 4-digit push-button selector on the unit or by the application of an external AC reference signal. The unit is powered via an external line power supply module.

Mounting holes are provided in the base and right-hand side of the housing (viewed from the front) to allow for mounting the model 197 onto an optical bench or support post.

Quartz Crystal Frequency Accuracy and Stability
The model 197, in common with all ***SIGNAL RECOVERY*** light choppers, uses a quartz crystal oscillator as its primary frequency standard. The oscillator signal is divided down to yield the required chopper frequency, and then the motor speed is continuously adjusted to phase lock the actual chopper frequency to this required value. The result is a chopper with an output frequency as stable as any other modern frequency source.

External Frequency Control
Like many other choppers, the frequency can be controlled externally. However, unlike other units the control is via an applied TTL reference rather than an analog voltage. This means that the modulation frequency generated is exactly that required which allows these choppers to be used in conjunction with the dual reference modes offered by our model 7124, 7265, 7270, and 7280 lock-in amplifiers to implement two-channel source compensation experiments - see applications notes AN1003 on page 135 and AN1005 on page 147.

Specifications

General

Dual-aperture self-contained chopper with internal or external reference frequency. Two sync outputs.

Frequency	15 Hz to 3000 Hz
outer sector	150 Hz to 3000 Hz
inner sector	15 Hz to 300 Hz

Control	
manual	Digital push-button
external	Application of 0.5 V to 10 V pk-pk sine or squarewave, 150 Hz to 3000 Hz to EXT SYNC BNC connector
Internal Frequency	
accuracy	±20 ppm at 25 °C
stability	±30 ppm/ °C (range 10 °C to 60 °C)

Specifications

Model 197 (continued)

Jitter (measured pk-pk and presented as a % of a full cycle)

outer sector

150 to 500 Hz	blade only: 0.5%; blade + electronics: < 1.5%
500 to 3000 Hz	blade only: 0.5%; blade + electronics: < 1%

inner sector

15 to 50 Hz	blade only: 0.5%; blade + electronics: < 1.5%
50 to 500 Hz	blade only: 0.5%; blade + electronics: < 1%
Lock indication	Bicolor LED - red when unlocked and green when locked

Settling Time

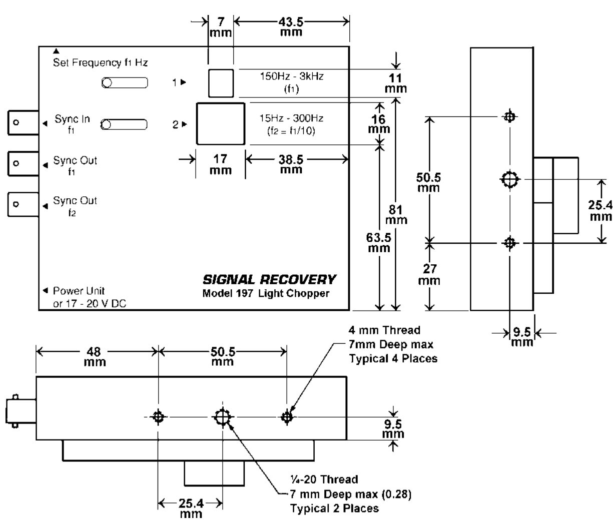
7 s nominal at 1 kHz from switch-on;
9 s nominal for frequency change from 150 to 3000 Hz;
30 s nominal for frequency change from 3000 to 150 Hz

Outputs

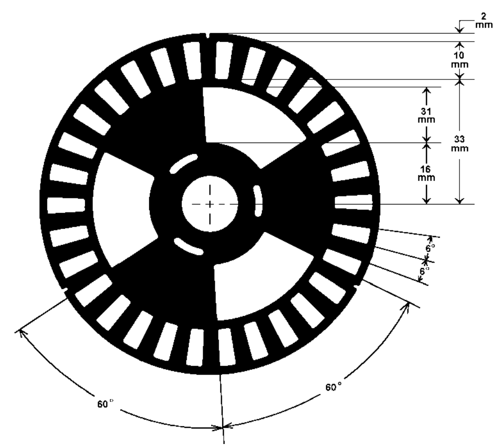
Sync Out 1	10 V pk-pk squarewave at outer sector chopping frequency, 150 - 3000 Hz
Sync Out 2	10 V pk-pk squarewave at inner sector chopping frequency, 15 - 300 Hz
Connectors	BNC
Impedance	10 kΩ. Note that although the output voltage is 10 V pk-pk, the high output impedance means that the outputs can be directly connected to the external reference input of any ***SIGNAL RECOVERY*** lock-in amplifier without causing problems.

General

Power Requirements	Via separate power adapter for 110 V AC, 50/60 Hz or 220/240 V AC, 50/60 Hz supply. State which voltage is required when ordering
Dimensions	
Width	4¾" (122 mm)
Height	4" (104 mm)
Depth	1¾" (44 mm)
Weight	1lb (0.45 kg) excluding power supply



Model 197 Mechanical Dimensions



Model 197 Chopper Blade Dimensions

Model 198A

Mixed Beam Light Chopper

FEATURES

- **Self contained chopper head**
- **Dual aperture with three SYNC outputs**
- **Quartz crystal frequency accuracy and stability**
- **Internal or external frequency reference**
- **Fully enclosed housings for safety and low noise**

APPLICATIONS

- **Pump-probe experiments**
- **Nonlinear optics**

DESCRIPTION

The model 198A is a dual frequency light chopper, using a thin rotating metal blade with an inner set of 11 apertures and an outer set of 18 apertures to simultaneously chop two light beams. The blade is driven by a precision DC motor whose speed is controlled via a phase locked loop that is referenced to either an internal quartz crystal oscillator or an external reference signal. The chopper has three reference frequency outputs, one at each of the frequencies generated by the inner and outer apertures, and one at a frequency equal to their sum. The three frequencies f_1, f_2 and $f_1 + f_2$ are chosen to be relative primes which significantly reduces mutual harmonic interference.

It is ideal for use in measurements where two modulated beams give rise to a third optical signal at a frequency which is the sum of the two chopped frequencies. The model 198A acts as a chopper for the incident light sources at frequencies f_1 and f_2 and generates a reference signal at a frequency $f_1 + f_2$ which can be used to drive a subsequent instrument, such as a lock-in amplifier.

In conjunction with the dual reference mode offered by the model 7124, 7265, 7270, and 7280 DSP lock-in amplifiers, the model 198A can also be used to implement a very cost-effective dual-beam ratiometric measurement system. This technique can eliminate variations in source intensity over several orders of magnitude. It is described further in an Applications Note AN1005 “Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper” on page 147.

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The chopper can also be used as a conventional single-beam unit since it also generates reference signals at f_1 and f_2.

Quartz Crystal Frequency Accuracy and Stability

The model 198A, in common with all ***SIGNAL RECOVERY*** light choppers, uses a quartz crystal oscillator as its primary frequency standard. The oscillator signal is divided down to yield the required chopping frequency, and then the motor speed is continuously adjusted to phase lock the actual chopping frequency to this value. The result is a chopper with an output frequency as stable as any other modern frequency source.

External Frequency Control

Like many other choppers, the frequency can be controlled externally. However, unlike other units the control is via an applied reference signal (TTL levels may be used) rather than an analog control voltage. The chopper locks to this applied reference, which is at the same frequency as that generated by the outer set of apertures (f_1), but for detection purposes the reference outputs it generates (i.e. at f_1, f_2 and $f_1 + f_2$) are of course also available.

Specifications

General

Dual-aperture self-contained chopper with internal or external reference frequency. Two sync outputs.

Frequency	55 Hz to 1500 Hz
outer sector	90 Hz to 1500 Hz, 18 apertures, f_1
inner sector	55 Hz to 917 Hz, 11 apertures, f_2
Control	
manual	Digital push-button
external	Application of 0.5 V to 10 V pk-pk sine or squarewave, 90 Hz to 1500 Hz to Sync In f_1 BNC connector
Internal Frequency	
accuracy	±20 ppm at 25 °C
stability	±30 ppm/ °C (range 10 °C to 60 °C)

Jitter (measured pk-pk and presented as a % of a full cycle)

outer sector	
90 to 140 Hz	blade only: 0.2%; blade + electronics: < 6%
140 to 1500 Hz	blade only: 0.2%; blade + electronics: < 1.5%
inner sector	
55 to 100 Hz	blade only: 0.2%; blade + electronics: < 6%
100 to 917 Hz	blade only: 0.2%; blade + electronics: < 1.5%
Lock indication	Bicolor LED - red when unlocked and green when locked

Settling Time

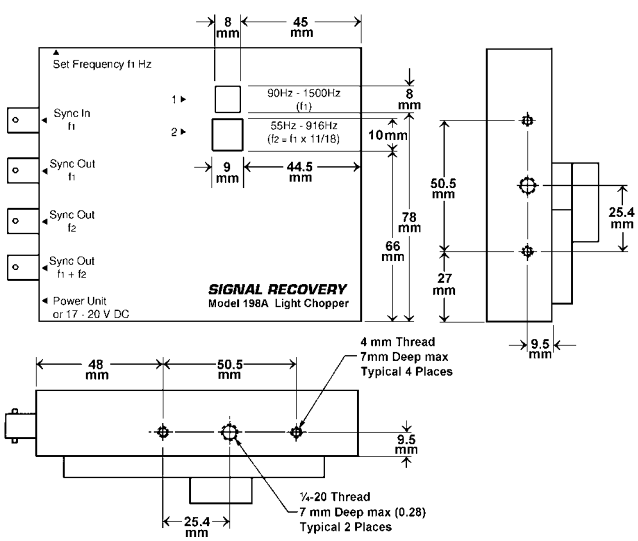
7 s nominal at $f_1 + f_2$ = 1 kHz from switch-on; 9 s nominal for $f_1 + f_2$ frequency change from 150 to 2400 Hz;
30 s nominal for $f_1 + f_2$ frequency change from 2400 to 150 Hz

Outputs

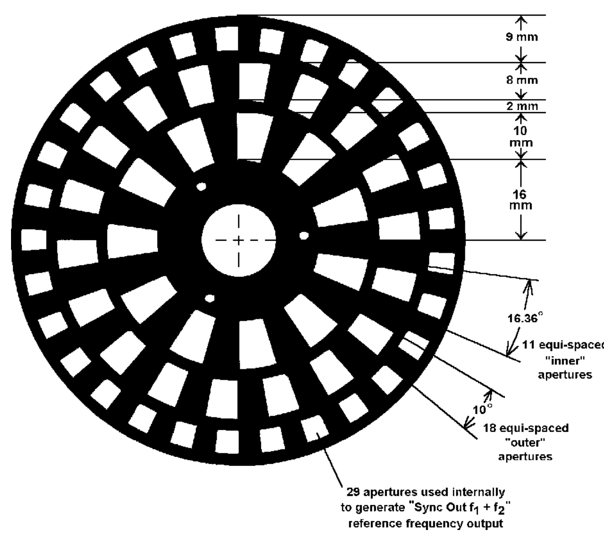
Sync Out f_1	10 V pk-pk squarewave at outer sector chopping frequency, 90 - 1500 Hz
Sync Out f_2	10 V pk-pk squarewave at inner sector chopping frequency, 55 - 917 Hz
Sync Out $f_1 + f_2$	10 V pk-pk squarewave at sum of chopping frequencies, 145 - 2417 Hz
Connectors	BNC
Impedance	10 kΩ. Note that although the output voltage is 10 V pk-pk, the high output impedance means that the outputs can be directly connected to the external reference input of any ***SIGNAL RECOVERY*** lock- in amplifier without causing problems.

Power & Mechanical

Power Requirements	Via separate power adapter for 110 V AC, 50/60 Hz or 220/240 V AC, 50/60 Hz supply. State which voltage is required when ordering
Dimensions	
Width	4¾" (122 mm)
Height	4" (104 mm)
Depth	1¾" (44 mm)
Weight	1lb (0.45 kg) excluding power supply



Model 198A Mechanical Dimensions



Model 198A Chopper Blade Dimensions

Signal Averager Selection Guide

When you need to recover non-sinusoidal signals, a lock-in amplifier is usually not suitable and a signal averager should be considered. Choosing the right model is not difficult if you follow this simple guide, but in case of doubt, simply contact us for further advice

First, decide whether you need details about the shape of the waveform you are recovering, or simply specific information about the size of peaks on it.

- If measuring the amplitude or integral of peaks is sufficient, then consider the **model 4121B** boxcar averager. If in addition you want to transfer the values it measures to a computer, then add a **model 4161A** dual channel ADC module as well.

- If you need all the information about the shape of the waveform, in a similar way to that obtained by using an oscilloscope, then you will need an instrument offering waveform digitization and averaging. If the experiment can be externally triggered then you should choose the ***FASTFLIGHT-2*** digital signal averager.

FASTFLIGHT-2

4 GSa/s Digital Signal Averager

FEATURES

- **4 GSa/s (250 ps per point) effective sampling rate (2 GSa/s real-time sampling rate)**
- **Long record lengths**
- **1 to 65,536 sweeps averaged per record**
- **<1 µs end-of-scan deadtime during averaging (<1 % idle time)**
- **Transfer rates to PC of up to 100 spectra (50 µs long and 500 ps per point) per second**
- **Precision Enhancer transforms 8-bit ADC into 12-bit ADC, for 16 times greater dynamic range**
- **Automatic correlated noise reduction algorithm**
- **Live or post-acquisition trend display**
- **Complete with software**
- **ActiveX Controls compatible with LabVIEW, C++, Visual Basic and other languages**

APPLICATIONS

- **Dielectric studies**
- **Time-of-flight measurements**
- **Fundamental particle studies**

DESCRIPTION

FASTFLIGHT-2® is a high performance digital signal averager in a compact benchtop console that is designed to be operated from a personal computer via its integral USB data link. Essentially, on receipt of a trigger pulse, it digitizes the applied analog signal at rates of up to 4 GSa/s (250 ps per point) using an 8-bit flash ADC and stores the resulting waveform into its internal memory. As such, it can be considered to be like a fast oscilloscope. But unlike most 'scopes, it includes a dedicated hardware averager, so that if the signal is repetitive, it is able to record and average successive waveforms into a 24-bit deep output memory with a deadtime between the end of one sweep and the start of the next of less than 1 µs.

This feature is the key figure of merit when comparing the *FASTFLIGHT-2* with other techniques, such as digital storage oscilloscopes, which often require significant times - up to milliseconds in some cases - after each sweep in order to perform the averaging process. Because of this low deadtime, the overall data throughput rate can be very high, allowing higher repetition rates and shorter experiment times than are possible when using other methods.

The rapid acquisition speed is matched by rapid data transfer to the host computer via the USB link of the resulting averaged spectra, with rates of up to 100 spectra per second being achievable. This in turn allows study of spectra that are changing over a period of time.

FASTFLIGHT-2 is supplied with a full applications software package, designed for Windows XP, which gives access to all its controls and graphically displays the acquired records, as well as allowing live or post acquisition trend analysis. Alternatively, users can develop their own software using the supplied ActiveX controls, which are compatible with most modern programming languages.

The instrument is suitable for use in any application requiring on-line averaging and/or high repetition rates, especially those with noisy repetitive signals of a transient nature. In these cases, where measurement times are necessarily short, the low deadtime and high data throughput will make it the instrument of first choice.

Specifications

General

Single-channel digital signal averager with 2 GSa/s 8-bit ADC capable of giving effective 4 GSa/s 12-bit performance. Benchtop console (with separate power supply module) controlled entirely from host computer via USB data link.

Full applications software package supplied. ActiveX controls for incorporating into custom programs.

Measurement Modes

The instrument can either respond to an external trigger pulse, or generate a trigger pulse, and then start digitizing the applied signal waveform for a preset period. The acquired waveform is stored to memory.

The cycle repeats for the preset number of sweeps to average, with each new record being added to those already in the buffer memory. On completion the averaged record is transferred to the PC for display and processing.

The supplied software can display the averaged record (waveform mode) or the history of a particular feature, such as pulse area or peak amplitude (trend mode).

Signal Channel

Channels	One
Modes	Single-ended voltage input
Full-scale Sensitivity	0 to -0.5 V
Impedance	50 Ω
Offset Control	
Range	-0.25 V to +0.25 V
Resolution	0.03 mV
Bandwidth	DC to >500 MHz
Rise/Fall Times	< 1ns
Equivalent Input Noise	
Uncorrelated	< 2 mV rms
Correlated	< 0.02 mV rms

Precision Enhancer

Extends the limiting ADC resolution to 12 bits (for input noise < 2 mV) when 256 or more records are averaged. May be turned on or off.

Analog to Digital Converter

Type	8-bit flash
Sampling period	500 ps, 1.0 ns or 2.0 ns real time; 250 ps interleaved sampling employing two scans per record
Differential non-linearity	Within ±0.1 LSB referred to the 8-bit ADC
Integral non-linearity	Within ±0.4% of full-scale

Trigger Input

Threshold	Adjustable from –2.5 to +2.5 V in 10 mV steps
Polarity	Positive or Negative
Max input	± 5 V DC
Min Pulse Width 2	5 ns
Timing Jitter	The first sampled point in the record is synchronized within ± 250 ps relative to the leading edge of the Trigger Input for real-time sampling

Trigger Output

Type	TTL
Impedance	50 Ω
Pulse Width	64 ns to 5120 ns
Pulse Polarity	Low to High signifies start of sweep
Timing Jitter	Synchronized to the first sampled point in the scan with a jitter <50 ps FWHM. The Trigger Output is alternately delayed by 0 and 250 ps relative to the sampling clock in the 250 ps interleaved sampling mode.

Spectrum Length

250 ps sampling	10.0 µs min; 375 µs max
500 ps sampling	10.0 µs min; 750 µs max
1 ns sampling	10.0 µs min; 1.5 ms max
2 ns sampling	10.0 µs min; 3.0 ms max

Data Acquisition Delay

Computer selectable digital delay after trigger from 0 to 1048.56 µs in 16 ns increments

End-of-Scan Dead Time

0.8 µs

End-of-Spectrum Dead Time

0.8 µs

Averaging Method Linear summation

Number of Records in Average

1 to 65,536

Timing Clock

Internal	2 GHz with stability of better than 2 ppm/ºC
10 MHz Clock Input	Allows instrument to be synchronized with a master timing source
10 MHz Clock Output	10 MHz signal phase-locked to the internal 2 GHz clock

Preamp Power Output

Type	9-pin subminiature D connector
Voltage	pin 4: +12 V, pin 9: -15 V, pins 1 & 2: ground

Interface USB

Maximum Spectral Transfer Rate

Up to 100 averaged spectra/s transferred to PC memory and hard disk for a 50 µs spectrum length and 500 ps sampling.

Software

Full operating package for Windows XP including programmer's toolkit. Package includes software controls to access every hardware control, averaged record and trend mode displays, and data storage to hard disk. Free LabVIEW driver available from **www.signalrecovery.com** website.

General

Power Requirements	
Voltage	110/120/220/240 VAC
Frequency	50/60 Hz
Dimensions - Chassis	
Width	12.9" (330 mm)
Depth	13.3" (340 mm)
Height	2.9" (74 mm)
Weight - Chassis	10.1 lb (4.9 kg)
Dimensions - PSU	
Width	5.2" (132 mm)
Depth	2.3" (58 mm)
Height	1.2" (30 mm)
Weight - PSU	1 lb (450 g)

Why should you choose *SIGNAL RECOVERY* products?

FASTFLIGHT2 Digital Signal Averager

SIGNAL RECOVERY Product Features	Benefit to you
• Very low end-of-scan deadtime	Experiments can be run at high repetition rates
• Rapid data transfer to PC	Eliminates data acquisition bottleneck common in transfer of data from oscilloscopes to computer
• 250 ps effective sampling time	Capture spectra at higher time resolution than other digitizers

Model 4121B

Gated Integrator

FEATURES

- **1 ns minimum gate width**
- **80 kHz max trigger rate**
- **Linear or Exponential averaging**
- **Input offset control**
- **Normal and Baseline sampling modes**
- **Built-in trigger generator**

APPLICATIONS

- **Pulsed laser experiments**
- **Phosphorescence decay time studies**
- **Precision signal sampling**

DESCRIPTION

This module is an ideal component for building boxcar averager systems. It includes a wide bandwidth variable gain AC/DC coupled input amplifier with offset adjustment and a high speed sampling gate with variable width and delay controls. It operates in normal or baseline sampling mode and features a switch-selected choice of how many samples are included in the averaging process. Separate outputs for the average and last sample taken are also provided. A gate monitor supplies a synchronized gate output pulse for application to an oscilloscope trigger or for referencing associated processing electronics. Trigger input is ECL or TTL or can be derived from the module's own adjustable trigger generator.

The module is packaged in a 2-unit wide NIM format and as such requires a suitable NIM rack and power supply to operate. The simplest single-channel system can therefore be produced with one model 4121B module and a suitable NIM rack and power supply (such as the ***SIGNAL RECOVERY*** model 4006 or 4001A/4002D. The addition of a second model 4121B and a model 4161A Display/ADC and control module provides a dual channel system with the added capability of allowing the transfer of output data to a computer for external analysis. Further modules can be added to increase the overall number of channels.

The unit can also be used with other ***SIGNAL RECOVERY*** instruments, such as our lock-in amplifiers, to build systems capable of swept-gate waveform recovery experiments, all controlled via the Acquire data acquisition software.

Specifications

General

Single-channel gated integrator module mounted in NIM enclosure with adjustable sensitivity, offset, gatewidth and output averager. Manual controls.

Analog gate delay generator with manual or DC voltage control.

Measurement Modes

On receipt of an external trigger, the instrument waits for the preset gate delay and then integrates the voltage present at its input for the preset gate width. On completion a DC voltage representing this integral is provided at the Last Sample Output connector and in addition fed forward into an analog integrator stage.

Signal Channel

Mode	Normal or Baseline Sampling
Sensitivity	±20 mV to ±2 V in 1-2-5 sequence
Coupling	AC/DC
Impedance	
DC only	50 Ω // 10 pF
DC or AC	1 MΩ // 30 pF
Maximum Safe Input	
50 Ω Input	±5 V
1 MΩ Input	±100 V
Offset	±10 × FS; non-removable
Overload Indicator	LED
Overload Level	Input (signal plus noise) > 1.1 × FS
Overload Recovery	Recovers after 1 sample for ×10 overload
Gain Drift	0.5% /°C, gate width > 30 ns; 1.0% /°C, gate width < 10 ns

DC Drift (referred to input)	0.2%/°C, gate width > 20 ns; 1.0%/°C, gate width < 20 ns
Bandwidth	
50 Ω input	DC to 450 MHz
1 MΩ DC input	DC to 100 MHz
1 MΩ AC input	1.5 Hz to 100 MHz
Signal Risetime	
50 Ω input	2 ns; 20% to 80%
1 MΩ input	10 ns; 20% to 80% from 50 Ω source.

Sampler and Timing

Gate Width	1 ns to 30 µs in 1-3-10 sequence, switch selectable with a continuously variable ×1 to ×5 multiplier
Sample Correlation	Less than 0.5% of the sample output due to trigger *t* remains at trigger *t* + 1
Gate Delay	
Input	0 to 10 V DC varies delay by 0.5% to 100% of range setting
Max delay	3 ns to 300 ns in a 1-3-10 sequence plus user options, which give 10 µs (default), and 1 µs, 100 µs, 1 ms or 3 ms by capacitor change.
Trigger Source	
Internal	0.5 Hz to 40 kHz selectable with range switches 0.5, 5, 50, 500, 5000, off. Vernier is 10× range.
External	
ECL	Positive edge, 5 ns min pulse width with termination of 50 Ω to -2 V; -5 V to +10 V pk-pk safe input.
TTL	Negative edge, 20 ns min pulse width; -5 V to +10 V safe input.
Max. Trigger Rate	80 kHz
Trigger Indicator	LED lights when unit is triggered
Trigger Generator	
Output	BNC TTL out on rear panel active in all trigger modes. Polarity set by jumper.
Frequency ranges	0.5, 5, 50, 500 Hz, 5 kHz and off with vernier to overlap ranges.
Baseline Input	TTL line to indicate whether sample is signal or baseline value.

Analog Output Averager

Mode	Linear or Exponential
Samples Averaged	1, 3, 10, 30, 100, 300, 1k, 10k
LSO Droop Rate	< 0.2% FS/s
Averager Droop Rate	When there are no triggers the droop rate is < 0.001% per minute for 10k samples

Outputs

Average Out	±10 V FS with 50 Ω output impedance and capable of driving 2 kΩ load
Last Sample Out	±10 V FS
Gate Monitor	0.3 V into 50 Ω to ground. Marker pulse-width equals gate width. Position is within 5 ns from actual gate
Trigger	TTL
Baseline Output	TTL output line that toggles with each trigger to indicate whether next sample is signal or baseline value.

General

Power Requirements	+24 V at 200 mA; -24 V at 150 mA +12 V at 300 mA; -12 V at 590 mA +6 V at 160 mA; -6 V at 630 mA
Dimensions	
Height	8¾" (222 mm)
Width	2¾" (70 mm)
Depth	9¾" (248 mm)
Weight	3 lb (1.4 kg)

Why should you choose *SIGNAL RECOVERY* products?

Model 4121B Gated Integrator

SIGNAL RECOVERY Product Features	Benefit to you
Higher maximum sensitivity	Sensitivity settings on 4121B are for a full 10 V output, not the 1 V of competing units, allowing you to measure smaller signals
High input bandwidth	Signals are less distorted before being sampled
1 ns minimum gate width	Isolate narrower features more easily. In scanned gate work obtain finer resolution of peaks
Built-in trigger generator	Use to trigger your experiment
Linear or exponential averaging	Linear averaging means that every sample contributes equally to the output
Baseline Out output	Will directly drive one of our light choppers for automatic baseline subtraction
Faster Triggering	80 kHz max trigger rate allows acquisition up to 4 times faster than competing instruments
Excellent reset of integrator between triggers	Ensures that each sample is essentially independent of previous samples

Model 4161A

Dual Channel ADC, Display and Control Module

FEATURES

- **Dual Channel 12-bit ADC with digital display**
- **RS232 and GPIB interfaces with GPIB status indicator**
- **Simple computer command set**
- **ADC trigger inputs**
- **Trigger hold-off output**
- **Independent analog panel meter**
- **2-wide NIM module**

APPLICATIONS

- **Digitize outputs of Model 4121B Gated Integrator module**
- **Computer-controlled boxcar averager systems using 9650A Digital Delay Generator**

DESCRIPTION

The model 4161A is a dual channel, analog to digital converter (ADC) module which will measure one or two analog voltages, display the result on a digital panel meter, and allow it to be read by an external computer connected to the module's RS232 or GPIB interface.

The module has two signal input channels, A and B, each with a full-scale sensitivity of ±10 V DC. On receipt of a trigger command at the appropriate channel the input voltage is digitized to a 5 mV resolution. A computer coupled to the module can determine the value of the input voltage by sending a simple ASCII command. The 3½ digit panel meter on the 4161A can be switched to monitor either of the signal channels.

The model 4161A is primarily intended to act as the interface between one or two model 4121B gated integrator modules (page 74) and a controlling computer. In multiple 4121B systems more than one 4161A can be used to digitize the data from several gated integrators, with all the results being read via the GPIB interface.

An edge-indicating analog panel meter is also incorporated into the module which is especially useful during the setup of boxcar systems.

Specifications

General

Two-channel ADC mounted in NIM enclosure with signal and trigger inputs and with trigger holdoff output. RS232 and GPIB (IEEE488) control. Separate analog edge-indicating panel meter.

Input

Channels	Two
ADC Inputs	BNC front-panel connectors, A and B
Input Impedance	1 MΩ
Input Full-Scale	±10 V
Accuracy	±5 mV
Linearity	±5 mV
ADC Trigger Inputs	BNC front-panel connectors, corresponding to channel A and channel B ADC inputs. Connectors are duplicated on rear panel
Trigger Thresholds	TTL. Triggers on rising edge of applied positive logic TTL pulse

Digital Display

Type	3½ digit LED display showing (Measured voltage / 20)

Display Selection	Switch selects channel A or channel B

Computer Interfaces

RS232	DIP switch selectable baud rate, terminator, character echo, parity and data bits.
GPIB	DIP switch selectable address and terminator
Status Indicators	Front panel LEDs indicate GPIB Talk, Listen, SRQ and Remote

Command Set

Twelve mnemonic type commands allowing both asynchronous and synchronous readings. Digitized voltages are reported back to the computer in integer format, with ±2048 corresponding to an input voltage of ±10.24 V

Software

A LabVIEW driver software suitable for version 4.01 and later of LabVIEW is available by download from our website at **www.signalrecovery.com**

Output

Busy Out

Rear-panel BNC connector generating TTL signal which under computer control will:-

1) Remain at logic 0 until a synchronized read command is issued by the computer.

2) Go to logic 1, releasing external trigger hold-off circuitry (such as can be provided by an external delay generator)

3) Return to logic 0 on receipt of a trigger signal at either the A or B ADC trigger inputs, and remain there while the measured value(s) are transferred back to the computer and thereafter until the next synchronized read command.

Analog Panel Meter

Type

Edge-indicating meter monitoring the voltage at the associated front-panel analog input BNC connector. This meter is completely independent of the analog to digital converter functions.

Input Impedance	10 kΩ
Full-scale sensitivity	±10 V

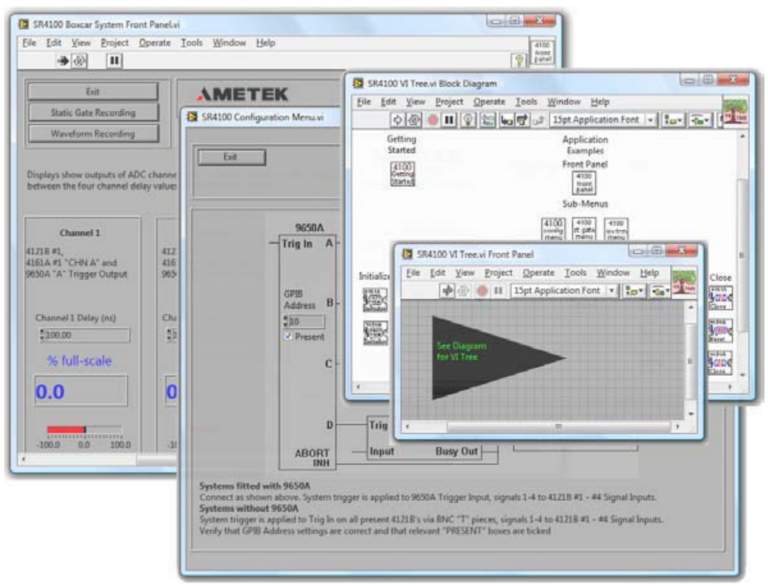
General

Power Requirements	
	+24 V at 50 mA; -24 V at 50 mA
	+12 V at 600 mA; -12 V at 30 mA
	+6 V at 550 mA; -6 V at 10 mA
Dimensions	
Height	8¾" (222 mm)
Width	2¾" (70 mm)
Depth	9¾" (248 mm)
Weight	2½lb (1.14 kg)

LabVIEW Driver Software

A LabVIEW driver for these modules is available from the **www.signalrecovery.com** website, offering example VIs for all their controls and outputs, as well as the usual Getting Started and Utility VIs. It also includes example soft-front panels built using these VIs, demonstrating how you can incorporate them in more complex LabVIEW programs.

Graphic display windows allow data curves to be plotted as a function of time, and the driver supports the model 9650A digital delay generator for use in waveform-recovery experiments.



Why should you choose *SIGNAL RECOVERY* products?

Model 4161A Dual Channel ADC

SIGNAL RECOVERY **Product Features**	**Benefit to you**
• Two channel ADC	Includes hold off circuit to prevent triggering until software is ready to read resulting data
• Digital panel meter	Accurate display of output voltages
• Analog panel meter	Eases setting of baseline zeros
• Excellent LabVIEW driver	Supports static gate experiments

Model 4006

NIM Bin & Power Supply

FEATURES

- **Accomodates up to 3 Model 4121B / 4161A NIM Modules**
- **Compact size**
- **Integral overvoltage protection**
- **LED fault indicators**
- **Up to 120 W of DC power available**

APPLICATIONS

- **Boxcar Averager systems using one, two or three modules**
- **Benchtop systems where a conventional NIM rack is too large**

DESCRIPTION

The compact Model 4006 Minibin and Power Supply is the ideal solution for building boxcar systems using our boxcar modules. Its compact 9½" x 12½" (240 mm x 320 mm) footprint minimizes the space required on the benchtop. It can operate at full power while sitting on a solid surface, because rear intake and exhaust of cooling air eliminates the need to provide free air flow from below. The Model 4006 accommodates up to three dual-width NIM modules such as the models 4121B and 4161A, although it can also be used with single-width modules as well.

In addition to the standard ±24 V and ±12 V dc power, ±6 V is provided to serve the high-current demands of TTL and ECL logic such as is used in the models 4121B and 4161A. The unit includes extensive protection for the power supply. Crowbar circuits are provided on the ±6 V power lines to protect TTL and ECL integrated circuits against overvoltage, while all six of the DC power lines incorporate protective fold-back circuits that automatically reduce the output voltage in case of an excessive load current or a short circuit.

Six status LEDs indicate the status of each DC output, glowing green when the supply is operating correctly and red under fault conditions. A further LED shows the temperature of the internal heatsinks, turning on when the temperature is within 15°C of its maximum safe value. All heatsinks are internal to the unit and are forced-air cooled, so there are no hot external surfaces that can be accidentally touched.

