Lock - in Amplifier and Applications

What is a Lock in Amplifier?

In a nut shell, what a lock-in amplifier does is measure the amplitude V_o of a sinusoidal voltage,

$$V_{in}(t) = V_o \cos(\omega_o t)$$

where $\omega_o = 2\pi f_o$ and f_o are the angular- and natural frequencies of the signal respectively. You supply this voltage to the *signal input* of the lock-in, and its meter tells you the amplitude Vo, typically calibrated in V-rms. What makes a lock-in different from a simple AC voltmeter, which is what I just described, is that you must also supply the lock-in with a reference input, that is, a decent size, say 1 volt p-p, sinusoidal voltage that is synchronized with the signal whose amplitude you are trying to measure. The lock-in uses this signal (like an external trigger for an oscilloscope) to "find" the signal to be measured, while ignoring anything that is not synchronized with the reference. In practice, the lock-in can measure voltage amplitudes as small as a few nano volts, while ignoring signals even thousands of times larger. In contrast, an AC voltmeter would measure the sum of all of the voltages at its input.

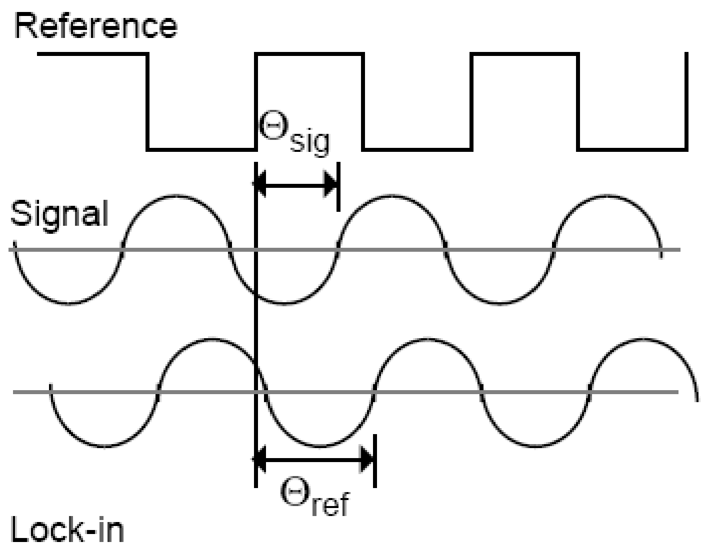
Let's consider an example. Suppose the signal is a 10 nV sine wave at 10 kHz. Clearly some amplification is required to bring the signal above the noise. A good low noise amplifier will have about 5 nV/ √Hz of input noise. If the amplifier bandwidth is 100 kHz and the gain is 1000, then we can expect our output to be 10μV of signal (10 nV x 1000) and 1.6 mV of broadband noise (5 nV/√Hz x √100 kHz x 1000). We won't have much luck measuring the output signal unless we single out the frequency of interest.

If we follow the amplifier with a bandpass filter with a Q=100 (a VERY good filter) entered at 10 kHz, any signal in a 100 Hz bandwidth will be detected (10 kHz/Q). The noise in the filter pass band will be 50 μV (5 nV/√Hz x √100 Hz x 1000) and the signal will still be 10 μV. The output noise is much greater than the signal and an accurate measurement can not be made. Further gain will not help the signal to noise problem.

Now try following the amplifier with a phase-sensitive detector (PSD). The PSD can detect the signal at 10 kHz with a bandwidth as narrow as 0.01 Hz! In this case, the noise in the detection bandwidth will be only 0.5 μV (5 nV/√Hz x √.01 Hz x 1000) while the signal is still 10 μV. The signal to noise ratio is now 20 and an accurate measurement of the signal is possible.

What is Phase Sensitive Detection?

Lock-in measurements require a frequency reference. Typically an experiment is excited at a fixed frequency (from an oscillator or function generator) and the lock-in detects the response from the experiment at the reference frequency. In the diagram below, the reference signal is a square wave at frequency ω_r. This might be the sync output from a function generator. If the sine output from the function generator is used to excite the experiment, the response might be the signal waveform shown below.



The signal is $V_{sig}\sin(\omega_r t + \theta_{sig})$ where Vsig is the signal amplitude, ω_r is the signal frequency, and θ_{sig} is the signal's phase. Lock-in amplifiers generate their own internal reference signal usually by a phase-locked-loop locked to the external reference. In the diagram below the external reference, the lock-in's reference and the signal are all shown. The internal reference is $V_L\sin(\omega_o t + \theta_{ref})$.

