IAN 47

INTRODUCTION TO LOCK–IN AMPLIFIERS

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What Does a Lock–In Amplifier Do?

The lock–in amplifier is an extremely selective ac voltmeter used to measure a single–frequency signal obscured by noise. A typical design might operate over a range extending from one cycle every ten seconds to two hundred thousand cycles per second. The lock–in rejects random noise, transients, incoherent discrete frequency interference and harmonics of the measurement frequency. The classical analog instrument produces a dc output proportional to the rms amplitude of the sinusoidal fundamental of the signal. This output may be indicated on an analog meter, recorded on a chart or digitized for display, storage or further processing.

The lock–in amplifier provides very high sensitivity, directly allowing accurate measurement of ac voltages of one microvolt with one nanovolt resolution. Under some circumstances, the use of an appropriate voltage preamplifier can yield a practical full scale sensitivity of one nanovolt with .02 nV resolution. When used in conjunction with a current preamplifier, ac currents in the sub–pico ampere region can be accommodated. It can also function as a phasemeter,spectrum analyzer, vector voltmeter (in–phase and quadrature components simultaneously) or noise- meter. The lock–in is widely used in low–level optical work, acoustical measurement, cross–talk measurement, electron spectroscopy, radio astronomy, neurological research, feedback control of lasers, complex impedance plotting and production signal–to–noise characterization of photo detectors. Among its myriad of other scientific and industrial uses to recover signal buried in noise are fiber optic loss measurement, optical pyrometry, superconducting squid measurement and hot wire anemometry.

How Does a Lock–In Amplifier Work?

The lock–in amplifier behaves as if it were a narrow bandpass filter followed by an rms meter which rectifies the "purified" sinusoidal signal to a smooth dc output. Unfortunately, such a straightforward circuit arrangement wouldn't work for two reasons. First, a practical bandpass filter can only achieve a selectivity factor Q (ratio of center frequency to bandwidth) on the order of 100 or so and this cannot adequately suppress high levels of interference. Second, the center frequency would somehow have to be tuned to the signal frequency to within a fraction of one percent to achieve acceptable accuracy, no mean feat at low frequencies or in the presence of heavy noise. A slight mistuning or drift would cause a gross error.

To solve these twin problems the lock–in amplifier reverses the order of signal processing; first rectifying, then filtering. To function it must receive a second input coherent with the frequency of interest. This ***reference*** input (F_r) establishes the frequency to which the instrument responds. The lock–in applies the signal and reference to a synchronous rectifier referred to as the ***phase sensitive detector*** (PSD), which acts as a frequency mixer. The PSD downshifts the signal input spectrum by exactly F_r, causing the frequency of interest to appear as a pure dc output level. All other frequency components appear as filterable ac fluctuations at the PSD output. They can be removed by a simple and utterly stable RC lowpass bandlimiting filter, the Time Constant (T) of which can be made almost arbitrarily long. Higher performance lock–ins employ two such lowpass filters in series, thus affording faster settling time of the dc output for a given degree of random noise smoothing and a much more effective attenuation of discrete frequencies near F_r (12 dB/octave rolloff, vs. 6 dB/octave for a single RC section).

If we view the lock–in as a "black box", the overall filtering effect of having a PSD followed by a lowpass filter is identical to having a tuned amplifier followed by a rectifier. The equivalent noise bandwidth (B), however, can be made exceedingly small. $B = 1/(8T)$ so that $T = 125$ seconds corresponds to 0.001 Hz, for example. At a reference frequency of 100 kHz, this yields a Q of approximately 100 million. The equivalent passband remains precisely locked to the reference input. As the signal and reference vary together the tracking characteristic of the PSD keeps the effective center frequency absolutely aligned with the signal frequency, hence the term "Lock–In Amplifier".

Viewing Figure 1, we see that the SENSITIVITY control sets the front end ac gain of the lock–in to provide adequate drive to the PSD (on the order of 10 millivolts).

The ***signal conditioning*** varies considerably in LIA designs, sometimes being omitted altogether. It serves to remove signal components which would cause errors if allowed to reach the PSD, such as odd harmonics of the signal ($3F_r$, $5F_r$, ...). It also

