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:-

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

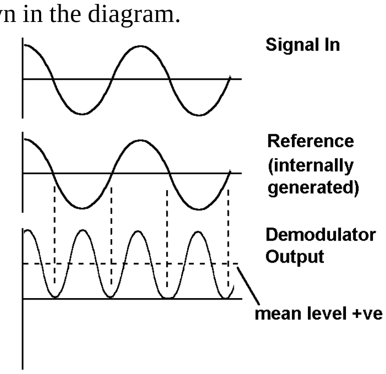

Figure 1

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.

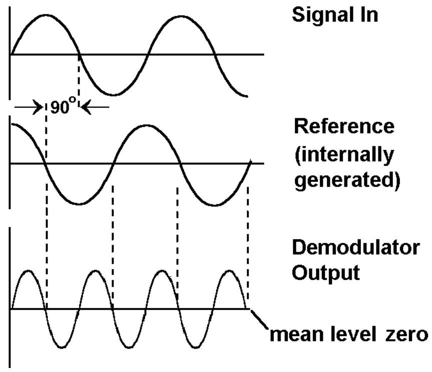

Figure 2

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

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

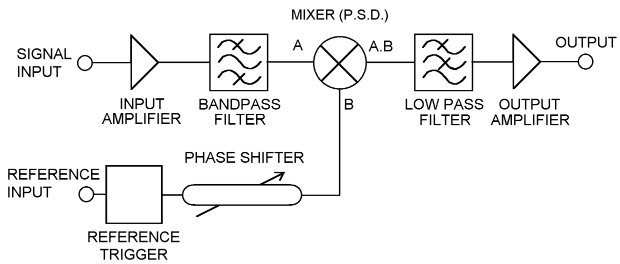

Figure 3

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.

PerkinElmer Instrument's 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:- $R = \sqrt{(X^2 + Y^2)}$

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 PerkinElmer Instruments (Signal Recovery) publications, which may be obtained from your local PerkinElmer Instruments (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 7260/7220 Synchronous Oscillator/Demodulator Monitor Output
AN 1002	Using the Model 7220 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 7220, 7265 and 7280 Lock-in Amplifiers

Appendix

Consider the case where a noise-free sinusoidal signal voltage Vin is being detected, where

$$V_{in} = A\cos(\omega t)$$

w is the angular frequency of the signal which is related to the frequency, F, in hertz by the equality:-

$$\omega = 2\pi F$$

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:-

$$V_{ref} = B\cos(\omega t + \theta)$$

where q 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:-

$$
\begin{aligned}
V_{psd} &= A\cos(\omega t) \,.\, B\cos(\omega t + \theta)\\
&= AB\cos\omega t\,(\cos\omega t\cos\theta - \sin\omega t\sin\theta)\\
&= AB(\cos^2\omega t\cos\theta - \cos\omega t\sin\omega t\sin\theta)\\
&= AB((\tfrac{1}{2} + \tfrac{1}{2}\cos 2\omega t)\cos\theta - \tfrac{1}{2}\sin 2\omega t\sin\theta)\\
&= \tfrac{1}{2}AB((1+\cos 2\omega t)\cos\theta - \sin 2\omega t\sin\theta)\\
&= \tfrac{1}{2}AB(\cos\theta + \cos 2\omega t\cos\theta - \sin 2\omega t\sin\theta)\\
&= \tfrac{1}{2}AB\cos\theta + \tfrac{1}{2}AB(\cos 2\omega t\cos\theta - \sin 2\omega t\sin\theta)\\
&= \tfrac{1}{2}AB\cos\theta + \tfrac{1}{2}AB\cos(2\omega t + \theta)
\end{aligned}
$$

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:-

- proportional to the magnitude of the input signal A
- proportional to the cosine of the angle, θ, between it and the reference signal
- modulated at 2ωt, 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 2ωt 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.

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