TABLE OF CONTENTS

SIGNAL PROCESSOR / LOCK-IN AMPLIFIER

The lock-in amplifier has been an important tool in the experimental physicists "bag of tricks" since the early 1950's. It would be unusual to find a research laboratory today that didn't have at least one lock-in amplifier in use and many more hidden within various measuring instruments. When the senior author of this manual (JFR) was a graduate student at Washington University (St. Louis, MO), he built every lock-in that was used in his magnetic resonance experiments himself. That was certainly not unusual, since all the lock-ins in the department were homemade.

All that has changed. Now, several companies produce very sophisticated lock-in amplifiers. These units are wonderful in almost every respect. They are ultra-low noise, have large dynamic range, are extremely stable, completely automatic, in short they are nearly ideal for enhancing the signal-to-noise of a "coded signal" buried in the noise. They are certainly much better instruments than we fabricated with transformers and vacuum tubes back 50 years ago. All you need is money and you can buy yourself a wonderful lock-in amplifier.

However, there is a price to pay for the ease and apparent simplicity of these devices. Our students today have only the vaguest idea of what the lock-in accomplishes and how it produces this signal-to-noise enhancement, much less how to build one. Signal processing is an important idea for physicists, electrical engineers, physical chemists, and biophysicists, as well as many others to understand, and even, for some, to master. Using a modern lock-in as a black box, will certainly not accomplish that.

We at TeachSpin believe it is essential that students who plan to be experimental scientists have hands-on experience with signal processing and understand, particularly, the workings of a lock-in amplifier. Thus, we have created the SPLIA1-A instrument, a modular device that can be used in a variety of ways to study signal processing. This device must be configured by the student and hand adjusted for the particular signal being studied. In several of its configurations, it can be used as a research grade lock-in with acceptable levels of stability, noise, and dynamic range. However, it was designed, by TeachSpin, as a teaching tool for advanced undergraduate or graduate students. That was our primary consideration when we designed the unit. It has, therefore, many, many more input and output connectors and the gain, phase shift, frequency, amplitudes, etc. are controlled by the student. Each function of the device is modular, self-contained, and can be examined by the student.

This instructor's manual attempts to provide a comprehensive overview of the instrument and possible student experiments that can be accomplished with the unit. This is not a student manual. Here we will include the data we acquired with the suggested experiments. The students, we believe, should take their own measurements and do their own analysis. As the instructor, you are free to use any material in this manual for your students. You are also free to make any modifications, additions, or subtractions you feel appropriate. We hope you and your students will enjoy working with this apparatus. Please let us hear from you if you have any suggestions for additional experiments or modifications that we might pass on to other users.

I. INTRODUCTION

A lock-in amplifier (phase-sensitive detector) is an electronic instrument designed to extract wanted signals which are accompanied by unwanted noise, disturbances, and /or background interference. These instruments reduce the unwanted noise, background, and disturbances and increase the desired signal to enhance the overall signal-to-noise ratio of the electrical signal being measured. This increases the precision of the measurement.

The lock-in can also be used in control circuits where the phase-sensitive detection is an essential component of the feedback loop. Although this is an important application of the lock-in amplifier, we will not consider it in this manual. The discussion of control feedback loops is usually beyond the needs of most advanced science students. This is the only place it will be mentioned in this manual.

The signal processor/lock-in amplifier you are about to use is more than a lock-in amplifier. It can be used in many different configurations to study signal-to-noise enhancement, to examine amplifier characteristics, filter characteristics, and noise characteristics. It can be configured not only as an amplitude detector, but also as a spectrum analyzer. In short, it is a versatile instrument that you can use in many different applications.

There are many different types of commercial lock-in amplifiers. Each has its own special characteristics that make it appropriate to a particular set of experiments. However, all lock-in have one thing in common, they all enhance the signal-to-noise for signals that are "coded". The word "coding" is not commonly used in the literature, but we believe it expresses the correct characteristics of a signal appropriate for a lock-in to process. We will explain more about what we mean by coding later in this manual.

Lock-in amplification is not the only method of signal processing that is available. At least four other types of signal processors have important applications in scientific experiments. We will not spend time describing these instruments in detail, but want to make students aware of their existence.

a. MULTICHANNEL SCALER

This device is basically a fast digitizer of the signal with multiple channels to add up the digitized signal as the entire experiment is repeated over and over again. It is particularly useful in Mossbauer spectroscopy, but can be used along with a lock-in for signal enhancement when there is long-term drift.

b. BOXCAR INTEGRATOR

This device is a gated analog integrator which is typically used in pulsed experiments, where a transient signal appears for an extremely short period of time. Optical echoes and other pulsed optics experiments have found this instrument helpful.

c. PULSE-HEIGHT ANALYZER

These instruments are used in all kinds of nuclear physics counting experiments where one needs to separate electrical pulses by amplitude and collect large numbers of counts to acquire reasonable statistics. They also can find application in light experiments where the detector is a photomultiplier and one is counting individual photons.

d. FAST-FOURIER ANALYZER

This is one of the most powerful ways to enhance the signal-to-noise of transient signals. It involves using a broadband receiver to accept all frequencies of a transient signal simultaneously and then performing a digital fast-fourier transform of the signal, followed by a digital averaging of the FFT signal. This is the standard way NMR signals from organic compounds are measured. It is truly the workhorse of both organic chemists and biologists.

II. NOISE

Noise is a vast and quite sophisticated subject. It could easily become the only subject of an advanced one-semester course in a physics or electrical engineering department. We will only scratch the surface of the subject, but hopefully we can provide the students with an introduction into the subject. Horowitz and Hill (p 430 2nd edition) will provide excellent supplementary reading on this important topic!

There are multiple sources and multiple types of noise, interference, and other unwanted, but often present, electrical signals. We shall divide noise into two general categories. These may not be universal definitions, but we shall use it in this manual.

1. RANDOM – stochastic, physical. One can apply statistical analysis to these signals.

2. SYSTEMATIC – such as gain changes, circuit parameters drift, vibrations, electromagnetic interference – particularly 60 Hz and 120 Hz from power lines, transformers, etc. and background fluctuations. Background noises are usually idiosyncratic to the apparatus, the local environment, and the experiment, and are, usually, not susceptible to mathematical statistical analysis, but are amenable to clever experimental adjustments and design.

All varieties of noise can be described in terms of three important characteristics.

a. Frequency distribution
b. Amplitude distribution
c. Physical mechanism

We will give a brief overview of several important types of noise considering these characteristics:

A. JOHNSON NOISE

Maybe the most common electronic component is a resistor. Although you may understand this device in terms of Ohms Law and its effect on the flow of electrical charge in a circuit, it is also an important source of noise. All by itself, with nothing attached, sitting on a table minding its own business, at room temperature (290°K), it is producing noise. That is, there are random fluctuating voltages appearing across its terminals due to fluctuations in the charge carriers at this temperature. It is clearly random in character and the mechanism is thermal in origin.

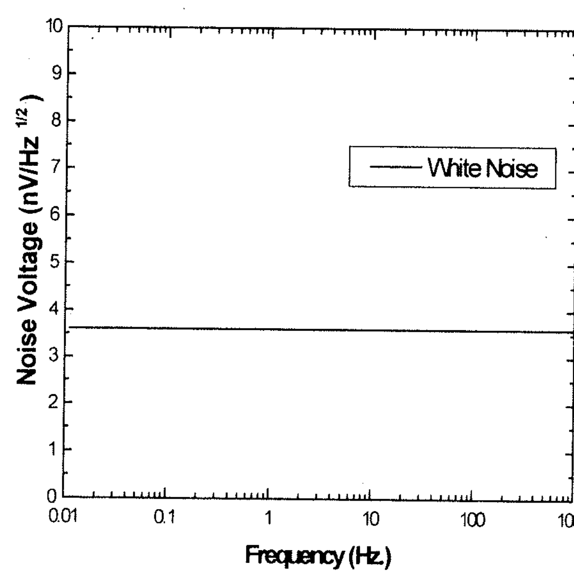

Figure 1

If one measures this noise voltage as a function of frequency, one gets a spectrum as shown in Figure 1: This "flat" spectrum is called "WHITE" Noise. It is characterized as having the same noise power per unit frequency range over all frequencies. (or at least over all frequencies that can reasonably be measured). The rms[1] Johnson noise voltage can be theoretically calculated from the expression:

$$V(rms) = (4kTR\Delta f)^{1/2}$$

Where

k = Boltzman's Constant = 1.38×10^{-23} J/K
T = Absolute Temperature of the Resistor
R = Resistance in Ohms
Δf = Effective Bandwidth of Noise Measurement in Hz

To reduce the noise voltage of a given resistor, one can either change the temperature (not usually practical) or reduce the bandwidth of the measurement. To give an example, if the range of frequencies that the measuring unit will accept is confined between 100 Hz and 1,000 Hz, the bandwidth will be 900 Hz, and there will be less noise measured than if the measurement was done between 100 Hz and 1 MHz.

Let's look at some numbers, to get a sense of the magnitude of this thermal noise.

At Room Temperature $(4kT)^{1/2} = (4 \times 1.38 \times 10^{-23}\ \text{J/K} \times 293°\text{K})^{1/2} = 1.27 \times 10^{-10} \sqrt{\text{J}}$
For a 1 KΩ Resistor with Δf = 10 Hz
V_{rms} (Noise) = 1.27×10^{-2} μV

A useful equation is: $V_{rms} = 1.27 \times 10^{-4} (R\Delta f)^{1/2}\ \mu V / Hz^{1/2}$

In Figure 2 we have plotted the open circuit rms noise voltage per $(\Delta f)^{1/2}$ as a function of resistance for an ideal resistor. Notice the units of noise in Figs. 1 and 2. They are in $nV/(Hz)^{\frac{1}{2}}$ a rather unusual unit. The reason for these units can be seen in the equation given rms voltage. A measurement

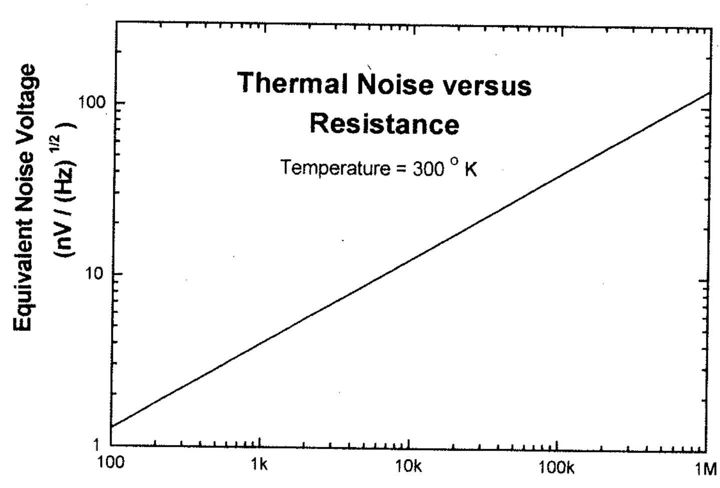

Figure 2

[1] RMS means root mean square defined as:

$$V_{rms} = \left[\frac{1}{2\pi}\int_0^{2\pi} V^2(t)dt\right]^{\frac{1}{2}} \qquad \text{For a sinusoidal } V = V_a \sin \omega t\ ,\ V_{rms} = \frac{1}{2\sqrt{2}} V_{p-p}$$

of noise voltage has physical meaning only if one specifies the bandwidth Δf over which the noise is being measured. A useful equation for ***room temperature*** noise is then:

$$\frac{V(rms)}{(\Delta f)^{\frac{1}{2}}} = 1.27x10^{-4} R^{\frac{1}{2}} \ in\, \mu V / Hz^{\frac{1}{2}}$$

The Johnson noise sets a lower limit on measurement on any signal source, detector, or amplifier that has internal resistance. This noise voltage comes from physical fluctuations that are fundamental to the resistance process.

The equation for calculating the Johnson noise might "ring a bell" in the student's head. If we let Δf get large without limit, then the noise voltage also becomes unphysically large. This is very reminiscent of the so called "ultra-violet catastrophe" in black-body radiation. Clearly these two phenomena are related. The classical theory that accurately predicts the Johnson noise for reasonable temperature and bandwidths, fails us for a large bandwidth.

The resolution of this problem can be seen by recognizing that any bound system has finite energy states and the thermal energy can only excite to the approximate frequency f_{max} where $kT \approx hf \ or \ f_{max} \approx \frac{kT}{h}$. For room temperature (300° K), f_{max} is approximately equal to 10^{12} Hz. So in some sense, there is a finite cutoff in the bandwidth of the noise spectrum.

The Johnson noise should not be confused with the <u>additional</u> noise that comes from the fluctuation in the resistance of the resistor when current passes through it. Such resistance fluctuation is <u>component construction</u> sensitive. That is, it depends on the kind of resistor, carbon composition, carbon film, metal film, wire wound, and it depends on the quality of the construction. This additional noise tends to have a noise spectrum which is not "white" but increases as 1/f at the low frequency end.

B. SHOT NOISE

Current is the flow of finite charge elements, usually electrons. Since the current is due to these discrete charges, there will always be a fluctuation of this current, or current noise. At large currents with many charge carriers, the percentage fluctuation is very small, but for very small currents, the fluctuations can become a large fraction of the current. The equation describing the noise current is:

$$I_{noise}\,(rms) = (2\, q\, I_{dc}\, \Delta f)^{1/2} \text{ in amperes}$$

Where

$q = 1.6 \times 10^{-19}$ C (electron's charge) Δf = Bandwidth in Hz

I_{dc} = current in amperes

Like the Johnson noise, shot or current noise is white noise and is fundamental to the physical processes, and thus irreducible in nature.

C. 1/f NOISE (FLICKER NOISE)

1/f noise, sometimes called flicker noise, has been the subject of intense experimentation and theoretical speculation for at least 20 years. There is evidence that 1/f noise may have some important universal properties, but the final chapter on this subject has not yet been written.