Specifications

General
NIM Bin and Power Supply with line input for accommodating and powering up to 6 single-width (or 3 dual width) NIM modules

Line Input
Standard IEC connector and selectable 100, 120, 220 and 240 V AC input voltages at 50 or 60 Hz

Output Voltages - NIM Connectors
±24 V DC, ±12 V DC, ±6 V DC and 115 V AC

Output Voltages - Preamp Power Connectors
Two 9-pin subminiature D type female connectors are also provided and can be used for powering auxiliary apparatus. These outputs are in parallel with the NIM connectors and supply the following voltages:
±24 V DC, ±12 V DC and ±6 V DC

Total Power Output
The total current on each rail must not exceed ±24 V DC @ 750 mA, ±12 V DC @ 1.5 A, ±6 V DC @ 4.0 A and 115 V AC @ 500 mA, subject to a total power output that is dependent on the ambient temperature, being 120 W DC at 23°C and 80 W DC at 50°C

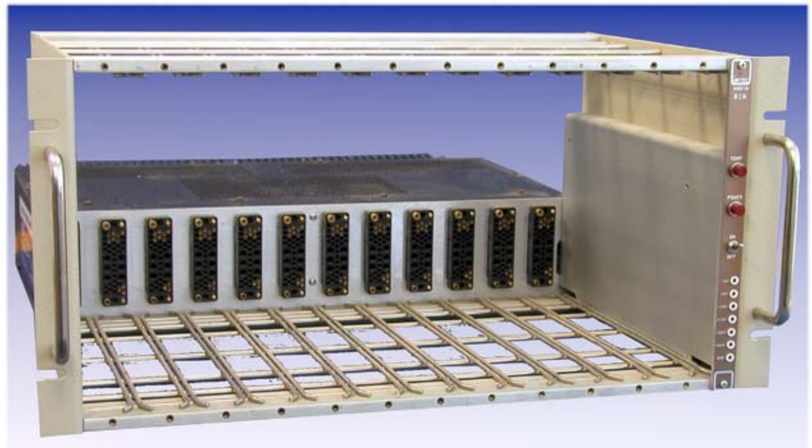
Indicators
LED indicators show status of DC output voltages, internal temperature and line input

Dimensions
9½" wide x 12½" deep x 14" tall
(240 mm x 320 mm x 352 mm)

Weight
26 lb (12 kg)

Model 4001A/4002D

NIM Bin & Power Supply



FEATURES

- **Accomodates up to 6 Model 4121B / 4161A NIM Modules**
- **Fits 19” rack**
- **Integral overvoltage protection**
- **LED power and temperature warning indicators**
- **DC voltage monitoring points**

APPLICATIONS

- **Boxcar Averager systems using up to six modules**
- **Rack mounted systems**

DESCRIPTION

The 4001A/4002D NIM Bin and Power Supply is suitable for accommodating multi-channel boxcar averager systems using up to six of our boxcar modules. The bins are constructed of wire-form grids to ensure unimpeded ventilation for the instruments operated within the enclosure. All DC and AC power levels from the power supply are distributed via a wiring harness.

The integral 4002D power supply supplies up to 160 W of DC power, and includes overload and overvoltage protection for all of the outputs.

Specifications

General

NIM Bin and Power Supply with line input for accommodating and powering up to 12 single-width (or 6 dual width) NIM modules

Line Input

Standard IEC connector and selectable 100, 120, 220 and 240 V AC input voltages at 50 or 60 Hz

Output Voltages - NIM Connectors

±24 V DC, ±12 V DC, ±5 V DC and 115 V AC

Total Power Output

The total current on each rail must not exceed ±24 V DC @ 1.5 A, ±12 V DC @ 3.0 A, ±5 V DC @ 10.0 A and 115 V AC @ 500 mA, subject to a total power output of 160 W DC

Indicators

Indicators show when unit is turned on and warn of temperature overrange conditions

Dimensions

19” wide x 21¼” deep x 8¾” tall (483 mm x 540 mm x 222 mm)

Weight

36 lb (16.3 kg)

Model 3820

Universal Counter

FEATURES

- **Measures Frequency, Period, Duty Cycle, Pulse High/Low Time, Logic Level**
- **Counts periodic or random pulses**
- **Complete with software that acquires, displays, and saves data under Windows XP/Vista**
- **USB interface for power and control**
- **ActiveX control for use with LabVIEW, C++, Visual Basic and VBA**

APPLICATIONS

- **Photon Counting**
- **Electronics R&D**
- **Logic testing**
- **Frequency monitoring**

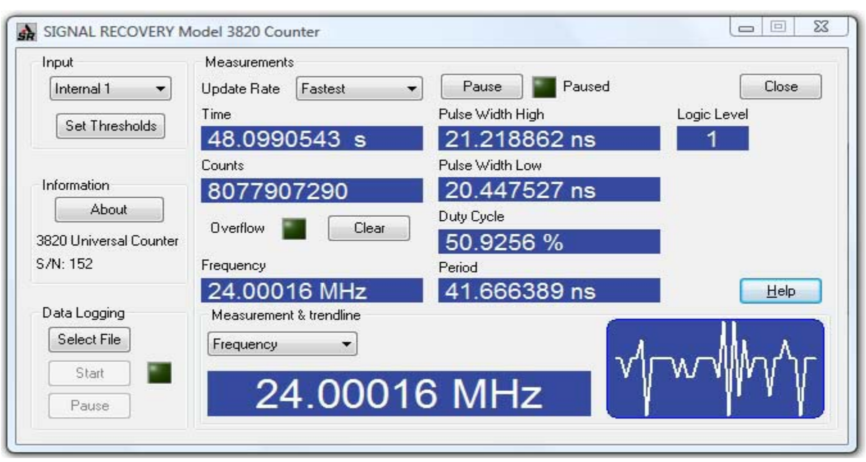
DESCRIPTION

The Model 3820 Universal Counter is a compact and cost effective tool for characterizing analog and digital pulses of a periodic or random nature. It measures frequency, period, duty cycle, pulse high and low times, event counts and logic level all as a function of elapsed time.

The counter is principally intended for counting bi-level signals, which have two distinct voltage levels and clean transitions between them. Such signals include those generated by all common logic families used in electronic circuits, as well as most Trigger or Sync outputs of common test instruments. Measurements are updated at one of five user selectable intervals in the range 5 ms to 100 ms. The module is powered and controlled directly from the PC's USB port, so requires no additional power source. Operation is entirely via software, with no manual switches or settings.

Two inputs are provided, each connected to a separate discriminator with adjustable threshold in the range –0.2 V to +0.5 V (–2.0 V to +5.0 V when used with a x10 probe). Following the discriminators, a multiplexer selects one of the signals for processing, allowing two different signals to be measured sequentially.

The supplied instrument control software consists of two layers. At the upper level, a simple Windows dialog application, "SR3820Counter", offers a convenient panel that allows the input (A, B or one of three internal test sources) to be selected, an update rate to be specified and all eight output measurements to be displayed. A further display area shows one of the measurements in a larger font size, as well as displaying a graphical trendline display.



3820 Control/Display Software

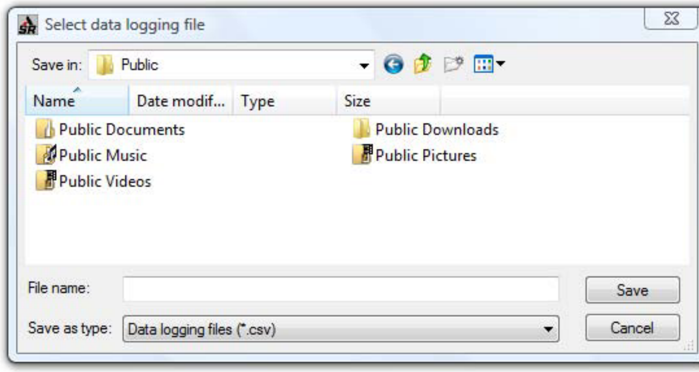
The program also supports data logging to text file of the output measurements, with data being written directly in CSV (comma separated value) format for easy import to other programs. The software includes a sub menu where the voltage input thresholds can be set for the two inputs.

At the lower level, a dedicated ActiveX control known as "SR3820Comms" takes care of all communications to and from the instrument. Two main modes of operation are therefore possible. Users who simply want to operate the counter "out of the box" need do no more than plug it in, install the driver and software, and then use the SR3820 Universal Counter software to control it. Alternatively, when the counter is to be used as part of a computer controlled test system, then the user can develop software to control it via the SR3820Comms ActiveX control. The control eliminates the need for users to write the low-level code needed to send commands to and receive responses from the counter, allowing them to concentrate on developing the higher level program to run their experiments. Typical applications include:

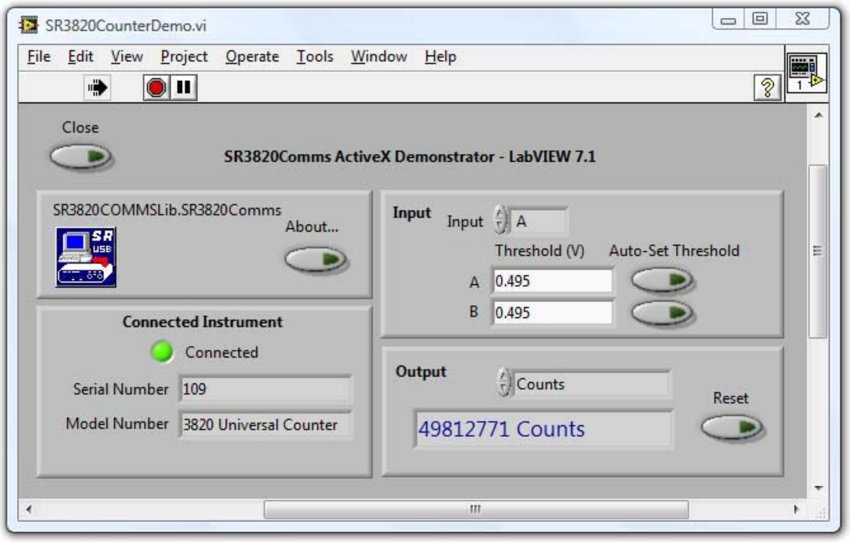
- Photon counting
- Frequency measurement
- Test and measurement systems implement in LabVIEW where a ***SIGNAL RECOVERY*** 3820 counter can be used at the same time as instruments from different suppliers.
- Measurement system using scripted web pages (HTML files) operated via Internet Explorer.

The SR3820Comms control can of course also be used at the same time as other ***SIGNAL RECOVERY*** software ActiveX controls, such as SR3830Comms, allowing sophisticated systems to be assembled. For example, five APD's (avalanche photodiodes) could each be connected to the inputs of a model 3830 multiplexer, with the output being in turn connected to the A input of a model 3820 counter. Using both controls a user-developed application program could sequentially count the pulses being generated by each APD.

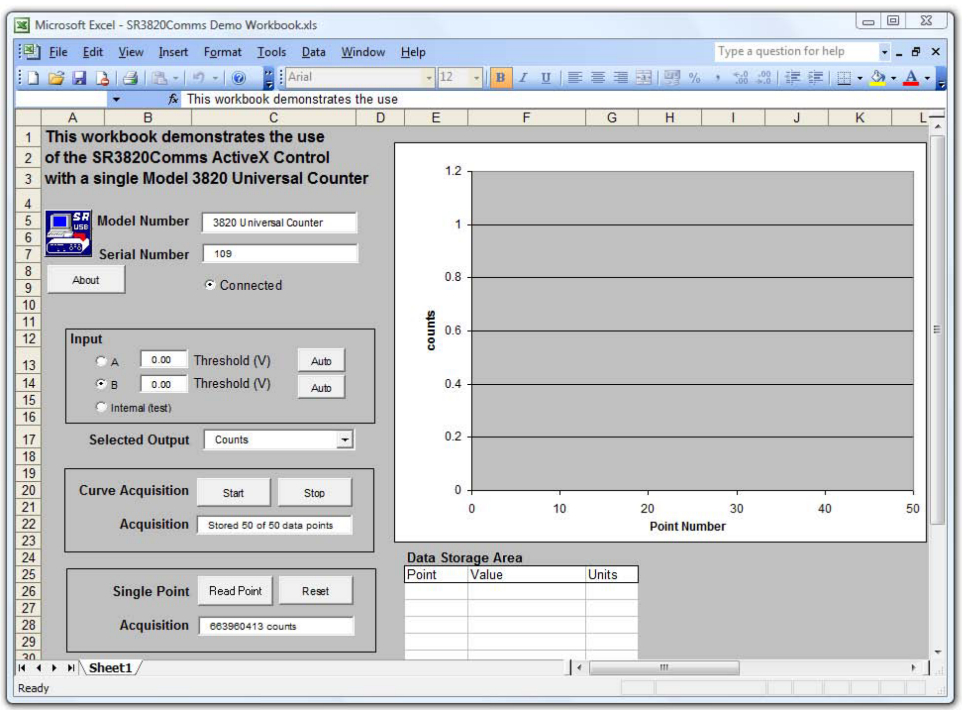
Both the top level SR3820 Universal Counter software and the lower-level SR3820Comms ActiveX control include comprehensive on-screen help files, while examples of how to use the control in LabVIEW, Visual Basic, VBScript, Visual C++, and Excel are also supplied.



Log Measured Data to File



LabVIEW Driver



Sample Excel Workbook using SR3820Comms

Specifications

General

Dual input discriminator, single channel counter measuring frequency, period, duty cycle, pulse high and low times, logic level and event counts as a function of time. Power and control via USB and supplied software. ActiveX control included.

Inputs

Impedance	1 MΩ, DC coupled
Threshold	
Direct	-0.2 V to 0.5 V in 1 mV steps
With x10 probe	-2 V to 5 V in 10 mV steps
Polarity	Event counter triggers on rising edge of signal
Sensitivity	
Direct	-15 dBm/50 Ω (23 mV rms) at 100 MHz, -10 dBm/50 Ω (0.7 V rms) at 120 MHz
With x10 probe	1 V pk-pk sinewave at 125 MHz
Absolute max input	≤ 50 V DC

Measurement

Frequency Range	DC to 125 MHz min, 160 MHz typ
Timebase accuracy	≤ 50 ppm, 0 to 50°C
Reporting Intervals	100, 50, 25, 10 and 5 ms
Functions	
Frequency	0 to 125 MHz
Avg Period	≥ 8 ns
Avg Duty Cycle	0 to 100%. Measured by sampling with a 65 MHz clock.
Avg Pulse High or Low Time	≥ 5ns. Computed from Duty Cycle and Frequency
Events	0 to 9,999,999,999 counts
Logic Probe	0 = input voltage below threshold, 1 = input voltage above threshold.

Indicators

USB	Front-panel LED turns on during USB communications
EVENT	Front-panel LED turns on when the input signal is above the threshold

USB Connector	Rear-panel, female USB connector for connection to the PC or a USB hub.

General

Power Requirements	
Voltage	<500 mA @ +5 V DC, supplied via USB
Dimensions	
Width	5½" (134 mm)
Depth	4½" (114 mm)
Height	1¼" (32 mm)
Weight	9.9 oz (280 g)

Software

A CD containing the full applications package for Windows XP/Vista (32-bit versions) allowing threshold and input to be adjusted, and measurements to be displayed and saved is supplied with each unit. SR3820Comms ActiveX control also included for use with compatible programming languages, and examples provided of its use in C++, VisualBasic, VBScript, LabVIEW and Excel. Both top-level and ActiveX software include on-screen help. Instruction manual supplied in both printed and PDF formats.

Model 3830

Multiplexer

FEATURES

- **Six BNC connectors for inputs and/or outputs**
- **Reed-relay DPST switching**
- **LED indicators show relays that are energized**
- **Complete with software for control from Windows XP/ Vista**
- **USB interface for power and control**
- **ActiveX control for use with LabVIEW, C++, Visual Basic, and VBA**

APPLICATIONS

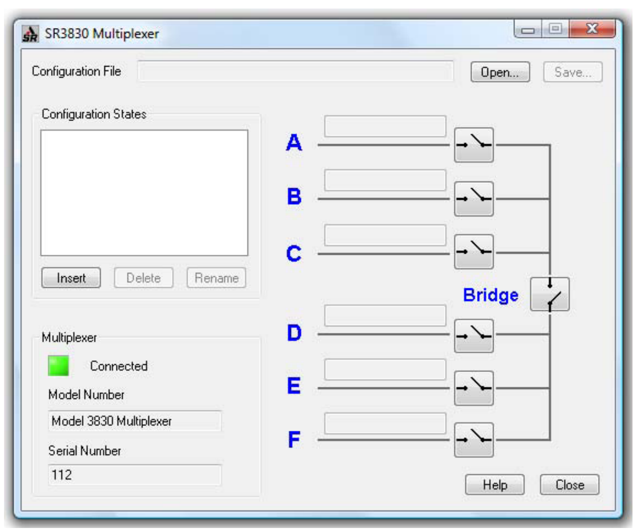
- **Input and output signal multiplexing**
- **Computerized test systems**

DESCRIPTION

The Model 3830 Multiplexer (mux) makes it easy to implement computer-selected interconnections between different instruments. Six floating BNC connectors can be used as either inputs or outputs and are coupled to one of two common buses via DPST reed-relay switches. This ensures that both inner and outer parts of the connector are isolated when the relevant relay is not energized. The two buses can also be connected together via a seventh "bridge" reed relay, allowing operation either as two, two-input, one-output muxes or as a five-input, one-output mux. Other configurations are of course possible.

The unit is powered and controlled directly from the PC’s USB port, so requires no additional power source. Operation is entirely via software, with no manual switches or settings on the module.

The supplied instrument control software consists of two layers. At the upper level, a simple Windows dialog application, "SR3830 Multiplexer", offers a convenient panel that allows switching patterns, known as "states", to be set up graphically and combined into "configuration" files.



Easy to use SR3830 Multiplexer applications software

Configuration files can be saved to and recalled from disk. Using these files makes replication of experiments very straightforward.

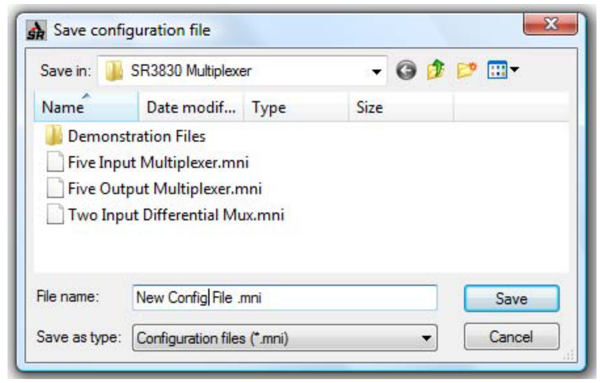
Two main modes of operation are therefore possible. Users who simply want to operate the multiplexer "out of the box" need do no more than plug it in, install the driver and software, and then use the SR3830 Multiplexer software to control it.

Alternatively, when the multiplexer is to be used as part of a computer controlled test system, then the user can develop software to control it via the SR3830Comms ActiveX control. The control eliminates the need for users to write the low-level code needed to send commands to and receive responses from the multiplexer, allowing them to concentrate on developing the higher level program to run their experiments. Typical applications include:

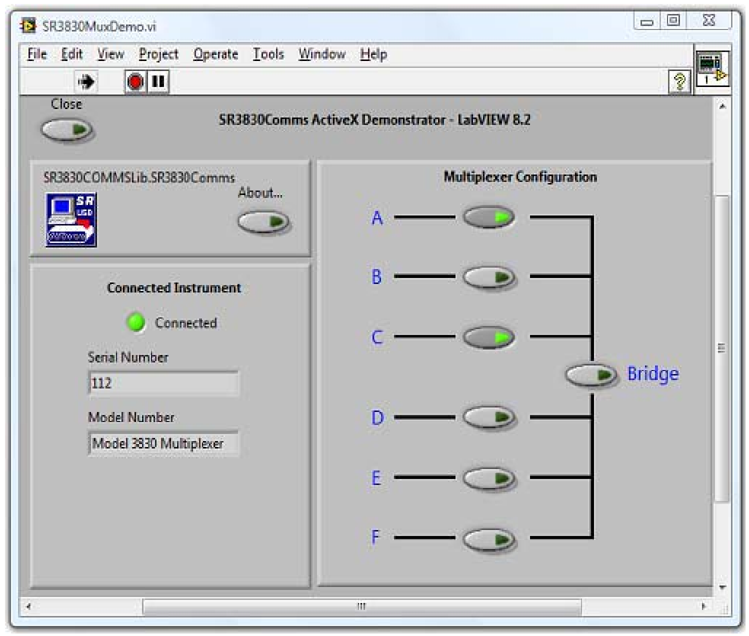
- Switching one of several sensor outputs to the input of a measuring instrument, such as a voltmeter or oscilloscope
- Connecting a signal source, such as an oscillator, to one of several actuators or drive coils
- Test and measurement systems implement in LabVIEW where a ***SIGNAL RECOVERY*** multiplexer needs to be controlled at the same time as instruments from different suppliers.
- Measurement system using scripted web pages (HTML files) operated via Internet Explorer.

The SR3830Comms control can of course also be used at the same time as other ***SIGNAL RECOVERY*** software ActiveX controls, such as SRInstComms, allowing sophisticated systems to be assembled. For example, a group of five photodiodes could be connected to the current preamplifier input of a model 7270 DSP lock-in amplifier via the model 3830 and then, using both controls a user-developed application program could sequentially measure the signal on each photodiode.

Both the top level SR3830 Multiplexer software and the lower-level SR3830Comms ActiveX control include comprehensive on-screen help files, while examples of how to use the control in LabVIEW, Visual Basic, VBScript, Visual C++, and Excel are also supplied. Also included is an instruction manual in both printed and PDF formats.



Save and Recall Switch Configurations



Sample LabVIEW Application Program

Specifications

General

Six way input/output multiplexer offering isolated switching via reed relays. Power and control via USB and supplied software. ActiveX control included.

Inputs/Outputs

Connectors	BNC, no internal load
Switches	Reed relay, DPST
Max Voltage	50 V between BNC inners and outers and from BNC to ground
Max Current	200 mA
Contact Resistance	< 0.15 Ω

Switching

Six DPST relays connect A, B, C to bus 1 and D, E F to bus 2. Seventh “bridge” relay connects bus 1 to bus 2

Indicators

LED indicates when corresponding relay is energized

USB Connector

Rear-panel, female USB connector for connection to the PC or a USB hub.

General

Power Requirements	
Voltage	<500 mA @ +5 V DC, supplied via USB
Dimensions	
Width	5½" (134 mm)
Depth	4½" (114 mm)
Height	1¼" (32 mm)
Weight	9.9 oz (280 g)

Software

A full applications package for Windows XP/Vista (32-bit versions) allowing any switching combination to be set up graphically is provided. Patterns can be saved and recalled from disk. ActiveX control also included for use with compatible programming languages. Both top-level and ActiveX software include on-screen help. Instruction manual supplied in both printed and PDF formats.

SIGNAL RECOVERY

What is a Lock-in Amplifier?

TECHNICAL NOTE
TN 1000

In its most basic form a lock-in amplifier is an instrument with dual capability. It can recover signals in the presence of an overwhelming noise background or, alternatively, it can provide high resolution measurements of relatively clean signals over several orders of magnitude and frequency. However, modern instruments offer far more than these two basic functions and this increased capability has led to their acceptance, in many scientific disciplines, as units which can provide the optimum solution to a large range of measurement problems. For example, the modern lock-in amplifier will function as:-

- È **an AC Signal Recovery Instrument**
- È **a Vector Voltmeter**
- È **a Phase Meter**
- È **a Spectrum Analyzer**
- È **a Noise Measurement Unit**
- È **…..and much more**.

It is this versatility, available in a single compact unit, which makes the lock-in amplifier an invaluable addition to any laboratory. This Technical Note describes the basic "building blocks" of the lock-in amplifier so that the user and potential user may better understand how the instruments work and how the choices made in their design affect their performance.

Introduction

A lock-in amplifier, in common with most AC indicating instruments, provides a DC output proportional to the AC signal under investigation. In modern units the DC output may be presented as a reading on a digital panel meter or as a digital value communicated over a computer interface, rather than a voltage at an output connector, but the principle remains the same.

The special rectifier, called a phase-sensitive detector (PSD), which performs this AC to DC conversion forms the heart of the instrument. It is special in that it rectifies only the signal of interest while suppressing the effect of noise or interfering components which may accompany that signal.

The traditional rectifier, which is found in a typical AC voltmeter, makes no distinction between signal and noise and produces errors due to rectified noise components. The noise at the input to a lock-in amplifier, however, is not rectified but appears at the output as an AC fluctuation. This means that the desired signal response, now a DC level, can be separated from the noise accompanying it in the output by means of a simple low-pass filter. Hence in a lock-in amplifier the final output is not affected by the presence of noise in the applied signal.

In order to function correctly the detector must be "programmed" to recognize the signal of interest. This is achieved by supplying it with a reference voltage of the same frequency and with a fixed phase relationship to that of the signal. This is most commonly done by ensuring that they are derived from the same source. The use of such a reference signal ensures that the instrument will "track" any changes in the frequency of the signal of interest, since the reference circuit is "locked" to it. It is from this characteristic that the instrument derives its name.

This inherent tracking ability allows extremely small bandwidths to be defined for the purpose of signal-to-noise ratio improvement since there is no frequency "drift", as is the case with analog "tuned filter/rectifier" systems. Because of the automatic tracking, lock-in amplifiers can give effective "Q" values (a measure of filter selectivity) in excess of 100,000, whereas a normal bandpass filter becomes difficult to use with Q's greater than 50.

Phase-Sensitive Detection

As mentioned above, the heart of the lock-in amplifier is the phase-sensitive detector (PSD), which is also known as a demodulator or mixer. The detector operates by multiplying two signals together, and the following analysis indicates how this gives the required outputs.

Figure 1 shows the situation where the lock-in amplifier is detecting a noise-free sinusoid, identified in the diagram as "Signal In". The instrument is also fed with a reference signal, from which it generates an internal sinusoidal reference which is also shown in the diagram.

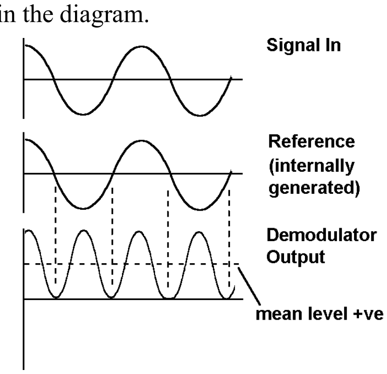


Figure 1

The demodulator operates by multiplying these two signals together to yield the signal identified in the diagram as "Demodulator Output". Since there is no relative phase-shift between the signal and reference phases, the demodulator output takes the form of a sinusoid at twice the reference frequency, but with a mean, or average, level which is positive.

Figure 2 shows the same situation, except that the signal phase is now delayed by 90° with respect to the reference. It can been seen that although the output still contains a signal at twice the reference frequency, the mean level is now zero.

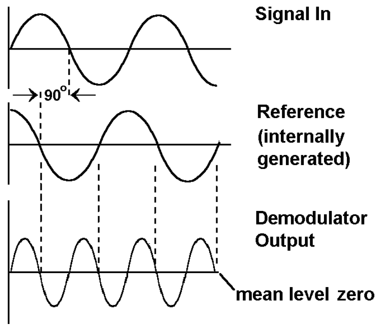


Figure 2

From this it can be seen that the mean level is:-

È proportional to the product of the signal and reference frequency amplitudes

È related to the phase angle between the signal and reference.

It will be appreciated that if the reference signal amplitude is maintained at a fixed value, and the reference phase is adjusted to ensure a relative phase-shift of zero degrees, then by measuring the mean level the input signal amplitude can be determined.

The mean level is, of course, the DC component of the demodulator output, so it is a relatively simple task to isolate it by using a low-pass filter. The filtered output is then measured using conventional DC voltmeter techniques.

The above discussion is based on the case of noise-free input signals, but in real applications the signal will be accompanied by noise. This noise, which by definition has no fixed frequency or phase relationship to the reference, is also multiplied by the reference signal in the demodulator, but does not result in any change to the mean DC level. Noise components at frequencies very close to that of the reference do result in demodulator outputs at very low frequencies, but by setting the low-pass filter to a sufficiently low cut-off frequency these can be rejected. Hence the combination of a demodulator and low-pass output filter allows signals to be measured even when accompanied by significant noise.

Those readers who are interested in a mathematical derivation of the same conclusions should refer to the Appendix at the end of this Technical Note.

The Typical Lock-In Amplifier

The block diagram of a typical lock-in amplifier is shown in figure 3. Readers should be aware that the following discussion makes no assumptions as to the technology used to implement each of the circuit elements and that analog, mixed technology and digital methods may be used.

Signal Channel

In the signal channel the input signal, including noise, is amplified by an adjustable-gain, AC-coupled amplifier, in order to match it more closely to the optimum input signal range of the PSD. Instruments are usually fitted with high impedance inputs for voltage measurements. Many also incorporate low impedance inputs for better noise matching to current sources, although in some cases the best results are obtained through the use of a separate external preamplifier.

The performance of the PSD is usually improved if the bandwidth of the noise voltages reaching it is reduced from that of the full frequency range of the instrument. To achieve this, the signal is passed through some form of filter, which may be simply a band rejection filter centered at the power line frequency and/or its second harmonic to reject line frequency pick-up, or alternatively a more sophisticated tracking bandpass filter centered at the reference frequency.

Reference Channel

It has been shown that proper operation of the PSD requires the generation of a precision reference signal within the instrument. When a high-level, stable and noise-free reference input is provided, this is a relatively simple task. However there are many instances where the available reference is far from perfect or symmetrical, and in these cases a well designed reference channel circuit is very important. Such circuits can be expensive and often account for a significant proportion of the total cost of the instrument.

The internally generated reference is passed through a phase-shifter, which is used to compensate for phase differences that may have been introduced between the signal and reference inputs by the experiment, before being applied to the PSD.

Phase-sensitive Detector

There are currently three common methods of implementing the PSD, these being the use of an Analog Multiplier, a Digital Switch or a Digital Multiplier.

Analog Multiplier

In an instrument with an analog multiplier, the PSD comprises an electronic circuit which multiplies the applied signal with a sinewave at the same frequency as the applied reference signal.

Although the technique is very simple in principle, in practice it is difficult to manufacture an analog multiplier which is capable of operating linearly in the presence of large noise, or other interfering, signals. Non-linear operation results in poor noise rejection and thereby limits the signal recovery capability of the instrument.

Digital Switching Multiplier

The switching multiplier uses the simplest form of demodulator consisting of an analog polarity-reversing switch driven at the applied reference frequency. The great advantage of this approach is that it is very much easier to make such a demodulator operate linearly over a very wide range of input signals.

However, the switching multiplier not only detects signals at the applied reference frequency, but also at its odd harmonics, where the response at each harmonic relative to the fundamental is defined by the Fourier analysis of a squarewave. Such a response may well be of use if the signal being detected is also a squarewave, but can give problems if, for example, the unit is being used at 1 kHz and there happens to be strong interfering signal at 7 kHz.

As discussed earlier, the use of a tuned low-pass or bandpass filter in the signal channel prior to the multiplier modifies the response of the unit so that it primarily detects signals at the reference frequency. However, in order to fully reject the 3F response, while still offering good performance at the reference frequency, very complex and expensive filters would be required. These are impractical for commercial instruments, so units fitted with filters tend to show some response to signals and noise at the third and fifth harmonics of the reference frequency and relatively poor amplitude and phase stability as a function of operating frequency.

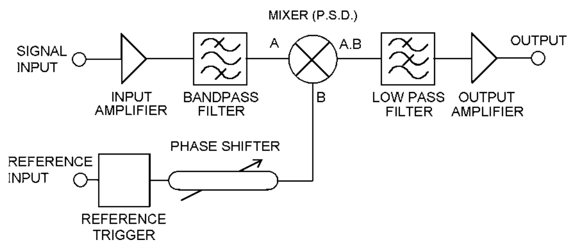


Figure 3

SIGNAL RECOVERY analog lock-in amplifiers use an alternative and more sophisticated type of switching demodulator which replaces the single analog switch with an assembly of several switches driven by a Walsh function. This may be thought of as a stepped approximation to a sinewave. Careful selection of components allows such a demodulator to offer all of the advantages of the switching demodulator with one additional benefit, which is the complete rejection of the responses at the third and fifth harmonics and reduced responses for higher orders.

Such a demodulator, when used with a relatively slow roll-off, 4th-order, low-pass filter in the signal channel, produces an overall response very near to the ideal. In this case the demodulator rejects the third and fifth harmonic responses and the higher orders are removed by the signal channel filter.

Digital Multiplier

In an instrument employing this type of multiplier the input signal is amplified and then immediately digitized. This digital representation is then multiplied by a digital representation of a sinewave at the reference frequency. A digital signal processor (DSP) is used for this task and the output is therefore no longer an analog voltage but rather a series of digital values.

The technique offers the advantages of a perfect multiplication with no inherent errors and minimizes the DC coupled electronics that are needed with other techniques, thereby reducing output drift. It has been used for a number of years in such applications as swept-frequency spectrum analyzers.

There are, however, a number of major problems with this method when applied to recovering signals buried in noise. The most important of these is dynamic range. Consider the case of an input signal in the presence of 100 dB (100,000 times larger) of noise. If the signal is to be digitized to an accuracy of "n" bits then the input converter must handle a dynamic range of 2n × 100,000 to fully accommodate the signal and noise amplitudes. With a typical value for n of 15, this equates to a range of 3.2 × 109:1, corresponding to 32 bits. An analog to digital converter (ADC) can be built with such an accuracy, but would be extremely expensive and quite incapable of the sampling rates needed in a lock-in amplifier operating to 100 kHz.

Practical digital lock-in amplifiers use a 16 or 18-bit ADC. Consequently, in the presence of strong interfering signals, the required signal may only be changing the least significant bits of the converter, and indeed may actually be so small that there is no change at all in the ADC output. Hence the measurement resolution of an individual output sample is very coarse. Resolution may be improved however by averaging many such samples. For example 256 samples of 1-bit resolution can average to 1 sample of 8-bit resolution, but this is at the expense of reduced response time.

This averaging only operates predictably if the spectral power distribution of the interfering noise is known. If it is not, then noise has to be added by the instrument from its own internal noise source to ensure that it dominates. The addition of this noise, which is only needed in demanding signal recovery situations, tends to lengthen the response time for a given measurement accuracy compared to an analog type of instrument.

Low-pass Filter and Output Amplifier

As mentioned earlier, the purpose of the output filter is to remove the AC components from the desired DC output. Practical instruments employ a wide range of output filter types, implemented either as analog circuits or in digital signal processors. Most usually, however, these are equivalent to one or more stages of simple single-pole "RC" type filters, which exhibit the classic 6 dB/octave roll-off with increasing frequency.

There is usually also some form of output amplifier, which may be either a DC-coupled analog circuit or a digital multiplier. The use of this amplifier, in conjunction with the input amplifier, allows the unit to handle a range of signal inputs. When there is little accompanying noise, the input amplifier can be operated at high gain without overloading the PSD, in which case little, if any, gain is needed at the output. In the case of signals buried in very large noise voltages, the reverse is the case.