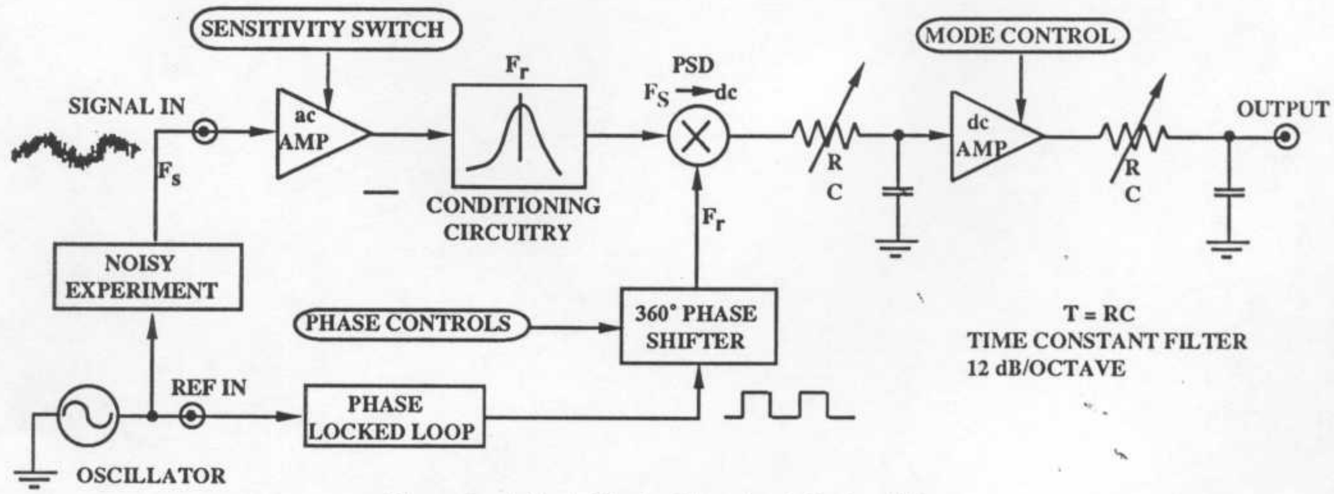

Figure 1 A Basic Single–Phase Lock–In Amplifier

increases the tolerance to high levels of interference (dynamic reserve) by excluding wideband components of the input spectrum and thereby conserving headroom.

The ***reference phase–locked loop*** conditions and stabilizes the external synchronizing waveform.

In general the phase of reference input will not be aligned with that of the signal input. As its name implies, the ***phase sensitive detector*** output decreases as the phase relationship of its inputs deviates from zero degrees (proportional to cosø). The PHASE control shifts the phase angle of the reference to maximize the dc output level and obtain a true reading.

The PSD multiplies the signal waveform by the reference waveform. Usually this is done by a switching circuit that, in effect, mathematically multiplies the input by a squarewave. This is the root of the odd harmonic responses, since a squarewave contains components at $3F_r$, $5F_r$, ...

The MODE control sets the post–detector dc gain of the lock–in for an output on the order of volts. Often this dc gain control is linked to the SENSITIVITY control and labeled "dynamic reserve". This allows one to apportion overall amplification between the ac and dc sections of the instrument and thereby trade decreased output stability for increased dynamic reserve. The higher the dc gain, the lower the ac drive to the PSD and the larger the allowable interference level before an overload occurs ahead of the PSD.

For more details on how lock–in amplifiers work, see ITHACO Application Notes IAN 23, *"The Heterodyning Lock–In Amplifier"* and IAN 35 *"The Evolution of the Modern Lock–In Amplifier"*.

Dual Phase Lock–Ins

The requirement for setting phase can become a very cumbersome impediment when long Time Constants are in effect or if the phase changes during a measurement. A dual–phase lock–in solves this problem by operating a second PSD in parallel with the first with a 90° shifted reference input. This yields two dc outputs (X and Y) which may be vector summed to attain a phase independent magnitude measurement (A).

As an added benefit, the dual phase lock–in can act as a vector voltmeter to measure both amplitude and phase simultaneously. Furthermore, since it yields an accurate measurement even when phase changes continuously, the signal need not be precisely