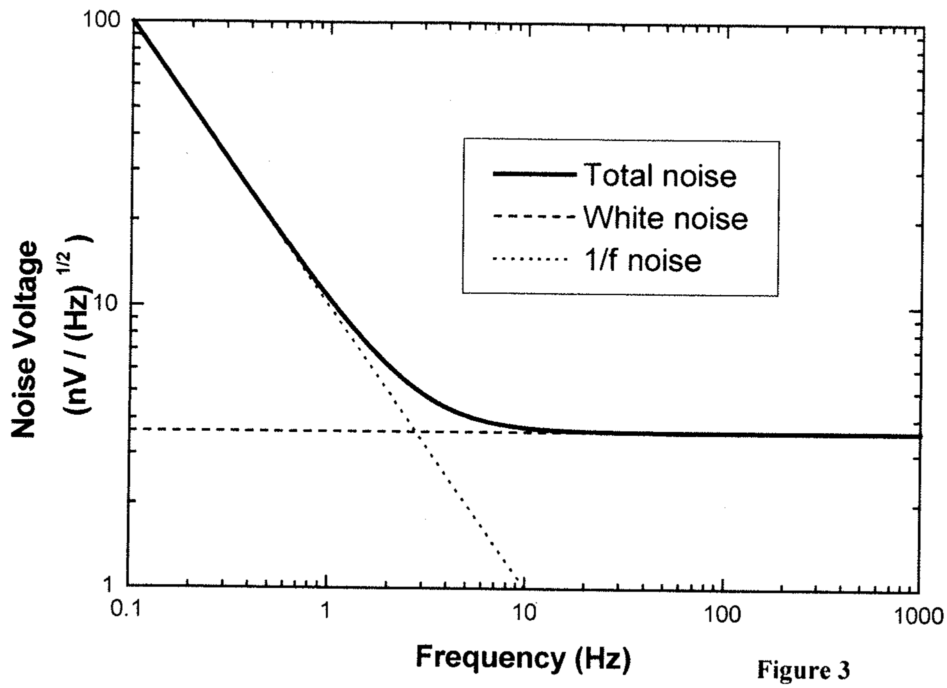

Figure 3

As its name implies, it is noise that increases as the reciprocal of the frequency, with a noise spectrum that appears approximately as shown in Figure 3. This noise appears in a wide variety of systems, some unexpected. Amplifier noise, the base current of transistors, and some photo detectors, are examples of where one might observe this noise. It also seems to depend upon the type and quality of construction.

Horowitz and Hill give the following data for resistor 1/f noise, in rms microvolts per volt applied across the resistor. They do not specify "the decade" over which these measurements were made. We present them here to give the students some feel for the size and variation of this kind of noise.

Carbon Resistor (Δf of one decade)	.1 μV - 3 μV
Carbon Film (Δf of one decade)	.05 μV - .3 μV
Metal Film (Δf of one decade)	.02 μV - .2 μV
Wire Wound (Δf of one decade)	.01 μV - .2 μV

These noise spectra are not exactly 1/f but more like $1/f^m$ where m can vary from .8 to 1.5. However, the experimenter should be aware that this noise can be dominant, particularly at low frequencies. Thus, it is advisable to recover the signal in a region where the white noise dominates. If the experimental noise spectrum is similar to Figure 3, then the coding signal should be above 10 Hz, and avoid the noise "spikes" at 60 and 120 Hz and their harmonics. This becomes important when one is choosing the "coding" frequency of the signal.

D. INTERFERENCE

There are many sources of interference in real experiments. We will only discuss some of the more common types of non-stochastic noise that experimenters deal with.

1. ELECTROMAGNETIC
   Any wire segment or copper trace on a circuit board constitutes an antenna. The environment most instruments find themselves in is, by any criteria, "hostile". These antennas are sensitive to both electrostatic and electromagnetic fields. AC power lines, transformer fields, radio and TV signals, cellular phones, motors, generators, atmospheric disturbances, and many others are usually available to interfere with the desired signal. If the desired signals are small, the experimenter is well advised to eliminate (as much as possible) the source of these fields and to shield as best as they can, against the rest.

2. DRIFT AND FLUCTUATIONS
   Real instruments have gain drift, offsets, and fluctuations of all their characteristics. These may be due to component change, temperature fluctuations, power line fluctuations, or aging. These can be a major source of effective noise, but generally cannot be analyzed mathematically.

3. MICROPHONICS
   Mechanical vibrations can be a major, even dominant, source of noise. Optics experiments are particularly susceptible to this form of interference. Today, a great deal of engineering expertise has gone into elimination of vibrational noise. A glance through a catalog of optical components shows expensive tables, hardware, and other components designed to minimize noise. The LIGO project to detect gravitational waves from collapsing stars might be the ultimate example of vibrational noise reduction.

4. BACKGROUND FLUCTUATIONS
   Sometimes, an experimenter has to deal with what we might describe as background noise. The best example we can think of is astronomical observations. Looking at stars always involves measuring starlight in the presence of light from our own earth sky. This can come from reflected city lights, moon light, clouds, upper atmosphere effects, etc. Clever experimental strategies can be very helpful here.

5. GROUND LOOPS
   This is a special, but important type of electromagnetic pickup of 60 and 120 Hz signals due to multiple grounding. This can easily be the dominant interference signal for low-level detection. This type of interference can usually be eliminated with proper design of the electrical grounds and, sometimes, careful shielding.

III. SIGNAL-TO-NOISE ENHANCEMENT

Now let's turn to the various strategies for signal-to-noise enhancement in real experimental situations. Since there are many techniques, we shall break them down into subsections and discuss them one at a time.

A. DISCRETE AVERAGING

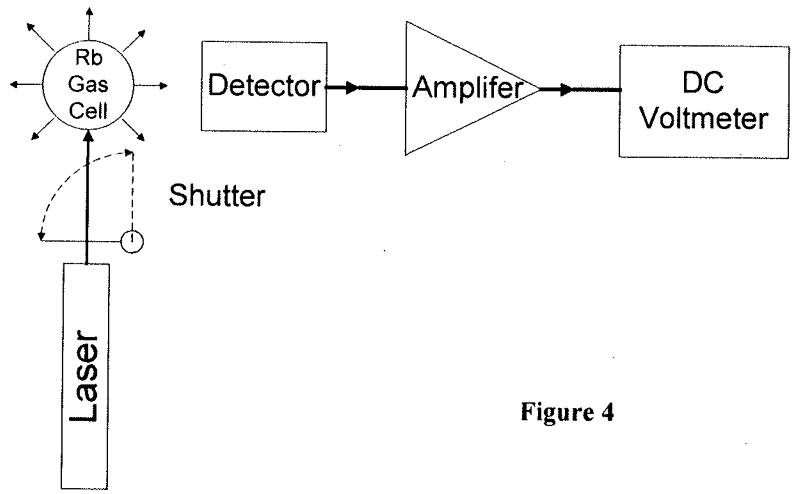

Figure 4

We will take as a given that the average value of any repeated measurement will be a number which more accurately represents the "true" value than any single measurement. This is, of course, only the case if each measurement has equivalent parameters. That is, there are no nonstochastic errors in each measurement. Let's examine a hypothetical experiment. Suppose we wish to study the fluorescence of rubidium atoms, in a vapor, which are being pumped by a CW laser. The block diagram is shown in Figure 4.

The fluorescent radiation appears in all directions, including at right angles to the incident radiation from the laser. With the shutter open, the light on the detector creates a dc voltage which is amplified and read by the voltmeter. The background can be taken care of by measuring the voltage with the shutter closed. As you watch the display on the dc voltmeter, it is always changing. What number should you write down? Well, the average number. But how does one determine the average? One method is to take a discrete average. Simply record the voltage a number of times.

Now we (or a computer in a real example, with hundreds of entries) do a simple mathematical average. We take the difference of the averages and obtain a reasonable average.

	Trial	Voltage Open	Voltage Closed
	1	6.50	.45
	2	7.15	.55
	3	7.01	.56
	4	6.89	.49
	5	6.96	.43
SUM		34.5	2.48
AVERAGE		6.90	.49
< Difference >	= 6.41		

Our intuition tells us that the more trials that are averaged, the more accurate the value obtained from the average. Of course, this is only true if the fluctuations are truly random in nature, and not some systematic variation in the apparatus, such as decrease in the laser power, or leaking gas cell, or fogging of the cell windows.

If the fluctuations are random stochastic noise, then it can be shown mathematically that the signal-to-noise ratio is proportional to $\sqrt{N}$, where N is the number of measurements sampled. This is because the signal sums as the number of samples, whereas the noise sums as the square root of N. The "random walk" problem has the same kind of behavior.[2] Thus:

$$\frac{Signal}{Noise} \propto \frac{N \cdot Signal}{\sqrt{N}} = \sqrt{N} \cdot Signal$$

So the signal-to-noise ratio improves as the square root of the number of samples measured. If each measurement takes a time τ, the total time T to do N measurement is:

$$T = N\tau \text{ or } N = \frac{T}{\tau}$$

The signal-to-noise is proportional to:

$$\frac{Signal}{Noise} \propto \sqrt{\frac{T}{\tau}} \cdot Signal$$

Thus, the signal-to-noise improves as the square root of the total time to do all the multiple measurement. If the experimenter averages over a hundred times longer time period, the signal-to-noise only improves by a factor of 10. This result is applicable even for electronic signal averaging, whether the signal is digitized first then averaged or is averaged with an analog signal averager.

B. ADDING MULTIPLE NOISE SIGNALS

All experiments have multiple noise sources, although often one of the noise sources is so large that the others can be neglected in any analysis. This large noise "dominates" the noise of the system. However, suppose we have a pair of comparable noise sources V_1 and V_2. How do we calculate the total noise from both sources?

If the noise sources are each truly stochastic and random, that is they are not correlated, then the noise adds in quadrature. The total noise is then given by:

$$V_{Total} = (V_1^2 + V_2^2)^{1/2}$$

The extension to three sources should be apparent. However, interference from 60 Hz pickup and 120 Hz pickup is usually not stochastic and <u>NOT</u> independent. These are typically highly correlated and thus do not add in quadrature as shown above.

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[2] A wonderful physical discussion of the random walk problem can be found in the famous Feynman Lectures Vol. 1, Chapters 6 and 41.

C. ANALOG AVERAGING

There is another, much more common way to average an analog signal. It needs neither an analog-to-digital converter nor a computer. It requires only two common passive elements, a resistor and a capacitor connected in the simple circuit shown in Figure 5. This circuit is found in some form in many kinds of electronic instruments. Let's examine it.

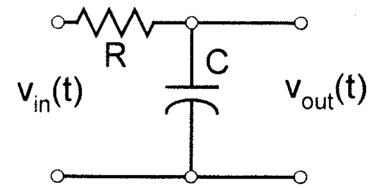

Figure 5

The differential equation describing the time dependence of the output voltage is

$$C\frac{dV_{out}(t)}{dt} = \frac{V_{in}(t) - V_{out}(t)}{R}$$

The solution for $V_{in}(t)$, an arbitrary function of time, (inhomogeneous differential equation) is:

$$V_{out}(t) = \frac{1}{RC}\int_{-\infty}^{t} V_{in}(t')e^{-\frac{(t-t')}{RC}}dt'$$

The equation shows that this circuit averages the part history of $V_{in}(t)$ with a weighting factor: $e^{-\frac{\Delta t}{RC}}$

As in the digital averaging case, the longer a signal with stochastic noise is averaged, the more the fluctuation in the measured average value is reduced. If the noise is truly random, this longer averaging produces a better average of the data. In effect, one is averaging over a longer past history of the signal. All this with two common, simple, passive components!

A second and equivalent way to think about this circuit is to examine its frequency response. In Figure 6 we have plotted the gain, $\frac{V_{out}}{V_{in}}$ as a function of frequency. The gain decreases by 3dB at a frequency $f_{3db} = \frac{1}{2\pi RC}$. This system of R and C is also called a Low-Pass filter. It "passes" (full gain) the low frequency components of the signal

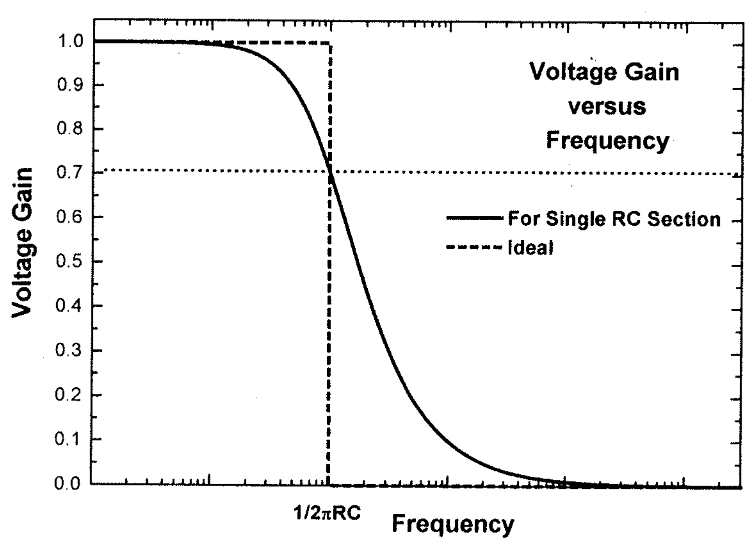

Figure 6

and reduces the amplitude of the high frequency components, where "high" and "low" are referenced to the frequency f_{3db}.

This filter enhances the signal-to-noise by reducing the high frequency components of the noise, while passing unaffected the low frequency signal (and noise). This low-pass filter is certainly capable of averaging our analog signals. The experimenter can use various values of R and C to change what is called the "TIME CONSTANT", RC. This changes the number of passed cycles averaged or the cut-off frequency of the filter (equivalent concepts). An ideal low-pass filter for reducing the bandpass yet retaining a dc and low frequency gain would look like the dashed line in Figure 6. It would behave like a step function. Such an ideal system can be approximated by the multiple section filter shown in Figure 7.

This is a three-stage filter with the unit gain operational amplifiers between the stages. Examining the filter characteristics of a one, two, or three stage filter in Figure 8 clearly shows the stepper "roll-off" response curve as the number of stages increases.