Output

The output from a lock-in amplifier was traditionally a DC voltage which was usually displayed on an analog panel meter. Nowadays, especially when the instruments are used under computer control, the output is more commonly a digital number although the analog DC voltage signal is usually provided as well. Units using an analog form of phase-sensitive detector use an ADC to generate their digital output, whereas digital multiplying lock-in amplifiers use a digital to analog converter (DAC) to generate the analog output.

Single Phase and Dual Phase

The discussion above is based around a single-phase instrument. A development of this is the dual-phase lock-in amplifier, which is not, as some people think, a dual channel unit. Rather it incorporates a second phase-sensitive detector, which is fed with the same signal input as the first but which is driven by a reference signal that is phase-shifted by 90 degrees. This second detector is followed by a second output filter and amplifier, and is usually referred to as the "Y" output channel. The original output being referred to as the "X" channel.

An advantage of the dual-phase unit is that if the signal channel phase changes (but not its amplitude) then although the output from one detector will decrease, that from the second increases. It can be shown, however, that the vector magnitude, R, remains constant, where:- $R = \cdot (X^2 + Y^2)$

Hence if the lock-in amplifier is set to display R, changes in the signal phase will not affect the reading and the instrument does not require the adjustment of the reference phase-shifter circuit. This capability has led to the dual-phase instrument becoming by far the most common type of unit.

Internal Oscillator

All lock-in amplifiers use some form of oscillator within their reference circuits. Many units however also have a separate internal oscillator which can be used to generate an electrical stimulus for the experiment, usually with user-adjustable frequency and amplitude.

Computer Control

Virtually all modern instruments include a microprocessor. This can simplify and automate manual measurements as well as supporting remote control of the instrument over common computer interfaces, such as the GPIB (IEEE-488) and RS232 links. The ability of the microprocessor to perform mathematical manipulations adds such useful functions as vector phase and noise measurements to the basic signal recovery capabilities of the lock-in amplifier.

Further Information

This Technical Note is intended as an introduction to the techniques used in lock-in amplifiers. Additional information may be found in other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative, or by download from our website at **www.signalrecovery.com**

TN 1001	Specifying a Lock-in Amplifier
TN 1002	The Analog Lock-in Amplifier
TN 1003	The Digital Lock-in Amplifier
TN 1004	How to Use Noise Figure Contours
AN 1000	Dual-Channel Absorption Measurement with Source Intensity Compensation
AN 1001	Input Offset Reduction using the Model 7265/7260/7225/7220 Synchronous Oscillator/Demodulator Monitor Output
AN 1002	Using the Model 7225 and 7265 Lock-in Amplifiers with software written for the SR830
AN 1003	Low Level Optical Detection using Lock-in Amplifier Techniques
AN 1004	Multiplexed Measurements using the 7225, 7265 and 7280 Lock-in Amplifiers
AN 1005	Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper

Appendix

Consider the case where a noise-free sinusoidal signal voltage Vin is being detected, where

$$V_{in} = A\cos(\varpi t)$$

ϖ is the angular frequency of the signal which is related to the frequency, F, in hertz by the equality:-

$$\varpi = 2\,\phi F$$

The lock-in amplifier is supplied with a reference signal at frequency F derived from the same source as the signal, and uses this to generate an internal reference signal of:-

$$V_{ref} = B\cos(\varpi t + \chi)$$

where χ is a user-adjustable phase-shift introduced within the lock-in amplifier.

The detection process consists of multiplying these two components together so that the PSD output voltage is given by:-

$$\begin{aligned}
V_{psd} &= A\cos(\varpi t)\,.\,B\cos(\varpi t + \chi)\\
&= AB\cos\varpi t\,(\cos\varpi t\cos\chi - \sin\varpi t\sin\chi)\\
&= AB(\cos2\,\varpi t\cos\chi - \cos\varpi t\sin\varpi t\sin\chi)\\
&= AB((\tfrac{1}{2} + \tfrac{1}{2}\cos 2\varpi t)\cos\chi - \tfrac{1}{2}\sin 2\varpi t\sin\chi)\\
&= \tfrac{1}{2}AB((1+\cos 2\varpi t)\cos\chi - \sin 2\varpi t\sin\chi)\\
&= \tfrac{1}{2}AB(\cos\chi + \cos 2\varpi t\cos\chi - \sin 2\varpi t\sin\chi)\\
&= \tfrac{1}{2}AB\cos\chi + \tfrac{1}{2}AB(\cos 2\varpi t\cos\chi - \sin 2\varpi t\sin\chi)\\
&= \tfrac{1}{2}AB\cos\chi + \tfrac{1}{2}AB\cos(2\varpi t + \chi)
\end{aligned}$$

If the magnitude, B, of the reference frequency is kept constant, then the output from the phase-sensitive detector is a DC signal which is:-

È proportional to the magnitude of the input signal A

È proportional to the cosine of the angle, χ, between it and the reference signal

È modulated at $2\varpi t$, i.e. it contains components at twice the reference frequency.

The output from the PSD then passes to a low-pass filter which removes the $2\varpi t$ component, leaving the output of the lock-in amplifier as the required DC signal.

In a practical situation the signal will usually be accompanied by noise, but it can be shown that as long as there is no consistent phase (and therefore by implication frequency) relationship between the noise and the signal, the output of the multiplier due to the noise voltages will not be steady and can therefore be removed by the output filter.

SIGNAL RECOVERY

Specifying Lock-in Amplifiers

TECHNICAL NOTE
TN 1001

Introduction

This Technical Note discusses the reasons why users choose lock-in amplifiers for their measurements and defines the terms used to describe the instruments' performance. It is assumed that readers already have a basic understanding of the operation of the lock-in amplifier, but if this is not the case then the ***SIGNAL RECOVERY*** Technical Note TN1000, "What is a Lock-in Amplifier?", may prove helpful.

Why use a Lock-In Amplifier?

The main reason for using a lock-in amplifier is to recover signals from noise. Consider the following example.

Let the signal to be measured be a:-

È 20 nV sinewave at 50 kHz.

Obviously some amplification is needed before it can be measured, whether on an AC voltmeter, an oscilloscope or a lock-in amplifier. Any amplifier will add noise to the signal, but for a good unit this might be:-

È Input noise Ω4 nV· Hz

at a:-

È Bandwidth of 1 MHz, and a

È Gain of 1000.

For the above signal the output signal of this amplifier will be:-

È 20 µV (1000 × 20 nV)

accompanied by:-

È 4 mV r.m.s. (· 1 MHz × 4 nV × 1000)

of broadband noise. Clearly it is impossible to measure the signal of interest in the presence of this level of noise, unless something is done to isolate it. One means of achieving this is the use of a filter before the amplifier.

If the signal is first passed through a bandpass filter with:-

È a center frequency of 50 kHz, and

È a *Q* of 100 (this implies a 3 dB bandwidth of 500 Hz - a specification which is difficult to attain),

then any signal within this 500 Hz bandwidth will be detected. The noise within this bandwidth is:-

È 89 µV (· 500 Hz × 4 nV × 1000)

but the signal is still only:-

È 20 µV

and so it will still not be possible to make a measurement.

Now consider feeding the same signal to a lock-in amplifier. bandwidth of 0.125 Hz, when the output time constant is 1 second and the filter slope is 12 dB/octave.

The noise accompanying the signal now will be only:-

È 1.4 µV (· (0.125 Hz × 4 nV × 1000))

implying that it will be possible to make a measurement of the 20 µV input signal. Performance can be further improved by using longer time constants, at the expense of increased time to complete the measurement. As specified, it would take around 5 seconds for the lock-in amplifier's output to stabilize following the application of the 20 µV signal. Hence the signal has *been recovered from the noise* by the lock-in amplifier.

What does a Lock-In Amplifier Measure?

Using Fourier's theorem, any input signal, including the noise accompanying it, can be represented as the sum of many sinewaves of different amplitudes, phases and frequencies. The phase-sensitive detector in the lock-in amplifier multiplies all these components by a signal at the reference frequency. In the case of a **sinewave-responding** (also known as a **fundamental responding**) instrument, the output is a DC signal proportional to the component of the input signal which is exactly in phase and frequency lock with the reference signal. **Squarewave-responding**, or **flat-responding**, lock-in amplifiers give a DC output proportional to the components of the input signal in phase lock with the reference signal and its odd harmonics, where the relative response at the "*n*"th harmonic is given by 1/*n*.

The two different types of response are important when comparing the measurements made using a lock-in amplifier with those made using other instruments, for example an oscilloscope. Consider the case of a 2 volt peak-to-peak noise-free sinusoidal signal, which can be expressed mathematically as:-

V = sin ϖ*t*

where ϖ = 2 φ

Both the sinewave-responding and the squarewave-responding lock-in amplifiers, when locked to the reference frequency *f* and correctly phase-adjusted for maximum response, would indicate a reading of:-

V = · 2/2 volts

= 0.707 volts

This is the r.m.s. value of a 2 volt peak-to-peak sinewave.

Now consider the case of a 2 volt peak-to-peak squarewave at frequency *f*, which according to Fourier's theorem can be expressed as:-

V = 4/ϕ sin ϖ*t* + 4/3ϕ sin 3ϖ*t* + 4/5ϕ sin 5ϖ*t* ...

If this signal were applied to a sinewave-responding lock-in amplifier, the instrument would extract the first (fundamental) component and the display would read:-

V = 4/ϕ · 2/2

= 0.9 volts

If the same signal were applied to a squarewave-responding lock-in amplifier, each of the harmonics would contribute to the output signal, and the display on the lock-in amplifier would read:-

V = 4/ϕ · 2/2 $((1/1)^2 + (1/3)^2 + (1/5)^2 ...)$

V = · 2/2 4/ϕ (1.23)

V = 1.11 volts

In other words, if the input signal is a squarewave, such as is the case in most chopped light signals, a squarewave-responding lock-in amplifier will detect about 23 % (i.e. 1.11/0.9) more signal than is the case when a sinewave-responding lock-in amplifier is used.

Peak-to-Peak or R.M.S.?

When measuring *sinusoidal* input signals, lock-in amplifiers generally display the measured value in volts r.m.s., so that if for example the lock-in amplifier shows a reading of 100 mV, the component of the input signal at the reference frequency is 100 mV r.m.s., or 283 mV peak-to-peak.

Phase Measurements

Lock-in amplifiers always use degrees as the unit of phase, although in some of the mathematics used to describe their operation radians are used. Similarly, frequency *f* is always measured in hertz, although the equations are often simpler if angular frequency, usually termed *"ϖ"*, is used, where:-

ϖ = 2 ϕ*f*

Lock-In Amplifier Specifications

The final section of this technical note seeks to explain some of the specification terminology that may be encountered when choosing a lock-in amplifier.

Input Amplifier

This may be either a voltage or a current input device. Voltage inputs offer the highest possible impedance to the signal to be measured so that they do not affect its level. Good lock-in amplifiers will present an input impedance of at least 10 MT .

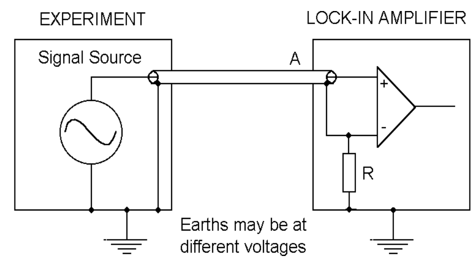


Figure 1, Single-ended Voltage Input

The **single-ended input** (figure 1) is the most commonly used configuration, in which the lock-in amplifier amplifies the difference between the signal at the inner and outer conductors of the input connector. The outer connector, or shield, is not forced to ground potential but is instead connected via a resistor of typically 100 to 1000 ohms. This avoids possible "ground loops" between the source and the instrument, due to differing ground potentials, by allowing the shield to "float" so that the lock-in amplifier can sense the signal source ground.

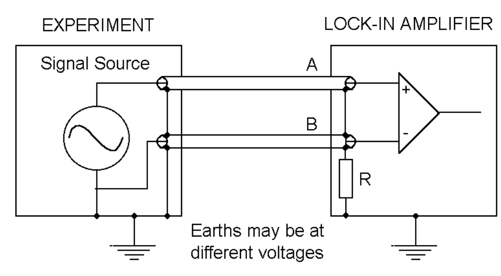


Figure 2, Differential Voltage Input

A **differential input** (figure 2) has two connectors and amplifies the difference in voltage between them. If the two cables used to connect them to the signal source are identical, then any spurious noise pick-up will affect both cables equally and be rejected by the common mode rejection capability of the differential input amplifier. The quality of this rejection is specified by the **Common Mode Rejection Ratio** or **C.M.R.R.**; a figure of at least 100 dB at 1 kHz should be expected, corresponding to the ability to measure a 10 µV signal in the presence of 1 volt of common mode interference.

A **current input** is designed to absorb all of the current offered to it, and as such should have as low an input impedance as possible. It should be used when the signal source impedance is high. Usually in lock-in amplifiers the current input is specified by a **conversion gain**, typically of 10^8 or 10^6 V/A. Hence to find the overall sensitivity of the lock-in amplifier in amps the current conversion range chosen must be multiplied by the full-scale voltage sensitivity being used. The **gain accuracy** of the amplifiers used in the signal channel is a measure of the initial accuracy of the gain calibration, so that for example if a 100 mV signal is being measured, a gain accuracy of 1 % implies that the lock-in amplifier will display a reading between 99 mV and 101 mV.

The **gain stability** may also be specified and defines how the initial gain may be affected by changes in both time and temperature. In a normal laboratory environment it is usual to consider the effects of a 10 °C change in temperature as being the largest that is likely to be encountered.

Dynamic reserve is a term which is often used to describe the signal recovery performance of lock-in amplifiers. It is a measure of how large a discrete asynchronous interfering signal can be before the lock-in amplifier starts to measure the required signal incorrectly. Typically this is determined as the point where the output is in error by 5 %, which is a more demanding specification than simply saying that it is the point where the lock-in amplifier is overloaded. It is usually expressed in decibels (dB), where

$$\text{Dynamic Reserve (in dB)} = 20\log_{10}\left(\frac{\text{Interfering Signal}}{\text{Required Signal}}\right)$$

The same dynamic reserve is not, of course, available at all sensitivity settings, since the peak input is restricted to the range of linear operation of the input amplifier, which is typically a few volts. For example, 60 dB of dynamic reserve is not available at a 1 V full-scale sensitivity since this would imply a 1 kV input capability.

Commercially available lock-in amplifiers offer dynamic reserves up to 130 dB, implying the ability to measure, for example, a 100 nV signal in the presence of a 300 mV interfering signal.

Often confused with dynamic reserve, **dynamic range** is the ratio of the peak signal input that can be measured without overloading to the minimum detectable signal.

Reference Channel

The reference input has to accept the applied reference signal and generate from it an accurate reference frequency for the lock-in amplifier to use. Reference circuits should be capable of responding to any periodic waveform, with two zero crossings per cycle, by adjusting the trigger threshold. The minimum and maximum reference levels are usually specified as well.

Any lock-in amplifier, when operating from an external reference, needs time to establish lock following a change in the applied reference. This is defined as the **lock acquisition time**.

The **phase-shifter resolution** defines the smallest phase increment with which the reference drive to the phase-sensitive detectors can be adjusted, and the **phase drift** specification shows how a specified phase will change with temperature and, possibly, time.

Some experiments call for the simultaneous measurement of signals that are in *quadrature* (i.e. shifted by 90° relative to one another), which requires the use of a dual phase lock-in amplifier. When the quadrature signal is very much smaller than the in-phase signal, **phase noise**, which is the random variation in the phase difference between the signal and reference inputs to the phase-sensitive detector, can become a problem.

As an example, consider that it is necessary to resolve:-

- an in-phase signal of 1 mV

and

- a quadrature signal of 1 μV

The phase noise of a good lock-in amplifier might be:-

- 10m° r.m.s.

at a

- time constant of 100 ms.

In this case the "breakthrough" from the in-phase to the quadrature component is:-

V = sin (0.010°) × 1 mV

= 174 nV

Hence the quadrature signal of 1 μV could easily be measured.

Orthogonality is a specification which applies only to dual phase lock-in amplifiers and refers to the accuracy of the nominal 90° phase shift between the reference drives to the two phase-sensitive detectors.

Output Channel(s)

In a single phase lock-in amplifier, there will be only one output channel, but in a dual phase unit there are two.

The output from the phase-sensitive detector passes to a low-pass filter. Usually filters are specified by the frequency at which their transmission is 3dB down on the passband, but since the value of this for a lock-in amplifier's output filter is very low, they are usually specified instead by a **time constant** which is inversely proportional to the -3 dB frequency.

Although the time constant defines the frequencies below which the filter will pass signals, the shape of the **roll-off** with increasing frequency is also important. Usually the filters used in lock-in amplifiers exhibit the same response as a resistor-capacitor (RC) filter, which shows a 6 dB/octave roll-off. This implies that for frequencies well above the passband, if the frequency doubles (an octave) the response falls by 6 dB, or halves.

Filter stages are often "stacked" to provide 12 dB, 18 dB and even 24 dB/octave roll-off. Generally the larger the roll-off, the shorter the time constant needed to achieve a given stability of output. In some cases, however, such as when the lock-in amplifier is part of a feedback control loop, a roll-off greater than 6 dB/octave should not be used since this can cause instabilities due to positive feedback.

In any lock-in amplifier with an analog output stage, the DC amplifier that is used exhibits drift with temperature. Usually it is possible to trade off this **DC stability** with dynamic reserve, but in any good lock-in amplifier even with the poorest settingof say 500 ppm/ºC, a 10 ºC temperature drift results in an error of less than 0.5 % of full-scale. These settings need only be used when there is a lot of interfering noise, in which case the errors are usually small when compared with the effects of the noise.

In some experiments coherent pick-up is a problem and the lock-in amplifier will indicate an output even if there is nominally no signal reaching the detector. This error may be removed by **offset** controls, which add to, or subtract from, the outputs a predetermined amount so as to bring them to zero.

Modern lock-in amplifiers invariably incorporate a microprocessor which allows them to offer the user automatic functions. **Auto-phase, auto-sensitivity, auto-measure** (which combines auto-sensitivity and auto-phase into a single operation) and **auto-offset** are examples which are commonly included. The microprocessor also facilitates instrument operation from a computer, typically via a GPIB (IEEE-488) and/or RS232 connection.

Many units provide an **internal oscillator** which can be used as a source of excitation and reference to the experiment. The oscillator usually provides a sinewave output with user-adjustable voltage and frequency.

Most lock-in amplifiers also include the capability of detecting signals at twice the reference frequency, which is called the **2F mode**. This is very useful in experiments using non-linear devices. Some units also provide for the detection of **higher harmonics** such as **3F, 4F, 5F** etc., although in all cases the highest harmonic frequency which can be measured is limited to the maximum detection frequency of the lock-in amplifier.

Auxiliary inputs and outputs are often provided so that if the instrument is used under computer control the user can take advantage of the analog to digital converter to digitize other experimental voltages and avoid the need for other instruments or interface cards. Inputs and outputs are usually limited to analog voltages which the user can read or set from the computer, but logic input/output ports are also sometimes provided.

The **ratio mode** usually uses one of these analog inputs to provide the denominator for a ratio calculation, where the numerator is the output from the lock-in amplifier. The lock-in amplifier displays the result of this calculation, which is particularly useful in correcting for source intensity changes in optical experiments. In this type of experiment a separate detector is used to measure the source intensity; the output from it, after amplification, being fed to the auxiliary input.

It has been shown that the output from a lock-in amplifier is a DC signal with an AC fluctuation superimposed on it. The amplitude of this fluctuation is dependent on the noise within a bandwidth defined by the output time constant and centered at the reference frequency. If the AC signal in the output is isolated and rectified then it is possible to measure the level of this noise. The **noise measurement mode**, which is provided on many instruments, performs this function.

Further Information

This Technical Note is intended to describe the specifications used in defining the performance of a lock-in amplifier. Additional information may be found in other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative, or by download from our website at **www.signalrecovery.com**

TN 1000	What is a Lock-in Amplifier?
TN 1002	The Analog Lock-in Amplifier
TN 1003	The Digital Lock-in Amplifier
TN 1004	How to Use Noise Figure Contours
AN 1000	Dual-Channel Absorption Measurement with Source Intensity Compensation
AN 1001	Input Offset Reduction using the Model 7265/7260/7225/7220 Synchronous Oscillator/Demodulator Monitor Output
AN 1002	Using the Model 7225 and 7265 Lock-in Amplifiers with software written for the SR830
AN 1003	Low Level Optical Detection using Lock-in Amplifier Techniques
AN 1004	Multiplexed Measurements using the 7225, 7265 and 7280 Lock-in Amplifier
AN 1005	Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper

SIGNAL RECOVERY

The Analog Lock-in Amplifier

TECHNICAL NOTE
TN 1002

For many years the lock-in amplifier was an all-analog instrument. As technology developed, digital electronics, in the form of microprocessor control, was introduced, although initially only in a supporting role. Later, the output filters were implemented using digital techniques, but the phase-sensitive detector (PSD), or demodulator, continued to use analog circuitry. In recent years instruments using Digital Signal Processing (DSP) techniques have been introduced and in these even the PSD is fully digital.

This Technical Note describes the technology used in those ***SIGNAL RECOVERY*** lock-in amplifiers which employ analog detection, such as the models 5109, 5110, 5209, 5210 and the model 5302 (when it is operating above 20 Hz). Technical Note TN1003 provides similar information for the DSP instruments.

Introduction

Lock-in amplifiers which use an analog signal processing channel are invariably known as analog instruments, even if they include digital output filters. The term “digital lock-in amplifier” usually refers to units which utilize a DSP demodulator.

DSP instruments will generally give better performance than their analog counterparts and have inevitably become the first choice for the user. It is worth remembering, however, that there are still some applications for which the analog instruments will offer distinct advantages. Three important examples are:-

È Improved performance when used as the first lock-in amplifier in a “dual demodulation” experiment. In this application, a high frequency “carrier” signal is amplitude modulated at another, lower, frequency. The overall signal is detected by a lock-in amplifier which must offer short output time constants to allow the modulation frequency to pass to the output. This is then detected by either a second lock-in amplifier or other instrument, such as a spectrum analyzer. A DSP based lock-in amplifier samples the input signal at a fixed rate, which is typically a few hundred kilohertz, so that as the reference frequency in increased towards this region, there are fewer samples per cycle from which to derive the output. The effect is particularly apparent at short time constants. Hence in these applications, analog units are usually better.

È Analog instruments provide true analog output filtering and output signals. These features are required for unconditional loop stability when the lock-in amplifier is used as a phase-sensitive detector within a feedback loop. For example, in scanning probe measurements, the probe is vibrated and the lock-in amplifier measures a signal generated as a result of this vibration. The output from the lock-in amplifier is used as a feedback signal to maintain the mean position of the probe at a constant level above the sample surface. Analog instruments usually perform better in this role.

È Higher operating frequencies. DSP units are currently restricted to operation at 2 MHz or below, whereas analog units can operate to many megahertz. Although there is a commercially available instrument described as a high frequency DSP lock-in amplifier, it is in fact an analog unit used as a “down converter” followed by a low frequency DSP final detector stage.

Instrument Description

Figure 1 shows the functional block diagram of a typical high performance, analog lock-in amplifier, such as the ***SIGNAL RECOVERY*** models 5109, 5110, 5209 and 5210. Dual phase instruments include all of the sections shown whereas those sections within the dotted line are omitted in single phase units.

Signal Channel

The input stage may be operated in one of three modes:-

Single-ended Voltage Mode

The signal to be measured is applied to one input connector which operates in single-ended mode and directly feeds the voltage input amplifier.

Differential Voltage Mode

Two input connectors are active and the instrument measures the difference in applied voltage between them.

Current Input Mode

A single connector is active which feeds a current-to-voltage converter, the output of which then drives the voltage input amplifier. The current conversion ratio is usually 10^6 V/A or 10^8 V/A, and the overall current sensitivity is given by multiplying this ratio by the full-scale voltage sensitivity setting.

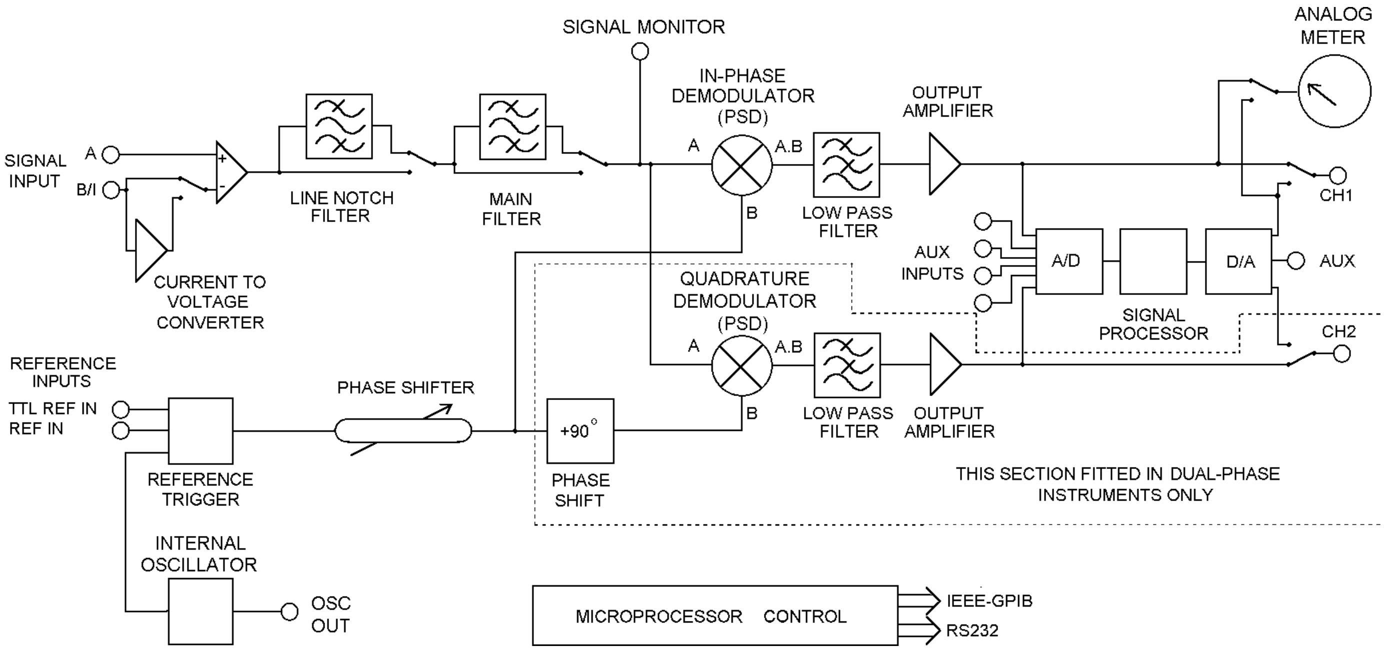


Figure 1

For example, if the voltage sensitivity is set to be 100 mV full-scale and the 10^8 V/A current conversion ratio is selected, then the instrument's full-scale current sensitivity will be 1 nA.

In the current measuring mode, the input impedance is low (typically less than 100 T) although it does rise as the operating frequency increases, and is higher for the 10^8 V/A than for the 10^6 V/A conversion setting. If the very best performance is needed then it may be better to use a separate dedicated current preamplifier.

The current input connector is often combined with the "B" voltage-mode input connector to simplify the layout of the front panel.

Line Notch Filter

The output of the input amplifier is optionally passed through a line notch filter. This is a band-rejection stage, designed to remove 50/60 Hz and/or 100/120 Hz interference from the input signal. Since the line frequency can vary by up to ±1 % of its nominal value, the Q-factor of the filter is only 1. Any higher value would not give satisfactory attenuation over the range of possible input frequencies. However this low Q value has the disadvantage of introducing significant attenuation and phase-shifts even at frequencies well removed from the set frequency.

Main Filter

Following the line notch filter, the signal passes to the main filter. This may be operated in the low-pass, bandpass or notch mode, or may be bypassed. When the user chooses to make it active, the phase-sensitive detector(s) in the following stage are switched to a special mode so that the instrument provides an overall sinewave, or fundamental, response.

The standard operating condition of the instruments is with the main filter set to the bandpass mode and tuned to the reference frequency. In this mode the roll-off of the filter both above and below the reference frequency is 12 dB/octave, giving rejection of interfering noise components in both frequency regions.

In many situations, particularly when using low reference frequencies, most of the interfering signals are at frequencies greater than the reference frequency, so that better performance is obtained by using the low-pass mode. In this case, the roll-off of the filter attenuating components above the reference frequency increases to 24 dB/octave.

The flat mode is often used for non-demanding applications and may give the best results when the input signal is a squarewave and the interfering noise is white.

The output from the main filter passes to the phase-sensitive detector(s) where it is multiplied by the output from the reference circuitry. In addition this signal is often brought out to a connector on the front panel of the instrument, allowing the user to monitor the effect of the filters on the input signal before it passes through the demodulator.

Reference Channel

The reference channel takes its input either from the external reference input connector, or from the internal oscillator and generates in-phase (and quadrature in a dual phase lock-in amplifier) switching waveforms which drive the demodulator(s).

Reference Trigger

The reference trigger circuit takes the applied reference signal and uses a threshold detector to produce an internal trigger signal. The only requirement imposed on the reference signal is that it has only two zero crossings per cycle and that it is of sufficient amplitude. This means that square, triangular and other waveforms can be used as well as the more usual sinusoidal form. The presence of the trigger threshold detector does however give rise to the possibility of phase-shifts at low frequencies, typically below 10 Hz, so the instruments also include a TTL, or logic level, reference input for use at these frequencies.

The reference channel can operate in either the F or 2F mode, which means that the output signal is either at the applied frequency or at twice this frequency. The ***SIGNAL RECOVERY*** model 5302 extends this range by allowing operation up to and including the seventh harmonic.

Phase-Shifter

The output from the trigger circuit is applied to a phase-shifting circuit, which allows the user to ensure that the phase difference between the signals at the reference and signal channel inputs to the in-phase demodulator is zero. Generally this phase-shifter may be set in increments of a few tens of a millidegree and it usually also incorporates the facility to add 90° steps. This feature is especially useful in single phase instruments where the phase is best adjusted by looking for a null at the output and then adding 90° or 270° to maximize the output.

90° Phase-Shift

In dual phase instruments, this section shifts the phase of the output of the phase-shifter by 90° to provide the reference signal drive to the quadrature phase-sensitive detector.

Phase-Sensitive Detector(s)

Single phase instruments have one phase-sensitive detector, or demodulator, whereas dual phase units have two. However the discussion that follows applies equally to both in-phase and quadrature detectors and hence does not differentiate between them.

The PSD serves to multiply the output of the signal channel by the reference signal and may be operated in one of two modes. In the squarewave, or flat, mode, in which the signal channel main filter is bypassed, the multiplying elements consist simply of reversing switches driven at the reference frequency. The Fourier analysis of this operation shows that this type of demodulator gives a steady-state output for any component of the signal channel output at the reference frequency or its odd harmonics, with the gain being inversely proportional to the harmonic number.

The presence of the odd harmonic responses in the squarewave demodulator is undesirable in the majority of experimental situations and is overcome by switching the demodulator to the Walsh mode.

In this mode, the demodulator is implemented by using a more sophisticated set of switches, that may be considered to perform a multiplication of the input signal by a stepwise approximation to a sinewave at the reference frequency. The effect of this is to provide a response to input signals at the fundamental reference frequency and its seventh and higher harmonics, with no response at the third and fifth harmonics.

Additionally, when switched to this mode, the main signal channel filter is switched into use. This gives excellent rejection of the responses at the seventh harmonic and above and hence, in conjunction with the demodulator, provides an overall lock-in amplifier response which is limited to signals at the reference frequency. Used in this manner the instrument is said to give fundamental, or sinewave, response and it is by far the most common mode of operation.

Output Stages

Output Low-Pass Filters

The output signal from the lock-in amplifier is required to follow in time the variation in the input signal magnitude and phase. The function of the output filter is to reduce the level of spurious time variations, which may be random or deterministic in nature and which are commonly referred to as output noise.

The output filters implement either first-order or second-order low-pass functions (6 dB/octave or 12 dB/octave roll-off respectively) by the use of a combination of analog and digital techniques and are normally specified by means of a time constant.

In traditional audio terminology, a first-order low-pass filter is said to have a roll-off of 6 dB/octave since at frequencies well beyond the passband its gain is inversely proportional to the frequency (a change of 6 dB corresponds to approximately a factor of 2 in amplitude and an octave represents a doubling of frequency); similarly a second-order filter is said to have a 12 dB/octave roll-off.

The 6 dB/octave option is not satisfactory for most purposes since it does not give good rejection of non-random interfering signals which may give aliasing problems in the analog-to-digital converters which follow. However, it is of use when the lock-in amplifier forms part of a feedback loop and when the time-domain response is critical.

Output Amplifier(s)

The instrument's overall full-scale sensitivity is affected by the gains of both the input and output amplifiers. The input amplifiers are AC-coupled, but the output amplifiers are DC-coupled and hence are subject to increasing drift with time and temperature as their gain is increased.

At the highest sensitivity settings, both input and output amplifiers will be operating near to their maximum gain. However, at mid-range sensitivity settings the same overall sensitivity can be obtained by different combinations of AC and DC gain. Lower values of DC gain give better output stability, but at the expense of reducing the instrument's dynamic reserve, since the corresponding higher AC gain results in the demodulator overloading at lower levels of interfering signal.

The instruments are therefore fitted with a control to adjust the gain distribution to best match the applied signals.

Signal Processor

The outputs from the PSDs can be taken directly to the output BNC connectors but are usually subjected to further processing in the digital signal processing section. This digitizes the PSD output signals and can then perform further filtering, and, in the case of dual phase units, derive the overall signal magnitude and phase angle. Once processed, analog outputs can be generated using digital-to-analog converters, although if the signals are to be read from the digital panel meters or via the computer interfaces then this is not of course necessary.

Microprocessor Control

All modern instruments incorporate a microprocessor which is used to implement a number of functions, such as the output low-pass filter and processing stages. However its most common purpose is to allow the instrument to be operated via a standard computer interface such as a GPIB (IEEE-488) or RS232 link.

The command set used in ***SIGNAL RECOVERY*** units is based on simple mnemonics which generally operate either to adjust or interrogate instrument settings depending on whether an additional optional parameter is sent.

As an example, to set the full-scale sensitivity of the model 5110 to the 2 mV setting requires the user to send the following ASCII character string:-

SEN 13 <CR>

where <CR>, the *terminator*, represents the carriage return character.