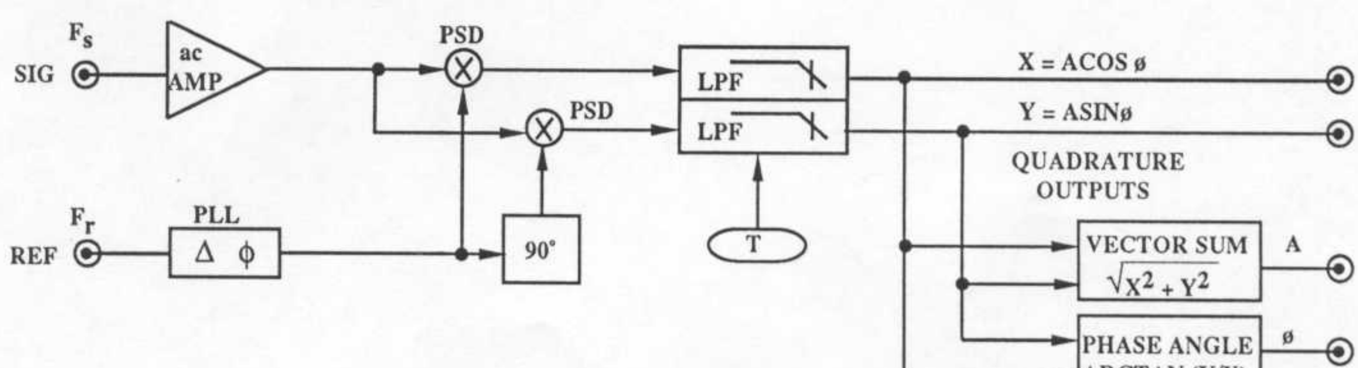

Figure 2 Dual Phase Lock–In Amplifier

synchronized with the reference. If F_s and F_r differ slightly, the phase will rotate at $(F_s - F_r)$ but A will remain true. By sweeping F_r, the dual phase lock–in can function as a spectrum analyzer with peak widths (selectivity) set by the Time Constant (bandwidth) control.

Automatic Phase Adjustment

There are several approaches to this function, which is of special importance for single phase lock–ins to increase the convenience and trustworthiness of measurements. Some single phase designs use a detector separate from the signal PSD to operate a phase setting loop. This has the advantage of a continuous, automatic, rapidly tracking phase adjustment that is independent of Time Constant setting and input level. The technique works admirably from low to moderately high interference levels.

Other single phase designs, on the other hand, sequentially measure the output at 0° and 90° phase settings then compute the proper angle for automated adjustment of the phase shifter. This yields a precise phase measurement in addition to the signal measurement. Use of the Time Constant control allows for accurate results even in very heavy noise (at the expense of slow automatic phase setting). These instruments cannot track a shifting phase and must be commanded to go through a new settling routine if the phase changes.

With dual–phase lock–ins (such as the ITHACO Model 3981), both vector components are available simultaneously, therefore it can respond instantly to an auto–phase command. It nulls the Y output by changing the reference phase shifter the requisite amount as computed from the previous X, Y and phase values. Typically this would be used to measure phase changes from an initial value.

Signal Conditioning

The absence of signal conditioning (i.e., "broadband" lock–in) in conjunction with the usual squarewave PSD usually limits measurement to noisy sinusoids, such as encountered in electro–chemical measurements. When the signal contains odd overtones, such as produced by electro–optical choppers, a phase dependent error on the order of 10% will occur. One can live with this only so long as one is making relative rather than absolute measurements and the phase of the reference input remains fixed relative to the signal input.

Historically, the odd harmonic problem was solved using a highly peaked (Q = 100) tuned ***bandpass*** amplifier front end ahead of the PSD. Some instruments still use this approach, despite its drawbacks such as severe phase shift, difficulty in tuning and poor frequency drift tolerance. A better approach, uses a ***peaked lowpass*** input filter, which supplies 12 dB/octave rolloff of odd harmonics in the ac signal input path rather than the 6 dB/octave rolloff provided by the bandpass amplifier. It efficiently removes signal overtones when set for moderate peaking (Q = 5) and thus is stable and easily settable. Note that any lock–in using conditioning filters cannot track a changing reference frequency unless the filter is re–tuned.

Some instruments sport additional 50/60 Hz line frequency notch filters and advertize them as a feature. Often these are "bandaid" fixes to cover up inadequate dynamic reserve problems. They add complexity and the potential for corrupting measurements.

The DYNATRAC® Lock–In Amplifier

ITHACO pioneered the heterodyning DYNATRAC lock–in amplifier, which provides an automatically tracking conditioning filter and inherent harmonic insensitivity without the need for extra user controls. In this scheme, the input spectrum is first upshifted to a fixed intermediate frequency (F_o) filtered by a fixed, peaked, lowpass filter, then downshifted to dc by the PSD. ***The PSD operates at a fixed F_o rather than a variable F_r*** and thus is both very stable and ***insensitive to harmonics of F_r***. This no–compromise design yields the ability to handle the really tough measurement problems, such as tracking a rapidly swept signal frequency.

PSD Approaches

Mathematically the PSD ought to be a linear multiplier driven by a sinusoidal reference input. The harmonic response problem would then disappear completely, making the conditioning circuitry non–mandatory. Unfortunately, actual analog multiplier circuits, have a number of poor characteristics, such as gain instability, non–linearity, distortion and offset drift, which become apparent when one attempts to make reproducible measurements in less– than– benign real world environments.

A much better approach employed by lower cost ITHACO instruments (Model 3981) involves using a stepped–waveform PSD which approximates a sinusoidal multiplication using digital techniques. This retains much of the accuracy characteristics of the squarewave PSD while suppressing all harmonics up to the 15th overtone (which always will be a trivial source of error). Again, conditioning filters are superfluous, allowing production of an inexpensive, simple, reliable and accurate instrument.