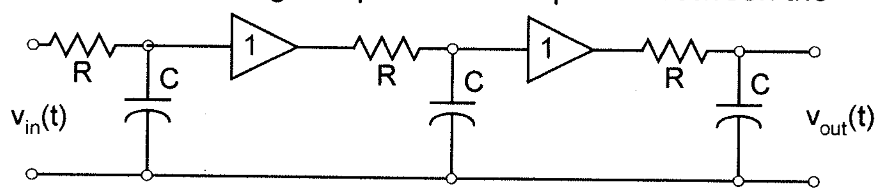

Figure 7

It would require an infinite number to produce the ideal filter.

1 stage	6 dB/octave roll off
2 stages	12 dB/octave roll off
3 stages	18 dB/octave roll-off

where an octave is a factor of 2 in frequency

Notice, in Figure 8, that the multistage filter decreases the high frequency response of the filter while not affecting the dc and low frequency response. Your lock-in has both a 6 dB and a 12 dB/ octave
filter which you will study.

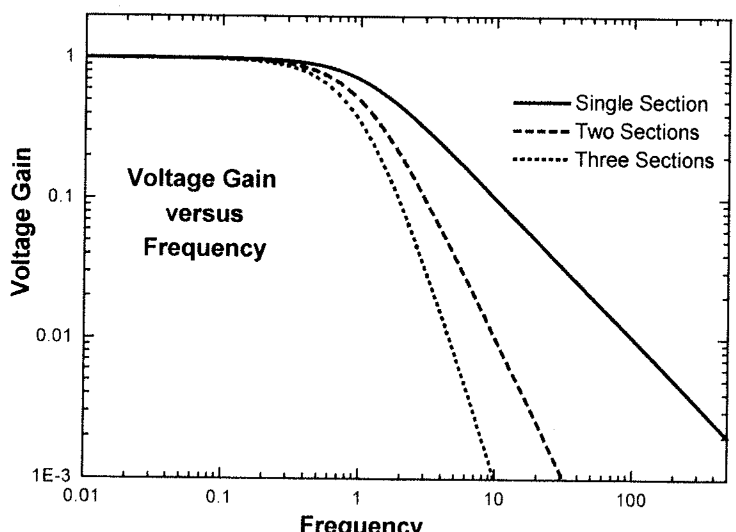

Figure 8

D. LIMITS

Certainly with modern computers one could digitize the signals and have a program that could do the arithmetic of averaging. Then why isn't this the preferred way to enhance signal-to-noise?

One answer is long-term drifts in the equipment. But changes in the background and transients in the interference signals are also a problem. These are not statistical fluctuations, and therefore violate the assumptions made in our analysis.

An even more fundamental problem with this simple approach is 1/f noise. This so-called flicker noise is ubiquitous in all kinds of electrical devices; detectors, amplifiers, discriminators, etc. It is clearly an advantage to process the signal in a frequency band where the noise from all components is minimum. DC is usually not that place (in frequency space). Usually, the experimenter can find a frequency space that will help optimize signal-to-noise and not violate any of the experimental parameters. These ac methods or coding techniques will be discussed in the section E.

E. AC METHODS

Suppose we collect data points from our Rb fluorescence experiment with our dc voltmeter. Then, we simply plot those points as a function of time on a graph. That graph might look like Figure 9. Notice that our imagined experimental signal is a composite of three signals: 1) The actual fluorescent light signal, 2) High frequency stochastic noise, and 3) Low frequency (dc) background drift. Now that's the signal we wish to process to enhance its signal-to-noise ratio.

Suppose we add another component to our fluorescence experiment. A chopper wheel is placed between the laser and the gas cell, so that the pump light is modulated on and off at 100 Hz. (Alternatively, one can imagine the shutter opening and closing at a rate of 100 Hz – not a practical idea, but this is a Gedanken experiment). The laser pumped light incident on the gas cell is now chopped.

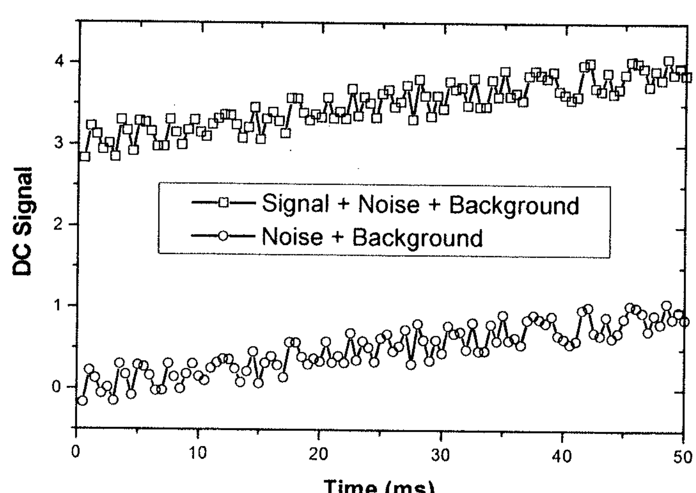

Figure 9

The signal output from the amplifier depends on the type of amplifier used. Let's assume we use a broad-band DC amplifier that has a bandwidth much larger than 100 Hz, so all frequencies pass. Then the output of the amplifier, with some attending noise, might look like Figure 10. There is some kind of signal even when the laser light is blocked off. This is a back-ground and noise signal.

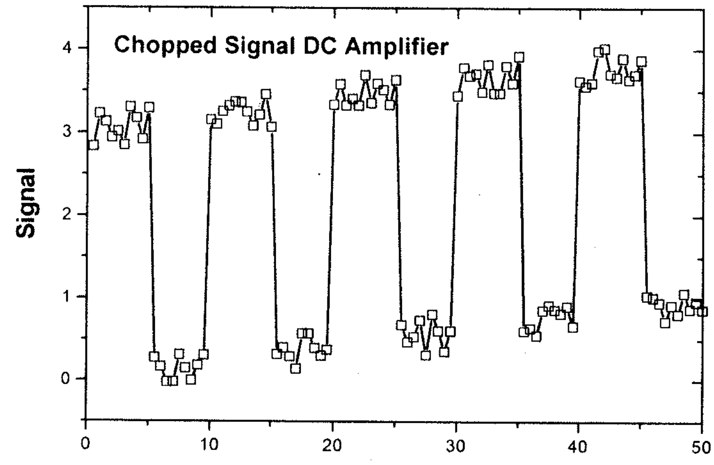

Figure 10

If we use an ac coupled amplifier the signal will look like Figure 11. The low frequency background drift has been removed, but the high frequency noise and signal are unchanged. Note the long-term time average of this signal must be zero, since it is the output of an ac coupled amplifier.

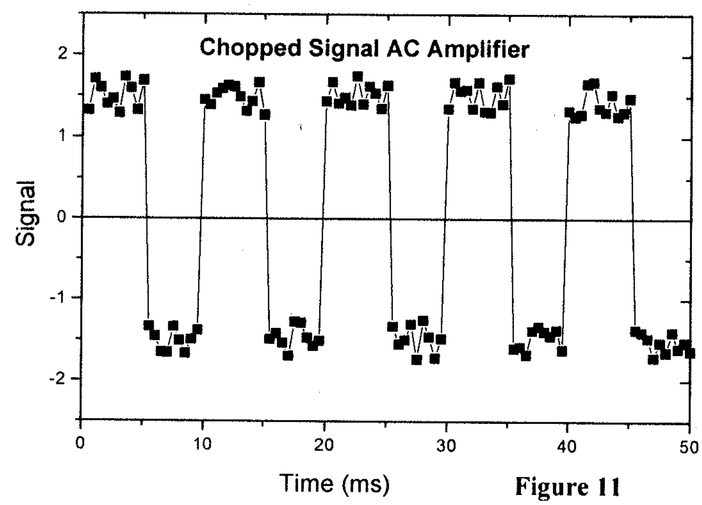

Figure 11

The question now becomes, how to measure the best value of the fluorescence light intensity using this chopped signal?

What are the advantages and disadvantages of using this ac signal? These are important questions whose answers require careful analysis. We will develop our answers in small steps.

Suppose we assume that flicker noise dominates this experiment, as it almost always does in most real experiments. If we had a spectrum analyzer that could measure the noise power as a function of frequency we might measure a noise spectrum like the one in Figure 3.

Examination of Figure 3 shows that the noise appears to become "white" at or beyond 10-20 Hz. A hundred hertz appears to be a reasonable choice for the modulation (coding) frequency. By chopping the laser light at 100 Hz we are coding the fluorescence signal at 100 Hz. As a result, the emitted light we are searching for will also be modulated at 100 Hz.

But if we are to average this chopped signal we must first develop an analog method to average an ac signal.

F. AC DETECTORS – PRECISION RECTIFIERS

Why did we bother to incorporate an expensive chopper in our laser physics experiment? Two basic reasons:

1. To get away from drift in the dc amplifier and light detector.
2. To try to reduce the 1/f noise inherent in various parts of the apparatus.

But we must be careful in our analysis, there are some subtleties which we must address:

What kind of amplifier are we using (ac or dc)?
What kind of averaging are we proposing to use at the output?
Are we incorporating an ac bandpass filter or any other kind of filter in the system?

1. DC AMPLIFIER

Suppose we use a dc amplifier in the experiment. Then the signal that passes to the low-pass filter looks like Figure 10. All the noise at all the frequencies passes to the low pass filter and there is no advantage, as far as signal-to-noise improvement, over the original system without the chopper using an analog (or digital) averaging system. In fact, the chopping actually degrades the signal-to-noise because the fluorescence is only present 50% of time. The average value will be about half of what one would measure with no chopper at all. This system does not enhance the signal-to-noise.

2. AC AMPLIFIER WITH PRECISION RECTIFIER

Suppose we use the apparatus shown in Figure 12. Here we have an ac amplifier, with some inherent bandpass, and a precision rectifier to convert the bipolar ac signal to unipolar ac. This rectified signal then passes through a low-pass filter.

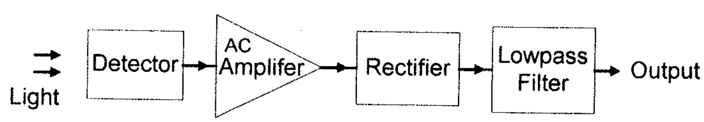

Figure 12

Let's examine the bandpass characteristics of our amplifier. They are shown in Figure 13:

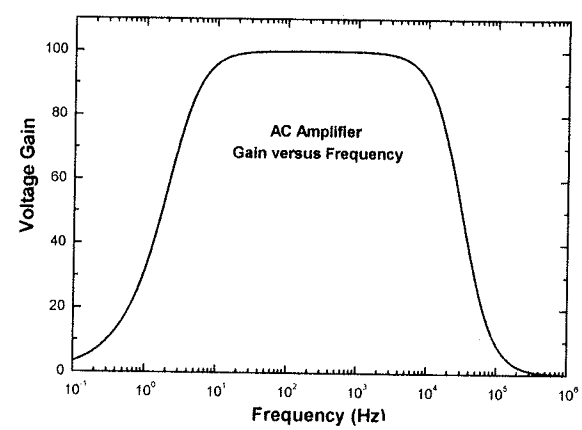

Figure 13

Since the amplifier has no gain at dc, it can only amplify the ac component of the signals it receives from the optical detector. The output of this amplifier might look like Figure 11. Now the advantage of the chopper can be seen. The dc drifts of the amplifier and other low frequency 1/f flicker noise is not amplified. Also, noise past about 10,000 Hz is not amplified. The ac amplifier acts, in fact, like a bandpass amplifier, reducing some of the noise.

However, the average of any ac coupled amplifier is exactly zero, since there must be equal areas under the curve for both positive and negative voltages. The precision rectifier (full wave) converts a bipolar signal to a monopolar as shown in Figure 14. This rectified signal has no dc drift, but the high frequency noise and the fluorescent signal are still present. However, it should be noted that the fluorescent signal has been reduced by a factor of 2, from about 3V to 1.5V. This signal can then be averaged by the final low-pass filter at the end of the instrument chain, which reduces the high frequency noise component.

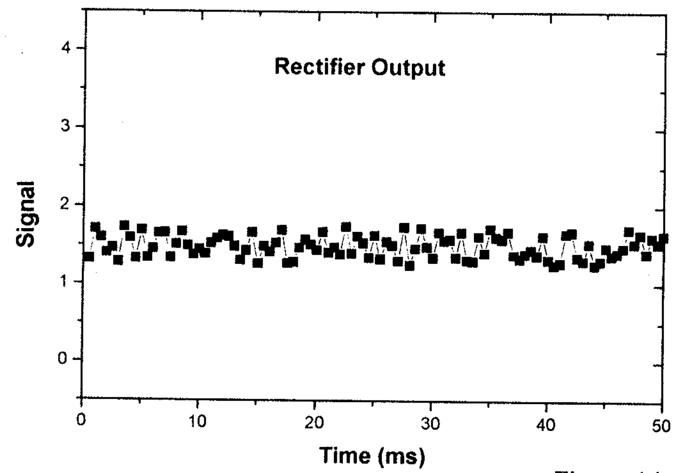

Figure 14

This system should improve the signal-to-noise of the experiment, with the improvement depending on the time constant of the final filter as well as the bandpass of the ac amplifier.

3. ADDITION OF A BANDPASS FILTER

Suppose that a narrow bandpass filter is now inserted in the system between the amplifier and the precision rectifier. The center frequency of this filter will be at the fundamental frequency of the light chopper. This new configuration reduces the bandwidth of the noise that is rectified and increases the signal-to-noise ratio.

The limitation of this method is due to practical consideration of a narrow band amplifier. A typical audio frequency narrow band amplifier has a quality factor between 20-100 where Q is defined as:

$$Q \equiv \frac{f}{\Delta f} \qquad \Delta f = BANDWIDTH$$

Not only are higher Q filters not practical, they have inherent problems of drift. That is, the center frequency changes with time, temperature, and power supply voltage. This drift creates another source of systematic non-stochastic noise, which degrades the signal. However, the reduction of noise bandwidth by the bandpass filter is an important and practical way to enhance the signal-to-noise ratio.