To interrogate the instrument's sensitivity setting, the user would send:-

SEN <CR>

and the instrument would respond with:-

13 <CR>

It is therefore very simple to program these instruments remotely and the logical nature of the mnemonics makes user programs easy to read.

Oscillator

Virtually all modern instruments include an internal oscillator which may be used as a source of excitation to the experiment. This generally has a sinusoidal output of adjustable voltage, typically 0 V to 2 V rms, and frequency over the operating range of the instrument. An internal switch allows the oscillator signal to be optionally connected directly to the reference channel, thereby eliminating the need for an external connection.

Other Features

The internal crystal clock and the microprocessor allow the units to measure the reference frequency very accurately and thereby offer the capability of being used as a frequency meter over their operating frequency range.

Many computer-controlled experiments require one or two analog or digital control signals and may generate analog signals which need recording at the same time as the output from the lock-in amplifier. Traditionally this was done by the provision of separate computer interface boxes, but modern instruments simplify the task by the inclusion of both analog and digital auxiliary inputs and outputs. These take advantage of the presence of the A/D and D/A converters and computer interface electronics already in the instrument and can often avoid the need to use separate A/D and D/A converter units.

Further Information

This Technical Note is intended to describe the overall structure of a modern analog lock-in amplifier. Additional information may be found in other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative, or by download from our website at **www.signalrecovery.com**

TN 1000	What is a Lock-in Amplifier?
TN 1001	Specifying Lock-in Amplifiers
TN 1003	The Digital Lock-in Amplifier
TN 1004	How to Use Noise Figure Contours
AN 1000	Dual-Channel Absorption Measurement with Source Intensity Compensation
AN 1001	Input Offset Reduction using the Model 7265/7260/7225/7220 Synchronous Oscillator/Demodulator Monitor Output
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SIGNAL RECOVERY

The Digital Lock-in Amplifier

TECHNICAL NOTE
TN 1003

In recent years the falling cost of digital signal processing circuitry has allowed its application in an ever wider field of instrumentation. In particular, it has become possible to use the technique in applications that hitherto have been wholly the preserve of analog electronics, such as the phase-sensitive detectors found in lock-in amplifiers.

This Technical Note describes the technology used in the **SIGNAL RECOVERY** digital lock-in amplifiers, such as the models 7220, 7220BFP, 7225, 7225BFP, 7260, 7265, 7280 and 7280BFP. Technical Note TN1002 provides similar information for instruments using analog technology.

Introduction

Lock-in amplifiers are usually only described as "digital" or DSP (Digital Signal Processing) instruments if their phase-sensitive detector is implemented in digital circuitry, even though traditional analog instruments have, for many years, used digital electronics extensively for instrument control and output processing.

Digital lock-in amplifiers have become very popular because in many cases they offer a number of advantages over analog units, including:-

È Better output stability. Since, unlike analog units, there are no DC-coupled output amplifier stages and therefore outputs are less prone to drift with time and temperature.

È Better internal oscillators. The crystal-stabilized internal oscillator is more stable with respect to time and temperature changes and hence gives better results in experiments which can be driven by it. Internal reference mode operation in DSP units also offers a very small, or even zero, lock acquisition time which is ideal for swept-frequency measurements.

È Perfect X and Y demodulator orthogonality. This improves the accuracy of measurements of weak in-phase components in the presence of strong quadrature signals.

È Better price/performance ratios. The reduced manufacturing and testing cost of DSP units delivers instruments with better price/performance ratios than the older technologies.

In spite of the above, there are still some applications for which analog techniques are better or indeed the only option. These include operating at higher frequencies, or when utilizing short time constants at mid-range frequencies (50 - 250 kHz). Requirements such as these are often encountered in dual demodulation experiments or when operating the lock-in amplifier as part of a control feedback loop. It should also be remembered that the input stages of a DSP lock-in amplifier still need to be implemented in analog technology, so that in reality the "all digital" instrument does not exist.

Instrument Description

Figure 1 shows the functional block diagram of a typical high-performance digital lock-in amplifier, such as those in the **SIGNAL RECOVERY** 72XX series.

Signal Channel

The input stage may be operated in one of three modes:-

Single-ended Voltage Mode

The signal to be measured is applied to one input connector which operates in single-ended mode and directly feeds the voltage input amplifier.

Differential Voltage Mode

Two input connectors are active and the instrument measures the difference in applied voltage between them.

Current Input Mode

A single connector is active which feeds a current-to-voltage converter, the output of which then drives the voltage input amplifier. The current conversion ratio is usually either 10^6 V/A (high bandwidth) or 10^8 V/A (low noise) although the user does not usually need to concern himself with the actual value used since when operated in the current mode the instrument directly displays the measured signals in terms of amperes.

In the current measuring mode, the input impedance is low (typically less than 100 T) although it does rise as the operating frequency increases, and is higher for the 10^8 V/A conversion setting than for the 10^6 V/A. If the very best performance is needed then it may be better to use a separate dedicated current preamplifier.

The current input connector is normally combined with the "B" voltage-mode input connector to simplify the layout of the front panel.

Line Frequency Notch Filter

The output of the input amplifier is optionally passed through a line notch filter. This is a band-rejection stage, designed to remove 50/60 Hz and/or 100/120 Hz interference from the input signal. Since the line frequency can vary by up to ±1% of its nominal value the filter has a low *Q*-factor to ensure satisfactory attenuation over the range of possible input frequencies. However, this does have the disadvantage of introducing significant attenuation and phase-shifts even at frequencies well removed from the set frequency.

AC Gain

The signal channel contains a number of analog filters and amplifiers whose overall gain is defined by the AC Gain function block. For each setting of AC Gain there is a corresponding level at which the instrument input will overload.

It is a basic property of the digital lock-in amplifier that the best demodulator performance is obtained by presenting as large a signal as possible to the main analog-to-digital converter (ADC). Therefore, in principle, the AC Gain setting is made as large as possible without causing amplifier or converter overload. This constraint is not too critical however and the use of a value 10 or 20 dB below the optimum makes little difference. Note that as the AC Gain value is changed, the demodulator (in-phase and quadrature multiplier) gain is also adjusted in order to maintain the selected full-scale sensitivity.

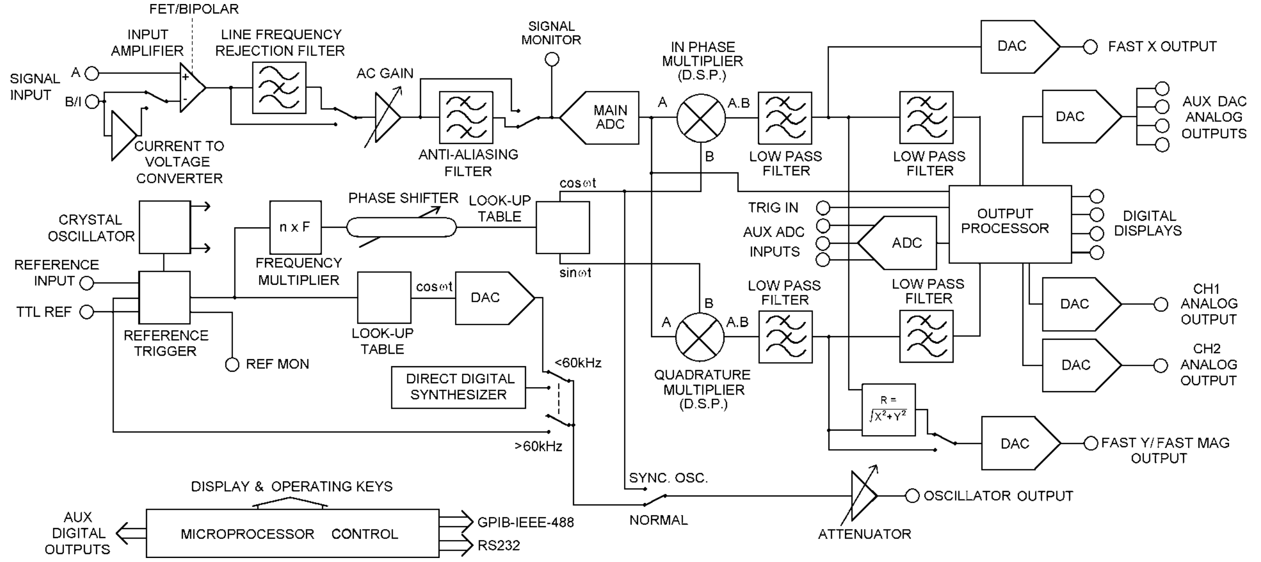

Fig 1, Block Diagram of Typical DSP Lock-in Amplifier

The instrument's full-scale sensitivity is set by a combination of analog AC Gain in the input circuits and digital output gain in the demodulator. Changes in AC Gain potentially affect other specifications, such as bandwidth and accuracy, but changes in digital gain have no such effects. Hence changes in full-scale sensitivity effected only by changes to the digital gain are free from these additional errors.

The AC Gain setting affects the instruments' dynamic reserve. This is a measure of its ability to make accurate measurements in the presence of interfering signals. If the AC gain is low but the full-scale sensitivity is high then the real signal will only occupy a few bits of the ADC's dynamic range, leaving plenty of headroom for stronger interfering signals without overload. However longer output time constant settings will be needed for a given output accuracy. Conversely, high AC Gain allows shorter time constants to be used for a given accuracy but results in lower dynamic reserve.

Anti-Aliasing Filter

Following the AC Gain amplifier stage, the signal passes to the anti-aliasing filter. This removes unwanted frequencies, which would cause a spurious output from the ADC due to the nature of the sampling process, by restricting the signal bandwidth reaching the ADC. If the instrument is being used to measure noise-free signals, as in the case of vector voltmeter measurements, then aliasing may not occur and slightly better performance can be achieved by bypassing the anti-aliasing filter.

A buffered version of the analog signal just prior to the main ADC is often made available at a connector on the rear panel of the instrument. This output can be viewed on an oscilloscope to monitor the effect of the signal channel filters and amplifiers.

Main Analog-to-Digital Converter

Following the anti-alias filter the signal passes to the main analog-to-digital converter which digitizes the input signal at the *sampling rate*. The output from this converter, which is a series of digital values representing the amplitude of the input signal, feeds the first of the digital signal processors. This implements the in-phase and quadrature digital multipliers, and the first stage of the output low-pass filtering for each of the X and Y channels.

In order to satisfy Nyquist's sampling theorem, the sampling rate generally needs to be at least twice the bandwidth of the anti-aliasing filter. If this bandwidth starts at DC, i.e. the filter is of a low-pass design, then this equates to a requirement that the sampling rate be at least twice the maximum reference frequency. However, the sampling rate can be reduced if the anti-aliasing filter can be operated in bandpass mode as in ***SIGNAL RECOVERY*** DSP lock-in amplifiers.

Reference Channel

The reference channel serves to provide the demodulator DSP with a stream of "phase values" at the sampling frequency of the main ADC. Each of these values represents the instantaneous phase angle of the reference frequency waveform at the sampling time.

For example, consider a 1 kHz reference frequency, either internally generated or from the reference input connector, with the unit set to fundamental reference mode (n = 1). If the main ADC sampling rate is 180 kHz then there will be 180 samples of the applied input signal for each reference period, which is one millisecond. Each one of these samples needs to be multiplied by the value of a cosine wave of unit amplitude at the corresponding phase position. Consequently, the reference DSP measures the reference frequency (1 kHz) and outputs 180 phase values during each reference period, which in this case would be at phase angles 0 to 360° in 2° (360/180) increments.

In external reference mode a second DSP, operating as a digital phase-locked loop (PLL), is used to measure the period (or frequency) of the signal applied to the TTL or analog reference inputs very accurately and generate the stream of phase values.

In internal reference mode the ideal situation would be to allow the reference processor to generate both the amplifier's internal oscillator output signal and the phase values for the demodulator at the selected reference frequency. In this case the reference channel is not dependent on a PLL, unlike the situation with analog lock-in amplifiers. Consequently the phase noise is extremely low and, because no time is required for a PLL to acquire lock, reference channel lock is immediate. However, technical limitations mean this is not yet possible at all reference frequencies, so that at the higher end of the frequency range the reference trigger input is provided by an internal link from the output of a separate direct digital synthesizer and the reference channel then operates as if in external reference mode.

The reference channel DSP is also utilized for implementing reference frequency multiplication, as is required for measurements made on the harmonics of the reference frequency. Normally, a lock-in amplifier measures the applied signal at the reference frequency but in some applications, such as Auger Spectroscopy or amplifier characterization, it is useful to be able to make measurements at some multiple "*n*", or harmonic, of the reference frequency "F". Digital lock-in amplifiers allow this multiple to be set to any value between 2 (i.e. the second harmonic) and 65535 (depending on which model is being used), as well as unity, which is the normal mode. The only restriction is that the product "$n \times$ F" cannot exceed the maximum normal reference frequency.

Phase-Shifter

The reference DSP also implements a digital reference phase-shifter, which adds or subtracts the set reference phase angle from the phase values being sent to the demodulator DSP. By adjusting this reference phase control, which has a typical phase-shift resolution of ten millidegrees, the signals at the reference and signal channel inputs to the X-channel demodulator can be brought into phase. The phase-shifter also incorporates the facility to add 90° steps, which is especially useful when the phase is adjusted by first looking for a null at the X-channel output and then adding 90° or 270° to maximize it.

SIGNAL RECOVERY lock-in amplifiers provide a TTL logic signal, at the reference frequency, at the REF MON connector, which allows the user to check that the reference channel is operating correctly.

Demodulator DSP

The main DSP takes each phase value from the reference DSP and uses it to find the amplitude of a cosine wave at the corresponding angle by means of a look-up table. It then takes this value and digitally multiplies it by the signal sample, the resulting number being the X-channel output sample. A second part of the DSP carries out a similar calculation, but uses the value of a sinewave at the same angle. This gives the Y-channel output sample.

First Stage Output Filters

The output from the multiplication process is a stream of digital X and Y channel output samples, at the sampling rate. These feed the first stage of the X channel and Y channel digital output filters, which implement the conventional output low-pass filter function of the lock-in amplifier. The filtered outputs are fed into the output processor which carries out further filtering and processing. In ***SIGNAL RECOVERY*** models these outputs are also used to drive two fast 16-bit digital-to-analog converters (DACs) to generate the signals at the instrument's FASTX and FASTY analog output connectors.

The demodulator output is digitally scaled to provide the demodulator gain control. As discussed earlier this gain is adjusted as the AC Gain is varied to maintain the overall full-scale sensitivity.

Output Processor

Although shown on the block diagram as a separate entity, the output processor is typically part of the instrument's main microprocessor. It provides more digital filtering of the X channel and Y channel signals, as well as carrying out vector magnitude and phase, noise, ratio and other calculations.

Second Stage Output Filters

Generally, digital lock-in amplifiers use Finite Impulse Response (FIR) low-pass filters offering 6, 12, 18 and 24 dB/octave roll-off with increasing frequency. These filters offer a substantial advantage in response time compared with analog filters or digital infinite impulse response (IIR) filters.

The 6 dB/octave filters are not satisfactory for most purposes because they do not give good rejection of non-random interfering signals, which can cause aliasing problems as a result of the sampling process in the main ADC. However, the 6 dB/octave filter finds use where the lock-in amplifier is incorporated in a feedback control loop, and in some situations where the form of the time-domain response is critical. Normally, the 12 dB/octave setting is used unless there is some definite reason for not doing so.

Following the output filters, an output offset facility is provided to enable an offset (up to ±300 % full-scale in ***SIGNAL RECOVERY*** units) to be applied to the X, Y or both outputs.

Magnitude and Phase Outputs

The processor calculates the vector magnitude and signal phase of the input signal. If the input signal, $V_s(t)$, is a sinusoid of constant amplitude at the reference frequency and the output filters are set to a sufficiently long time constant, then the filtered demodulator outputs are constant levels V_x and V_y. The vector magnitude · $(V_x^2+V_y^2)$ is dependent only on the amplitude of the required signal, $V_s(t)$ (i.e. it is not dependent on the phase of $V_s(t)$ with respect to the reference input) and is computed by the output processor in the lock-in amplifier. The phase angle between $V_s(t)$ and the applied reference signal is called the signal phase, it is the arctangent of the ratio V_y/V_x and is also computed by the processor.

Noise Calculation

The noise measurement facility uses the output processor to perform a noise computation on the Y output where it is assumed that the amplitude distribution of the waveform at this point is Gaussian with a mean value of zero. The zero mean is usually obtained by using the reference phase control or the Auto-Phase function with a comparatively long time constant (say 1 s). The time constant is then reduced (to say 10 ms) for the noise measurement.

The instrument takes into account the equivalent noise bandwidth of the measurement, which is set by the output time constant and slope, and displays the noise value directly in V/· Hz or A/· Hz either on the digital panel meters or via the computer interfaces.

Analog Outputs

Any of the outputs which are digitally displayed are also available in analog form at connectors on the rear panel. These analog outputs are generated by using two further 16-bit digital-to-analog converters (DACs).

Internal Oscillator

As mentioned earlier, when used in internal reference mode, the reference DSP generates phase values, representing the required reference frequency, to drive the main DSP. The sinusoidal values from the look-up table are applied to a fast DAC and the resulting analog signal, after filtering, is the unit's "internal oscillator" output. The great advantage of this technique is that lock acquisition is instantaneous, since no time is needed for a reference channel PLL to acquire lock.

However, as the oscillator frequency is increased towards the instrument's sampling frequency, the stepped approximation to a sinewave, that is generated by this technique, becomes more obvious. To overcome this effect a dedicated direct digital synthesizer running at a much higher sampling frequency is used at higher frequencies to generate the oscillator signal. This is then internally coupled back to the "external" reference input of the lock-in amplifier when internal reference mode is selected.

Microprocessor Control

In addition to the digital electronics used for the phase-sensitive detector, the digital lock-in amplifier incorporates a microprocessor which, as has been seen, is used to implement a number of signal processing functions. It also allows the instrument to be operated via a standard computer interface such as a GPIB (IEEE-488) or RS232 link.

The command set used in ***SIGNAL RECOVERY*** units is based on simple mnemonics which generally operate either to set or interrogate instrument settings depending on whether an optional parameter is sent. As an example, to set the full-scale sensitivity of the model 7260 or 7265 to the 20 nV setting requires the user to send the following ASCII character string:- SEN 4 <CR>

where <CR>, the *terminator*, represents the carriage return character. To read the instrument's present sensitivity setting, the user would send:-

SEN <CR>and the instrument would respond with:-

4 <CR>

Ease of use is further improved over earlier analog instruments by offering the "floating point" mode for reading instrument outputs. Conventionally, for example, determining the true level of the in-phase component of an input signal in volts rms has required the programmer to know the present sensitivity setting and then to send a command to the instrument to read the output value. The response is sent as a percentage of the present full-scale sensitivity setting, and so a look-up table is needed to convert this response and the sensitivity setting into a value expressed in volts.

The floating point response mode causes the instrument to carry out this translation, although the older technique is retained for compatibility.

Other Features

SIGNAL RECOVERY digital lock-in amplifiers offer many features beyond those strictly needed for signal recovery but which nonetheless make the experimenter's job easier. These include the following:-

Auxiliary ADC, DAC and Digital Outputs

Many computer-controlled experiments require one or two analog or digital control signals and may generate analog signals which need recording at the same time as the output from the lock-in amplifier. The instruments therefore include analog and digital auxiliary inputs and outputs taking advantage of the presence of the ADCs, DACs and computer interface electronics already in the instrument and often avoiding the need to use separate units.

Data Storage

A 32k, 16-bit data buffer allows selected instrument outputs to be stored prior to transfer to an external computer, or displayed on the graphical display panel where available.

Transient Recorder

The auxiliary ADC inputs may be used in conjunction with the curve storage capability to allow the units to function as 16-bit, 40 kSa/s transient recorders.

Frequency Measurement

The internal crystal clock and the microprocessor allow the units to measure the present reference frequency very accurately and thereby offer the capability of acting as a frequency meter over their operating frequency range.

Unique Features

The ***SIGNAL RECOVERY*** DSP lock-in amplifiers provide the following three modes of operation, believed to be unique.

Virtual Reference Mode™

In the virtual reference mode, the Y channel output is used to make continuous adjustments to the internal oscillator frequency and phase to achieve phase-lock with the applied signal, such that the X channel output is maximized and the Y channel output zeroed. This mode of operation allows signal recovery measurements to be made without the use of a reference signal.

Dual Reference Mode

The dual reference mode allows the instrument to make simultaneous measurements at two different reference frequencies, an ability that previously required two lock-in amplifiers. This flexibility imposes a few restrictions, such as the requirement that one of the reference signals be external and the other is derived from the internal oscillator, a limit on the maximum operating frequency and that both signals be passed through the same input signal channel. This last restriction implies either that both signals are derived from the same detector (for example two chopped light beams falling on a single photodiode) or that they can be summed prior to measurement, either externally or by using the differential input mode of the instrument. Nevertheless, the mode proves invaluable in many experiments.

Dual Harmonic Mode

Dual harmonic mode allows the simultaneous measurement of two different harmonics of the input signal. As with dual reference mode, there are a few restrictions, such as a limit on the maximum operating frequency.

Further Information

This Technical Note is intended to describe the overall structure of a modern digital lock-in amplifier. Additional information may be found in other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative, or by download from our website at **www.signalrecovery.com**

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SIGNAL RECOVERY

How to use Noise Figure Contours

TECHNICAL NOTE
TN 1004

Introduction

A variety of techniques can be used to measure and specify the noise characteristics of amplifiers, which may be stand-alone units or functional blocks in more sophisticated signal recovery equipment, such as lock-in amplifiers. Once these measurements have been made, the results need to presented in a format which allows the user of such equipment to easily select the optimum instrument for his application. The Noise Figure Contour format has proved popular for such use, and the purpose of this Technical Note is to describe how such contours are generated and how to interpret them.

Measuring Amplifier Noise

One of the best known, and certainly the simplest, method of measuring amplifier noise is to short the amplifier input to ground and measure the rms (root mean square) output voltage over a specified bandwidth. The value is divided by the square root of this bandwidth and then by the amplifier gain to give a result which is quoted as the "noise referred to the input" in units of volts per root hertz. The quoted figure for a given instrument is usually the best attainable and is that obtained when using the highest gain settings.

Typical values for this figure, for a variety of ***SIGNAL RECOVERY*** products, are given in table 1.

Model Number	Minimum Input Noise at 1 kHz (nV/· Hz)
5113	4
5183	2
5184	0.8
5185	10 or 30 *
5186	4
5187	1.2
5320	4
5109/5110	5.5
5209/5210	5
7220/7260/ 7225/7265	5 or 2 *

* Figure depends on input configuration

Table 1, Typical Input Noise for various *SIGNAL RECOVERY* products

Comparing various products by using figures measured in this way is probably satisfactory where the frequency to be used is close to that at which the noise was measured and the measurement bandwidths are similar.

If an accurate determination of instrument noise is to be made, then it should be realized that this noise is a function of both the operating frequency and source resistance. The "noise referred to the input" technique ignores the fact that source resistance exists in every application. This source resistance ranges from several tenths of an ohm, in the case of devices such as thermocouples, to many megohms for devices such as photomultiplier tubes or vibrating capacitors.

In order to fully specify the noise performance of an amplifier, the noise must be measured at a range of different frequencies and source resistances. This can be done directly using the test circuit shown in figure 1.

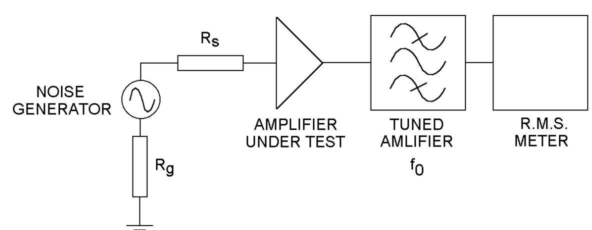

Figure 1, Noise Measurement Test Circuit

The calibrated white-noise generator (equal power per unit bandwidth) is set for zero output and the source resistance R_s is inserted into the circuit. The tuned amplifier is adjusted to the desired center frequency. Under these conditions the reading on the rms meter is read and recorded. The noise generator output is then increased until the rms meter reading is 1.414 (i.e. $\sqrt{2}$) times its former value. (Noise powers add directly but noise voltages add vectorially so that 1 mV of noise plus 1 mV of noise equals 1.414 mV of total noise.) At this point the reading on the calibrated front-panel meter of the noise generator equals the total noise due to the amplifier plus that due to the thermal (or Johnson) noise in the sum of the two resistances, R_g and R_s.

Varying the source resistance whilst maintaining a constant center frequency allows the total noise as a function of source resistance to be measured. Varying the center frequency whilst maintaining a constant source resistance allows total noise as a function of frequency to be determined.

Once the entire range of frequencies and source resistances has been measured, the resulting data needs reducing to a format suitable for presentation. One method is to calculate the *noise figure* for each frequency and source resistance combination.

Noise Figure

The noise figure of an amplifier relates the amount of noise added by that amplifier to the amount of thermal, or Johnson, noise inherent in the source resistor.

Thermal noise is an rms voltage generated by any resistor due to random electron movement present at any temperature above absolute zero. It can be calculated from the following equation:

$$E_n = \sqrt{4KTR_sB_n} \qquad (1)$$

where:

- E_n = rms voltage noise within the bandwidth of measurement
- K = Boltzmann's constant = 1.38×10^{-23} Joules per Kelvin
- T = Absolute temperature in Kelvin
- R_s = Resistive component of the impedance across which the voltage is measured, in ohms
- B_n = Bandwidth over which the noise voltage is measured

In the case of a source at room temperature, taken as 300 K, equation 1 simplifies to:

$$E_n = 1.28 \times 10^{-10}\sqrt{R_s} \quad \text{V}/\sqrt{\text{Hz}} \qquad (2)$$

The value of equation 2 over a range of typical source resistances is plotted in figure 2.

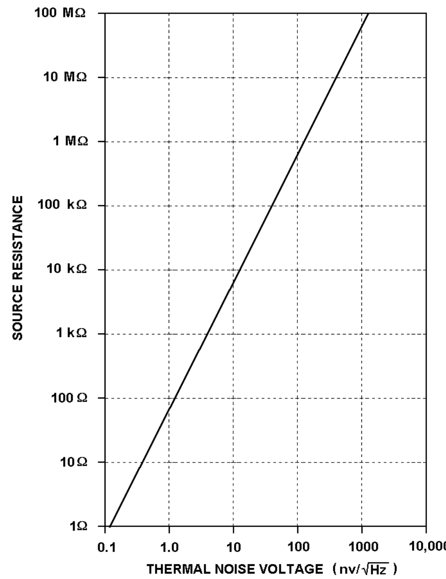

Figure 2, Thermal Noise Voltage vs. Source Resistance

Since the theoretical noise of the source resistance is known and the total of this noise plus that from the amplifier has been measured, the noise figure, *NF*, can now be calculated for each measured point using the following equation:

$$NF = 20\log_{10}\frac{[\text{Thermal Noise} + \text{Amplifier Noise}]\,(\text{Measured})}{\text{Thermal Noise}\,(\text{calculated from eqn } 1)} \qquad (3)$$

Noise Figure Contours

The three variables of frequency, source resistance and noise figure can best be shown by plotting the contours of constant noise figure on a full logarithmic frequency versus source resistance scale.

A typical set of such noise figure contours is shown in figure 3. As can be seen, this noise figure specification completely describes the noise performance of the amplifier for **every** source resistance and frequency combination over which it was designed to operate.

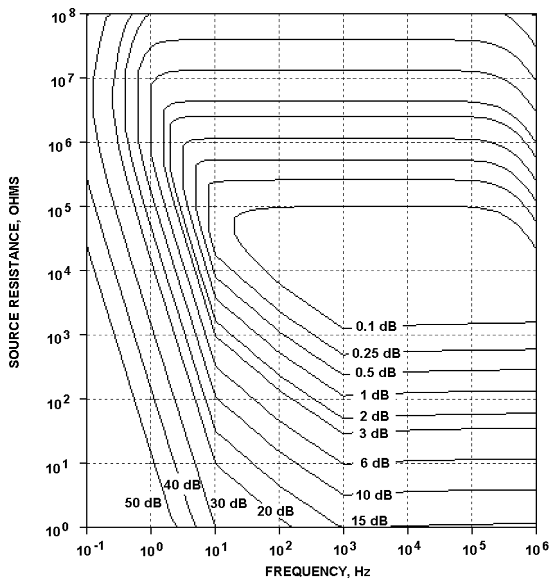

Figure 3, Typical Noise Figure Contours for *SIGNAL RECOVERY* Model 5184 Preamplifier

Noise figure contours are extremely useful because of the variety of information which they can provide, including the following:

1) **Instrument Selection**
   When the source resistance and operating frequency are fixed by experimental limitations, the contours of several amplifiers can be compared to determine the correct instrument to use for minimum noise. For example, in an optical experiment the frequency may be limited to 300 Hz, because of the limitations of a mechanical light chopper, and the available infrared semiconductor detector may fix the source resistance as 80 kT . Since neither of these parameters can be varied, a review of the contours of all the available amplifiers would allow selection of the one with the lowest noise figure and hence the lowest noise performance.

2) **Determining Optimum Operating Frequency**
   If a particular amplifier is already available, it may be possible to design the experiment so that the operating frequency appears at the lowest noise figure point on the plot. Assume, for example, that the ***SIGNAL RECOVERY*** model 5184 preamplifier were available, the detector had a source resistance of 10 T and there was a relatively wide latitude on operating frequency. From the model 5184 contours shown in figure 3, it is evident that best performance would be obtained in the frequency range from 1 kHz to 1 MHz. The user would then try to arrange the experiment to run at a frequency in this range and thereby obtain the lowest possible noise.

3) **Determining Optimum Coupling Transformer Ratio**
   If it is possible to choose the detector source resistance then this should be as low as possible. The thermal noise is proportional to the square root of the source resistance (figure 2) and the improvement obtained by using a lower source resistance is greater than the resulting deterioration in noise figure of the amplifier. For example, a source resistance of 1 T gives 30 dB lower thermal noise than that generated by a 1 kT source resistance. In the case of the 5184, the noise figure deterioration at 10 kHz caused by using a 1 T rather than a 1 kT source resistance is only 15 dB, so using a 1 T source resistance is still 15 dB better overall than using one of 1 kT .

   If the source resistance is very low, such as when measuring the impedance of superconducting materials, then it may be possible to reduce the noise penalty caused by the amplifier by using a signal coupling transformer, such as the ***SIGNAL RECOVERY*** model 1900, between the source and amplifier. The transformer increases both the source signal and thermal noise by the turns ratio and in addition increases the effective source resistance seen by the preamplifier by the square of the ratio. If the ratio is chosen to yield an effective source resistance which gives the lowest amplifier noise figure at the operating frequency, then the additional noise contributed by the amplifier is minimized.

   Note that this assumes that the transformer is noiseless and that its frequency response is adequate. However with careful transformer selection it is nonetheless still possible to obtain a significant improvement using this technique, even after allowing for such penalties.

4) **Determining Minimum Signal Level for a minimum Signal-to-Noise Ratio**
   Since noise figure contours provide complete information on internal amplifier noise, they can readily be used to determine the signal level required to achieve a given signal-to-noise ratio.

   As discussed earlier, the noise figure relates total amplifier noise plus thermal noise to thermal noise alone. The thermal noise for a given value of source resistance can be found from equation 1, allowing the total noise to be obtained using the following equation, which is simply a rearranged version of equation 3:

$$\mathit{Total\ Noise} = \left(10^{\frac{NF}{20}}\right) \times \mathit{Thermal\ Noise} \qquad (4)$$

If the required minimum signal-to-noise ratio is *S,* then:

$$Signal \varnothing S \Delta Total\ Noise \qquad (5)$$

For example, the minimum signal level giving a signal-to-noise ratio of 20, for a model 5184 preamplifier with a source resistance of 600 T and at a frequency of 20 Hz can be calculated as follows:-

1) From equation 1, the thermal noise due to the 600 T source resistance at 300 K is approximately 3.2 nV/· Hz
2) From the noise figure contour (figure 3) the noise figure at this combination of source resistance and frequency is approximately 3 dB
3) Hence, solving equation 4, the thermal noise plus amplifier noise is 4.5 nV/· Hz
4) If the measurement bandwidth were 1 Hz then the actual noise would be 4.5 nV rms
5) The signal level therefore needs to be 20 × 4.5 nV, or 90 nV, to give the required signal-to-noise ratio.

5) Determining Equivalent Input Noise Resistance (R_e)

The “equivalent input noise resistance” of an amplifier is the lowest value of resistance for which the thermal noise is equal to the amplifier noise at a given frequency. Using equation 3, it can be seen that when this is the case, the overall noise figure will be 3 dB, and hence it is possible to use the plotted 3 dB contour to determine the equivalent input noise resistance. For example, to find the value for a model 5184 preamplifier at 1 kHz, follow the 1 kHz ordinate to the lower 3 dB noise figure contour. The source resistance on the abscissa (–30 T) is then the “equivalent input noise resistance”.

Summary

Noise figure contours offer an extremely useful tool for evaluating the noise performance of amplifiers used in low-level signal recovery. In addition to completely describing the noise performance of an amplifier over its entire operating frequency range, they provide the researcher with the information necessary to determine the ultimate performance of his or her system.

Further Information

This Technical Note is intended to explain the use of noise figure contours. Additional information may be found in other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative, or by download from our website at **www.signalrecovery.com**

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TN 1001 Specifying Lock-in Amplifiers
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SIGNAL RECOVERY

What is a Boxcar Averager?

TECHNICAL NOTE
TN 1005

Introduction

The *boxcar averager*, also known as a *boxcar integrator, boxcar detector,* or *gated integrator* is a sampling instrument that integrates the applied input signal during a predefined *gatewidth* or *aperture width,* starting at a predefined *trigger, gate,* or *aperture delay* after an applied trigger. Each of these integrated samples of input signal can then be averaged, using either an analog averager or by digitizing each sample and then averaging the resulting digital values.

The boxcar therefore performs signal recovery by three methods. First, the input signal only affects the output during the period in which it is being sampled; at all other times its level is unimportant, other than the need to avoid causing an input overload for which the recovery time might affect a subsequent sample. The sampling window achieves temporal separation of the signal from the noise, and it is often the biggest single contributor to improving signal to noise ratio. For example, in experiments using pulsed laser sources the signal being measured may only be a few nanoseconds wide, while the repetition rate is at most a few tens of hertz, yielding a signal duty factor of the order of 10^{-8}. In such cases the boxcar gives a convenient way of extracting the signal of interest.