The lock in amplifies the signal and then multiplies it by the lock-in reference using a phase-sensitive detector or multiplier. The output of the PSD is simply the product of two sine waves.

$$V_{psd} = V_{sig}V_L\sin(w_r t + \theta_{sig})\sin(\omega_L t + \theta_{ref})$$

$$= 1/2\ V_{sig}V_L\cos([\omega_r - \omega_L]t + \theta_{sig} - \theta_{ref}) - 1/2\ V_{sig}V_L\cos([\omega_r + \omega_L]t + \theta_{sig} + \theta_{ref})$$

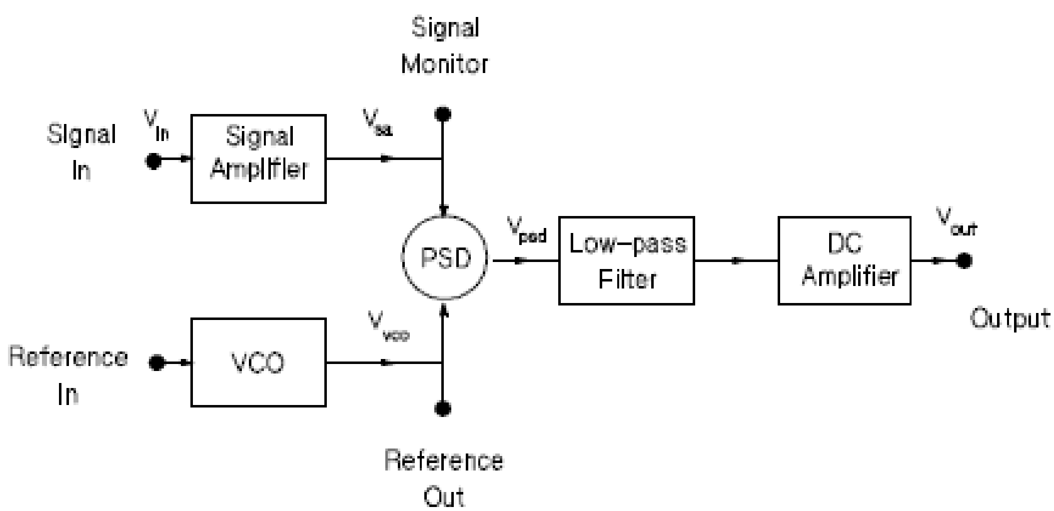
The PSD output is two AC signals, one at the difference frequency $(\omega_r - \omega_L)$ and the other at the sum frequency $(\omega_r + \omega_L)$. If the PSD output is passed through a low pass filter, the AC signals are removed. What will be left? In the general case, nothing.

However, if ω_r equals ω_L, the difference frequency component will be a DC signal. In this case, the filtered PSD output will be $V_{psd} = 1/2\ V_{sig}V_L\cos(\theta_{sig} - \theta_{ref})$.
This is a very nice signal — it is a DC signal proportional to the signal amplitude. It's important to consider the physical nature of this multiplication and filtering process in different types of lock-ins. In traditional analog lock-ins the signal and reference are analog voltage signals. The signal and reference are multiplied in an analog multiplier, and the result is filtered with one or more stages of RC filters. In a digital lock-in such as the SR850, the signal and reference are represented by sequences of numbers. Multiplication and filtering are performed mathematically by a digital signal processing (DSP) chip. We'll discuss this in more detail later.

Block Diagram

While a lock-in amplifier is an extremely important and powerful measuring tool, it is also quite simple. The block diagram of a lock-in amplifier is shown in Figure 1. The lock-in consists of five stages:

1. AC amplifier, called the signal amplifier;
2. Voltage controlled oscillator (VCO);
3. Multiplier, called the phase sensitive detector (PSD);
4. Low-pass filter; and
5. DC amplifier.



The signal to be measured is fed into the input of the AC Amplifier. The output of the DC amplifier is a DC voltage proportional to V_0. This voltage is displayed on the lock-in's own meter, and is also available at the output connector. I now elaborate on the functions of the five stages.

The ***AC amplifier*** is simply a voltage amplifier combined with variable filters. Some lock-in amplifiers let you change the filters as you wish, others do not. Some lock-in amplifiers have the output of the AC amplifier stage available at the signal monitor output. Many do not.