G. LOCK-IN DETECTION

Robert Dicke was one of the truly great experimental physicists of the twentieth century. He did major work in many fields but was most famous for his work on general relativity and radio astronomy. Many physicists (including the senior author) believe he should have received at least one Nobel Prize. He developed the lock-in amplifier originally for improving the signal-to-noise for radio astronomy signals. He helped start PAR (Princeton Applied Research) which was the first company to manufacture a lock-in amplifier. This instrument has become so important that it is reasonable to argue that he should have been given a Nobel on the basis of this device alone.

Before the lock-in was invented, one could build a system that would reject almost all "information" that was not at a particular frequency by inserting a very narrow band, stable, bandpass filter. But a coded signal has more information in it than just the frequency of the code; it contains <u>phase information</u>. That is, our fluorescent light not only is modulated at 100 Hz by the laser chopper, it has a <u>definite phase</u> relationship with respect to the phase of the chopper. To be specific, if the fluorescence occurs at a small delay time after the laser light strikes, then all the fluorescence light signal will be out of phase by a fixed phase angle, for all time.

Instruments, Prof. Dicke realized, could be designed to be selective on the phase as well as the frequency of the signal. The more the instrument is selective, the greater reduction in the noise, since the noise voltage is random in frequency <u>as well as phase</u>. He designed and built the first phase sensitive detector, the "father" of all modern lock-in amplifiers. How does it work?

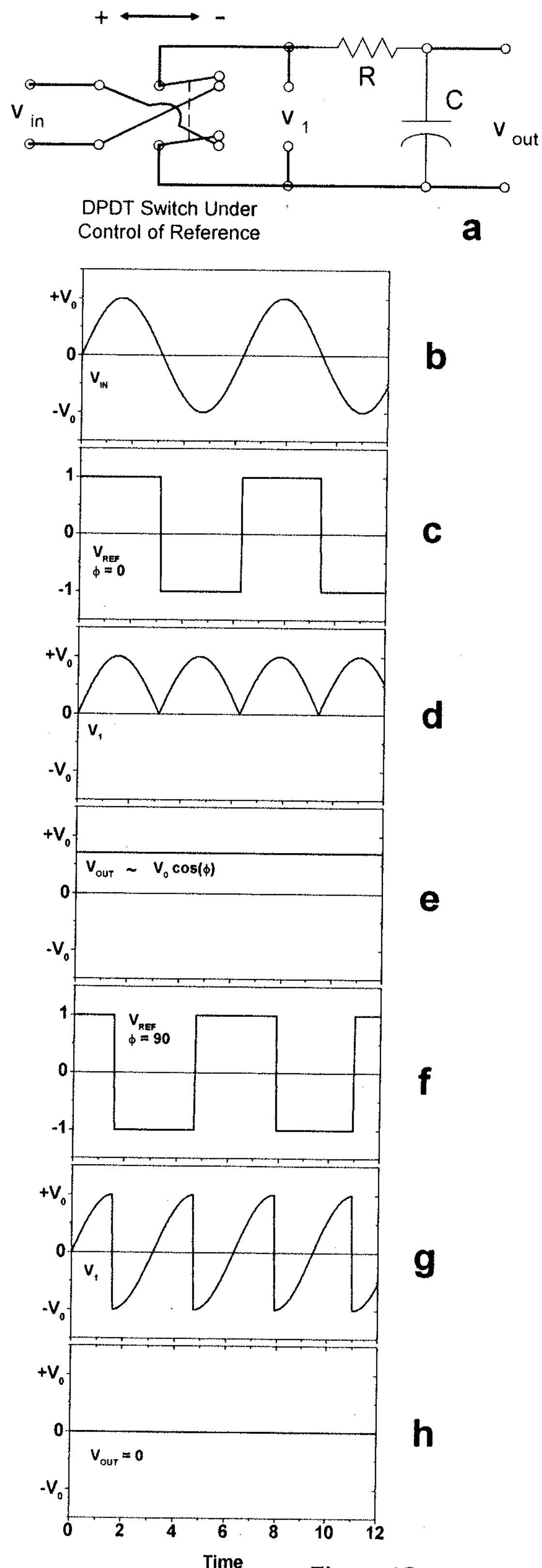

Figure 15

Consider the very simple circuit shown in Figure 15a. It consists of a double pole double throw switch and a single resistor and capacitor. The switch is wired to reverse the polarity of the incoming signal with every throw and this signal is presented to the resistor-capacitor combination. (This you recognize as a low-pass filter).

Suppose a signal of the form

$$V_{in} = V_0 \sin 2\pi f_{\text{mod}} t$$

appears on the input. If the switch is thrown as shown in the third drawing (c) where it is in the (+) position for the first half cycle and in the (-) position for the second half of the cycle and this pattern is repeated, then the voltage at V_0, will <u>always</u> be positive and will appear like Figure 15d.

If $RC \gg \frac{1}{2\pi f_{\text{mod}}}$, then the output signal will average the voltage V over many cycles and yield a dc level. This dc voltage is proportional to the magnitude of the input voltage as:

$$V_{out} \propto V_0$$

Now, suppose we change the timing on the switch. Consider the case where we delay the switching by ¼ of a cycle. That's equivalent to a 90° phase shift. This is shown in Figure 15f. What does the signal look like at V_1. We are now switching at the maximum and minimum of a cycle so that V_1 looks like Figure 15g. Notice that the average voltage at V_1 over many cycles is now exactly zero.(Figure 15h)

If one delays the timing by ½ of a cycle (180°) the average becomes $\propto$ - V_0; ¾ - cycles it again becomes zero. In all these experiments the input signal has remained the same. The only thing that was changed was the start time of the switch (its phase), not its frequency. A more mathematical analysis shows that the output voltage is given by

$$V_{out} \propto V_0 \cos\phi$$

where ϕ is the relative phase angle between the input and reference signals. This is a phase-sensitive or lock-in detector. ***It is sensitive to phase, even at the correct frequency. The output depends on this phase angle.***

How does the lock-in act selectively with respect to the frequency of the signal? That is, how does it reject a signal whose frequency is not the same as the reference frequency signal? We will not work out the mathematics of this process, but merely show two output signals from the lock-in when the signal frequency is greater than the reference ($f_{signal} > f_{ref}$) Figure 16 and when the reference frequency is larger than the signal ($f_{ref} > f_{sig}$) Figure 17. Close examination of these two signals will reveal that the average over many cycles of the output of the lock-in is zero. (Figure 17 is easier to interpret, with channel 2 the lock-in output) So the lock-in has both phase and frequency selectivity.

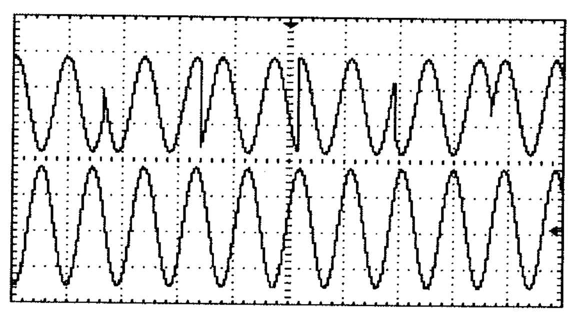

Figure 16

All this is nice for this hand-operated switch, but, you might ask, what this has to do with a "real" lock-in amplifier? Everything! Most modern electronic lock-in amplifiers are essentially the same circuit, except of course the switch is electrically operated by the reference input signal. So understanding this circuit is helpful to understanding a modern lock-in.

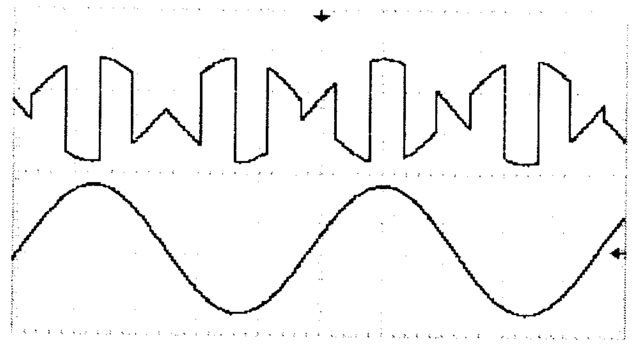

Figure 17

How does the experimenter "know" what the correct phase is for the experiment being performed? How is the reference phase set? Here is the procedure:

You code (modulate) the signal of interest – so you have access to the reference signal, both its frequency and its phase.
You have, in your lock-in, a phase shifter, so you can vary the phase of the reference signal with respect to the phase of the coded signal.
You must adjust the phase on a real signal to obtain the maximum output signal. This is the correct phase. To check, switch the phase by 90°. You should obtain zero output. Make fine adjustments of the phase to obtain the best zero.

In some experiments the coded signal phase can be calculated and the lock-in reference phase can be set from this calculation. In some modern lock-ins, the instrument itself can determine the correct phase by an internal searching procedure.

How are the ***noise voltages*** processed by the lock-in detector? In other words, what is the output of the lock-in in the absence of a signal? The lock-in rejects noise that is not at the reference frequency, with a bandwidth given by $\frac{1}{4RC}$ (6dB/oct) and $\frac{1}{8RC}$ (12 dB/oct). It, in effect, has a "built in" bandpass filter. What about the noise that is at the coding frequency (or more precisely within the bandwidth as set up by the low-pass filter)? Because the noise has a random phase with respect to the coding reference, the instantaneous output of the lock-in, after the low-pass filter, could be positive (in phase), zero (90° or 270° out of phase) or negative (180° out of phase). Given the random nature of noise phase, the average value of the output will be **zero!**

Contrast this with a detector scheme that has a bandpass filter, rectifier, low-pass filter configuration. In the absence of a fluorescent signal, the output has a finite value equal to the rms noise voltage present at the input to the rectifier. Comparing the amplitude and the lock-in detection schemes with the same bandwidth and with the rms signal voltage and rms noise voltage equal as well, the signal-to-noise of the lock-in detection over the amplitude detection is better by about a factor of 2. The signal output for both systems will be the same, with the same amount of output noise. But the noise in the absence of a fluorescent signal, will average to zero with the lock-in detector. For the amplitude detection, however, it will average to one-half of the signal.

Let's state all of this in another way. Given equal bandwidths, the noise output of each of detector is the same. However, because the average output of the lock-in is zero for pure noise, the *change* in the output when there is a "real" signal is larger for lock-in detection.

A third way to think about this lock-in enhancement of signal-to-noise is to consider the information in the signals. The real signal has both frequency and phase information, whereas the noise signal has only frequency, no phase information. Since the lock-in detector provides both phase and frequency discrimination, the signal passes through two "tests", and only the true signal passes both "tests". The true signal meets both criteria.

In many experimental situations, the signal is much bigger than the noise. Is there any advantage here to use lock-in detection? In theory, we would say no, but in practice, there certainly are advantages. There are serious technical problems with building stable high Q bandpass filters. Even if your filter is stable, your reference oscillator may not be. A frequency change in either the reference oscillator or the filter will severely corrupt the signal. However, the bandwidth of the lock-in is determined by the time constant of the low-pass output filter and one can easily configure the lock-in so that it has an effective Q of 1000. For example, a 1 second time constant and a coding frequency of 1kHz has an effective Q of 1000.

The students will actually perform a set of experiments which will demonstrate the advantage of a lock-in over a precision rectifier detector, examining a signal embedded in noise. Both types of detectors will be used, and the final output signal-to-noise compared.

There is a "price to pay" for this lock-in or amplitude detection signal-to-noise enhancement. The ac signal is gone at the output. The resulting signal is a dc voltage (with some low frequency fluctuation) proportional to the ac amplitude at the input. Using only a narrow bandpass filter, the ac signal would be preserved but the signal-to-noise enhancement would not be as great.

One last note. A single oscillator should be used for both the reference coding (modulating) and for the reference to the lock-in switch. This insures that there is no phase slippage between the reference and signal channel. TeachSpin's SPLIA-1 provides such an oscillator, but it is possible to use an external oscillator, so long as its frequency is within the bandwidth of the phase shifter.

IV. ELECTRONIC TERMINOLOGY

Before we examine the specifics of the TEACHSPIN apparatus, we want to assure that the students understand the common terminology used to describe electronic components and modules. We highly recommend reading section 6.4 of Moore, Davis, and Coplan, which is in the references. This gives an excellent discussion of amplifiers, which is more complete than this brief overview. We will give here a kind of glossary of important terminologies.

GAIN: This refers to voltage gain, the ratio of the output voltage to the input voltage. It is dimensionless.

INPUT IMPEDANCE: For this instrument we are only considering input resistance – which indicates how much current is drawn from the signal source. However, for ac coupling, the instrument uses a coupling capacitor, which could have a significant effect on the input impedance at very low frequencies.

DC OFFSET: This is an equivalent voltage, which, if applied to an ideal amplifier, would produce the measured output of the real amplifier. For example if the input of an amplifier with gain of 1,000 is grounded, yet the output was 5 volts, the dc offset would be 5×10^{-4} volts or 50 μV.

AMPLIFIERS: Beside the ac and dc coupled amplifiers, there are two types of amplifiers commonly used. They are voltage-to-voltage and current-to-voltage. In the first case, a voltage (or low impedance) source is amplified to give a voltage output. The second type, a high impedance or current source has a low impedance voltage output.

The frequency response or gain-bandwidth is also a very important characteristic of an amplifier. This can best be depicted by a plot of gain versus frequency as shown in Figure 18 for two amplifiers.