Second, the signal is integrated during the gate width, unlike common sample and hold circuits that simply take a "snapshot" measurement of the signal level at one point in time. Hence if there is noise or other interference present at the input at frequencies that are much higher than the reciprocal of the gatewidth then these will be suppressed.

Finally, the measured samples are themselves averaged, ensuring that low frequency fluctuation or noise, which would cause sample-to-sample variation, is also diminished.

Operating Modes

The boxcar is normally used in one of two modes. In *static gate* work, the time position of the sampling gatewidth relative to the applied trigger (i.e. the gate delay) is fixed, so that the instrument monitors the same point in time on the applied signal. This mode is commonly used to determine how a single feature (for example a peak) in a signal changes as a function of time, typically as some other experimental parameter is adjusted. For example, if the decay time of a fluorescent material were dependent on temperature, then this could be studied by using a pulsed laser to excite it and a photodetector to detect the resulting optical signal. The boxcar averager would be set to measure this photodetector's output signal at a fixed gate delay after the laser pulse, so that by changing the temperature of the material while recording the boxcar output the required information could be obtained.

In *waveform recovery* mode the boxcar operates rather like a sampling oscilloscope, with the gate delay being swept over a range of values while the output is recorded. The result is a record of the signal waveform.

In commercial boxcar instruments, the output was traditionally recorded using an analog chart recorder. The recorder's Y input was connected to the output from the boxcar's output averager, and its X input was driven either by a voltage proportional to the experimental parameter being studied (in static gate work) or by one proportional to the gate delay (in waveform recovery). Nowadays the boxcar's output is usually digitized for recording by a computer and the trigger delay is set using a separate digital delay generator. This makes it possible to use a computer to run the whole experiment and record the resulting data.

Boxcar averagers can recover very fast waveforms and resolve features down to sub nanosecond level. However they can be time inefficient if used for waveform recovery and in such applications signal averagers and storage oscilloscopes may offer better performance. They remain very useful for static gate work.

General Principles of Operation

The heart of any boxcar is the *gated integrator* circuit, shown in simplified form in Figure 1. This circuit is simply an RC low-pass filter gated by switch S_1, (the sampling gate). As shown, the gated integrator has unity DC signal gain.

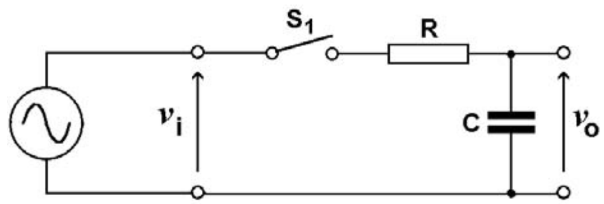


Figure 1, Gated Integrator

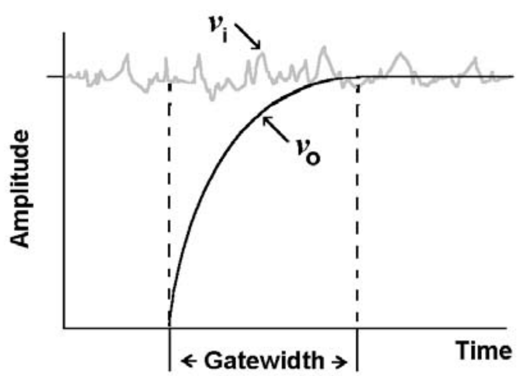


Figure 2, Gated Integrator Operation

When the gate opens (switch S_1 closes) the output voltage ν_o starts to rise exponentially towards ν_i, as shown in Figure 2, where the gatewidth is given as being small in relation to long term changes in ν_i. The gate time constant (set by the product RC) is adjusted so that the ν_o is typically within a few percent of the input ν_i by the end of the selected gate width. Consequently high frequency components of the input signal are removed. The equivalent noise bandwidth is simply $B_{no} = \frac{1}{4RC}$

The integral of the gate sample is the voltage ν_o at the end of the gatewidth. This can either be integrated in an analog output averager or immediately digitized for averaging, after which the sampling circuit is reset ready for the next trigger.

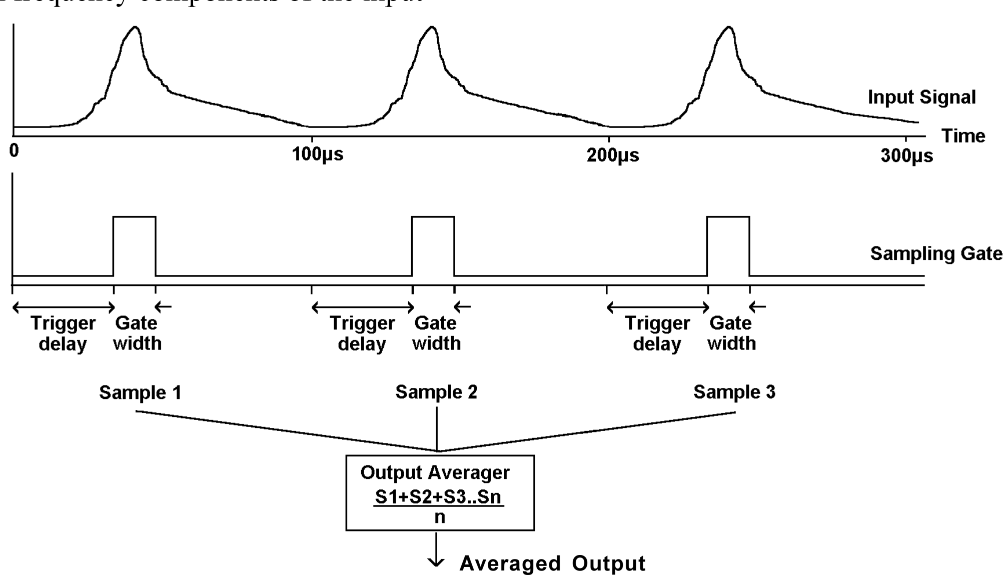
The rest of the boxcar consists of the output averager, discussed later, and trigger processing and delay circuits. The ***SIGNAL RECOVERY*** model 4121B boxcar averager has a trigger input that is similar in design to those used on oscilloscopes, allowing triggering on a variety of waveforms. Following the trigger, a delay circuit allows the position in time start of the gate to be adjusted while a variable pulse generator creates the gate pulse.

The overall maximum trigger rate of the instrument is set by the gate delay, gate width, averager processing and reset times, with the model 4121B offering the fastest rate of commercially available units of up to 80 kHz; competitive instruments are limited to about 20 kHz.

Static Gate Boxcar Averaging

In the *static gate* boxcar averager the length of the trigger delay is fixed and the intention is usually to determine the amplitude of some 'spike' or narrow feature of a waveform that is typically much narrower than the repetition period set by the overall trigger rate.

Consider the situation shown in Figure 3 below.



The input signal consists of a repetitive waveform triggered at a 10 kHz rate, so that each cycle lasts 100 µs. The trigger delay is set to open the gate just before the feature of interest and the gatewidth is set to "bracket" this feature. Each sample results in an integral representing the area under the signal curve for the duration of the gate, and these samples are then averaged in the output averager.

When using a linear output averager, all samples have equal weight and so the output will rise in a linear staircase fashion as shown in Figure 4, curve A.

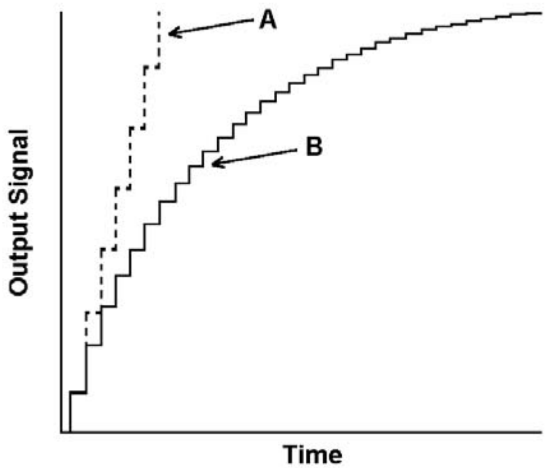


Figure 4, Gated Integrator Output Averager A) linear B) exponential

Each step in this curve represent a new sample becoming available, which in turn corresponds to a trigger cycle. In this *linear averaging* mode, the desired number of signal samples (*n*) is selected; after *n* triggers have occurred, a switch or other method is used to reset the output averager. Since the signal component of the samples will add linearly, but random noise samples add vectorially, after *n* samples of a constant amplitude signal (S) plus white noise (*N*), and after maximizing the gate width to suit the signal waveshape, the output Signal to Noise ratio (SNR_{Out}) is given by:

$$\mathrm{SNR}_{\mathrm{Out}} = \frac{S_1 + S_2 + S_3 + \cdots + S_n}{\sqrt{\left(N_1^{\,2} + N_2^{\,2} + N_3^{\,2} + \cdots + N_n^{\,2}\right)}}$$

$$\mathrm{SNR}_{\mathrm{Out}} == \frac{nS}{\sqrt{\left(nN^2\right)}} = \frac{S}{N}\sqrt{n} = \mathrm{SNR}_{\mathrm{in}}\sqrt{n}$$

so that the Signal to Noise Improvement Ratio (SNIR) is:

$$\mathrm{SNIR} = \frac{\mathrm{SNR}_{\mathrm{out}}}{\mathrm{SNR}_{\mathrm{in}}}\left(= \frac{\mathrm{SNR}\ (n\,\mathrm{samples})}{\mathrm{SNR}(1\ \mathrm{sample})}\right) = \sqrt{n}$$

Note that in this operating mode, it is easiest to think in terms of time averaging since the equivalent noise bandwidth of the gated integrator circuit is not constant but will decrease with increasing *n*.

The linear averager suffers from the disadvantage of needing to be reset after each set of triggers. Although this is not difficult if the output averager is implemented digitally (for example by using a program running in an attached PC), it is not as easy when using analog techniques. Historically, analog versions were more common and so in such cases the output used *exponential averaging.* Essentially this consists of nothing more than a further gated integrator stage, but this time with a time constant much longer than the gate width. The gate on this stage is operated for a short fixed period (typically a few microseconds) to apply the sample voltages out of the input averager to the output averager, with one such gate per trigger cycle.

With exponential averaging, if the samples from the input averager are similar in amplitude then the output from the output averager will rise exponentially as shown in Figure 4, curve B. Again, each step corresponds to a new sample becoming available, and hence a trigger cycle.

Waveform Recovery Boxcar Averager

Boxcar averagers can also be used for *waveform recovery,* where the intention is to record the waveform of the input signal. In this mode of operation the trigger delay is not fixed but rather is incremented by a fixed amount on successive groups of *n* triggers so that it sweeps between *initial delay* and *final delay* values. In this mode, the boxcar output is a replica of the signal waveform and the boxcar can be regarded as a time-translation device that can slow down and recover fast waveforms.

Figure 5 (overleaf) gives a simplified view of this mode. The gate width is now set to be much shorter than the signal period. The first set of *n* samples is taken after a trigger delay set to the initial delay setting. This group of samples is then averaged by the output averager and appears as point A on the output plot. The boxcar counts the applied triggers and after the first set of *n* have been detected, the trigger delay is incremented and the cycle repeats, resulting in point B. The process continues (generating point C and further.

points) until the trigger delay is equal to the preset final delay value. The number of points *m* on the output waveform record is:

$$m = \frac{\text{Final Delay - Initial Delay}}{\text{Delay Increment}}$$

It will therefore be seen that as the delay increment is reduced the time resolution of the recorded waveform will improve, since there are more points. The downside is that the time to record the waveform will also increase.

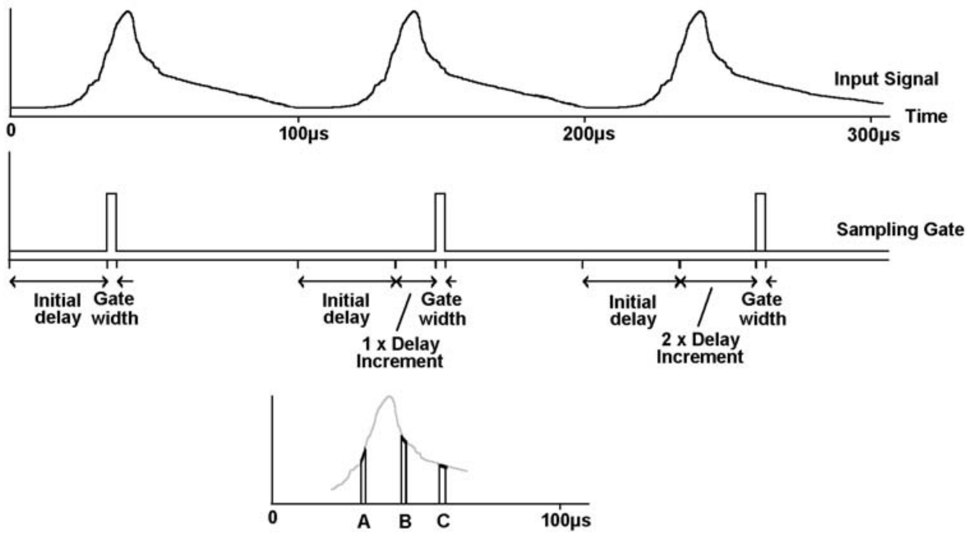


Figure 5, Boxcar Averager Operation - Waveform Recovery Mode

Further Information

This Technical Note is intended as an introduction to the concept of boxcar averaging. Additional information may be found in other ***SIGNAL RECOVERY*** publications, all of which may be downloaded from our website at **www.signalrecovery.com**

In addition staff at any of our offices or those of our distributors and representatives will be happy to answer any questions you may have. For contact details, please visit our website at **www.signalrecovery.com**

SIGNAL RECOVERY

Boxcar Averager Specification Comparison

TECHNICAL NOTE
TN 1006

Introduction

This Technical Note compares key specifications of the **SIGNAL RECOVERY** model 4121B Boxcar Averager with those of the SRS SR250 Gated Integrator. The following sections discuss some key specification differences in detail, which are then summarized at the end of the document.

Time Response

An amplifier followed by a sampling gate and an integrator make up what is known as a *gated integrator* or *boxcar averager*. The time response of the amplifier (t_{Amp}) and sampling gate (t_{Gate}) determine the time resolution of the instrument (t_{GI}). To a first approximation, the time response of an amplifier is 0.35 divided by its 3 dB bandwidth ($t = 0.35/f_{-3dB}$). The overall time response is calculated by summing the squares of the individual responses and then taking the square root ($t_{GI}^2 = t_{Amp}^2 + t_{Gate}^2$). To achieve the best performance/cost ratio, the time response of the amplifier must be just fast enough so that it adds very little to the response time of the gating circuit. Design it with a much faster response and the cost increases with only marginal improvement in resolution. Design it with a slower response and the effort and cost put into the gate design is wasted, giving poorer resolution.

The gatewidth of the **SIGNAL RECOVERY** model 4121B is 1.5 ns, with an amplifier bandwidth of 400 MHz, giving a response time of 0.875 ns. Hence the overall response is **1.7 ns**. The SR250 is specified with sampling gate response time of 2 ns, which by itself, is meaningless. Its input amplifier has a 3 dB bandwidth of only 100 MHz, giving a 3.5 ns response time and limiting its true time resolution to **4.0 ns**.

Amplitude Response and Stability of the Sampling Gate

An important measure of a sampling gate's stability is how its amplitude response changes as a function of its width. The model 4121B gate's amplitude response is stable over its entire range of widths. On the model SR250, we have measured a drop in the response of 30% when the gatewidth is reduced from 100 ns to 10 ns, and an additional drop of 29% when it is reduced further, from 10 ns to 2 ns. This is an overall drop of 50%, not only affecting stability but also significantly reducing the input sensitivity specification at short gatewidths. Table 1, discussed in the next section, makes this clear.

Input Sensitivity

The term *sensitivity* is often thought of as the input voltage (or current) required to produce a full-scale output. Hence when comparing sensitivity specifications expressed simply in terms of input signal level, it is important that the full-scale output is the same, which is the case with the 4121B and SR250. Alternatively, instrument gain can be calculated (normally quoted in reciprocal units of millivolts of input per volt of output) to eliminate the need to take the full-scale output voltage into account.

Table 1 below gives the sensitivity characteristics of the model 4121B. From this it can be seen that the **model 4121B is five times more sensitive than the SR250** at short gatewidths.

	Model 4121B at all gatewidths	SRS SR250 at > 100ns gatewidths	SRS SR250 at 10 ns gatewidth	SRS SR250 at 2 ns gatewidth
Input Sensitivity	20 mV	50 mV	71.2 mV	100 mV
1/Gain	2 mV per Volt	5 mV per Volt	7.12 mV per Volt	10 mV per Volt

Table 1, Instrument Sensitivity and 1/Gain Comparison

Input Impedance

The model 4121B offers input impedances of 1 MΩ or 50 Ω via two separate connectors. The 50 Ω input amplifier is DC coupled and gives a true 50 Ω impedance, giving both high bandwidth and excellent voltage standing-wave-ratio (VSWR). VSWR indicates how much of the input pulse reflects from the input amplifier, back into the input cable, with a perfect figure implying that the amplifier completely absorbs the signal presented to it. In this case there is no reflection back towards the source that might cause distortion or an erroneous signal to appear.

Some experiments need a high input impedance, but in these cases the signal bandwidth will be limited by the input time constant, which is given by the product of the cable capacitance and the input impedance. Such experiments can use the 4121B's 1 MΩ input, which is connected to a unity gain buffer amplifier. The bandwidth of this amplifier is lower than that of the 50 Ω input amplifier, but this is not a restriction since the bandwidth will in this case be limited by the input time constant.

The SR250, on the other hand, has only one input amplifier with a 1 MΩ input impedance, restricting the signal bandwidth. This also means that when a 50 Ω input is required then the user must fit an external coaxial terminator.

Hence the 4121B gives the user the option of using a true 50 Ω input when maximum bandwidth is required (typically for the narrowest input signals and short gatewidths), while still offering a 1 MΩ input when this is needed. The SR250, on the other hand, only offers a 1 MΩ input that has limited bandwidth.

Trigger Rate

The repetition rate of lasers and other signal sources is constantly rising. As a result, the maximum trigger rate of boxcar averagers is increasingly important, since it determines whether the required data can be collected in the time available. For example, assume that the experiment is capable of running at the maximum trigger rate of the boxcar. In this case, the 4121B's 80 kHz trigger rate will allow data to be taken **four times faster** than when using the SR 250, which is limited to a 20 kHz maximum trigger rate.

Inter-Sample Correlation

Both the 4121B and SR250 provide a Last Sample Output signal. This is an analog voltage that represents the integral of the input signal during the gatewidth for the sample initiated by the previous trigger pulse. Clearly it is desirable that the voltage at this output should relate only to the sample corresponding with the last trigger pulse, but in practice this may not be the case. In the SR250, for example, this output is not fully reset between triggers, and up to 5% of the sample corresponding to trigger t_{n-1} can remain and thereby affect the sample corresponding to trigger t_n. The equivalent figure for the model 4121B is less than 0.5%.

One way of trying to obtain the same performance from the SR250 as that from the 4121B is to ignore every other sample at the last sample output. Hence, for example, the sample taken at t_{n-2} is used, but that at t_{n-1} is ignored. The next sample, at t_n is also used, since this now has a maximum error due to the last sample which was used, at t_{n-2} of 5% × 5%, or 0.25%. However, this technique clearly results in the maximum data collection rate for the SR250 being half of that for the 4121B, when using the Last Sample Output. When the maximum trigger rate is also taken into account, this gives a maximum data acquisition rate for the 4121B which is eight times that provided by the SR250.

Trigger to Sample Time

Boxcar averagers always take time to respond to a trigger, which is known as the *intrinsic delay*. This delay is the time from the receipt of a trigger to the point at which the sampling gate opens. Designers struggle with keeping this time as short as possible, since the signal of interest often starts at the same time as, or only shortly after, the trigger.

In the model 4121B the minimum intrinsic delay is 20 ns, while that in the SR250 is 25 ns. Although this difference of 5 ns is small, it can still make a difference between measuring the signal and not finding it at all. In really difficult situations, the 4121B's internal delay board can be bypassed to reduce its intrinsic delay even further, down to 15 ns

Long Gatewidths

The model 4121B has gatewidths that are continuously variable from 1.5 ns to 150 µs, while the SR250's longest gatewidth, without dismantling it and changing a capacitor, is 15 µs

Analog Averaging Mode

Both the 4121B and SR250 include an analog output averager, with both offering an exponential averaging mode that is good for following changing signals. This is because it "forgets" older data in an exponentially weighted fashion. The weighting factor for a given output sample is $e^{1/-t}$, where time *t* is the time between this sample and the most recently acquired one. Because of its forgetful nature, the exponential averaging is not the optimum choice when measuring the smallest signals.

The 4121B offers a second mode of analog averaging that is not provided in the SR250. This is linear averaging, in which each output sample contributes equally to the output, ensuring that none of the precious signal is "thrown away". The result is the best signal-to-noise improvement in the shortest amount of time.

Baseline Subtraction

In some experiments using laser sources it is useful to be able to sample both the signal of interest and the baseline signal, subtracting one from the other to eliminate baseline drift. The baseline subtraction mode of the boxcar averager can help in such cases. In this mode, both the experiment and boxcar are triggered at twice the required rate, but the boxcar generates an output signal that indicates whether the given trigger will be treated as a signal or a baseline value. This signal can then be used to driver a shutter or light chopper, such as the ***SIGNAL RECOVERY*** models 197 or 651-1, which effectively block alternate laser pulses from exciting the experiment.

Both the 4121B and SR250 include this Baseline Subtraction mode. But the model 4121B also includes another baseline mode, which can be used at high trigger rates when the chopper or shutter used in the normal mode cannot respond fast enough. In this second mode the boxcar accepts a TTL "steering" input that indicates whether a sample is to be treated as a signal or a baseline value. Hence, for example, in an experiment running at 20 kHz trigger rate, an external chopper, running at say 1 kHz could be used to drive this input. The effect would be that groups of 20 samples would be treated alternately as signal and baseline values. The SR250 does not provide this extra flexibility.

Summary

Table 2 below summarizes the above discussion, from which it will be seen that the ***SIGNAL RECOVERY*** model 4121B Boxcar Averager offers a number of significant advantages over the SRS SR250

Specification	***SIGNAL RECOVERY*** model 4121B	SRS SR250
Input Time Response	1.7 ns	4.0 ns
Max. Input Bandwidth	400 MHz	100 MHz
Amplitude Response	Flat over all gatewidths down to 2 ns	Flat to gatewidths to 100 ns Drop by 30% at gatewidths down to 10 ns Drop by 50% at gatewidths down to 2 ns
Max. Sensitivity	All gatewidths: 20 mV	> 100 ns gatewidth: 50 mV 10 n gatewidths: 71.2 mV 2 ns gatewidth: 100 mV
Input Impedance	1 MΩ or 50 Ω	1 MΩ only
Max. Trigger Rate	80 kHz	20 kHz
Inter-Sample Correlation	less than 0.5%	less than 5%
Min. Trigger to Sample Time (Intrinsic Delay)	With Delay Board in circuit: 20 ns Delay Board bypassed: 15 ns	25 ns
Standard Gatewidth Range	1.5 ns to 150 µs	1.5 ns to 15 µs

Table 2, Comparison of Key Specifications - Model 4121B vs. SRS SR250

Further Information

This Technical Note compares the specifications of two commercial boxcar averagers. Additional information may be found in other ***SIGNAL RECOVERY*** publications, all of which may be downloaded from our website at **www.signalrecovery.com**

In addition, staff at any of our offices or those of our distributors and representatives will be happy to answer any questions you may have. For contact details, please visit our website at **www.signalrecovery.com**

The Incredible Story of Dr D. P. Freeze

TECHNICAL NOTE
TN 1007

I'm sure you've all heard of the astounding event that took place recently. Dr D. P. Freeze, a well known experimental physicist from the 1960's, who had been lost and presumed dead during an Antarctic expedition many years ago, was discovered entombed in a huge block of ice.

To the amazement of his rescuers, Freeze was not dead and after thawing out was able to resume his post at the University.

A few days after being given his first assignment, Freeze visited his supervisor Dr W. I. Thit, radiating gloom and despondency. "Bill", said Freeze, "You've given me an impossible task. This measurement you want me to make needs an incredibly sophisticated piece of equipment. Just look at the specification we need."

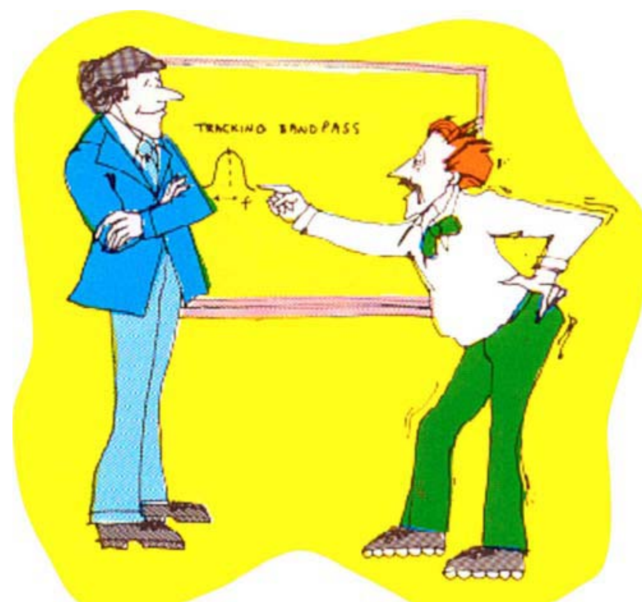
"First of all, the system must amplify a 1 nV AC signal and turn it into a 10 V DC signal. Bill, do you realize that's a gain of 10^{10}, or 200 dB? Think of the shielding we'll need!"

"The huge gain wouldn't be so bad if we had a clean signal, but look at the input signal to noise ratio we can expect. I calculate that our 1 nV signal will be drowned out by an interfering signal that's bigger by five orders of magnitude."

"For God's sake, Bill, don't you realize that means an input dynamic reserve of at least 10^5 if the system is not going to overload. And look at the dynamic range that implies. We need to resolve our signal to 10 pV or one part in a hundred, and that means an input dynamic range of 10^7 or 140 dB!"

By this time, Freeze was pacing backwards and forwards, his face pink with passion.

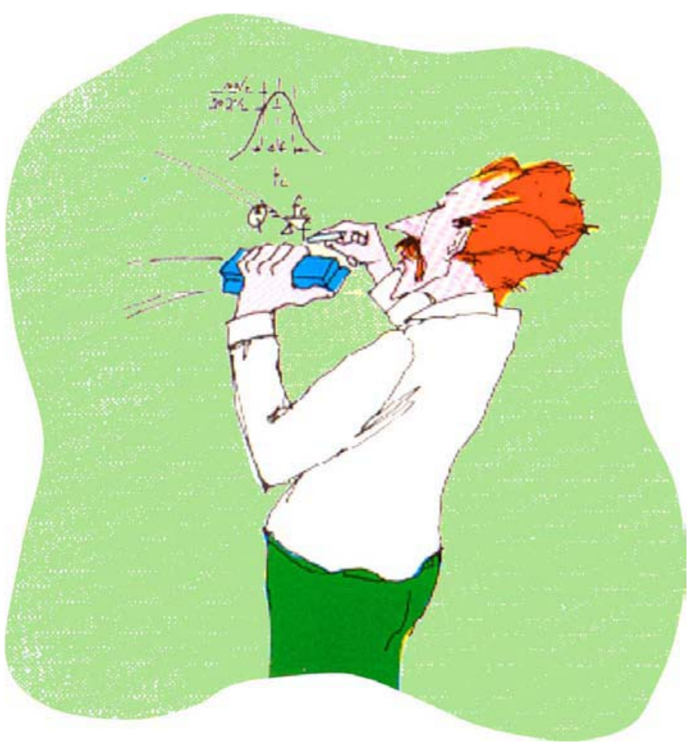
"And, Bill," he said in a choked voice, "the worst is yet to come - look at the filtering performance we need."



"Our system must lock in, in both frequency and phase, to a reference signal and not only can the waveshape of our reference be sinusoidal, square, triangular, narrow pulses, or anything in between, but its frequency need not be constant - it can change continuously over a 10^6 range of frequency."

At this point, Freeze took a grip on himself and continued more calmly. “You see, Bill,” he said, “I don't think you fully appreciate the problem.”

“Our system must act as a selective or tuned amplifier and amplify only a selectable narrow band of frequencies centered on the frequency of the reference signal. In other words, Bill, our system needs to be a frequency-tracking bandpass filter with enormous gain and with selectable bandwidth or Q-factor.”



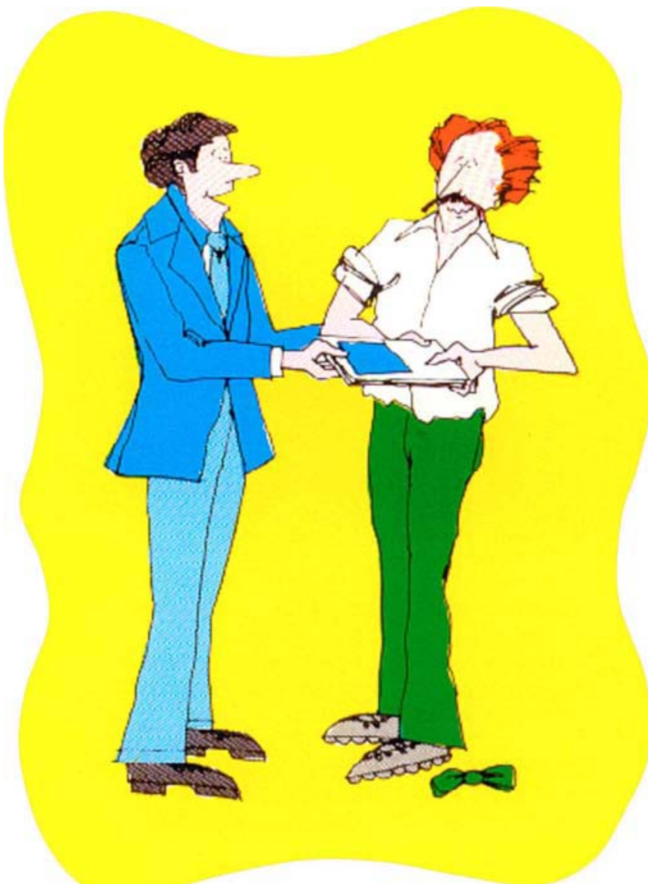
“If nothing else will convince you,” said Freeze, “just look the Q-factor requirements. We both know what Q is – the filter center frequency divided by the bandwidth, right? You remember when we were students learning circuit design – with either tuned L-C circuits or op-amp R-C circuits, a Q of 100 is about as high as you can go and still have acceptable frequency and amplitude stability, not to mention phase stability.”

“All right,” said Freeze, taking a deep breath, “this system you're asking for needs a bandwidth 0.001 Hz at a centre-frequency of 100 kHz. That implies a Q of 10^8, or 100 million - now do you believe it's impossible!” He slumped tiredly into a chair and waited for his friend to reply.

“It’s understandable, of course,” said W. I. Thit with a smile, “in view of your forty-five years of hibernation.

continues over....

"What you have described requires an instrument called a lock-in amplifier and there's a company called ***SIGNAL RECOVERY*** which for the last forty years, both under their present and former names of Princeton Applied Research and Brookdeal Electronics, has specialized in their design, manufacture and application. Here's their latest catalog – take it with you, or visit their website at **www.signalrecovery.com**, and choose the model you need.



"Hello, ***SIGNAL RECOVERY***, I have this measurement problem. A colleague gave me a copy of your catalog, and I've looked at your website and..."

Further Information

The following Technical Notes give further information about the selection and operation of lock-in amplifiers. They may be downloaded from our website at **www.signalrecovery.com**

TN 1000 What is a Lock-in Amplifier?
TN 1001 Specifying a Lock-in Amplifier
TN 1002 The Analog Lock-in Amplifier
TN 1003 The Digital Lock-in Amplifier

Company Names

SIGNAL RECOVERY is part of AMETEK Advanced Measurement Technology and includes the businesses formerly trading as EG&G Princeton Applied Research, EG&G Brookdeal Electronics, EG&G Instruments (Signal Recovery), EG&G Signal Recovery and PerkinElmer Instruments (Signal Recovery). The brand name Princeton Applied Research is also part of AMETEK Advanced Measurement Technology, but is now only used for electrochemistry equipment and accessories.

SIGNAL RECOVERY

Digital Noise at Lock-in Amplifier Input Connectors

TECHNICAL NOTE
TN 1008

Overview

Since their invention in the early 1960's, lock-in amplifiers have been used whenever the need arises to measure the amplitude and/or phase of a signal of known frequency in the presence of noise. Unlike other AC measuring instruments they have the ability to give accurate results even when the noise is much larger than the signal - in favorable conditions even up to a million times larger.

Early instruments used analog technology, with manual controls and switches, and with output readings being taken from large panel meters. Later, microprocessors were added to give more user-friendly operation, digital output displays, and to support computer control. More recently still the analog phase sensitive detectors forming the heart of the instrument have been replaced by DSP (digital signal processing) designs, further improving performance.

But the addition of this digital technology has had one unfortunate side effect, which is that the instrument itself can act as a source of digital clock and switching noise that is typically coupled back into the experiment via the signal input or internal oscillator output connectors. This noise is of course rejected by the lock-in and generally does not impair its performance, but the power it dissipates in the sample or device under test can cause serious problems. This is particularly the case in low temperature physics and semiconductor research.

The ***SIGNAL RECOVERY*** model 7124 precision lock-in amplifier has been designed to be particularly suited to such work. It uses a unique analog fiber optic link to interconnect a remote connection unit (RCU), to which the experiment is connected, with the main instrument console. In normal operation there are no digital clock signals within the RCU, and so it can emit no switching noise. This architecture gives an instrument with all the advantages of the latest DSP technology for signal detection, and a powerful processor for easy user operation, as well as the low noise performance that until now has only been available in instruments of all-analog design.