For rock solid stability, accuracy in the presence of heavy interference, linearity over a wide dynamic range, and operation over the widest frequency range the classical square wave–mixer PSD far outperforms other analog techniques. However this design demands added complexity in the form of signal conditioning circuitry to prevent harmonic responses.

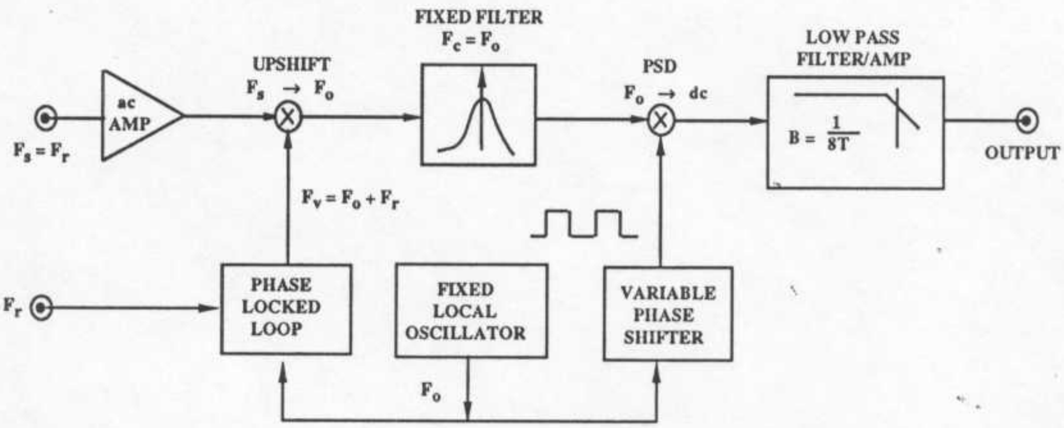

Figure 3 Heterodyning Lock–In Amplifier

Noise Measurement

Since the lock–in acts as a narrowband filter centered on the reference frequency, it passes a slice of the input noise density spectrum. This noise manifests itself as random fluctuations superimposed on the ideal dc output of the instrument. The rms amplitude of this variation equals the noise within the bandwidth (B) of the lock–in. It can be measured by an analog rms meter contained in the lock–in (e.g., ITHACO Noise Option 01). Better results can be obtained by digital means as described in ITHACO IAN 38, *"Method for Lock-In Amplifier Noise Measurement Using Digital Integration"*. This technique is implemented in the Model 3981 PC Plug–In Lock–In Amplifier.

Signal, Noise Bandwidth and Speed

(See IAN 49 for more detail)

The penalty for applying a long Time Constant (narrow bandwidth) to suppress noise is a long settling time and the consequent low measurement speed. Given the signal and random noise, for example, one can readily calculate the time required to attain a given accuracy. The fractional rms noise output fluctuation equals the inverse of the SNR (peak–to–peak excursion will be six times this value). If e_n is the rms 1 Hz bandwidth noise and e_s is the signal, then for the usual 12 dB/octave rolloff Time Constant filter:

$$\sigma_x = \text{fractional uncertainty in reading} = \frac{1}{SNR} = \frac{e_n \sqrt{B}}{e_s} + \frac{e_n}{e_s \sqrt{8T}}$$

Since roughly eight Time Constants are required for settling we have:

$$\text{Measurement time} \approx 8T = \left[\frac{e_n}{e_s \ \sigma_x} \right]^2$$

The use of linear signal averaging of digital samples by an instrument such as the Model 3981 will cut this time approximately in half.

High Frequency Lock–Ins

Most conventional lock–ins are limited to frequencies of 200 kHz or less. To get to measurement frequencies in the megahertz range, a mixing technique must be employed which will downshift both the signal and reference inputs to a range which can be handled by an ordinary lock–in PSD circuit. The circuitry bears much in common with ordinary AM radio components. The ITHACO/HMS Model 5311 for example, acts as a front end to translate inputs in the 100 kHz to 10 MHz range down to 20 kHz.

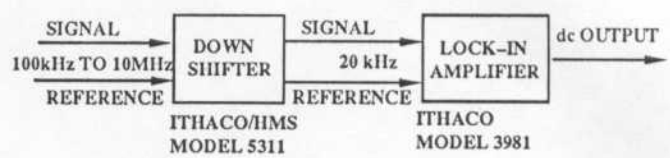

Figure 4 High Frequency Down Converter

Conclusion

The lock–in amplifier is a sophisticated and highly flexible signal recovery and analysis instrument. This is why such a "voltmeter" has a complex control panel. Proper use requires a degree of understanding of its operating principles. The subtle aspects of its performance which characterize a superior instrument which you can rely on for reproducible results are not readily apparent when reading specification brochures. ITHACO, a leader in the development of lock–in amplifiers, provides instruments designed to provide satisfaction consistently year after year.