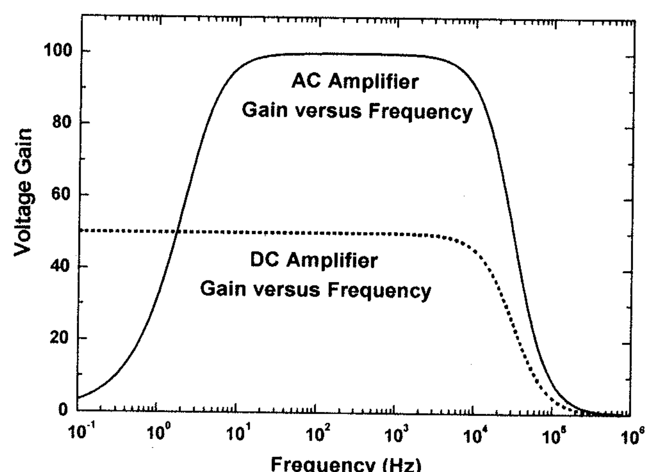

Figure 18

KRAMERS - KRONIG RELATIONS: This is the name of an important and fundamental relationship for all linear systems which has application for filters, amplifiers, detectors and other devices. When a linear system has a gain (or response) that changes with frequency, there is, usually, an accompanying change in the phase shift by the same instrument. (There are some exceptions, but this is a good rule of thumb) It is best understood by an example. In Figure 19 we have plotted both the amplitude and the phase shift response of a filter. Notice that when the amplitude or gain is independent of frequency, the phase shift is constant.

These ideas are important for lock-in detector, since the reference phase stability and signal phase stability are essential for accurate detections.

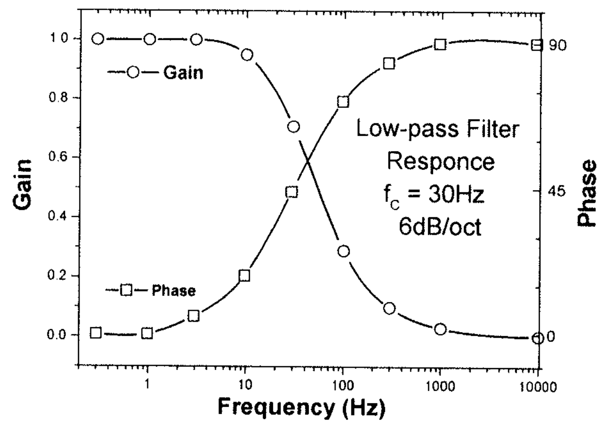

Figure 19

MAXIMUM INPUT VOLTAGE:
The preamplifier, filter, detectors, low-pass/output, and noise generator all have a maximum input voltage that they can accept without causing damage to the electronics.

MAXIMUM OUTPUT VOLTAGE:
The preamplifier, filter, detectors, low-pass/output, oscillator and noise generator all have a maximum output voltage before the output signal "***clips***". This causes a serious distortion of the signal and a corruption of the signal processing. A clipped output appears as in Figure 20.

Care must be exercised when noise voltages are being considered, because of the stochastic nature of the signal. Average noise voltages can be misleading, since it is the noise spikes that might be distorted.

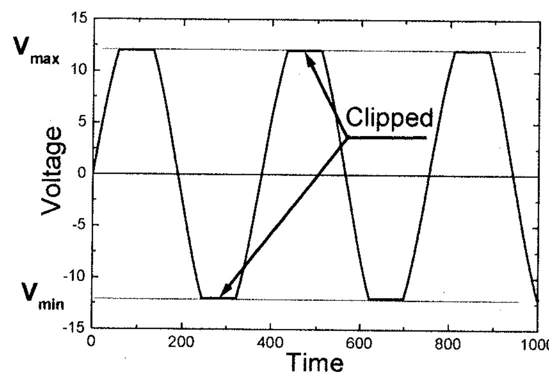

Figure 20

MAXIMUM OUTPUT CURRENT: The active electronic components have a maximum amount of current they can supply to the load on their output. This can be a serious problem with the reference oscillator, if the student tries to drive a low impedance load (such as the windings on an electromagnet) directly from the reference oscillator. The current can clip, just like the voltage.

SHORT CIRCUIT PROTECTION: The SPLIA-1 is protected against the student putting a very low impedance or a short circuit across the output of any module. Although this might cause the unit to exceed its current limit; it cannot damage the circuit. They are all protected against this error.

NOISE REFERRED TO INPUT: This is noise as if came from the input into an ideal device, particularly amplifiers. For example, if we had an amplifier with a gain of 10^3 and a noise voltage of 50mV, then the noise referred to the input would be 50μV.

V. SPLIA1-A, THE INSTRUMENT

A. MODULAR STRUCTURE

The SPLIA-1 is constructed with a modular format in a single box. Each module can be accessed individually. Each has its own input and output, although they all share a common power supply and common ground (except the output from the reference oscillator). One must treat each module individually, and must be aware of its specifications, and its limits, as well as its functionality. The student must make all connections between the modules using coaxial cables with BNC connectors. The cable provided for the output of the reference oscillator is supplied with ground isolation.

There are seven modules. This section will describe, in detail, each of the seven modules. The format for this will be (a) the Purpose (b) the Characteristics (c) the Electronic Specifications. Here, we will treat each module separately. In Section VI, we will discuss how to connect them and how to use them in various experiments.

1. PREAMPLIFIER

a. Purpose: To give voltage amplification to the experimental signal to a level where the noise voltage of any subsequent unit will not significantly effect the signal-to-noise of the final output.

b. Characteristics: This preamplifier can be operated in two different methods:

Single Ended – either input (+) or (-) can accept the signal and the other input is grounded.

With (+) input (-) ground it is a non-inverting amplifier

With (-) input (+) ground it is an inverting amplifier

Differential – In this mode the signal is fed into both the (+) and the (-) input and the output is proportional to the DIFFERENCE between the two inputs. The ground here acts only as a shield. The most common example of the usefulness of this operating configuration is the signals from a bridge circuit. This signal for the unbalance of the bridge is proportional to $V_+ - V_-$.

Another use for the differential input is when there is a large common mode interference signal on both sides of the line. The differential mode can greatly help reduce this unwanted signal by taking the difference.

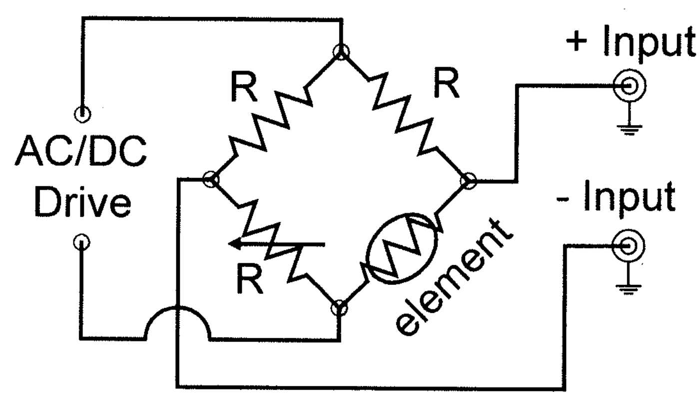

Figure 21

This unit can be operated as a dc or ac amplifier. In the ac mode the input is ac coupled with a series capacitor. Very low frequency (< 3 Hz) signals are amplified with reduced gain in this mode.

The voltage gain is variable over three orders of magnitude in steps clearly indicated by the dial on the unit. What is not so obvious is that the bandwidth of the unit changes when the gain changes. We will revisit this property when we measure the bandwidths of the preamplifier in Section VI.

c. Specifications

Input Impedance	1MΩ	Noise (@1kHz) RTI* 9nV / Hz ^ ½ (typical)
Maximum Input Voltage	±15V	DC Offset Voltage RTI* 30μV (typ.), 125μV (max.)
Maximum Output Voltage	±10.0V	Oscillator Feed Through @3kHz 20 nV rms
Maximum Output Current	18mA	Oscillator Feed Through @1kHz and less <10nV rms
High Frequency 3dB point		Common Mode Rejection (Gain = 10) 100 dB (min)
Low Frequency 3dB point (AC Coupling)	3Hz	*RTI = Referred to Input

2. FILTER

a. Purpose: To select the frequencies that will pass on to the detector (or to other modules) and to provide additional voltage gain.

b. Characteristics: This module has three types of filters, each with its own output BNC panel connector. These filters act like their name, as can be seen by studying their gain vs. frequency characteristics, shown in Figure 22.

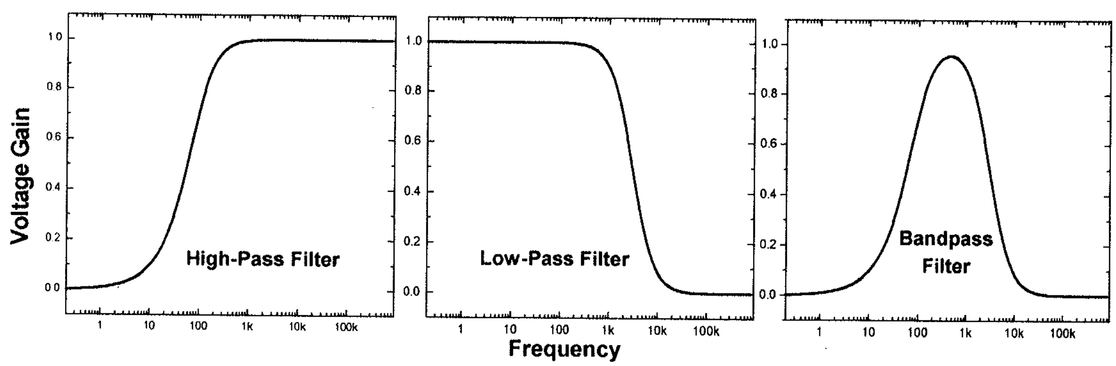

Figure 22

Which filter is appropriate to a given experiment depends upon the nature of the noise and interference one is attempting to minimize.

The dial on the center left side of the module marked Q allows the student to change the detailed characteristics of the response curves for all three types of filters. The number, 2, 5, 10, 20, 50 for the bandpass filter refer to the "Q" of the filter.

Q is defined as

$$Q \equiv \frac{f_c}{\Delta f}$$

where f_c is the center frequency and Δf is the full width at half power points of the band-pass filter as shown in Figure 23:

The higher the Q, the narrower the bandpass, and the more selective the filter.

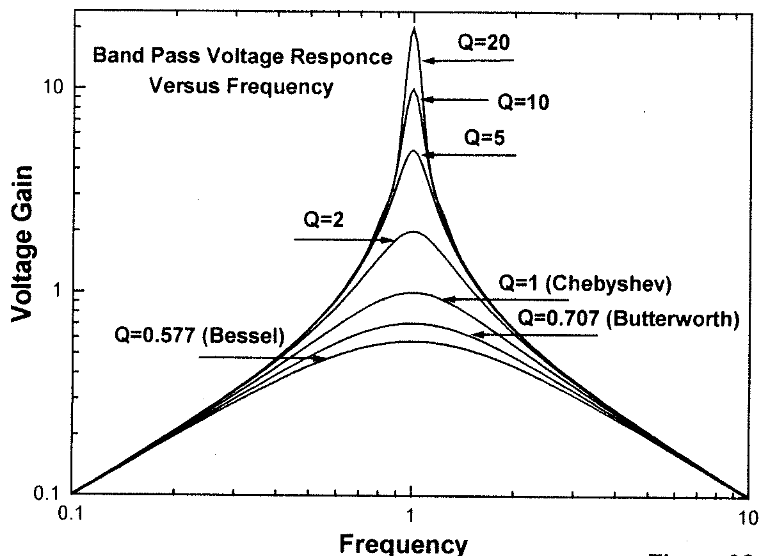

Figure 23

In Figure 24a, we have plotted the response curves for the bandpass filter for two values of Q. The relative phase shift of the signal as a function of frequency is shown in Figure 24b.

The "price" one pays for having a highly selective (high Q) bandpass filter should be clear from these two diagrams. In 24b we have plotted the relative phase shift of the bandpass filter as a function of frequency. Note the extremely rapid change in phase shift around the center frequency for a high Q filter. Phase stability is essential for phase-sensitive (lock-in) detection. These data clearly show that small fluctuations in the center frequency of the signal will cause large fluctuations in the phase shifts of the signal as it

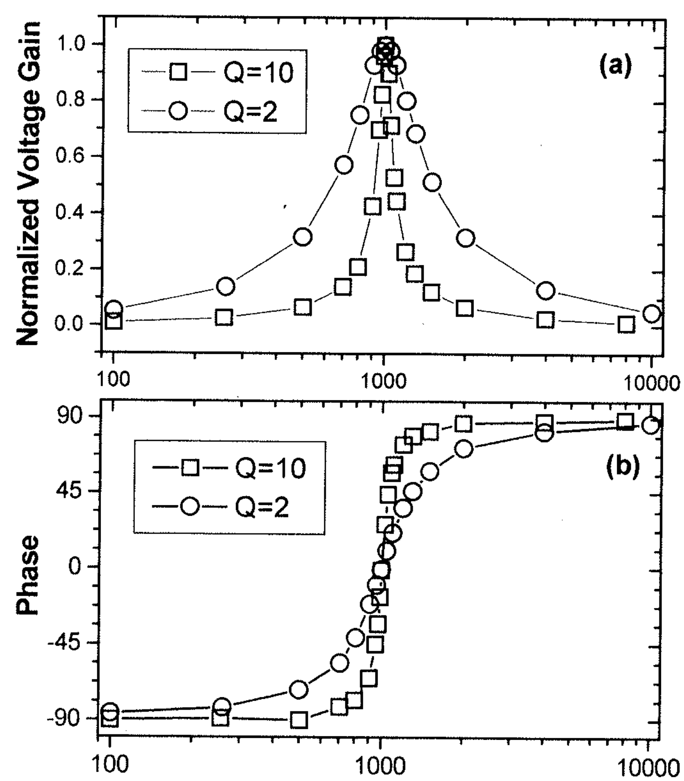

Figure 24

passes through the bandpass filter. This will corrupt the signal-to-noise enhancement. Practical considerations like this usually limit the usefulness of high Q filters in phase-sensitive detection.

CHEB, BUTT, and BESS stand for Chebyshev, Butterworth and Bessel. These are types of filters. Each has its own characteristic response curve and has different applications. These are best described in graphs of their response curves.

All the filters in SPLIA1-A are two-pole (also called second order) filters. In practical terms this means they all have two sections, each with a capacitor. The gain vs. frequency characteristics of second order filters "fall off" at a rate of 40dB/decade (12dB/oct) and the bandpass filter falls off at 20dB/decade (6dB/oct). (See Figure 25).

Note that we have plotted only the low-pass data since the high-pass curves are the mirror image of them.