This technical note describes measurements of the noise emitted by the signal input connectors on a number of lock-in amplifiers to demonstrate the superior performance of the Model 7124. The following instruments are considered:

Model Number	Supplier	Design
SR830	Stanford Research Systems	DSP
7265	***SIGNAL RECOVERY***	DSP
124A	Princeton Applied Research, the brand name used historically for ***SIGNAL RECOVERY*** products. The product was discontinued in 1994 but is still widely cited in the technical literature	Analog
7124	***SIGNAL RECOVERY***	DSP with separate all-analog front-end connection unit connected to main console via analog fiber links

The Model 7265 and the SR830 are general purpose DSP lock-in amplifiers, and were chosen for these tests in order to illustrate the performance of the Model 7124 compared with the most commercially popular instruments. The Model 124A is a sought after, but now obsolete all-analog instrument, which has attained legendary status in low temperature physics research because of the complete absence of digital switching noise at its input connectors.

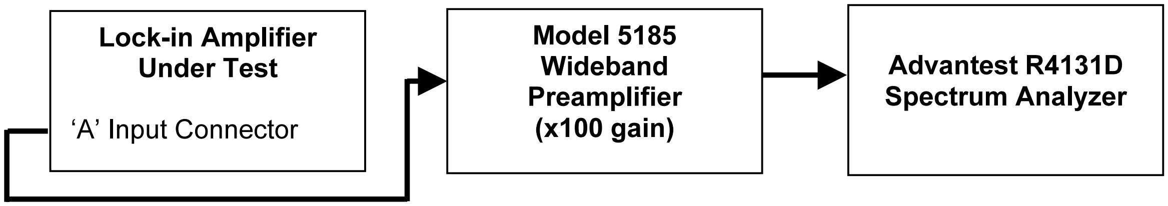


Figure 1, Spectral Measurement Test Setup

Spectral Measurements

In the first set of tests the input connector on each instrument was connected to a general-purpose spectrum analyzer via a low noise wideband preamplifier, in order to improve the overall sensitivity of the measurement. The preamplifier was set to ×100 gain with an input impedance of 50 Ω, and had a bandwidth of greater than 200 MHz. The test setup is shown in figure 1.

It might be expected that if the lock-in amplifier were turned off then there would be no measurable signal on the spectrum analyzer. However, the presence of interconnecting cables and ground connections to the line power source mean that this is not the case, so two sets of measurements were therefore taken. In the first, the instrument was connected to the line power supply but turned off, giving a "background" measurement, and in the second it was turned on. The intention was to identify the additional energy generated by the instrument when it is turned on and which is therefore properly attributed to its operation. In many real experiments researchers use a Faraday cage and extensive RF filtering on the line power supply to significantly reduce the background level.

Figure 2 shows the background spectrum for the Stanford Research Systems model SR830 when it is turned off, and figure 3 the same measurement when it is turned on. The significant increase in energy above 40 MHz that is output from the input connector is very apparent.

Figures 4 and 5 show the results for the same measurement using the ***SIGNAL RECOVERY*** model 7265. The two spectra are similar for frequencies up to 100 MHz, but there is some additional energy in the region above this when the instrument is operating, although very much less than in the case of the SR830.

The benchmark against which the model 7124 will be compared is, though, the model 124A. Figures 6 and 7 show the results for this unit. There is some increase in signal in the 40 MHz to 80 MHz region, which given that this unit has no digital clock signals cannot be caused by these breaking into the signal channel. Rather, it is most likely to be due to changes in the impedance of the input circuits between their powered and unpowered states. Most noticeably, though, in the region above 80 MHz there is no significant difference in the spectra, and there is less energy than in the case of the Stanford Research Systems model SR830 or ***SIGNAL RECOVERY*** model 7265.

Figures 8 and 9 show the results for the ***SIGNAL RECOVERY*** model 7124. In this case there is no additional noise when the unit is turned on, and indeed the rejection of background interference in frequencies up to 40 MHz actually improves, again

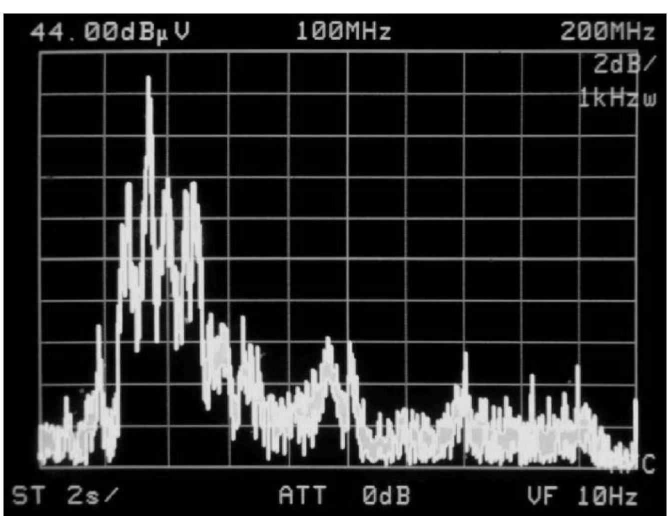


Figure 2, Background Spectrum when turned Off - Model SR830

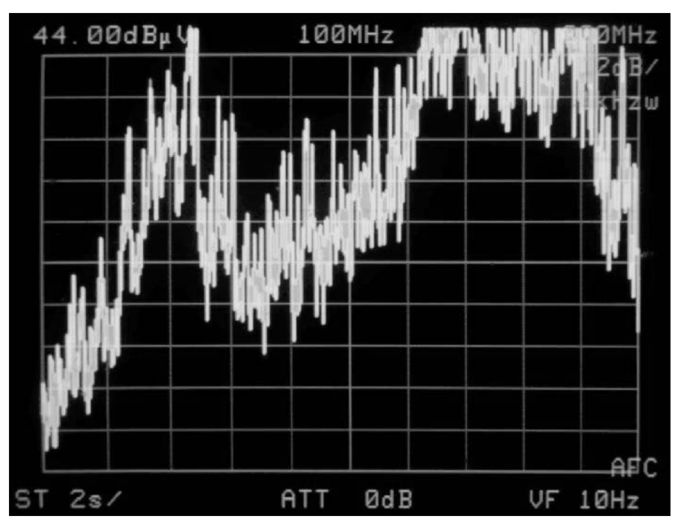


Figure 3, Spectrum when turned On - Model SR830

most probably due to changes in the impedance of the input circuits between their powered and unpowered states.

In conclusion, the spectral power tests clearly indicate that of the three instruments in current production, the model 7124 has the lowest emission of interfering signals from its input connectors, and furthermore, its performance matches or even exceeds that of the now-obsolete model 124A.

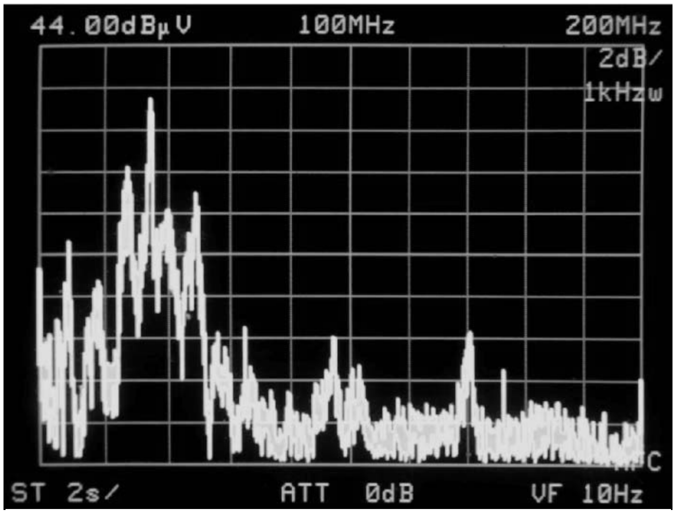


Figure 4, Background Spectrum when turned Off - Model 7265

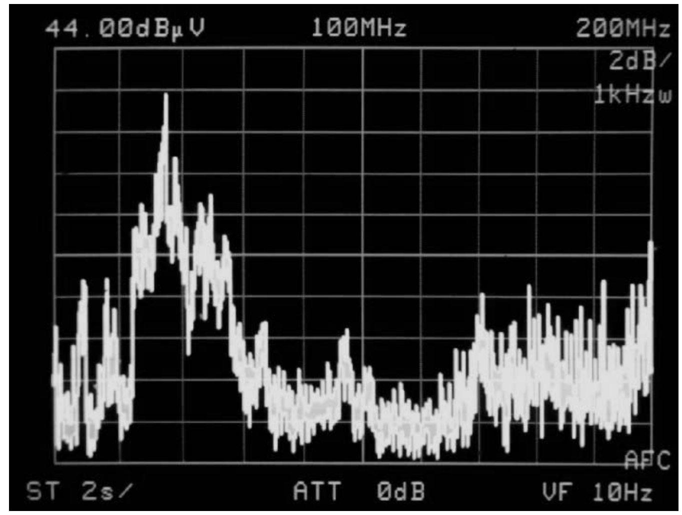


Figure 5, Spectrum when turned On - Model 7265

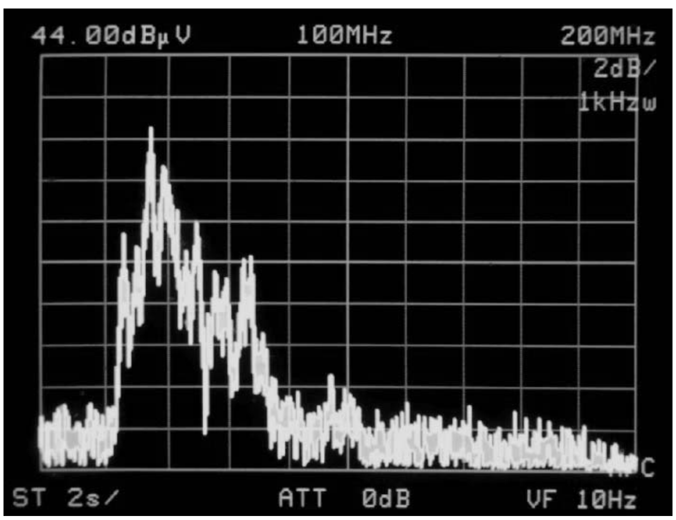


Figure 6, Background Spectrum when turned Off - Model 124A

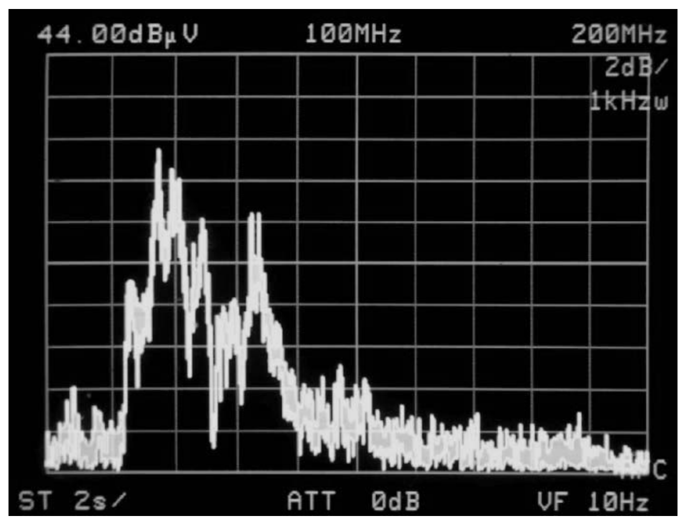


Figure 7, Spectrum when turned On - Model 124A

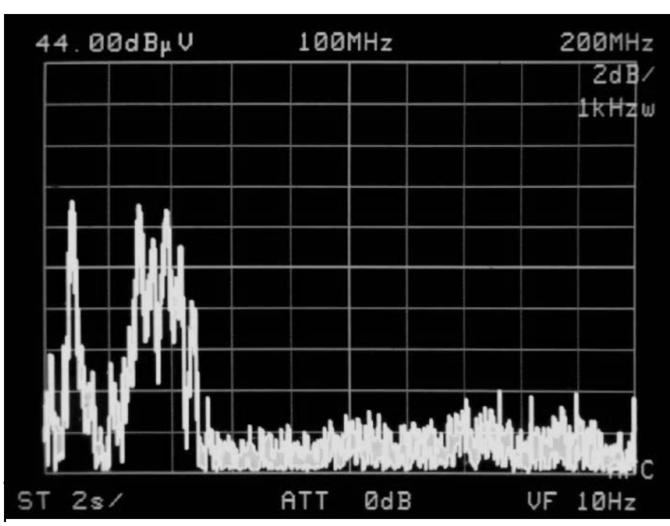


Figure 8, Background Spectrum when turned Off - Model 7124

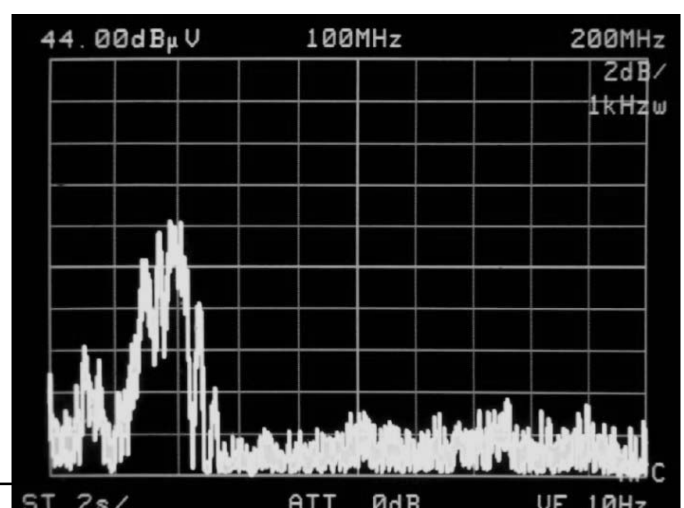


Figure 9, Spectrum when turned On - Model 7124

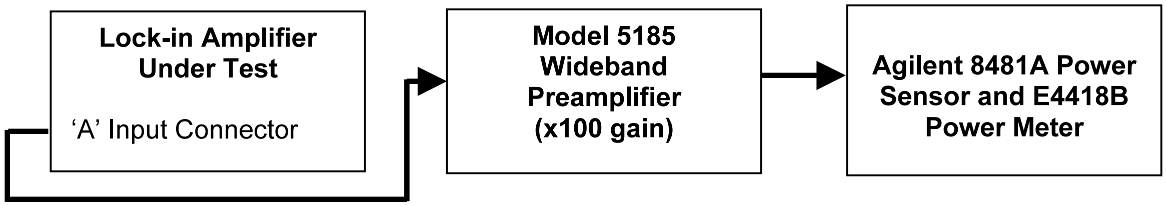


Figure 10, Power Measurement Test Setup

Power Measurements

The actual spectral density of the power emitted from the input connectors does not normally matter. What is of most interest is the total power, since it is this that causes sample heating and affects experimental results.

A second set of tests was therefore performed. The input connector on each instrument was connected to an Agilent power meter via a low noise wideband preamplifier, again in order to improve the overall sensitivity of the measurement. The preamplifier was set to ×100 gain with an input impedance of 50 Ω, and had a bandwidth of greater than 200 MHz. The test setup is shown in figure 10.

The results of these measurements are given below in Figure 11. In this chart, power expressed in dBm is the power expressed in decibels with respect to a power of 1 mW; hence the lower the figure, the lower the power.

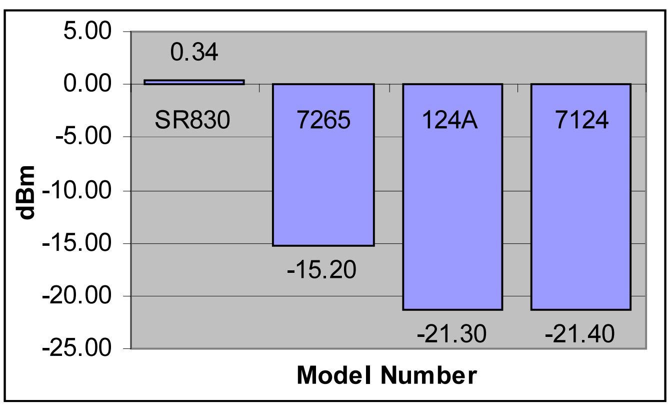


Figure 11, Power Measurement Results

It can clearly be seen from these results that the noise emitted from the input of the model 7124 is more than 21 dBm (125 times) lower than that from the SR830, and 6 dBm (4 times) lower than that from the model 7265. It also matches the performance of the model 124A.

Conclusions

Digital switching noise emitted from the input connectors of a lock-in amplifier can cause problems where the power it dissipates affects the experiment. Of the currently commercially available instruments, the ***SIGNAL RECOVERY*** model 7124 offers the best performance in this respect, and matches that delivered by the now-obsolete model 124A. It is therefore the optimum choice of instrument for any research where this key specification is critical.

Equipment Tested

The results reported herein were measured using the following instruments: Model SRS830 S/N 21378, Model 7265 S/N 08028799, Model 124A S/N 98103, and Model 7124 S/N 08199246.

Further Information

The following Technical Notes give further information about the selection and operation of lock-in amplifiers. They may be downloaded from our website at **www.signalrecovery.com**

TN 1000 What is a Lock-in Amplifier?
TN 1001 Specifying a Lock-in Amplifier
TN 1002 The Analog Lock-in Amplifier
TN 1003 The Digital Lock-in Amplifier
TN 1004 How to Use Noise Figure Contours
TN 1007 The Incredible Story of Dr D.P. Freeze

Company Names

SIGNAL RECOVERY is part of AMETEK Advanced Measurement Technology and includes the businesses formerly trading as EG&G Princeton Applied Research, EG&G Instruments (Signal Recovery), EG&G Signal Recovery and PerkinElmer Instruments (Signal Recovery). The brand name Princeton Applied Research is also part of AMETEK Advanced Measurement Technology, but is now only used for electrochemistry equipment and accessories

SIGNAL RECOVERY

Dual-Channel Absorption Measurement with Source Intensity Compensation

APPLICATION NOTE

AN 1000

Introduction

The **SIGNAL RECOVERY** (formerly EG&G/PerkinElmer) models 7260, 7265 and 7280 Dual Phase DSP Lock-in Amplifiers include a dual reference mode which allows the independent, but simultaneous, measurement of two signals of different frequencies. One application for which this mode is ideally suited is the removal of errors due to source intensity fluctuations in optical measurements.

The few restrictions which are imposed by the use of the dual reference mode, such as a normal maximum operating frequency of 20 kHz, do not usually cause any problems in this type of experiment. This is because the signals to be detected are caused by chopped light beams, typically generated by mechanical rotating-blade light choppers, and hence the maximum frequencies encountered are only a few kilohertz.

Problem

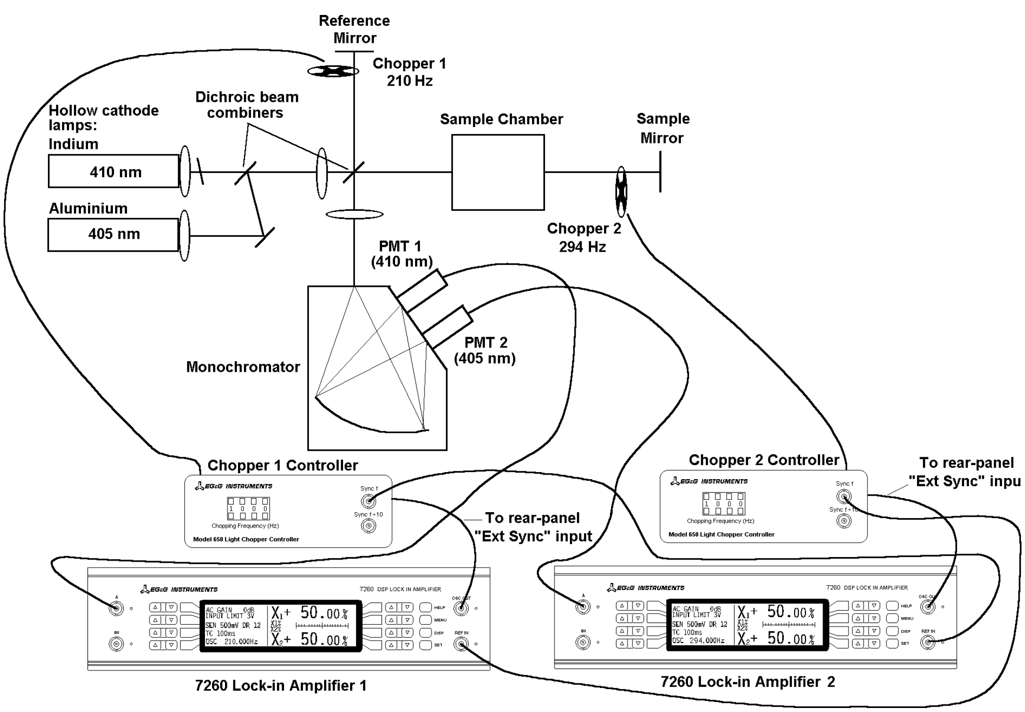
A customer wished to measure simultaneously the optical absorption of a sample at two wavelengths, namely 410 nm and 405 nm. However the available light sources at these wavelengths were not stable enough to use without compensation for their intensity fluctuations, and so the system as shown in figure 1 was utilized.

Description of Solution

The outputs of the two hollow-cathode lamps are combined, using a dichroic beam combiner, to form a single beam containing both 405 and 410 nm. This beam strikes a second dichroic beam combiner/splitter, and is split so that it passes via the reference and sample paths. The reference beam passes via the first of two light choppers, which is running at 210 Hz, and is then reflected back via the reference mirror. The sample beam passes through the sample chamber, is chopped by the second light chopper at 294 Hz, reflected back via the sample mirror and makes a second pass through the sample chamber.

Both sample and reference beams are recombined at the beam combiner and pass to a monochromator which is equipped with two exit slits adjusted to 410 nm and 405 nm center wavelengths. Each slit is fitted with a photomultiplier tube (PMT) detector, shown as PMT1 and PMT2 in the diagram.

Hence the electrical signal generated by PMT1 has a component at 210 Hz, resulting from that part of the output from the 410 nm lamp which passed via the reference mirror, and a component at 294 Hz, resulting from that part which passed via the sample chamber and mirror. In a



similar way, the signal at the output of PMT2 has a component at 210 Hz, resulting from that part of the output of the 405 nm lamp which passed via the reference mirror, and a component at 294 Hz, resulting from that part which passed via the sample chamber and mirror.

Two model 7260 lock-in amplifiers, both operating in dual reference mode, are used to detect these signals with the connections made, and controls set, as follows:-

1) The first instrument's internal oscillator is set to 210 Hz and a cable is connected between its **OSC OUT** connector and the external sync input of the chopper 1 controller.
2) The switch on the rear panel of the chopper 1 controller is set to external sync mode, and so the chopper 1 frequency is also 210 Hz.
3) The SYNC output of chopper 1, which is therefore also at 210 Hz, is connected to the **REF IN** reference input connector on the front panel of the second 7260 lock-in amplifier.
4) This second 7260's oscillator is set to 294 Hz and, in the same way as for the first unit, a cable is connected between its **OSC OUT** connector and the external sync input of the second chopper (chopper 2) controller.
5) The switch on the rear panel of the chopper 2 controller is also set to external sync mode, and so the chopper 2 frequency is also 294 Hz.
6) Finally, the loop is completed by a cable linking the **SYNC** output of chopper 2, which is therefore also at 294 Hz, to the **REF IN** reference input connector on the front panel of the first 7260 lock-in amplifier.

Hence the first lock-in amplifier runs at an internal reference frequency of 210 Hz and an external reference frequency of 294 Hz, whilst the second runs at an internal reference frequency of 294 Hz and an external reference frequency of 210 Hz. The signal inputs of the two instruments are connected to PMT1 and PMT2 respectively.

When all four reference phases are adjusted for maximum X-outputs, the following outputs are generated:

Lock-in amplifier 1:-

È X1 output corresponding to the 410 nm, 294 Hz signal at PMT1 which passed through the sample

È X2 output corresponding to the 410 nm, 210 Hz signal at PMT1 which passed through the optical reference path.

Lock-in amplifier 2:-

È X1 output corresponding to the 405 nm, 210 Hz signal at PMT2 which passed through the optical reference path.

È X2 output corresponding to the 405 nm, 294 Hz optical at PMT2 which passed through the sample

These four outputs are transferred to a computer via each of the instruments' RS232 interfaces using the compound command X1;X2 which reports the present value of the X1 and X2 outputs respectively.

Any change in the absorption of the sample in the sample chamber affects only the intensity of the 294 Hz signals at PMT1 and PMT2, whereas any change in the intensity of the hollow cathode lamps affects the signals at both 294 Hz and 210 Hz. Hence by calculating the ratio X1/X2 of the outputs of lock-in amplifier 1, and X2/X1 of the outputs of lock-in amplifier 2, the effect of these fluctuations can be removed and the absorption measured at each wavelength independently.

The calculations are performed by a user-written program running on a controlling computer (not shown in figure 1) which operates both instruments via their RS232 interface(s).

Conclusion

The combination of the unique dual reference mode provided by the models 7260, 7265 and 7280 and a simple optical design for the experiment allows two independent source-compensated optical absorption measurements to be made using only two lock-in amplifiers. Traditional approaches to the same experiment would require four instruments. In addition to this saving in equipment, the dual reference mode provides more accurate measurements since both the signal and reference are detected by the same detector and follow the same signal path, thereby avoiding problems caused by differential drift between two different detectors and instruments.

Acknowledgement

SIGNAL RECOVERY acknowledges the assistance of Dr P Brewer of Hughes Research Laboratories, Malibu, California, USA, in the preparation of this applications note.

SIGNAL RECOVERY Input Offset Reduction using the model 7265/7260/7225/7220 Synchronous Oscillator/ Demodulator Monitor Output

APPLICATION NOTE

AN 1001

Introduction

The **SIGNAL RECOVERY** (formerly EG&G/PerkinElmer) models 7220, 7225, 7260 and 7265 Digital Signal Processing (DSP) lock-in amplifiers, when used in the external reference mode, are able to generate a sinusoidal signal of variable amplitude and variable phase with respect to the applied reference. This signal can be used to null out the "steady state" value of signals at their inputs, allowing small signal changes to be more easily measured.

This input offset suppression is useful in such applications as Hall effect studies where it offers better performance than the output offset capability also available in the instruments.

In the models 7260 and 7265, the term Synchronous Oscillator is used to describe this feature, but in the models 7220 and 7225 it is known as the Demodulator Monitor. In order to avoid confusion, throughout the rest of this Technical Note the former term will be used.

Synchronous Oscillator Operation

The operation of the Synchronous Oscillator is best understood by reference to the simple block-diagram of part of a lock-in amplifier, shown in figure 1.

The diagram shows the signal and reference channels, and the in-phase mixer, or phase-sensitive detector (PSD) found in the models 7220, 7225 7260 and 7265 dual phase lock-in amplifiers. The quadrature mixer has been omitted for the sake of clarity.

When the synchronous oscillator output is activated, the output at the **OSC OUT** connector on the front panel becomes an analog representation of the sinusoidal drive signal to the in-phase PSD, rather than simply the internal oscillator output. Its amplitude can still be controlled, using the oscillator's level control, to voltages between 1 mV and 5.000 V, but its frequency is now set by the external reference frequency input signal. In addition, its phase relative to the applied phase-shifter can be adjusted in 10 m° increments using the instrument's reference phase-shifter.

Note that due to a lack of sufficient connectors on the instrument, it is not possible to make both the oscillator output and the synchronous oscillator output available simultaneously. Hence the technique cannot be used for input offset reduction when using internally referenced experiments and so the synchronous oscillator output is not available when the instrument is set to internal reference mode.

Operation is generally easier if the instrument is set to R-χ (vector magnitude and phase) display mode, although it is still possible to use the technique in the X-Y display mode.

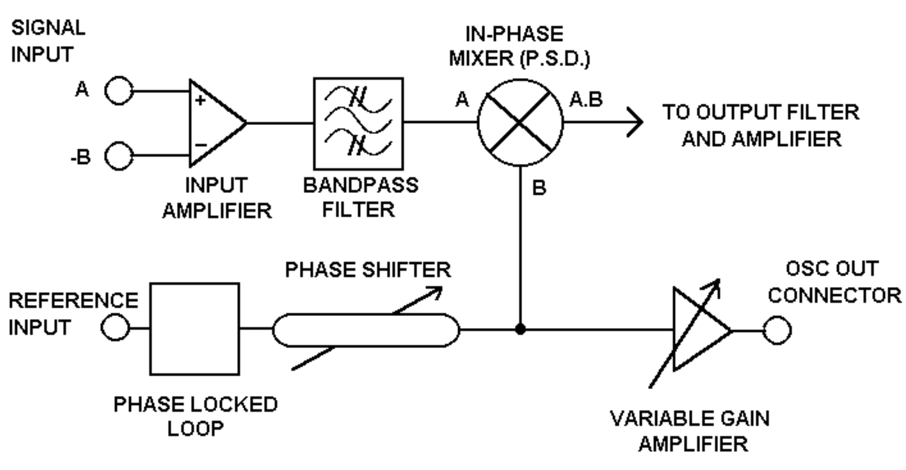


Figure 1

Offsetting a Voltage Signal

In order to use the input offset reduction technique, it is necessary to couple the synchronous oscillator output signal back to the instrument's input. There are various ways in which this can be done, and two of these are shown in figures 2 and 3.

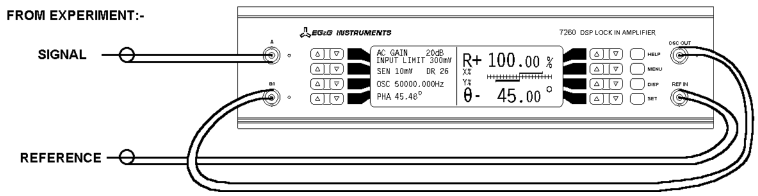


Figure 2
Connecting the Synchronous Oscillator Signal using the 7260 Differential Input Amplifier

Figure 2 shows the easiest way to make the required connections and is suitable for use when the signal to be measured is a single-ended voltage. The instrument controls are then set as follows:-

a) Perform an auto-default operation to set the instrument to a defined state. This operation automatically sets the outputs to R-χ mode.

b) Set the unit to external reference mode and connect the reference signal to the selected input, either **REF IN** on the front panel or **TTL REF IN** on the rear panel. Connect the signal to the **A** input connector and connect a cable between the **OSC OUT** connector and the **B/I** connector.

c) Adjust the experiment so that the measured signal is at the level that it is required to be offset to zero, i.e. establish the "steady" state value from which changes will be measured.

d) Perform an auto-sensitivity function.

e) Perform an auto-phase function. The result of steps d) and e) will be to set the instrument so that the signal at the reference input to the X-channel PSD is in phase with that at the signal input.

f) Note the measured signal magnitude. In order to offset this signal, the synchronous oscillator amplitude needs to be set to exactly the same level. Since the synchronous oscillator output can only be adjusted between 1 mV and 5 V, it may be necessary to use of a coaxial attenuator inserted in the BNC cable between the **OSC OUT** and **B** connectors in order to obtain the correct level.

In addition, the resolution with which the amplitude can be adjusted is better at larger output amplitudes. For example, with no attenuator and if the measured signal were 800 mV then the oscillator amplitude control provides sufficient resolution, but if the signal level were 2 mV then it will not. However, if a ×1000 attenuator (60 dB) is fitted then the oscillator amplitude control will provide microvolt resolution, which is sufficient.

g) Set the oscillator amplitude control so that the amplitude following the attenuator, if fitted, is the same as the measured signal.

h) Turn the synchronous oscillator output on and set the signal input mode to differential (A-B) mode. This will cause the input to the signal channel to become equal to the input signal minus the synchronous oscillator output level. Since these two signals are nominally equal, the displayed magnitude will decrease.

i) Use the instrument's reference phase control to obtain a minimum output. Do not try to use the auto-phase function since it will not work correctly when using the instrument in this mode.

j) Make fine adjustments to the oscillator amplitude and reference phase controls to further reduce the output. It should also be possible to increase the full-scale sensitivity range.

k) The system is now set up to measure small changes in the applied signal. Because the offset level has been removed by the above procedure, the full input dynamic range is available for the measurement, giving the best possible accuracy.

Offsetting a Current Signal

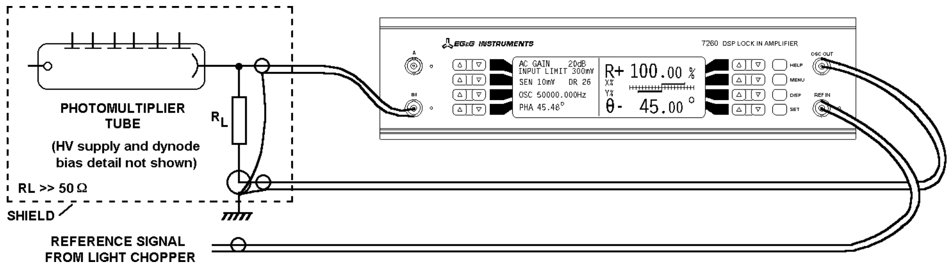


Figure 3
Connecting the Synchronous Oscillator Signal using an External Resistor

Figure 3 shows the use of the technique when measuring a current signal, such as that from a photomultiplier tube. The resistor R_L injects an offset current, derived from the synchronous oscillator output, into the lock-in amplifier's current input. It needs to be chosen in a similar way to that described above for choosing an attenuator so that the injected current equals the offset level to be removed.

The controls are adjusted in a similar way as described for the first example, except that because the injected current is now in phase with the signal to be measured, the reference phase control needs to be adjusted to bring the signal and reference inputs to the in-phase PSD to be 180° out of phase. This causes the synchronous oscillator output, and hence the injected current, to also be 180° out of phase with the signal as is required for offset reduction.

The overall procedure to adopt is therefore:

a) Perform an auto-default operation to set the instrument to a defined state. This operation automatically sets the outputs to R-χ mode.

b) Set the unit to external reference mode and connect the reference signal to the selected input, either **REF IN** on the front panel or **TTL REF IN** on the rear panel.

c) Set the signal channel input to current mode and connect the signal input to the B/I input connector. At this stage there should be no connection to the input of the current injection resistor, R_L

d) Adjust the experiment so that the measured signal is at the level that it is required to be offset to zero, i.e. establish the "steady" state value from which changes will be measured.

e) Perform an auto-sensitivity function.

f) Perform an auto-phase function. The result of steps e) and f) will be to set the instrument so that the signal at the reference input to the X-channel PSD is in phase with that at the signal input.

g) Change the reference phase shift control by 180°. This is done most easily by pressing either the +/-90° key on the 7260 or 7265's Reference Menu, or the 90° key on the 7220 or 7225's front panel, twice.

h) Note the measured signal magnitude. In order to offset this signal, the injected current given by the synchronous oscillator amplitude and the injection resistor, R_L, needs to be set to exactly the same level. Since the synchronous oscillator output can only be adjusted between 1 mV and 5 V, it may be necessary to use a coaxial attenuator inserted in the BNC cable between the **OSC OUT** and injection resistor connectors in order to obtain the correct level.