The ***voltage controlled oscillator*** is just an oscillator, except that it can synchronize with an external reference signal (i.e., trigger) both in phase and frequency. Some lock-in amplifiers contain a complete oscillator and need no external reference. In this case they operate at the frequency and amplitude that you set, and you must use their oscillator output in your experiment to derive the signal that you ultimately wish to measure. Virtually all lock-in amplifiers are able to synchronize with an external reference signal. The VCO also contains a phase-shifting circuit that allows the user to shift its signal from 0-360 degrees with respect to the reference.

The ***phase sensitive detector*** is a circuit which takes in two voltages as inputs V1 and V2 and produces an output which is the product V_1*V_2. That is, the PSD is just a multiplier circuit.

The ***low pass filter*** is an RC filter whose time constant may be selected. In many cases you may choose to have one RC filter stage (single pole filter) or two RC filter stages in series (2-pole filter). In newer lock-in amplifiers, this might be a digital filter with the attenuation of a "many pole" filter.

The ***DC amplifier*** is just a low-frequency amplifier similar to those frequently assembled with op-amps. It differs from the AC amplifier in that it works all the way down to zero frequency (DC) and is not intended to work well at very high frequencies, say above 10 KHz.

Time Constant and DC Gain

The output of the PSD contains many signals. Most of the output signals have frequencies which are either the sum or difference between an input signal frequency and the reference frequency. Only the component of the input signal whose frequency is exactly equal to the reference frequency will result in a DC output. The low pass filter at the PSD output removes all of the unwanted AC signals, both the 2F (sum of the signal and the reference) and the noise components. This filter is what makes the lock-in such a narrow band detector.

Time Constants

Lock-in amplifiers have traditionally set the low pass filter bandwidth by setting the time constant. The time constant is simply 1/2pf where f is the -3 dB frequency of the filter. The low pass filters are simple 6 dB/oct roll off, RC type filters. A 1 second time constant referred to a filter whose -3 dB point occurred at 0.16 Hz and rolled off at 6 dB/oct beyond 0.16 Hz. Typically, there are two successive filters so that the overall filter can roll off at either 6 dB or 12 dB per octave. The time constant referred to the -3 dB point of each filter alone (not the combined filter). The notion of time constant arises from the fact that the actual output is supposed to be a DC signal. In fact, when there is noise at the input, there is noise on the output. By increasing the time constant, the output becomes more steady and easier to measure reliably. The trade off comes when real changes in the input signal take many time constants to be reflected at the output. This is because a single RC filter requires about 5 time constants to settle to its final value. The time constant reflects how slowly the output responds, and thus the degree of output smoothing. The time constant also determines the equivalent noise bandwidth (ENBW) for noise measurements. The ENBW is NOT the filter -3 dB pole, it is the effective bandwidth for Gaussian noise.

DC Output Gain

How big is the DC output from the PSD? It depends on the dynamic reserve. With 60 dB of dynamic reserve, a noise signal can be 1000 times (60 dB) greater than a full scale signal. At the PSD, the noise can not exceed the PSD's input range. In an analog lock-in, the PSD input range might be 5V. With 60 dB of dynamic reserve, the signal will be only 5 mV at the PSD input. The PSD typically has no gain so the DC output from the PSD will only be a few mill volts! Even if the PSD had no DC output errors, amplifying this mill volt signal up to 10 V is error prone. The DC output gain needs to be about the same as the dynamic reserve (1000 in this case) to provide a 10 V output for a full scale input signal. An offset as small as 1 mV will appear as 1 V at the output! In fact, the PSD output offset plus the input offset of the DC amplifier needs to be on the order of 10 μV in order to not affect the measurement. If the dynamic reserve is increased to 80dB, then this offset needs to be 10 times smaller still. This is one of the reasons why analog lock-ins do not perform well at very high dynamic reserve. The digital lock-in does not have an analog DC amplifier. The output gain is yet another function handled by the digital signal processor. We already know that the digital PSD has no DC output offset. Likewise, the digital DC amplifier has no input offset. Amplification is simply taking input numbers and multiplying by the gain. This allows the SR830 to operate with 100 dB of dynamic reserve without any output offset or zero drift.

THE BASIC LOCK-IN

This measurement is designed to use the internal oscillator to explore some of the basic lock-in functions. You will need BNC cables.

Specifically, you will measure the amplitude of the Sine Out at various frequencies, sensitivities, time constants and phase shifts.

1. Disconnect all cables from the lock-in. Turn the power on while holding down the [Setup] key. Wait until the power-on tests are completed.

When the power is turned on with the [Setup] key pressed, the lock-in returns to its standard default settings. See the Standard Settings list in the Operation section for a complete listing of the settings.