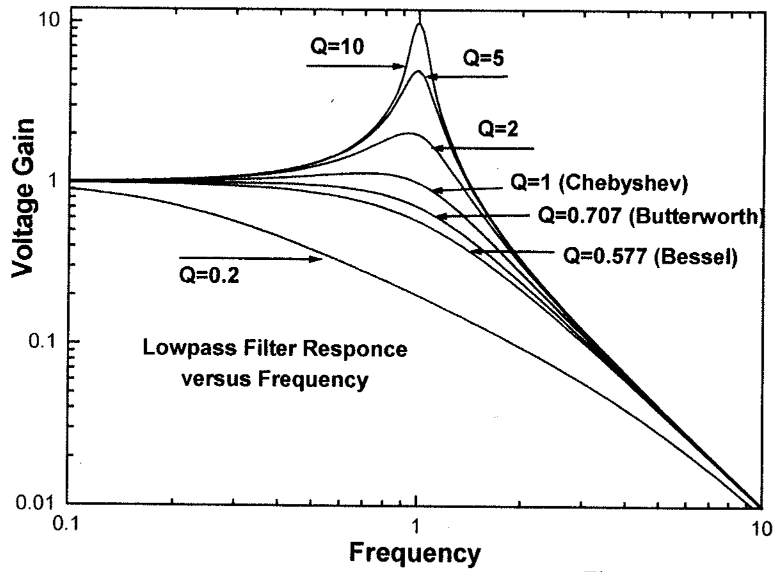

Figure 25

The Q determines the sharpness of the cutoff in both the high-pass and low-pass filter, as well as the narrowness of the peak in the bandpass mode.

Real experiments often have peculiar, idiosyncratic noise spectrums which must be addressed by the experimenter. Sometimes, it is essential to have a very sharp roll-off of the frequency response so that a noise signal can be reduced without affecting the signal transmission. In some cases, it is more important to have a flat response in a given region.

In applications where the sharp response is more important, a Q >.707 is used in both high-pass and low-pass filters. These are called Chebyshev. Although the scale of Figure 25 does not make it apparent, these response curves have some ripple. For Q =1 the Chebyshev has a ripple of about 1.25 dB.

For applications where flatness of the response is essential, then the Butterworth configuration is the proper choice.

Gain vs. frequency response is not the only consideration. Filters also have frequency dependant phase shifts. We have already discussed this issue with regard to lock-in detection and phase stability. But non-linear phase response can also cause problems for transient signals. The various Fourier components of these transients will have different time delays when passing through a non-linear phase response region of a filter, and exit distorted. In general, the sharper the cutoff, the larger the distortion.

A Bessel filter has the most linear response and thus produces the least distortion of the transient signal. In Figure 26 we show a square wave input signal and the output of the Bessel and Chebyshev low-pass filters.

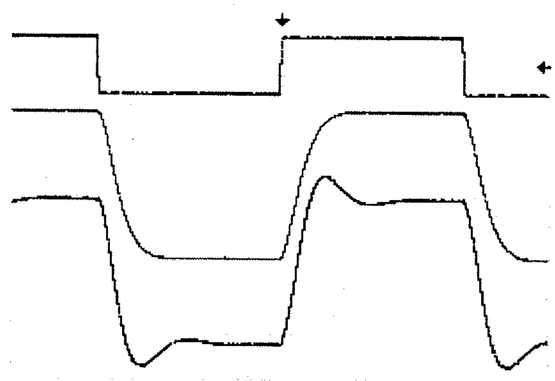

Figure 26

Notice that there is no ringing on the Bessel figure output, but the rise time of the signal does not match the input signal. The Chebyshev reproduces the rise better, however it overshoots and shows significant transient oscillations.

c. Specifications

Input Impedance 5 M Ω	Voltage Gain in Pass-Band of LP & HP -1V/V (inverting)
Maximum Input Voltage ± 12.5 V	Voltage Gain at Center Freq. of BP = Q V/V (inverting)
Maximum Output Voltage ± 10.0 V	LP and HP 2 Pole Filter Falls off at 20dB/dec.
Maximum Output Current ± 45 mA	BP Filter Falls off at 10dB/dec.
High Frequency 3dB point HP 2mHz	Input Offset Voltage (RTI) 300μV
Frequency Range 3Hz – 3kHz	Oscillator Feed Through 2μV rms @ 3kHz
Q .577, 0.707, 1, 2, 5, 10, 20, 50	

3. DETECTORS

a. Purpose: To rectify the ac signal, either synchronously, as a lock-in detector, or asynchronously, as an amplitude detector (sometimes called a precision rectifier).

b. Characteristics: With the flick of the toggle switch in the lower right hand corner of the module, one can change this unit from an amplitude detector to a lock-in detector. Let's examine the characteristics of each.

As an ***amplitude detector,*** the unit simply rectifies the ac signal, without any phase coherence to the coding (the reference oscillator). Consider the ac signal, Figure 27a. The output of the amplitude detector is a full wave rectification of this waveform, Figure 27b. It should be noticed that the output is negative.

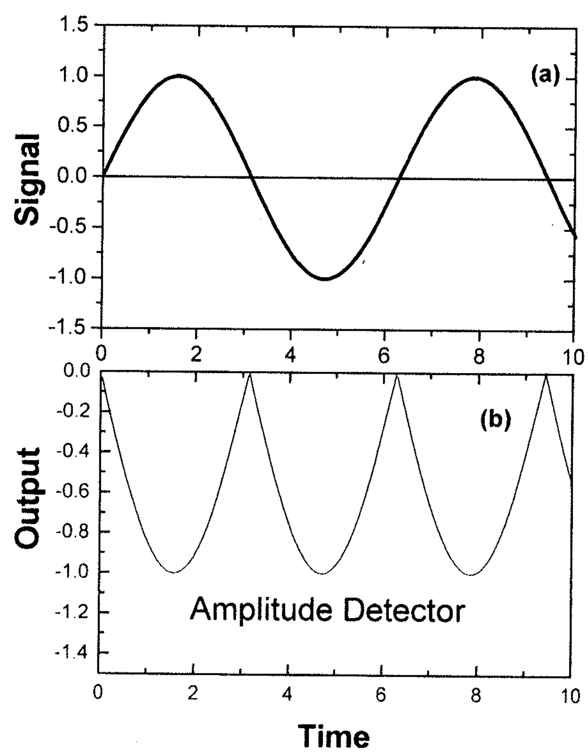

Figure 27

This unit has variable voltage gain from 2 to 2×10^3. It can be ac or dc coupled, so it can rectify signals down to ultra-low frequencies. In most applications, ac coupling is used. The gain-bandwidth plot is similar to that of the preamplifier. As the gain increases, the high frequency 3dB roll-off point decreases in frequency. (see specifications) Thus, the bandwidth of the amplifier does depend on the gain.

To use the unit in ***lock-in*** mode requires a flip of the switch as well as the introduction of the essential reference signal. The SPLIA1-A provides that signal from the reference oscillator, or it can be provided by an external oscillator. In either case, the reference signal must be sine wave, which has passed through the phase shifter module. This signal produces the coherent switching signal for the lock-in detector.

The output of the lock-in depends on the phase of the reference signal with respect to the phase of the input signal. If the two are in phase, the lock-in output looks the same as the amplitude detector and if the two are 180° out of phase, the output signal has the same appearance, but is of opposite polarity. For signals 90° out of phase, the detector output appears as shown in Figure 28.

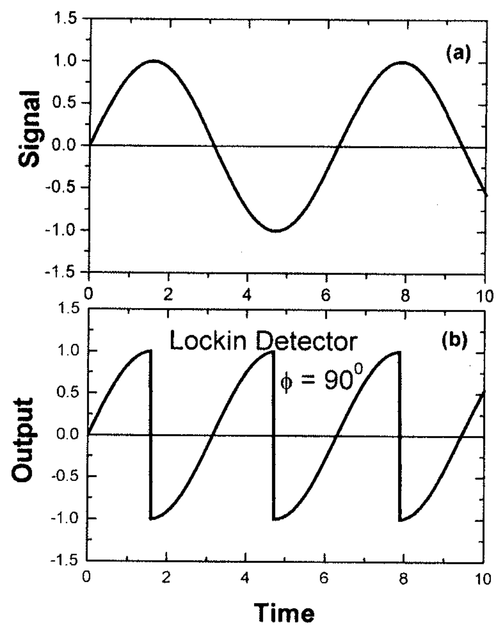

Figure 28

The student must take care not to overload either detector with too large of a signal. This can be a source of systematic errors, particularly when the signal has a large random noise component. Noise spikes not only occur randomly in time, but also in amplitude. A large noise spike that overloads the detector can corrupt the signal averaging process. It is the peak noise signal, not the rms voltage that needs to be considered.

To understand the working of the Lock-In/Amplitude (Figure 15a) switch we recall the DPDT switched lock-in example in Section III-G. With the Lock-In/Amplitude switch on lock-in, the DPDT switch is under the control of the reference input, and changes state (positive to negative) when the reference passes through zero. With the lock-in/amplitude switch on amplitude, the DPDT switch is under the control of the input signal itself. When the input signal crosses zero, the DPDT switch changes state, and the output changes polarity. With a little thought you can understand how this action rectifies the input signal.

When the DPDT switch changes polarity, there is a spike generated at the output. (There is not really a DPDT switch within the detector module, but electronics which mimic the DPDT switch). The switching spike is positive going, has a peak height of about 50 mV and lasts about 0.5 μs. These spikes do not cause much of an error when the detector is in lock-in mode. This is because of the low duty cycle. (Spikes last for 0.5 μs out of minimum period of 330 μs (3kHz. modulation frequency). However, when the detector is in Amplitude mode, and is used to measure high frequency noise voltages, the spikes can cause an error. Having high frequencies present when measuring noise means that the detector will be switching many more times per second, up to 300kHz. For this reason, the output from the detector should be kept above 100mV when measuring noise. Because the spikes are mostly positive going, if the output voltage is not kept above 100mV, the amplitude detector can have a positive average output voltage. The output is supposed to be negative! The presence of a positive average output voltage indicates that the switching spikes are dominating the noise measurement and more gain needs to be applied before the detector. This is one reason why the output voltage from the amplitude is configured to have a negative polarity.

c. Specifications

Input Impedance	100k Ω			Low Frequency 3dB point (AC coupling) = 3Hz
Maximum Input Voltage	± 12.5V			Reference Switch Window ± 2mV
Maximum Output Voltage	± 10V			Reference Switching Spikes 50 mV pp for 0.5 μs
Maximum Output Current	± 35mA			Input Offset Voltage 30μV
High Frequency 3dB point				Oscillator Feed Through 2 μV rms @ 3kHz
Gain = 2	650 kHz	Gain = 100	44 kHz	
Gain = 5	420 kHz	Gain = 200	22 kHz	
Gain = 10	250 kHz	Gain = 500	9.8 kHz	
Gain = 20	175 kHz	Gain = 1000	4.8 kHz	
Gain = 50	88 kHz	Gain = 2000	2.4 kHz	

4. LOW-PASS FILTER/OUTPUT

a. Purpose: To average the rectified signals coming from the detectors. This can be done with various time constants (averaging time) and with two different roll-off characteristics (6dB/oct and 12dB/oct). Because this module has a dc amplifier as well as a dc offset control, it is possible to study small changes in large signals. This can be accomplished by subtracting out the large dc signal, and amplifying the difference signal around zero to search for small changes. The unit also has a meter drive circuit along with an analog meter to monitor and measure the dc output of the signal processor.

b. Characteristics: The low-pass filter is characterized by the gain vs. frequency plot. In Figure 29, we have plotted these for both the 6dB/oct and 12dB/oct roll off settings on the unit.

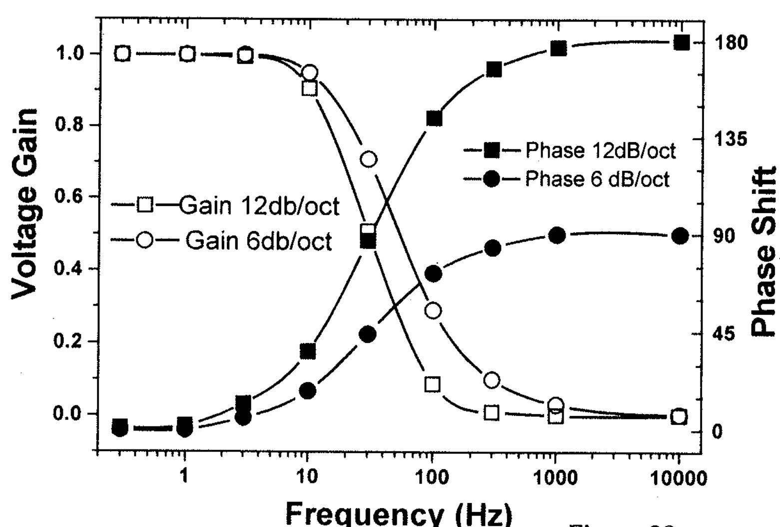

Figure 29

It is clear from this plot that the 12dB/oct filter reduces the bandwidth more than the 6dB/oct setting. The 12dB/oct filter has the sharper cutoff, but retains the same gain at dc and ultra low frequencies. Changing the time constant even more significantly changes the bandwidth of the signal and thus changes the averaging time.

The 6dB/oct roll-off still has important applications. In Figure 29, we have plotted both the gain and the phase shift as a function of frequency for the low-pass filter. Notice that past 1000 Hz the phase shift is flat, never exceeding 90°. That's important and unique to the 6dB/oct one-pole filter. The 6dB/oct filter is used in feedback circuits where stability against oscillation is crucial. A phase shift of 180° (with a gain greater than one) will produce feedback oscillation. Thus, a 6dB/oct roll-off filter will help protect against transient oscillation of the entire feedback loop.