   In addition, the resolution with which the amplitude can be adjusted is better at larger output amplitudes. For example, with no attenuator and if the measured signal were 800 mV then the oscillator amplitude control provides sufficient resolution, but if the signal level were 2 mV then it will not. However, if a ×1000 attenuator (60 dB) is fitted then the oscillator amplitude control will provide microvolt resolution, which is sufficient.

i) Set the oscillator amplitude control so that injected current given by the amplitude following the attenuator, if fitted, and the injection resistor, R_L, is the same as the measured signal.

j) Turn the synchronous oscillator output on and connect a cable, and the attenuator if used, between the **OSC OUT** connector and the injection resistor connector.

   This will cause the input to the signal channel to become equal to the input signal minus the synchronous oscillator output level. Since these two signals are nominally equal, the displayed magnitude will decrease.

k) Use the instrument's reference phase control to obtain a minimum output. Do not try to use the auto-phase function since it will not work correctly when using the instrument in this mode.

l) Make fine adjustments to the oscillator amplitude and reference phase controls to further reduce the output. It should also be possible to increase the full-scale sensitivity range.

m) The system is now set up to measure small changes in the applied signal. Because the offset level has been removed by the above procedure, the full input dynamic range is available for the measurement, giving the best possible accuracy.

Conclusion

The use of the Synchronous Oscillator output, believed to be unique to the ***SIGNAL RECOVERY*** models 7220, 7225, 7260 and 7265, for input offset reduction is invaluable for those experiments where small changes in large signals must be measured. Although it is a little more complex to use than the output offset controls, which are also provided on these instruments, it offers better performance because the full dynamic range of the instrument is available for the measurement.

SIGNAL RECOVERY

Using the Model 7225 and 7265 Lock-in Amplifiers with software written for the SR830

APPLICATION NOTE

AN 1002

Introduction

The majority of user-developed software programs written to operate the SR830 lock-in amplifier are easily modified to use the ***SIGNAL RECOVERY*** models 7225 and 7265 Digital Signal Processing (DSP) instruments instead. This Application Note describes the changes required by first considering how the behavior of the GPIB and RS232 interfaces differs between the instruments and then listing the most commonly used SR830 commands and their 7225/7265 equivalents.

The ***SIGNAL RECOVERY*** models 7225 and 7265 are both dual phase lock-in amplifiers and share the same command set, except that some of the command parameters have a wider range for the latter unit, to accommodate its increased operating frequency range. There are also a few commands which are exclusive to the 7265. In the rest of this Application Note the term 7265 can be taken to include the 7225, except where otherwise noted.

General

Perhaps the most important difference between the instruments is that in the SR830 the user needs to specify the interface to which the response to a command should be sent, using the **OUTX** command as the first command in a program. There is no equivalent command for the 7265 or 7225 since they always generate a response to the same interface port at which they received a command.

GPIB Interface

Both the SR830 and ***SIGNAL RECOVERY*** 7265 instruments support the IEEE-488.1 (1978) standard, but only the SR830 supports the common standards of the IEEE-488.2 (1987) standard. However this is rarely a problem since the commands associated with the latter are not particularly useful when controlling a specialized instrument such as a lock-in amplifier.

The 7265's GPIB address is set using the GPIB SETTINGS MENU in the range 0 to 31.

The serial poll status byte bit allocations differ between the instruments, as follows:

Bit	SR830 Serial Poll Status Byte	***SIGNAL RECOVERY*** 7265 Serial Poll Status Byte
bit 0	no scan in progress	command complete
bit 1	no command execution in progress	invalid command
bit 2	bit in error status byte has been set	command parameter error
bit 3	bit in LIA status byte has been set	reference unlock
bit 4	data available in output buffer	input or output overload
bit 5	bit in standard status byte has been set	new ADC values available after external trigger
bit 6	asserted SRQ	asserted SRQ
bit 7	not used	data available in output buffer

Hence the following bits are equivalent:-

SR830 Serial Poll Status Byte		***SIGNAL RECOVERY*** 7265 Serial Poll Status Byte
bit 1	no command execution in progress	bit 0 - command complete
bit 4	data available in output buffer	bit 7 - data available
bit 6	asserted SRQ	bit 7 - asserted SRQ

Information indicated by the assertion of the other four functional bits in the SR830's Status Byte can generally be obtained from the 7265's Status Byte (accessed via a serial poll or **ST** command and by using the commands **N** (Overload Byte) and **M** (Monitor Curve Acquisition).

Although both instruments will allow the user to simply send a command over the GPIB and then read the response, the recommended method with the 7265 is to use the serial poll status byte as part of a single write/read routine. This ensures that the response read back from the instrument is that generated by the command sent to it and in addition allows any error conditions which might be present to be immediately identified.

For example, with the 7265 this recommendation translates into the following Visual Basic code element, based on the use of a National Instruments PCI-GPIB interface card:-

```
' send the command
Call ibwrt(devAssign, sndcmd)          ' sends command to lock-in
Call ibrsp(devAssign, SPSB)            ' reads serial poll status byte
cmddone = SPSB And 1                   ' command complete = LSB of SPSB

cmdrsp = "No response to this command"  ' initializes command response string

' next section performs repeated serial polls in a loop, waiting for
' command done to clear, implying that command is being processed. But on slow
' computers and with write-only commands, command done might clear and be
' reasserted without the serial poll detecting it. Hence a timeout loop using
' k as a counter is included.
k = 0                                  ' k is used to implement timeout loop
While k < 1000 And cmddone = 1         ' start loop with timeout
Call ibrsp(devAssign, SPSB)            ' read SPSP
cmddone = SPSB And 1                   ' wait for command done to clear
k = k + 1                              ' increment timeout counter
Wend                                   ' loop

' command done should now be cleared, implying instrument is processing command
' next section performs repeated serial polls in a loop until command done
' is re-asserted. Each time the data available bit is also detected and if
' asserted a GPIB read is performed.
While cmddone = 0
Call ibrsp(devAssign, SPSB)            ' serial poll
cmddone = SPSB And 1                   ' command done = bit 0
dataavail = (SPSB And 128)             ' data available = bit 7
If dataavail = 128 Then                ' if data is available
cmdrsp = Space$(32)                    ' define buffer into which to read response
Call ibrd(devAssign, cmdrsp)           ' read response
cmdrsp = Left(cmdrsp, ibcnt)           ' ibcnt is a global variable returned by
                                       ' the NI GPIB handler software containing
                                       ' the number of bytes read. Use this to
                                       ' trim response to that number of bytes
End If
Wend                                   ' loop until command done is asserted
Invalidcmd = (SPSB And 2)
Paramerr = (SPSB And 4)
Refunlock = (SPSB And 8)
Overload = (SPSB And 16)
```

The 7265 can be set to accept and generate a Carriage Return character, Carriage Return/Line Feed character pair or a GPIB EIO as input and output terminators for GPIB communications. These three options include the two available on the SR830.

RS232 Interface

Both instruments are fitted with RS232 interfaces, although that on the SR830 is configured as a DCE and has a 25-pin connector, whilst the 7265 is a DTE and uses the more modern 9-pin connector. Consequently the user will need a different type of RS232 cable to couple the 7265 to his computer; the connection diagram for the most popular combination is given in Figure 1.

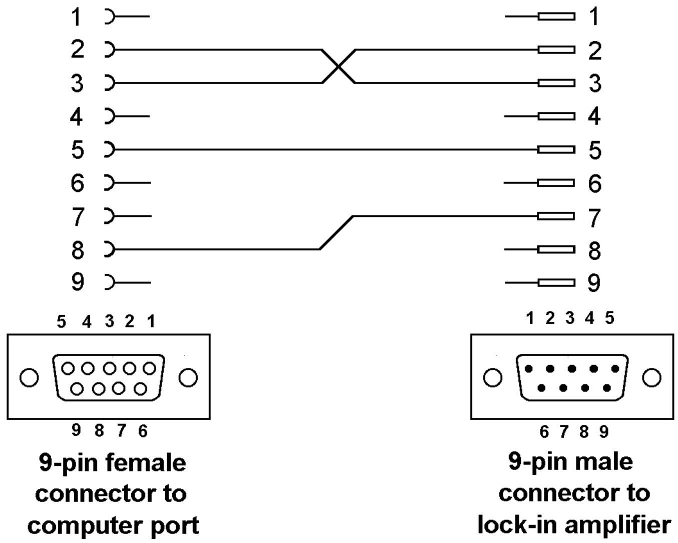


Figure 1, RS232 Cable to Connect 7265 to PC Computer

The other important difference between the instruments' RS232 interfaces is that the 7265 uses a character-by-character software handshake rather than CTS/DTR hardware or DRQ software handshaking. Hence the user's program must send each command one character at a time and wait until the instrument echoes it back before sending the next character.

For example, a Qbasic code segment to achieve this is:

```
'send the command cmd$ - which includes CR terminator - one character at a time.
'Note ; after print statement avoids QBasic sending extra CR or LF characters.
   FOR charcount = 1 TO LEN(cmd$)
   sentchar$ = MID$(cmd$, charcount, 1)
   PRINT #1, sentchar$;

   'check for each character being echoed by the lock-in
   DO
   WHILE LOC(1) = 0: WEND
   recdchar$ = INPUT$(1, #1)
   LOOP UNTIL recdchar$ = sentchar$
   NEXT charcount
```

Like the SR830, the 7265 accepts a Carriage Return or Carriage Return/Line Feed character pair as input terminator and normally generates a Carriage Return/Line Feed character pair as an output terminator. Unlike the SR830, the 7265 also normally sends a prompt character after the terminator, either an asterisk "*" or question mark "?". The former indicates that the instrument is functioning correctly, whilst the latter implies that an error condition is present. The user's program can then issue **ST** (status) and **N** (overload status) commands to determine the source of error.

Multiple Commands and Delimiters

Both instruments will accept multiple commands separated by semi-colons on a single line. Multiple responses are normally separated by a comma delimiter, although the 7265 allows the user to change this to any other printable ASCII character.

Communications Monitor

The 7265 includes a communications monitor display. Unlike the SR830 receive input buffer display, this shows not only the commands received from the computer, but also responses generated by the lock-in amplifier, making program debugging easier. The 7225 does not have such a display.

Command Set

In both instruments, GPIB and RS232 commands are identical as far as possible, and have the general format of a command mnemonic followed by one or more parameters.

In the SR830, the present setting of an instrument control is determined by adding a question mark "?" character after the command mnemonic and omitting any parameter(s), whereas in the 7265 the same effect is achieved by simply omitting the command parameter(s). The instrument then responds with the present value(s) of the requested command's parameter(s). Commands with multiple parameters require commas between parameters in the SR830, but not with the ***SIGNAL RECOVERY*** 7265.

The following table lists in alphabetical order the most commonly used SR830 commands and their 7265 equivalents, together with important notes about the parameter range. Note that [n] indicates that the parameter n is optional. Although the table makes translating programs easier, the reader may need to refer to the instruments' instruction manuals as well.

SR830 Command	***SIGNAL RECOVERY*** 7265 Command	Purpose	Notes
***CLS**	No equivalent command	Clear Status Registers	
***ERRS? [i]**	No equivalent command	Read Error Status Byte	
***ESR? [i]**	No equivalent command	Read Standard Event Status Byte	Information corresponding to bits 4 and 5 of standard event status byte can be obtained from the 7265's response to a serial poll or **ST** command
***IDN?**	**ID**	Read Instrument Identity	7265 returns only "7265", 7225 returns only "7225"
***LIAS? [i]**	No equivalent command	Read LIA Status Byte	Information corresponding to bits 0 to 3 of the LIA status byte can be obtained from 7265's response to a serial poll or **ST** command and /or the response to an **N** command
***RST**	**ADF 1**	Reset all instrument settings to default values	Communications interface settings are not changed
***STB? [i]**	**ST**	Read Serial Poll Status Byte	Note bit allocations differ between the SR830 and 7265 as described earlier in this document
AGAN	**ASEN**	Auto Sensitivity/Gain	**AGAN** is inoperative at time constants longer than 1 second
AOFF i	**AXO**	Auto-Offset	In the SR830, i selects output to be offset. In the 7265 both X and Y are auto-offset
APHS	**AQN**	Auto Phase	Operates once per call
ARSV	**AUTOMATIC 1**	Auto Reserve - AC Gain	**ARSV** operates once per call; Automatic AC Gain on (n = 1) adjusts AC Gain with FS sensitivity in the 7265
AUXV i, x	**DAC.** $\mathbf{n_1\ n_2}$	Set Auxiliary DAC Voltages	i = n_1 = 1 to 4 sets DAC 1 to DAC 4 outputs. x and n_2 are in volts, with 1 mV resolution
AUXV? i	**DAC.** $\mathbf{n_1}$	Read Auxiliary DAC Voltages	As above
FMOD i	**IE n**	Set Reference Channel Source	i = 1 and n = 0: internal i = 0: external SR830 n = 1: external TTL 7265 n = 2: external analog 7265

SR830 Command	***SIGNAL RECOVERY*** 7265 Command	Purpose	Notes
FMOD?	**IE**	Read Reference Channel Source	As above
FPOP 1, j	**CH 1 n**	Set CH1 Analog Output Type	j = 1, n = 0: X%, see manuals for other outputs
FPOP? 1	**CH 1**	Read CH1 Analog Output Type	As above
FPOP 2, j	**CH 2 n**	Set CH2 Analog Output Type	j = 1, n = 0: Y%, see manuals for other outputs
FPOP? 2	**CH 2**	Read CH2 Analog Output Type	As above
FREQ f	**OF. n**	Set Internal Reference (Oscillator) frequency	f and n are in hertz. Ranges are limited as follows:- SR830: 1 mHz to 102 kHz 7225: 1 mHz to 120 kHz 7265: 1 mHz to 250 kHz
FREQ?	**FRQ.**	Read Reference Frequency	As above. Note that in the 7265 the oscillator frequency is not necessarily the same as the reference frequency.
HARM i	**REFN n**	Set Reference Harmonic Control	i and n indicate reference harmonic number, with range limited as follows:- SR830: 1 to 19999 7225: 1 to 32 7265: 1 to 65535
HARM?	**REFN**	Read Reference Harmonic Control	As above
ICPL i	**CP n**	Set Input Coupling mode	i = n = 0 : AC i = n = 1: DC
ICPL?	**CP**	Read Input Coupling mode	As above
IGND i	**FLOAT n**	Set Input Connector Float/Ground Control	i = 0 and n = 1: Float i = 1 and n = 0: Ground
IGND?	**FLOAT**	Read Input Connector Float/Ground Control	As above
ILIN i	**LF n_1 n_2**	Set Line Frequency notch filter	i n_1 Function 0 0 No filters 1 1 F 2 2 2F 3 3 F & 2F n_2 sets the filter frequency in the 7225/7265 n_2 = 0: 60 Hz, n_2 = 1: 50 Hz
ILIN?	**LF**	Read Line Frequency notch filter	As above
ISRC 0	**IMODE 0; VMODE 1**	Set Input Mode to A input, voltage mode	
ISRC 1	**IMODE 0; VMODE 3**	Set Input Mode to A-B input, voltage mode	
ISRC 2	**IMODE 1**	Set Input Mode to current - high bandwidth	
ISRC 3	**IMODE 2**	Set Input Mode to current - low noise	
ISRC?	**IMODE; VMODE**	Read Input Mode	As above

SR830 Command	***SIGNAL RECOVERY*** 7265 Command	Purpose	Notes
LOCL i	**REMOTE n**	Set Local/Remote status	i = n = 0: Front Panel Active i = n = 1: Remote Operation Only
LOCL?	**REMOTE**	Read Local/Remote status	As above
OAUX? i	**ADC. n**	Read Auxiliary ADC Voltages	i = 1 to 4, n = 1 or 2: Response is in volts, with 1 mV resolution.
OEXP 1 [, j, k]	**XOF 1 n_2; EX n_3**	Set X Output Offset and Expand	j = n_3 =0: No expansion j = n_3 = 1: Δ10 Expansion k = n_2/100 = Percentage offset
OEXP? 1	**XOF 1; EX**	Read X Output Offset and Expand	As above
OEXP 2 [, j, k]	**YOF 1 n_2; EX n_3**	Set Y Output Offset and Expand	j = n_3 =0: No expansion j = n_3 = 1: Δ10 Expansion k = n_2/100 = Percentage offset
OEXP? 2	**YOF 1; EX**	Read Y Output Offset and Expand	As above
OFLT i	**TC n**	Set Output Filters Time Constant	The available SR830 time constants are in a 1-3-10 sequence from 10 µs (i = 0)to 30 ks (i = 19), whilst the ***SIGNAL RECOVERY*** 7265 range from 10 µs (n = 0) to 100 ks in a 1-2-5-10 sequence.
OFLT?	**TC**	Read Output Filters Time Constant	As above
OFSL i	**SLOPE n**	Set Output Filters Slope	n or i — Selection 0 — 6 dB/octave 1 — 12 dB/octave 2 — 18 dB/octave 3 — 24 dB/octave
OFSL?	**SLOPE**	Read Output Filters Slope	As above
OUTP? 1	**X.**	Read X Output Value	Response is in volts
OUTP? 2	**Y.**	Read Y Output Value	Response is in volts
OUTP? 3	**MAG.**	Read Magnitude (R) Value	Response is in volts
OUTP? 4	**PHA.**	Read Signal Phase (χ) Value	Response is in degrees
OUTX [?] [i]	No equivalent command	Select Output Interface	The 7265 always responds to the port at which it received a command.
OVRM [?] i	No equivalent command	Set/Read Front Panel Lockout	Use 7265's **REMOTE** command
PHAS x	**REFP. n**	Set Reference Phase	x and n are in degrees
PHAS?	**REFP.**	Read Reference Phase	Response is in degrees
PAUS	**HC**	Pause data acquisition to buffer	
REST	**NC**	Reset Data Buffers	Commands erase data buffers
RMOD i	**ACGAIN n**	Set Dynamic Reserve or AC GAIN	i is limited to 0, 1 or 2. n ranges from 0 to 9
RMOD?	**ACGAIN**	Read Dynamic Reserve or AC GAIN	As above
RSET i	No equivalent command	Recall setup from buffer i	Use series of instrument commands with required settings instead

SR830 Command	SIGNAL RECOVERY 7265 Command	Purpose	Notes
RSLP[?]	No equivalent command	Set/Read External Reference Trigger Threshold	Use 7265 **IE n** and appropriate reference connector
SEND 0;STRT	**TD**	Start data acquisition to buffer - one-shot	In the SR830, acquisition may not start until trigger is received
SEND 1;STRT	**TDC**	Start data acquisition to buffer - loop mode	In the SR830, acquisition may not start until trigger is received
SENS i	**SEN n**	Set full-scale sensitivity	When the 7265 and SR830 are in voltage input mode or wide bandwidth current mode (**IMODE 1**), then n = i +1. Hence for a full scale sensitivity of 100 nV, i = 5 and n = 6. When the 7256 is in low noise current mode (**IMODE 2**), then n = i + 6. Hence for full-scale sensitivity of 10 nA, i = 21 and n = 27
SENS?	**SEN**	Read full-scale sensitivity	As above
SLVL x	**OA. n**	Set Oscillator Amplitude	x and n are in volts rms, with range limited as follows:- SR830: 0.004 to 5.000 7225: 0.001 to 5.000 7265: 0.000001 to 5.000 (accuracy is not specified for <100 µV
SLVL?	**OA.**	Read Oscillator Amplitude	Response is in volts rms
SNAP i [,j,k,l,m,n]	**STAR n** followed by ***** commands	Read Output(s) sampled at same time	Use **STAR n** to select 7265 outputs equivalent to those specified by parameters i to n to be read and then use ***** commands to read data
SPTS?	**M**	Read Number of points in buffer	The **M** command returns four values; the number of points stored is the fourth response
SRAT i	**STR n**	Set Curve Buffer Storage Rate Control	i sets time per point according to table in range 1.9 ms to 16 s per point. n sets time per point in milliseconds in range 1.25 ms (n = 0) to 1E6.
SRAT?	**STR**	Read Curve Buffer Storage Rate Control	As above
SSET i	No equivalent command	Save setup to buffer i	
STRT	**TD**	Start data acquisition to buffer	
SYNC i	**SYNC n**	Set Synchronous Time Constant	n = i = 0: Sync filter off n = i = 1: Sync filter on
SYNC?	**SYNC**	Read Synchronous Time Constant	As above
TRIG	No equivalent command	Software Trigger	
TRCA? i, j ,k	**DC. n**	Read points stored in buffer i/n	Response is in ASCII floating-point numbers
TSTR 1; STRT	**TDT 0**	Start curve acquisition on-trigger	

Example - Basic Signal Recovery

This section shows how a typical program coded to operated the SR830 needs to be modified to operate the 7265. The program, which is listed simply as the required sequence of commands, excluding details of the write/read subroutine to send them, sets the lock-in amplifier's controls and then records the chosen outputs, perhaps as a function of time.

Assuming the output sampling rate is less than a few points per second then there is no need to use the internal curve buffers. The commands to achieve this would therefore be similar to the following sequence:

SR830	***SIGNAL RECOVERY* 7265**	**Notes**
OUTX 1	No equivalent command	Set GPIB as the interface to which responses are sent
FMOD 0	IE 2	Set reference source to external front panel input
ISRC 0	IMODE 0; VMODE 1	Single-ended voltage input mode
IGND 0	FLOAT 1	Float input connector shell with respect to chassis ground
ILIN 0	LF 0 0	Turn off line frequency rejection filter
OFLT 8	TC 11	Set time constant to 100 ms
AGAN	AS	Auto-Sensitivity/Gain
APHS	AQN	Auto Phase

Then the outputs could be read as follows:

OUTP? 1	X.	Reads X output in volts
OUTP? 2	Y.	Reads Y output in volts
OUTP? 3	MAG.	Reads Magnitude in volts
OUTP? 4	PHA.	Reads Phase in degrees
FREQ?	FRQ.	Reads reference frequency in hertz

The controlling program would send a new output command each time a new reading were required. Note that at an output filter slope of 12 dB/octave a good “rule of thumb” is to wait for a period of five time-constants after the input signal has changed before recording a new value. Hence when recording the measured signal as an experimental parameter is changed, the program should issue the commands to whatever equipment causes this change, wait for five time-constants, and then record the required output.

Conclusions

In the majority of cases, programs written to control the SR830 lock-in amplifier use only a small number of the available instrument commands. In such cases, modifying the program to operate the ***SIGNAL RECOVERY*** 7225 or 7265 instruments is not nearly as large a task as might at first be thought.

Changing to the ***SIGNAL RECOVERY*** units allows the user to take advantage, maybe at a later date, of the richer feature set of these instruments, including such items as the extended frequency range, dual reference and harmonic modes, the transient recorder facility and the more powerful output data curve buffer.

SIGNAL RECOVERY

Low Level Optical Detection using Lock-in Amplifier Techniques

APPLICATION NOTE
AN 1003

Introduction

This note describes a number of techniques that can be used to detect low level optical signals. It starts by considering the problems inherent in the use of DC techniques and how these may be reduced by using AC methods instead. It then discusses a range of different experimental approaches using lock-in amplifiers, pointing out the advantages as well as any disadvantages of each method. Finally, it outlines the important specifications of the mechanical light choppers that are often used as part of such systems.

The note is written primarily for scientists, students and laboratory personnel who have little or no experience with low level light measurements. It is assumed however that readers have some basic knowledge of lock-in amplifiers, but if this is not the case then they may refer to the ***SIGNAL RECOVERY*** Technical Notes TN1000 "What is a lock-in amplifier?" and TN1001 "Specifying Lock-in amplifiers". Further references are given at the end of this note.

The DC Approach

In the simplest form of light measurement, a suitable current meter measures the DC current generated in an optical detector as a result of the incidence of a steady state light source. Such a detection system has its use in applications such as camera light meters, sensors for switching on or off outdoor lighting fixtures or other applications where high levels of light are detected or measured. At much lower light levels, errors will begin to appear as the measurement becomes more susceptible to random events and noise from various sources. Figure 1 illustrates a DC technique for measuring low light levels in a typical experiment. The output from the detector is taken to a current to voltage converter, implemented using a low-drift operational amplifier. The voltage output from the amplifier is then measured using a conventional voltmeter (not shown). An offset control is used to compensate for the detector's DC leakage current.

This approach has some merit in that it can be used in situations where the "photodiode and current meter"

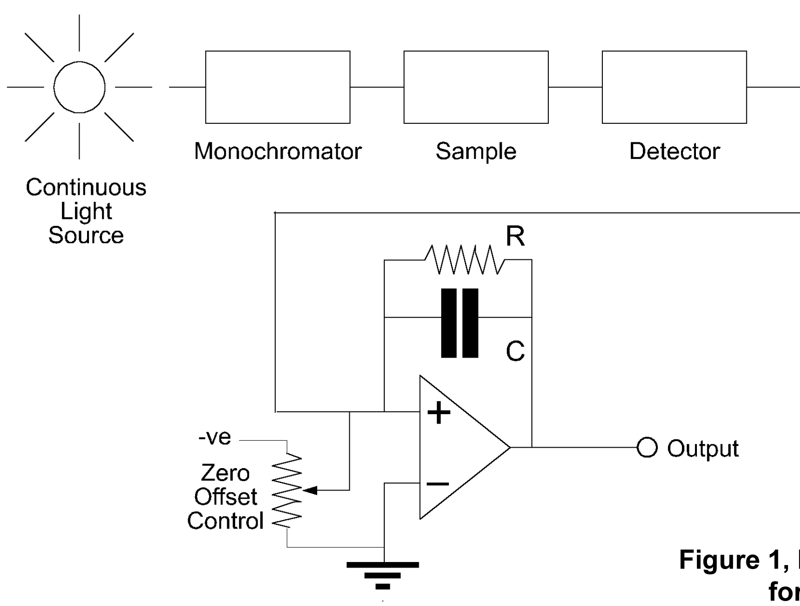


Figure 1, DC Measurement System for Low Light Levels

approach doesn't provide adequate results. Although low-cost and simple, it has a number of disadvantages which limit its usefulness. If the offset control is not adjusted correctly, then the DC signal from the detector due to leakage current will give a non-zero output even with no optical input. Another problem is that there is no way of separating the output signal caused by the wanted input signal from that due to stray light entering the detector. Discerning the signal of interest from these sources of error can be a real challenge. Even when the offset control is correctly adjusted, subsequent readings will still be subject to drift as temperature changes affect the leakage current.

The Modulated Light Approach

The most widely used method of measuring a low level optical signal is to apply a modulation to the light source and then recover the signal at the modulation frequency. The modulation can be of any periodic form, but sinusoidal and square waves are most commonly used. It is generated either by direct application of the output of signal generator to a light source, such as a laser diode, or by using some form of light chopper to periodically interrupt a continuous light source. In either case, an AC signal is generated at the detector output which allows the experimenter to use any one of a variety of AC measurement techniques and, as a result, greatly reduces some of the problems which plague the DC method.

In figure 2, the optical signal generated by a laser is modulated at the frequency output by the signal generator. The output of the detector is therefore now the unwanted DC signal caused by the thermal leakage and an AC signal at the same frequency as the modulation. The signal then passes to a tuned amplifier, which consists of a signal filter and amplifier stage. The filter is set to a bandpass mode, which limits the bandwidth of the measuring system to those frequencies close to the modulation frequency, and its output is then measured using an AC voltmeter.

Although still relatively inexpensive, the lower limit of light detection using this method represents a significant improvement when compared to the DC system. With careful choice of modulation frequency, the lower detection limit may increase by more than an order of magnitude over the DC method. The second advantage is that some stray light can fall on the detector and not influence the voltmeter reading. Still, there are some shortcomings in this method. The minimum signal that can be detected is primarily determined by the selectivity of the detection system, which in this case is set by the Q-factor of the filter. For example, with a band pass filter of Q equal to ten and a modulation frequency of 1 kHz, the signal bandwidth would be 100 Hz. Thus, noise components 50 Hz on both sides of the center frequency of 1 kHz could still make a relatively large contribution to the output. If these same noise voltages were large enough, an error in the measurement would occur since the AC voltmeter would measure not only the 1 kHz modulation signal, but the noise as well. One possible solution is to further limit the bandwidth by increasing the Q of the filter. However, there is a practical limit to the ultimate selectivity of a tuned amplifier. At Q's of 100 for example, it becomes difficult to implement analog filters of sufficient frequency stability, possibly resulting in the pass band of the tuned amplifier shifting away from the wanted frequency. Once this happens, the output signal-to-noise ratio will degrade, requiring the experimenter to retune the filter frequency.

Another problem is that tuned amplifiers are not the best instruments to use in the "front end" of an experiment. Many filters are not optimized for the best noise performance and a low noise preamp ahead of a tuned amplifier is almost always recommended.

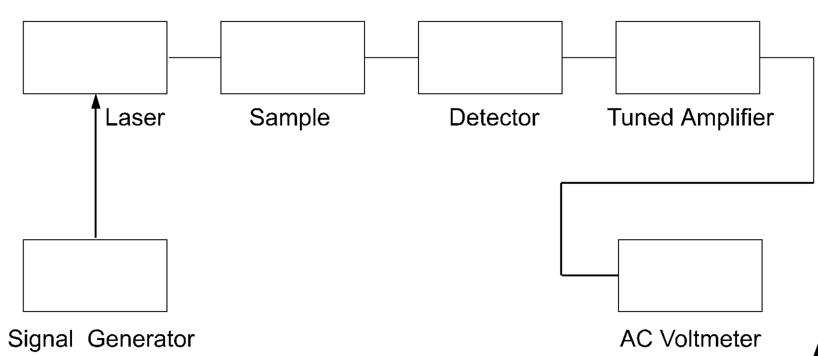


Figure 2,
AC Measurement System using Tuned Amplifier

However, for applications not requiring the ultimate in signal recovery performance, the AC filter method may still be preferred over more sophisticated techniques. ***SIGNAL RECOVERY*** make the model 7310 noise-rejecting voltmeter which is an instrument ideally suited to such use, consisting of a tunable band-pass filter followed by precision AC voltmeter all in one box. Since the filter is implemented using digital circuitry, it does not suffer from the frequency drift problems of analog designs, whilst the front-end amplifier stage is of the same high quality as used in ***SIGNAL RECOVERY*** lock-in amplifiers.

The Lock-In Amplifier Method

A much better approach to the AC detection method is to use a lock-in amplifier to measure the AC signal from the detector. Like the tuned amplifier approach previously outlined, the lock-in amplifier uses a frequency-selective technique. However, when using a lock-in, a much smaller bandwidth can be easily achieved without the inherent frequency-drift problems associated with the tuned amplifier. One can think of a lock-in amplifier as a specialized AC voltmeter, which measures only the amplitude of signals at a frequency equal to the applied reference frequency.[1]

Once set up in the experiment, the lock-in amplifier will display the measured input on a panel meter or make it available over a computer interface. Furthermore, it will provide a DC output voltage which is proportional to the AC voltage appearing at its input, which can be used for such purposes as driving a strip chart recorder or serve as the input to another instrument.

Figure 3 illustrates a basic optical detection setup using a mechanical light chopper and a lock-in amplifier. The light chopper consists of a motor, speed control mechanism, and a rotating blade or chopper wheel. In some cases, all three of these components are in one assembly. In other choppers, the control unit may be in a separate housing. The chopper wheel is a rotating metal disk which contains one or more sets of equally spaced apertures which allow the light source to pass through or be blocked altogether. The number of apertures and the wheel rotation speed determine the chopping frequency. Since the rotation of the blade causes the optical signal path to be interrupted, the light source that stimulates the experiment is in the form of an AC excitation. One could visualize this excitation as an optical equivalent of a square wave, although this is only true if the aperture size is large compared to the beam diameter. The signal appearing at the detector output may or may not be a good representation of the optical stimulation since factors such as detector response time and cable capacitance must be considered.

In addition to modulating the light source, the chopper also provides a synchronous reference signal capable of driving the reference channel of a lock-in amplifier. This reference output voltage is a square wave, usually in the order of a few volts peak to peak.

The optical signal stimulating the experiment and thus falling on the optical detector generates an electrical current which can be measured by the lock-in. Any

1 The lock-in amplifier will respond to spectral noise voltages very close to the reference frequency as well. Such noise voltages at the input will appear as random fluctuations on the lock-in's output.

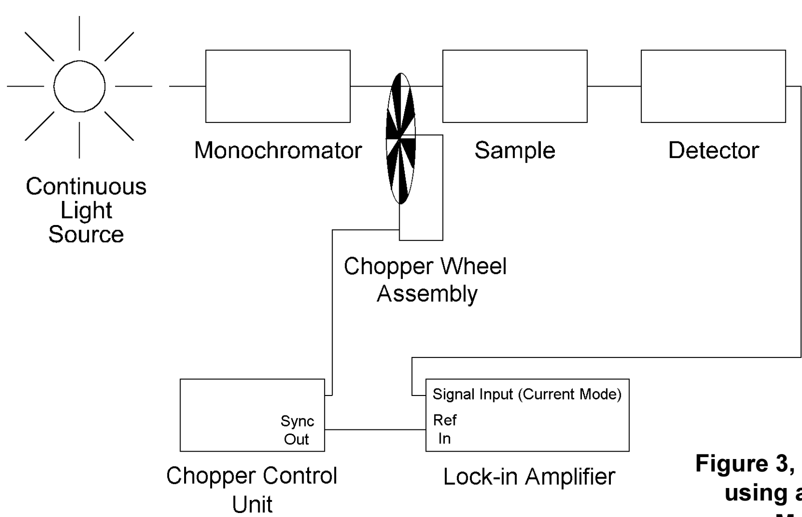


Figure 3, AC Measurement System using a Lock-in Amplifier and Mechanical Chopper

discrete frequencies or noise voltages not equal to the reference frequency will be rejected by the lock-in amplifier. The end result is a much lower limit on signals which can be measured. In fact, it's possible that the signal of interest may be completely obscured by noise if one were to view the detector output with an oscilloscope. Again, stray light falling on the detector is usually not a problem as long the magnitude is insufficient to saturate the detector. However, the user still needs to insure that stray light does not enter into the experiment via the chopped light path.