The Channel 1 display shows X and Channel 2 shows Y.

2. Connect the Sine Out on the front panel to the A input using a BNC cable.

The lock-in defaults to the internal oscillator reference set at 1.000 kHz. The reference mode is indicated by the INTERNAL led. In this mode, the lock-in generates a synchronous sine output at the internal reference frequency.

The input impedance of the lock-in is 10 MΩ. The Sine Out has an output impedance of 50Ω. Since the Sine Output amplitude is specified into a high impedance load, the output impedance does not affect the amplitude.

The sine amplitude is 1.000 Vrms and the sensitivity is 1 V(rms). Since the phase shift of the sine output is very close to zero, Channel 1 (X) should read close to 1.000 V and Channel 2 (Y) should read close to 0.000 V.

3. Press [Auto Phase]

Automatically adjust the reference phase shift to eliminate any residual phase error. This should set the value of Y to zero.

4. Press [Phase]

Display the reference phase shift in the Reference display. The phase shift should be close to zero.

5. Press the [+90°] key.

This adds 90° to the reference phase shift. The value of X drops to zero and Y becomes minus the magnitude (-1.000 V).

Use the knob to adjust the phase shift until Y is zero and X is equal to the positive amplitude.

The knob is used to adjust parameters which are shown in the Reference display, such as phase, amplitude and frequency. The final phase value should be close to zero again.

Press [Auto Phase]

Use the Auto Phase function to return Y to zero and X to the amplitude.

6. Press [Freq]

Show the internal oscillator frequency in the Reference display.

Use the knob to adjust the frequency to 10 kHz.

The knob now adjusts the frequency. The measured signal amplitude should stay within 1% of 1 V and the phase shift should stay close to zero (the value of Y should stay close to zero).

Use the knob to adjust the frequency back to 1 kHz.

The internal oscillator is crystal synthesized with 25 ppm of frequency error. The frequency can be set with 4 1/2 digit or 0.1 mHz resolution, whichever is greater.

7. Press [Ampl]

Show the sine output amplitude in the Reference display.

Use the knob to adjust the amplitude to 0.01 V.

As the amplitude is changed, the measured value of X should equal the sine output amplitude. The sine amplitude can be set from 4 mV to 5 V rms into high impedance (half the amplitude into a 50 Ω load).

8. Press [Auto Gain]

The Auto Gain function will adjust the sensitivity so that the measured magnitude (R) is a sizable percentage of full scale. Watch the sensitivity indicators change.

9. Press [Sensitivity Up] to select 50 mV full scale.

Change the sensitivity back to 20 mV.

Parameters which have many options, such as sensitivity and time constant, are changed with up and down keys. The sensitivity and time constant are indicated by leds.

10. Press [Time Constant Down] to change the time constant to 300 μs.

The values of X and Y become noisy. This is because the 2f component of the output (at 2 kHz) is no longer attenuated completely by the low pass filters.

Press [Time Constant Up] to change the time constant to 3 ms.

Let's leave the time constant short and change the filter slope.

Parameters which have only a few values, such as filter slope, have only a single key which cycles through all available options. Press the corresponding key until the desired option is indicated by an led.

11. Press the [Slope/Oct] key until 6 dB/oct is selected.

The X and Y outputs are somewhat noisy at this short time constant and only 1 pole of low pass filtering.

Press [Slope/Oct] again to select 12 dB/oct.

The outputs are less noisy with 2 poles of filtering.

Press [Slope/Oct] twice to select 24 db/oct.

With 4 poles of low pass filtering, even this short time constant attenuates the 2f component reasonably well and provides steady readings.

Press [Slope/Oct] again to select 6 db/oct.

Let's leave the filtering short and the outputs noisy for now.

12. Press [Freq]

Show the internal reference frequency on the Reference display.

Use the knob to adjust the frequency to 55.0 Hz.

At a reference frequency of 55 Hz and a 6 db/oct, 3 ms time constant, the output is totally dominated by the 2f component at 100 Hz.

13. Press [Sync Filter]

This turns on synchronous filtering whenever the detection frequency is below 200 Hz.

Synchronous filtering effectively removes output components at multiples of the detection frequency. At low frequencies, this filter is a very effective way to remove 2f without using extremely long time constants.

The outputs are now very quiet and steady, even though the time constant is very short. The response time of the synchronous filter is equal to the period of the detection frequency (18 ms in this case).

This concludes this measurement example. You should have a feeling for the basic operation of the front panel. Basic lock-in parameters have been introduced and you should be able to perform simple measurements.