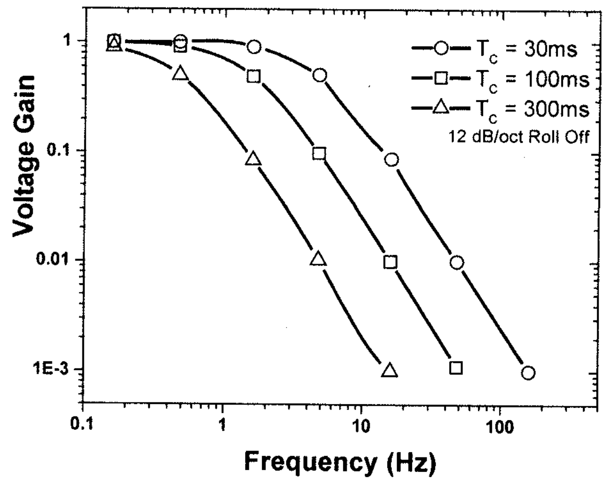

Figure 30

Figure 30 shows the gain-bandwidth of this low-pass filter for three time constants. All the curves are for a 12dB/oct roll-off setting. These characteristics are important because they determine the time necessary to remain at one experimental "setting" to keep the output of the filter in quasi-equilibrium.

The unit has a dc amplifier as well as a dc offset control. The offset bias is inserted before the dc amplifier so that the dc signal at that point can be adjusted to zero. That allows the experimenter to use large dc gain and look for small changes in large dc signals that might be of physical origin. *This module can be used to average any dc signal from any experiment.* (Remember, the average of any ac coupled signal must be zero, since a capacitor conducts no dc current across it).

c. Specifications

Input Impedance	1 MΩ	DC Offset Voltage	±10V
Maximum Input Voltage	± 12.5V	Input Offset Voltage	100 μV
Maximum Output Voltage	± 10.5V	60 Hz Pickup RTI	60 μV p-p
Maximum Output Current	± 35mA		

5. REFERENCE OSCILLATOR

a. Purpose: It provides the "coding" or modulation signal for the experimental apparatus. This oscillator is also the source of the reference signal for the lock-in detector. It can also provide a test signal that students may use to study a variety of signal processing configurations.

b. Characteristics: This module provides a variable frequency (3Hz to 3kHz) as well as a variable amplitude sine or square wave signal. The frequency is changed by a combination of a step switch and a 300° rotation of an analog dial. Note that there are two different scales, coaxially printed on the analog dial. A toggle switch in the lower right side of the module changes the signal outputs from a sine to a square wave.

Two outputs are provided. One goes directly to the phase shifter, and the second can be used either for external modulation (coding) of experimental apparatus or to provide a test signal. A toggle switch attenuator provides additional attenuation by a factor of 100 for the left side output. This switch is particularly useful when this output is being used as a test signal.

IMPORTANT NOTE: The left side output BNC connector of this module (to either the noise generator or to the experimental apparatus) is isolated from the common ground of all the connector bases. It is not metallic.

Do not connect the "ground" (return) of this output to the other common ground of any of the other modules.

Although violating the above injunction will not damage the SPLIA1-A, it will cause pickup of the modulation signal in the other modules. This may create unintended signals which will be phase coherent to the modulation and will corrupt the signal-to-noise enhancement.

The reason for this corruption can be traced to significant ground currents which produce microvolt signals in the ground lines if the oscillator signal returns through the common ground. Keeping the high level output signals isolated from common ground allows all the return current to flow back to the oscillator through its isolated ground, and eliminates this pick-up problem.

It is important to design the experimental apparatus in such a way that the modulation signal does not have a ground return to the SPLIA1-A. For example, such an "accidental" ground return might be through the preamplifier input. Check your circuit carefully. One way to eliminate such a ground return is by using a transformer to couple the modulation into experimental apparatus. There are also optocouplers as well as other tricks to isolate one ground from another.

TeachSpin provides a special coaxial cable in which the ground shield is broken. The distinctively marked cable is to be used in experiments where the output is a test signal.

c. Specifications:

Frequency 2.6 Hz to 3.2 kHz	Amplitude vs. Freq. 5% within range / 10% over entire range
Harmonic distortion 0.3% @ 3Hz < 0.03% 30 Hz and greater	Frequency Stability 200 ppm / C° at HF end of range
Maximum Output Voltage 4 Vp-p (sine) 8.8 Vp-p (square)	800 ppm/ C° at LF end of range
Maximum Output Current 35mA	

6. PHASE SHIFTER

a. Purpose: To provide a shift in the relative phase of its input to its output of 0°- 360° for ac signals from 3 Hz to 3 kHz.

b. Characteristics: The only function of this module is to shift the phase of the ac reference signal. The unit has a step switch that changes the phase in steps of 90°. It also has a fine control that changes the phase continuously from 0° to 90°. This unit does not provide an absolute phase reference. That is, it only provides a relative phase shift from the input to the output. The absolute phase shift depends on the reference frequency. The relative phase shift is independent of frequency. It is worth the student's time to examine this module using a dual trace oscilloscope, with channel one on the input and two on the output. The phase shifts created by the module are immediately apparent.

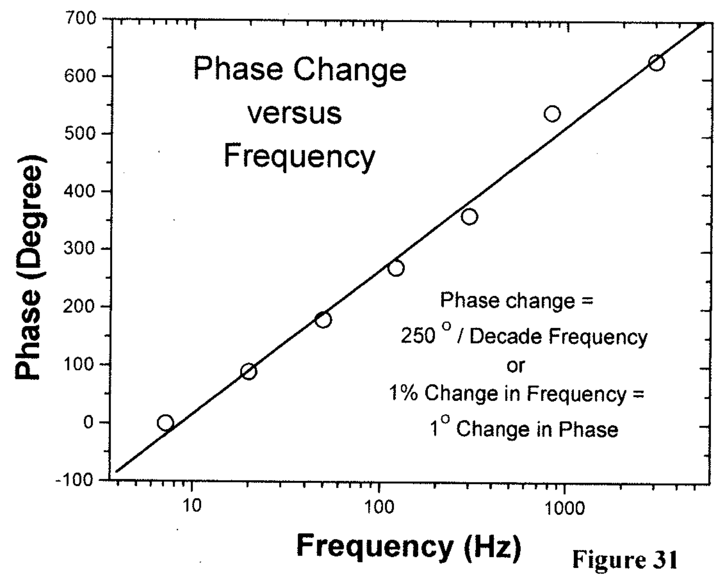

Figure 31

c. Specifications:

Input Impedance	50k Ω	Maximum Output Current	±35mA
Maximum Input Voltage	±12.5V	Frequency	3Hz – 3kHz
Maximum Output Voltage	±10.5V	Quadrature Phase Accuracy	±2 degrees

7. NOISE / ATTENUATOR

a. Purpose: To provide a noise signal, a calibrated attenuator, and a test signal (combining the reference oscillator signal with the noise source) for various experiments.

b. Characteristics: This unit provides an approximately white noise source whose amplitude can be varied over five orders of magnitude. The noise is only "white" over a frequency range shown in Figure 32.

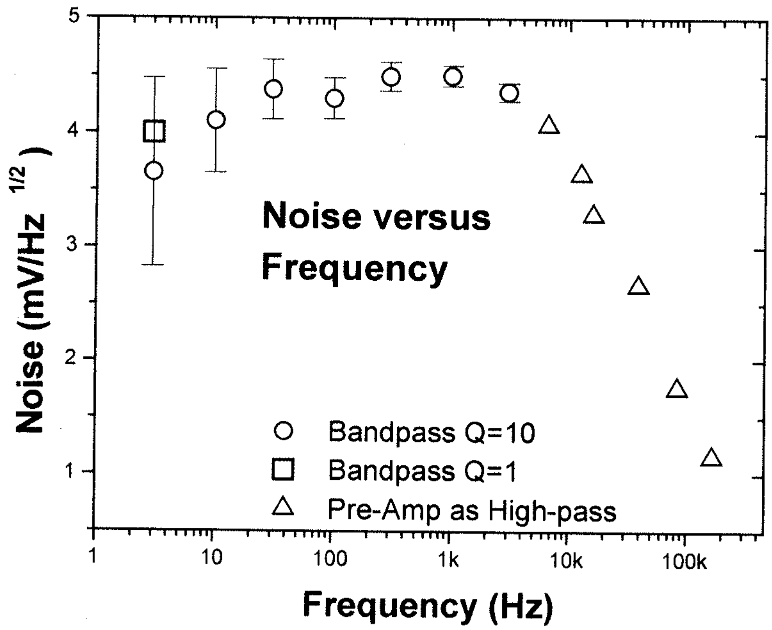

Figure 32

The test signal is obtained by summing the reference oscillator's output (through the ground isolated cable) with the Zener diode noise source inside the module. Both the signal and the noise can be varied over a wide range. A signal-to-noise ratio of 10^{-3} is easily obtained and examined on an oscilloscope. Although the signal cannot be seen on the oscilloscope, the lock-in detection configuration can extract the signal from the noise.

c. Specifications:

Input Impedance	1.1M Ω	High Frequency 3 dB. 130 kHz
Maximum Input Voltage	± 12.5V	Noise Bandwidth – See Figure 32.
Maximum Output Voltage	± 10V	Oscillator Feed Through 3.0 μV rmsl
Maximum Output Current	± 45mA	

VI. EXPERIMENTS WITH VARIOUS CONFIGURATIONS

In this section we will describe various experiments that students can perform with the SPLIA1-A. They require a dual trace oscilloscope, which could be either analog or digital, with a bandwidth of at least 500 KHz. For some experiments, an external oscillator function generator, whose frequency range is from 1 Hz to .5 MHz, is necessary. An ac multimeter is sometimes useful, but most digital scopes have the necessary built-in measuring capabilities.

We will divide this section into two parts. In the first part we the experiments will involve only individual modules. Measurements will be made with each module to provide hands-on understanding of its function. Students can compare their own measurements with the specifications in this manual.

The second part will focus on modules connected together to form some type of a signal processor. The student will study the efficiency of each configuration and how well it enhances signal-to-noise.

A. INDIVIDUAL MODULE EXPERIMENTS

1. PREAMPLIFIER:

Gain-Bandwidth Characteristics:

We will measure the frequency response of the preamplifier for various gain settings. It is important for the student to understand that the preamplifier's frequency response depends on the gain settings. Since the frequency response of this module exceeds the range of the reference oscillator, an external signal generator is required. A block diagram of the setup is shown in Figure 33:

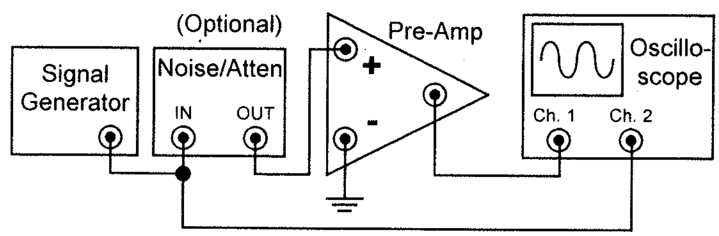

Figure 33

With this setup, the student can measure both the small signal gain and the phase shift of the preamplifier. The signal is fed into the + input and the – input is grounded. The + input is dc coupled. The data should be similar to our measurements shown in Figure 34. (Please note that the optional signal attenuator has a high frequency 3dB roll-off at 100kHz, so it cannot be used for high frequency measurements.)

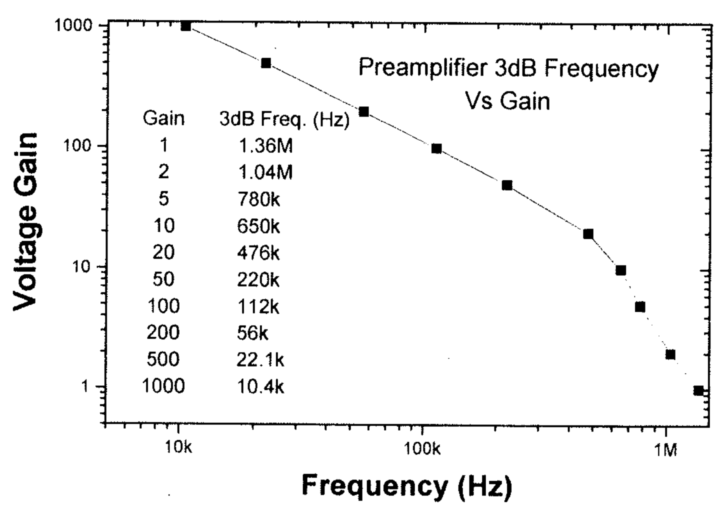

Figure 34

These measurements could also be made for large amplitude input signals.

Without changing the experimental setup, a student can look at:
1. Clipping of the output when amplifier is overdriven
2. DC offset, with inputs shorted

Common mode rejection can be studied by putting the same signal into both (+) and (-) BNC inputs (using a BNC Tee) and measuring the output as a function of input voltage. If it were perfect, the output would always be zero.

2. FILTER

This is one of the most important modules of this instrument. However, the switch labels are not entirely self-evident. It is essential that the students become familiar with the operations of this module. A dual trace oscilloscope, and a function generator are required to examine either the very low frequency or very high frequency response of the filter. The experimental setup is shown in Figure 35.

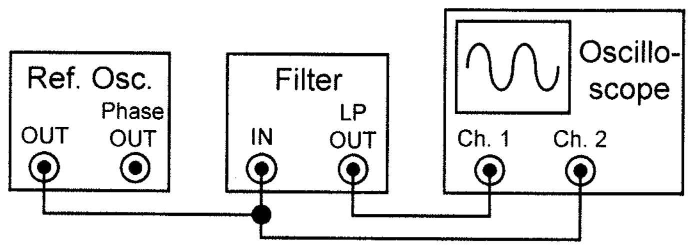

Figure 35

BANDPASS MODE

Students should measure the gain as a function of frequency at some midrange frequency, such as 300 Hz for various values of Q = 2, 10, 50. On a parallel graph they should plot the relative phase shift as a function of frequency. This will confirm the Krammers Kronig relationship for linear systems.

Students should also examine the effect of changing frequency. These same measurements should be made for the settings CHEB, BUTT, and BESS.