Source Compensation - Ratiometric Spectroscopy

Although the use of a lock-in amplifier dramatically enhances the ability to measure a signal buried in noise, there can be sources of measurement errors other than noise and background voltages. In optical measurements, an often troublesome source of error can be attributed to variations in light source intensity, since the output of many light sources vary over time. Moreover, the efficiency of a scanning monochromator may vary as a function of wavelength. If, in the previous example, the output of the lock-in were to change, the experimenter would not be able to discern between changes in the optical properties of the sample or source variations. This is a common problem that can only be addressed by introducing a second detection path which measures the optical output of the excitation source and by using a ratiometric technique to normalize for source fluctuations.

In figure 4, the optical output from the monochromator is split off and sent to a separate detector and preamplifier (not shown). This generates a DC voltage, the magnitude of which is determined by the intensity of the source as well as the relative efficiency of the monochromator. This optical path is usually referred to as the "normalizing signal" or "optical normalizing path".

It was mentioned earlier that the lock-in generates a DC voltage at its output as part of the detection process. In this configuration, a second DC voltage is now available which represents only the optical signal from the monochromator. By calculating the ratio of the DC output of the lock-in amplifier to the DC voltage generated as a result of the normalizing beam, a third DC voltage is generated which is proportional to only those changes due to properties in the sample path. The block labeled "Ratiometer" may be an analog circuit, or more likely a digital system that calculates the ratio of the two DC voltages and provides an output in some digital form. The neutral density filter is used to adjust the level of the normalizing beam for the appropriate nominal input voltage to the "B" input of the ratiometer. When using most lock-in amplifiers manufactured since the late 1980s, a separate ratiometer is usually not necessary. Such instruments have built-in auxiliary analog to digital converters (ADCs) whose inputs are accessible on the rear panel of the instrument. For example, one could apply a DC voltage from the preamp output to one of the rear panel ADCs (typically **ADC1**), then invoke the lock-in amplifier's ratio mode. The lock-in front panel will then display the ratio of the measured "X" value to the DC level applied to the rear panel ADC1 input.

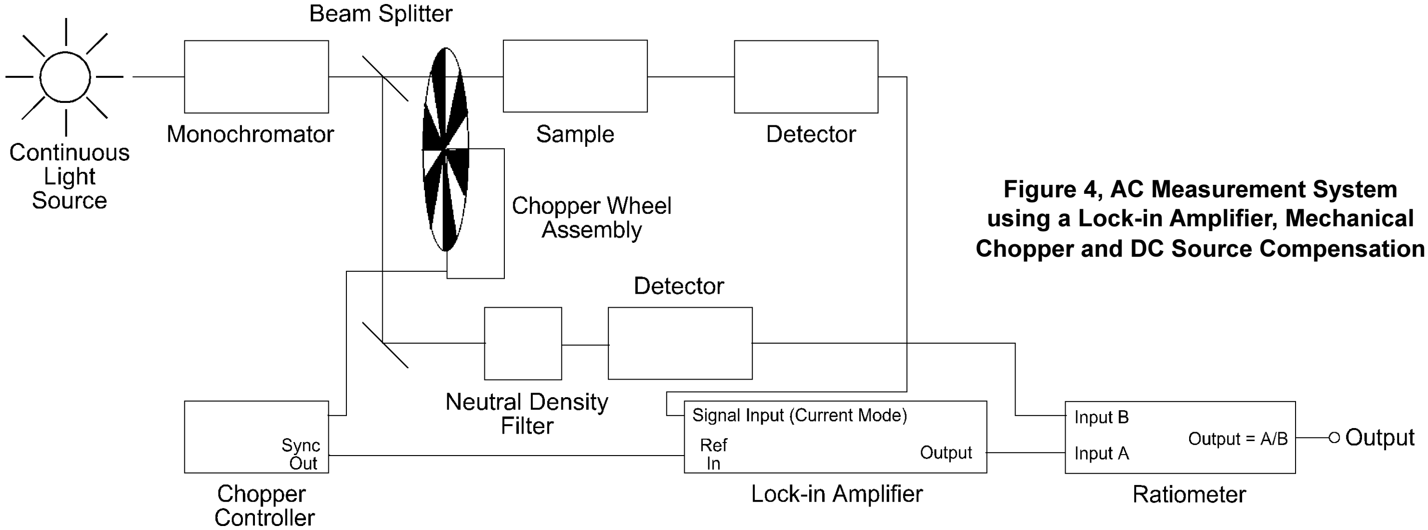


Figure 4, AC Measurement System using a Lock-in Amplifier, Mechanical Chopper and DC Source Compensation

Again referring to figure 4, note that it is essential not to allow stray light to fall on the detector in the normalizing path. Also, there may still be a source of error due to any mismatch in the performance of the two photodetectors.

Source Compensation Using Two Lock-Ins

An improved version of the ratiometer approach is shown in figure 5. Basically, the difference is that both the normalizing and signal beams are chopped, and the two beams are recombined into a single detector. This eliminates any error due to mismatching of optical detectors. Although one could use two separate light choppers, a more practical and economical approach is to use a dual beam chopper such as the ***SIGNAL RECOVERY*** Model 651-1 which has the capability of chopping two light beams simultaneously. Since the Model 651-1 uses a dual aperture blade, two reference signals are available; one for the outer set of apertures, the other for the inner set of apertures.

As shown in Figure 5, a second lock-in amplifier, #2, is used to detect the normalizing beam. The second lock-in's output is used for the denominator of the ratio calculation. Since the magnitude of the signal in the normalizing beam is usually quite large, a low cost instrument will almost always suffice for this path. The output of the lock-in used in the normalizing path is fed into the rear panel A-D converter 1 of the signal path lock-in, #1, which is configured for the ratio mode.

Another approach, which is perfectly acceptable, is to take output readings from both lock-ins into a computer, and have it perform the ratio calculation.

This system generally provides the best solution to a low level optical experiment.

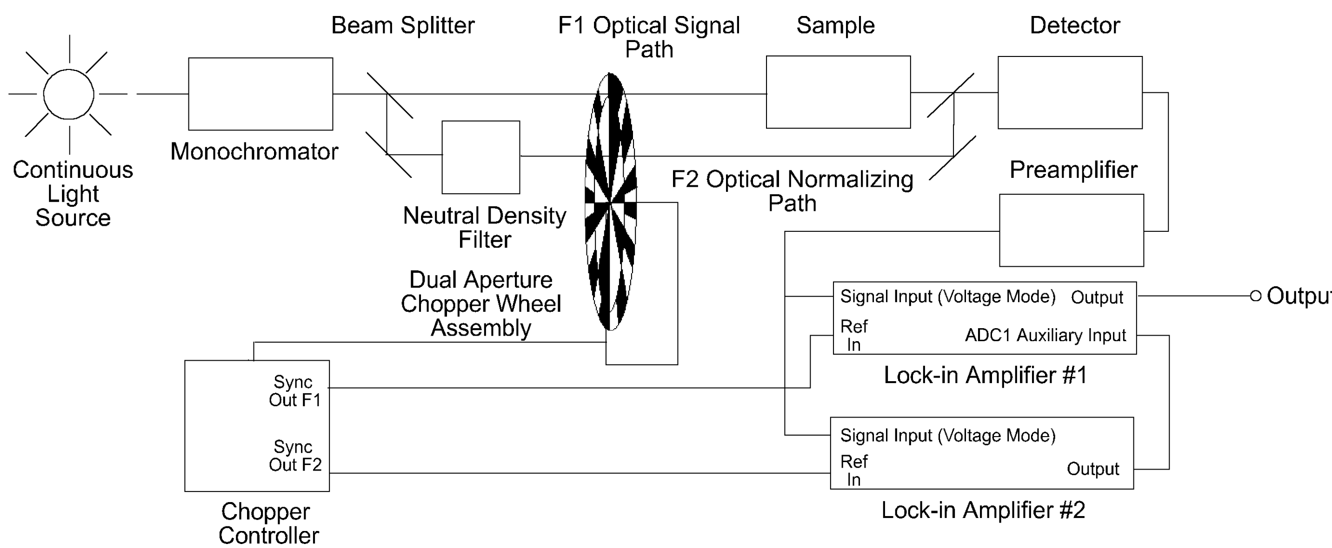


Figure 5, AC Measurement System using two Lock-in Amplifiers, Dual-Beam Mechanical Chopper and AC Source Compensation

Dual Reference Lock-In Amplifiers

Since cost is always a consideration when setting up a new experiment, one should consider the purchase of a lock-in amplifier which has a dual reference capability. Such a lock-in can detect two signals simultaneously, and as a result, only one lock-in is needed for a ratiometric experiment. The ***SIGNAL RECOVERY*** Models 7260, 7265, and 7280 all have dual reference capabilities.

Figure 6 illustrates how the single ***SIGNAL RECOVERY*** Model 7265 can be used in a ratio experiment. As in the case of the two lock-in approach, the output from the monochromator is split off and the Model 651-1 Optical Chopper is used to chop the light source at the two chopping frequencies. The chopping frequency, F, is controlled by the Model 7265s internal oscillator, which means that the outer set of apertures is synchronized with the internal oscillator of the Model 7265. The chopper also generates a reference signal, this time at F/10, synchronous with the inner set of apertures. This signal is fed back to the reference input connector of the Model 7265. The Model 7265 now has the two required reference signals, one at frequency F corresponding to the signal channel path and one at F/10 relating to the optical normalizing path.

The Model 7265 can now measure and display the amplitude of both signals appearing at its input connector. In the dual reference mode, two complete sets of output signals are available; X_1, Y_1, X_2, Y_2, MAG_1, MAG_2, etc. Any of these outputs may be displayed on the front panel. In a ratio experiment, the user may prefer to perform the ratio function using the 7265's firmware simply by taking advantage of its "user equations" menu. Once the user equation is setup, the result of the ratio calculation can be displayed on the front panel, accessed by a host computer, or the user can specify that a DC voltage proportional to the ratio be available on a rear panel BNC connector.

It is important to note that when using the dual reference mode, there may be other features of the lock-in used which may not be available. For example, in the Model 7265, the dual reference capability is limited to frequencies below 20 kHz. Moreover, one of the reference frequencies must be derived from the internal oscillator of the 7265 (a requirement satisfied by a chopper which can be externally synchronized). Also when using the dual reference mode, output time constants less than 5 ms are not available. These restrictions usually have little or no impact on a chopped light experiment, but the user should still be aware of any performance differences when operating in dual reference mode.

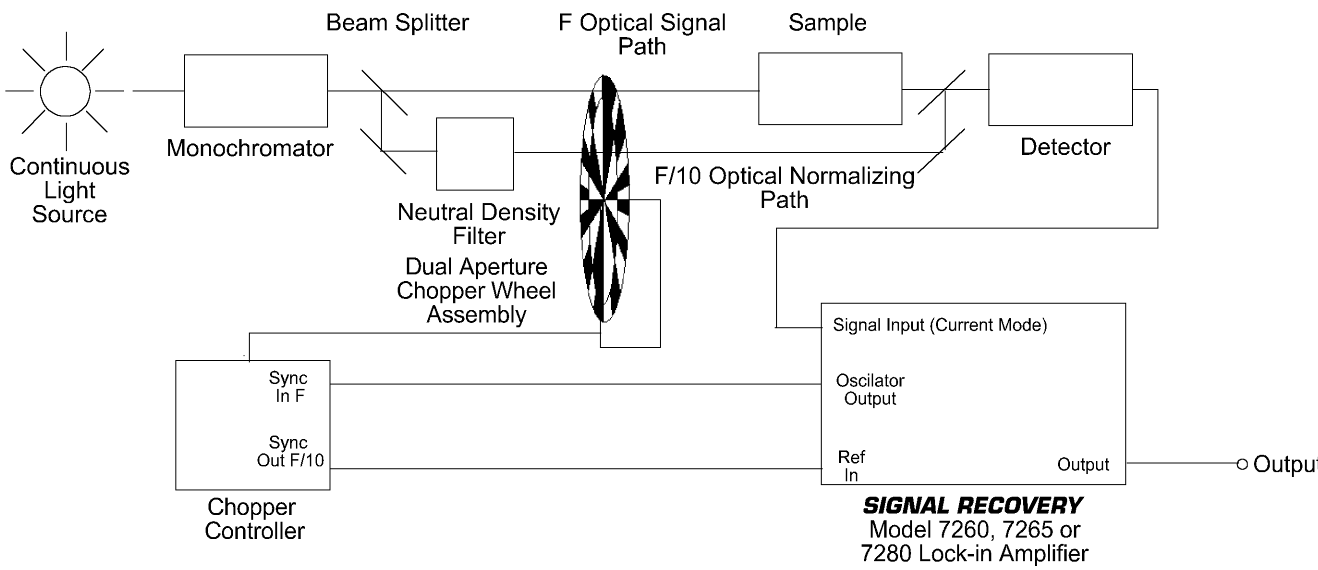


Figure 6, AC Measurement System using Dual-Reference Lock-in Amplifier, Dual-Beam Mechanical Chopper and AC Source Compensation

Mechanical Light Choppers

The experimenter is often faced with the task of determining which light chopper is best suited for a particular experiment. Part of the problem in selecting a chopper is interpreting what each specification means and how it will impact a particular experiment. In this section, some of the more pertinent specifications will be explained and defined.

Frequency Range

The chopping frequency is variable to allow the user the ability to both select a frequency which is optimum for the detector as well as avoid troublesome frequencies.

It's usually a good idea to chop at frequencies above the 1/f noise level (typically 100 Hz) unless there is a more important criterion calling for a lower chopping frequency. In addition, a chopping frequency near the power line frequency or any harmonic of it should be avoided. The ***SIGNAL RECOVERY*** Models 7265 and 7280 both have a built in FFT display which can aid in selecting a chopping frequency. For choppers using a dual aperture blade, two sets of frequency ranges may be specified since at any given wheel rotation speed, one has the ability to chop at two different frequencies.

External Frequency Control

In addition to a frequency control on the chopper itself, the frequency of most choppers can be controlled externally. In some choppers this is done by the application of a DC voltage. Other choppers, including all the ***SIGNAL RECOVERY*** models, are controlled by applying an external reference frequency signal. If the dual reference capability of ***SIGNAL RECOVERY*** lock-ins is to be taken advantage of, it is essential that a chopper of the latter type be used.

Most modern lock-in amplifiers incorporate an internal oscillator, the output of which can be connected to the chopper's reference frequency input. If this is done, then changing the oscillator frequency also changes the frequency of the modulation generated by the chopper. If the lock-in is operated under computer control then the oscillator frequency can be set by the program, allowing for example, with suitable software, automatic selection of an operating frequency where any interfering signals are smallest. Another use of this technique is to prolong the chopper's motor life by reducing its speed whenever measurements are not being taken.

Jitter

Jitter is the short term variation in the period of one chopping cycle to the next. Its effect is to add noise to a measurement. The source of jitter is twofold; one is the mechanical imperfections in the chopper blade, the other is from the speed control electronics and motor combination.

Figure 7 illustrates graphically how jitter manifests itself. Jitter can be expressed in either degree rms values or peak-to-peak units as a percent.

A difficulty may arise when comparing two chopper jitter specifications where the two different values are specified. For example, one chopper might be specified to exhibit 0.5% peak-to-peak jitter. This is to be compared to another manufacturer who might publish a jitter specification of 0.5 degree rms under similar operating conditions.

Naturally, a comparison must be made in the same mathematical units. The first step is to convert the peak-to-peak percent specification to degree rms units. The 0.5% peak-to-peak specification refers to the percentage of a complete wheel rotation or 360 degrees. In this case, the peak-to-peak jitter is (.005 X 360) or 1.8 degrees. The calculated 1.8 degrees is still in peak-to-peak units, so it is necessary to convert to rms values. Peak-to-peak values are 2.8 times larger than rms values. In this case, it is necessary to divide the calculated 1.8 degrees by 2.8 in order to arrive at a rms value. In this case, a conversion to rms will yield 1.8/2.8 = 0.64 degrees rms. In this case, the two choppers have a very similar jitter specification.

Although jitter specifications are almost always specified by chopper manufacturers, the effect of blade jitter is usually too small to have any significant impact except in those cases where extremely small signals are to be measured.

Other factors to consider when choosing a chopper are mechanical configuration, beam size to be chopped, and how one wishes to externally control the speed.

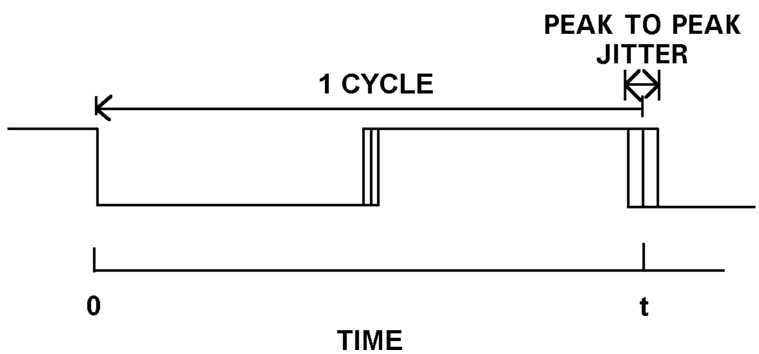


Figure 7, Definition of Chopper Jitter

Further Information

This application note is an introduction to the techniques used in low level light measurements. Additional information may be found in other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative, or by download from our website at **www.signalrecovery.com**

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AN 1004 Multiplexed Measurements using the 7225, 7265 and 7280 Lock-in Amplifiers
AN 1005 Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper

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Multiplexed Measurements using the 7225, 7265 and 7280 DSP Lock-in Amplifiers

APPLICATION NOTE

AN 1004

Introduction

There are many experiments in which the researcher would like to be able to use lock-in amplifier detection techniques to measure more than one signal. This application note describes the options available to the users of **SIGNAL RECOVERY** products, from the simplest case of two signals through to an example of a system requiring ten measurement channels.

Simultaneous Measurements

When two or more signals are to be measured simultaneously then in general the same number of lock-in amplifiers are required. This may be achieved either through the use of multiple instruments, or, if all signals are at the same reference frequency, using a multi-channel instrument.

However, if there are only two signals that are at different frequencies and of comparable magnitude then the **SIGNAL RECOVERY** models 7265 or 7280 lock-in amplifiers may well be suitable. This is because these instruments include both Dual Harmonic and Dual Reference modes of operation, allowing independent measurement of each signal. Further information about an experiment that takes advantage of this capability to use just one instrument where previously two would have been required is given in Application Note AN1000.

Sequential Measurements

In many cases is not necessary to measure the required signals simultaneously. An example of this is in the study of photosynthesis in leaf tissue, where several detectors are monitoring the absorption of light at different points on the leaf during an experiment lasting one day. All that is required is that each detector be connected in turn to the lock-in amplifier's input, the output be allowed to settle and the value recorded. Clearly this could be done manually but this is often inconvenient and is prone to error. The solution is to automate the system and use some form of input multiplexer to connect the required detector to the instrument's input.

If there are just two such signals and they are single-ended voltages then the **SIGNAL RECOVERY** model 7265 or7280 lock-in amplifiers can perform the required switching without the need for any further equipment. One signal is connected to the **A** and the second to the **B** channel inputs and the selection of which one is used can then be made via the front-panel, but more usefully in a computerized system, by computer command. Figure 1 shows how this can be done with the model 7265. The only correction which might need to be applied is to allow for the signal inversion of the **B** input channel.

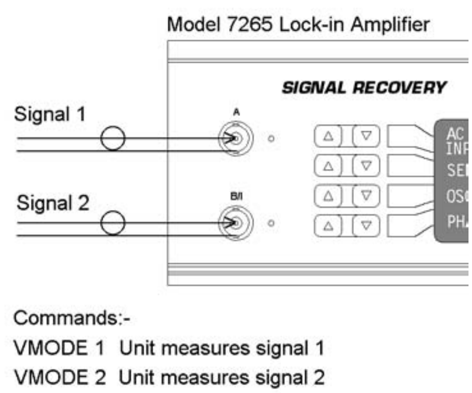

Figure 1, Two-Channel Input Multiplexing with the model 7265

When three or more signals need to be measured then an external multiplexer is required. Such devices are available from a number of sources, but the rest of this application note describes a ten-channel system built using a standard model 7265 lock-in and two model 7200 ten-channel multiplexers. These were designed and manufactured as special items by **SIGNAL RECOVERY** for this particular experiment.

10-Channel System for Measuring Critical Current Density in Superconductors

In the manufacture of superconductive probes from coated wafers the measurement of the critical temperature T_c and the critical current density J_c above which superconductivity ceases is important for the following processing stages. T_c can be determined by making contact with the wafer at its periphery, but the traditional measurement method for J_c implies forming a narrow bridge structure on the wafer. The technique is well established but for testing larger areas or higher quantities a non-destructive approach is preferable.

Claasen[1] et al (1991) proposed such a technique which generates an eddy current in the sample by inductive coupling and uses a pick-up coil to measure the resulting field. When the sample current density is below the critical level the system remains linear and the signal at the pick-up coil is at the same frequency as that applied, but as soon as the critical level is reached the resulting non-linearity causes a signal at the third harmonic to be generated. By detecting when a signal at this frequency occurs as the applied signal is increased it is possible to determine J_c

The ten-channel system, built by Zaitsev[2] et al (1999), uses this same technique but extended to ten measurement channels to measure the characteristic properties of YBCO films, intended for use as passive microwave devices for which a critical current density of 3 MA/cm^2 is required.

The experimental system is shown diagrammatically in Figure 2.

The 7265 is operated in internal reference mode at a frequency of 1.1 kHz, but set to the 3*f* detection mode so that it will measure signals at 3.3 kHz. The oscillator output signal, **OSC OUT**, is taken to an audio power amplifier which is operated at low output power to minimize distortion, and its output is fed to the **Common** input of the first of two ten-way multiplexers. The multiplexer connects the amplified oscillator signal to the driving side of one of the ten measurement coils, of which only two are shown in the diagram for the sake of simplicity.

Each coil consist of two concentric windings of 50 µm diameter wire, the inner one of 1100 turns being used as the driving element and the other one of 170 turns for the pick-up. The power amplifier is adjusted so that the driving current lies in the range 1 to 120 mA.

The overall diameter of each coil is about 6 mm and ten of them are distributed evenly over the surface of the 3" diameter wafer. The wafer and the coils are of course mounted in a liquid nitrogen dewar that is also not shown on the diagram.

Each pick-up coil is connected to the corresponding input of the second ten-channel multiplexer, which in turn connects one input to its **Common** connector. This is then connected to the **A** input connector to the lock-

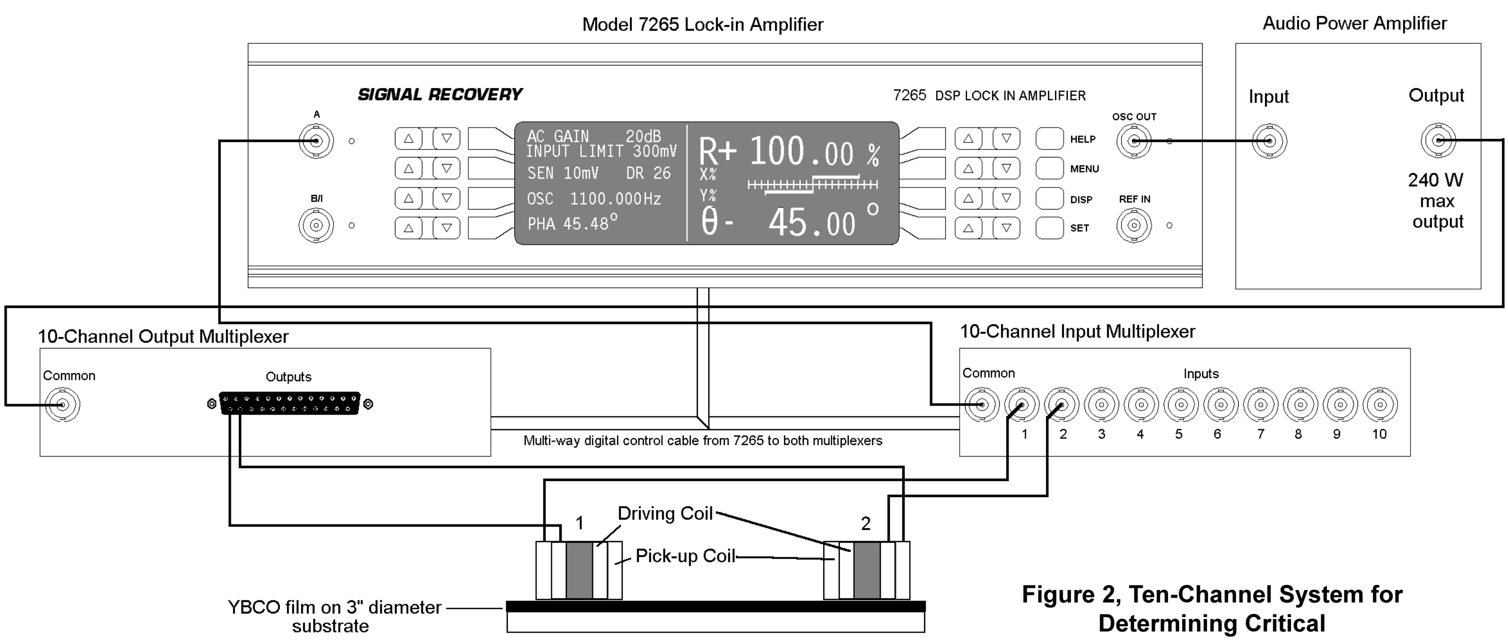

Figure 2, Ten-Channel System for Determining Critical Current Density in Superconductors

in amplifier. With the given driving current the typical signals at this point lie in the range of 1 to 50 µV.

The two multiplexers are controlled via the 8-bit digital output port of the lock-in amplifier, so that under computer control the system can connect the amplified oscillator signal to one of the driving coils and the corresponding pick-up coil back to the lock-in amplifier. The lock-in is operated in the 3*f* detection mode, in which it detects signals at the third harmonic of its reference frequency, or in other words at 3.3 kHz. Hence by increasing the oscillator output signal amplitude and monitoring the 3*f* amplitude the critical current density can be determined.

The system is calibrated by comparison with measurements taken on test samples using the traditional method.

Results

Figure 3 shows the plot of the output of one pick-up coil as the excitation current is increased, for the case of a wafer substrate only and one coated with a YBCO film. It can be seen that for the former sample the 3*f* detected signal is close to zero throughout. When, however, the superconducting film is present the 3*f* signal rises strongly at excitation currents greater than about 70 mA, which point corresponds to the critical current density for the sample.

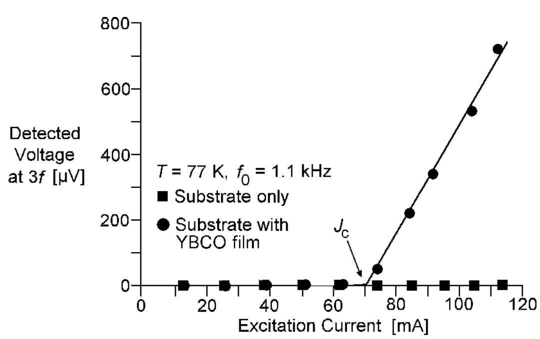

Figure 3, 3*f* Signal as a Function of Excitation Current for Superconducting Sample on Substrate and Substrate only conditions

Advantages of 10-Channel System

The major advantage offered by the use of the multiplexers is the speed with which the experiment can be performed. Setting up a single coil on the sample and cooling it down to 77K can take 10 to 15 minutes, so measuring ten points required several hours of time simply to allow the sample to heat up to room temperature, reposition the coil and cool it down again for each measurement. Using the multiplexed method, all ten coils are set up at the same time, which adds only a few minutes to the process, and then all ten measurements are made without needing to remove the sample from the dewar. This has the added advantage of reducing the risk of damage to the superconducting material that can be caused by water condensation during sample cooling and heating process.

Conclusions

In cases where several signals need to be measured using a lock-in amplifier the user is often not restricted to the obvious, but costly, solution of using multiple instruments in parallel. In some cases the use of the special detection modes offered only by the ***SIGNAL RECOVERY*** models 7265 and 7280 lock-in amplifiers may provide a solution, and in others the use of multiplexing techniques will give a cost-effective system.

Acknowledgement

SIGNAL RECOVERY acknowledges the assistance given by Dr Zaitsev of Forschungszentrum Karlsruhe, INFP, 76021 Karlsruhe, Germany, in the preparation of this note.

References

1 Claasen J H, Reeves M E and Soulen R J, Jr. 1991, Rev.Sci.Instr. 62, 996
2 AG Zaitsev, R Schneider, J Geerk et al., European Conf. on Appl. Superconductivity, 1999 Sitges (Spain)

Further Information

This application note is an introduction to the concept of input signal multiplexing. Additional information may be found in the following and other ***SIGNAL RECOVERY*** publications, which may be obtained from your local ***SIGNAL RECOVERY*** office or representative or by download from **www.signalrecovery.com**

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AN 1005 Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper

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Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper

APPLICATION NOTE

AN 1005

Introduction

The model 198A mixed beam light chopper can be used in conjunction with the dual reference mode provided by the **SIGNAL RECOVERY** model 7260, 7265 and 7280 DSP lock-in amplifiers to build a very cost-effective dual-beam measurement system. This technique can eliminate variations in source intensity over several orders of magnitude, which is especially useful in two common situations:

È If the source output is unstable over time, such as with some discharge lamps.

È If the "source" is the output of a spectrometer with a tungsten-halogen or other lamp as its input and the spectrometer center wavelength is being scanned as part of the experiment.

This application note describes how such a system can be configured.

Experimental Setup

A typical experimental arrangement is shown in Figure1 below.

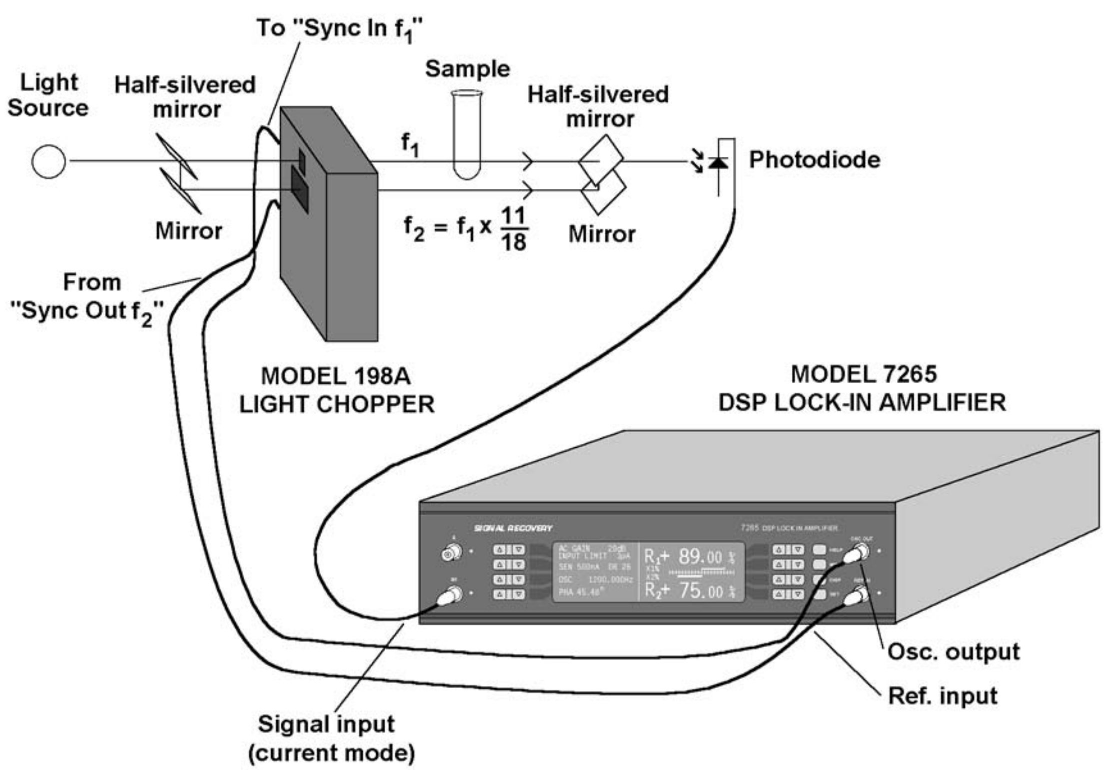

Figure 1, Dual Beam Ratiometric System using Model 198A Light Chopper and Model 7265 DSP Lock-in Amplifier

Description

A focussed light source is split into two beams using either a half-silvered mirror or beam-splitter, with the two beams being aligned so that they pass through the inner and outer apertures of the model 198A. One beam passes through the sample, while the other passes via a reference path that may include ND (neutral density) filters (not shown). Both beams are recombined back onto a single photodiode using a second mirror/beam-splitter combination. Hence there will be two signals at the detector, one at frequency f_1 that is due to the light that passes through the sample and the second at f_2, due to light travelling via the "reference" path.

A 7265 lock-in amplifier is set to the dual reference mode with the internal oscillator set to 360 Hz. The oscillator amplitude is set to 2 V rms and the **OSC OUT** connector coupled to the **Sync In f_1** input on the 198A. The chopper locks to this input, causing the f_1 frequency also to be 360 Hz and thereby satisfying the criterion that one of the signals being detected by the lock-in must be at the internal reference frequency.

The chopper also generates a reference signal at the f_2 frequency, which is coupled back to the lock-in's **REF IN** input. Hence the external reference channel operates at this frequency, which is in this case $360\ \Delta\ 11/18 = 220$ Hz.

Once the lock-in amplifier's sensitivity and phase controls have been properly adjusted then it indicates in the $\mathbf{R_2}$ channel (i.e. the internal reference frequency) the magnitude of the signal due to light through the sample, while the $\mathbf{R_1}$ channel (the external frequency) shows that due to light via the reference path.

The ratio of these two signals is independent over several orders of magnitude to changes in the source intensity, so if the ratio is calculated and displayed using the 7265's **User Equations** menu such variations can be eliminated.

Advantages

The system described is considerably more cost-effective than the traditional approach which required two lock-in amplifiers. Since the same detector and analog signal channel is used for both signals, differential drift between channels is also eliminated. Note, however, that limitations in the performance of the dual reference mode mean that best results are obtained when the experiment is arranged so that the absolute levels of the two electrical signals at f_1 and f_2 differ by no more than a factor of one hundred.

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