The bandpass filter can also be used as a *"Poor Man's Spectrum Analyzer"*. In case the student is not familiar with a spectrum analyzer, the faculty should explain that it is an electronic instrument which processes electrical signals and presents them in the form of spectral power as a function of frequency. That is, it will indicate how much of the signal being analyzed is concentrated in a given frequency range. These are wonderful, but very expensive devices.

Manually scanning the frequency of the bandpass filter, a student can obtain the frequency spectrum of an unknown signal over the frequency range from 3 Hz to 3KHz. This is not a large range, but it can demonstrate the usefulness of spectrum analyzer data by obtaining important information about 1/f noise, as well as 60 Hz and 120 Hz interference.

TUNING THE BANDPASS FILTER

Because the dials on the filter and oscillator controls are only approximately calibrated, it will often be necessary to tune the bandpass filter to the correct frequency using an oscilloscope. The oscillator is feed into the filter and both the oscillator and filter outputs are observed on the scope. When the filter is tuned to the oscillator frequency, the two signals will be in phase. (Actually because the filter inverts the signal, the two signals will be 180° out of phase). This is only true for the bandpass output. The low-pass and high-pass outputs will lead or lag the input by 90° when tuned to the same frequency. For this reason it is always easiest to tune the filter with the bandpass output, even if the low-pass or high-pass output is going to be used in the measurement.

LOW-PASS MODE

The curve of the gain as a function of frequency should be measured by students for the three settings CHEB, BUTT, and BESS. The same gain vs. bandwidth could be measured for a Q = 20 in the mid-frequency range.

HIGH-PASS MODE

Ideally, all of the gain vs. frequency curves determined for the low-pass mode should be measured for the high-pass mode. At a minimum, one or two values should be measured carefully. The rest can then easily be inferred.

3. DETECTORS

This module contains two detectors that can be accessed by the flick of a switch.

AMPLITUDE DETECTOR

This is nothing more than a precision rectifier. It can be examined with a dual trace oscilloscope, using the setup shown in Figure 36.

The reference oscillator will be used to provide a dummy signal. Students should examine the waveform of the input and output of the detector at a few settings of the gain.

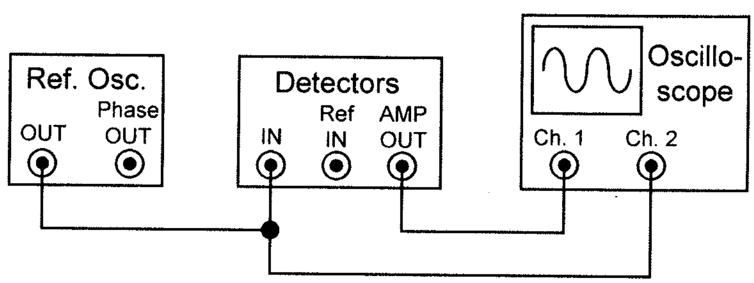

Figure 36

The output of the amplitude detector should appear as shown in Figure 27.

If the input is shorted, the dc-offset voltage can be measured. The overload characteristic should also be looked at.

LOCK-IN DETECTOR

The lock-in detector has a more complicated behavior, since its output depends on the relative phase of the input signal and the reference signal. The experimental setup is shown in Figure 37.

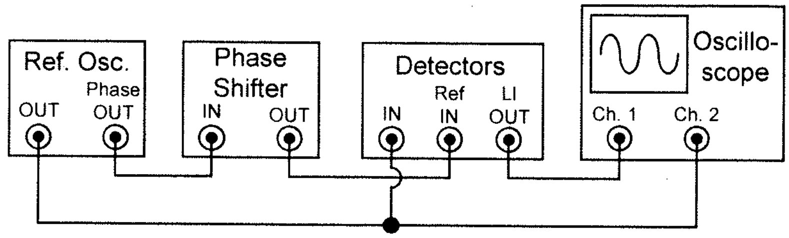

Figure 37

Students should examine the output waveform of the lock-in detector for phase shifts of 0, 45°, 90°, 180°. The lock-in output for 90° is shown in Figure 28.

Students might also examine the overload voltage, where the input voltage distorts the output signal, thus corrupting the lock-in detection.

4. LOW-PASS FILTER/OUTPUT

In order to examine the frequency response of this module the students will need a standard function generator. This will have a very low frequency signal which will be needed to look at long time constants. Students should measure the gain of the filter for various time constants and for both the 6dB/out and 12dB/oct roll off. The experimental setup is shown in Figure 38.

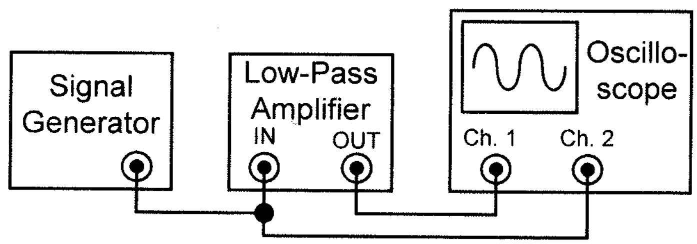

Figure 38

The data should be similar to the curves shown in Figure 39.

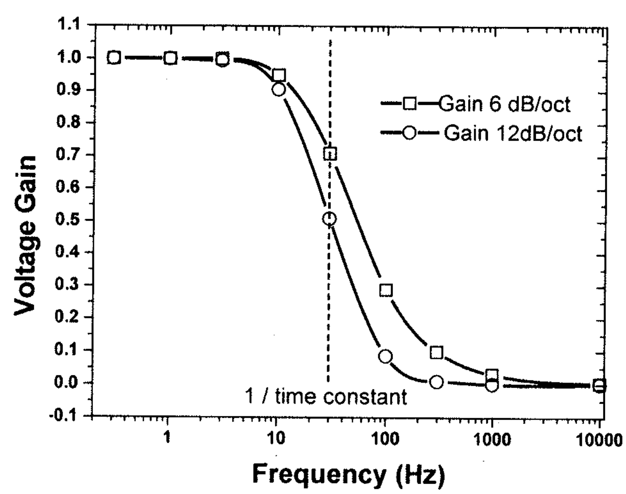

Figure 39

5. PHASE SHIFTER

Although the students have used the phase shifter in the lock-in detector experiments, they may wish to check the modules calibration with an oscilloscope. The setup is straight forward:

This same experimental setup can be used to study the phase shift as a function of frequency for a given setting of the relative phase.

The data for this measurement should look like Figure 31.

This curve is important for phase detection experiments because it shows that the relative phase shift of the phase shifter does depend on frequency. If the reference oscillator shifts in frequency, the phase shift drifts also. This effects lock-in detection.

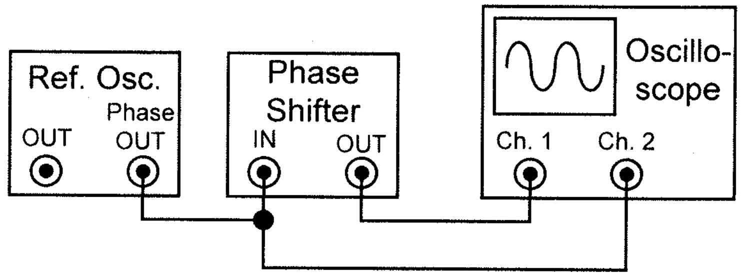

Figure 40

6. NOISE GENERATOR

Students should first look at the output of the noise generator signal on an oscilloscope. They should acquaint themselves with what a noise signal looks like on a scope.

Next, the reference output should be connected to the noise generator and the summed signal of noise and reference oscillator should be examined on the scope. This should be done for a wide range of signal-to-noise ratios.

Quantitative measurements of noise power or voltage are not easy for a variety of reasons. However, reasonably accurate measurements can be made with the SPLIA1-A in the following configuration:

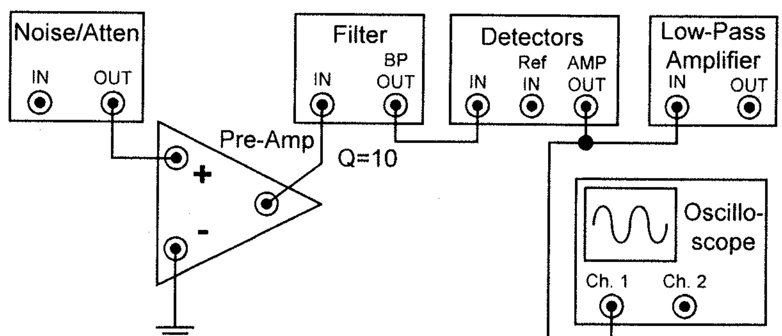

Figure 41

Using various settings of the bandpass filter to change the bandpass of the noise, students can quantitatively measure the noise spectrum of the noise source. We have explicitly included the oscilloscope in this figure to remind you that, when measuring noise voltages, it is advisable to keep the rms voltage level at the output of the detector above 100mV. This will help avoid errors due to the switching spikes (See Section V-3).

7. REFERENCE OSCILLATOR

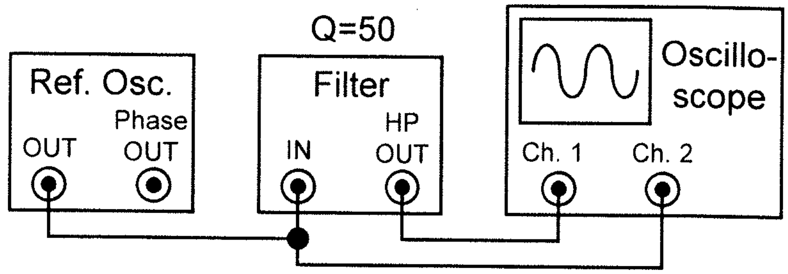

Figure 42

The students have already used the reference oscillator for many experiments. They probably have checked the frequency of the output signal against the instrument's calibration. An interesting experiment is to look for harmonic distortion in this module. There is very little distortion at the high end of the output frequencies, but a small amount of distortion can be detected at the low end around 15-20 Hz.

To measure the harmonic distortion, set up the modules as shown in Figure 42. We have set the Q to 50 and are using the high-pass output because it attenuates the fundamental more effectively than the bandpass filter does. The distortion is largest at low frequencies, and the largest component is the third harmonic.

Set the oscillator at 3Hz. Slowly sweep the filter's fine control near 9Hz. (With a Q of 50 at 9Hz, it will take tens of seconds for the filter to reach its equilibrium value.) You should observe the amplification of the third harmonic as you sweep. Carefully adjust the fine control for a maximum third harmonic signal.

Notice that the fundamental is always present. For a relatively accurate determination of the distortion, one needs to subtract the fundamental from the filter output signal. We do this visually on the oscilloscope by using the reference output on one channel and the filter output on the other. (See Figure 43). If your scope has a subtraction feature, it is possible to adjust the relative levels of the two signals so that the scope can do the subtraction. The percentage of distortion is then defined as the ratio of the input voltage (fundamental) to the third harmonic.
Do not forget that there is a voltage gain of about 50 in the filter section.

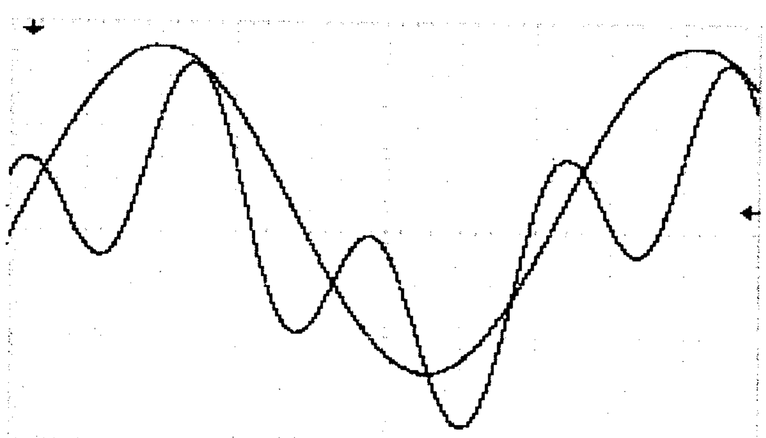

Figure 43

B. SYSTEMS EXPERIMENTS

1. TEST SIGNAL

The SPLIA1-A, as you now know, has a built-in test signal created by the combination of the reference oscillator, the attenuator, and the noise generator. This system of modules can be used to generate a dummy test signal whose signal-to-noise can be enhanced in many different ways. These various methods of S/N enhancement can be studied and compared. There is certainly a wide range of experimental parameters that might be used, so we will only suggest a few possibilities.

The wonder of a lock-in is how it can "pullout" a signal apparently completely buried in the noise. It is our recommendation that the student select a dummy signal with those characteristics. This signal is created by connecting the left output of the reference oscillator to the noise generator's signal input, ***using the special BNC cable with the ground isolation.***

First, construct a test signal at 50 Hz with a signal-to-noise ratio 1/1, where we define the signal and noise by their rms voltage. Configure the modules as shown in Figure 44. With the attenuator in the off position and the noise amplitude set to 1, measure the rms noise voltage. Now, turn the noise signal off, and set the signal attenuator to 1. Adjust the oscillator amplitude until the rms voltage and the noise voltage are the same. Now, turn on both the noise and the signal.

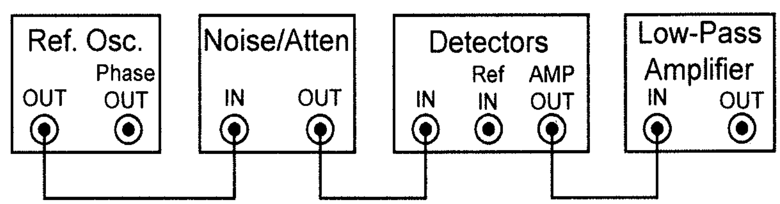

Figure 44

(If the signals are too big, you can attenuate both the signal and noise by a factor of 10 or more and still keep the same signal-to-noise ratio.)

a. Look at the output (signal + noise) on an oscilloscope. See if you can observe the 50 Hz signal.

Now, attenuate the signal by a factor of 10. Can you still see it?

b. With the 1/10 S/N, trigger the scope from the phase shifter output of the reference oscillator. Now see if you can detect the signal in the noise.

c. If you have a digital scope, you can use your signal averaging mode to look for the buried signal. Try signal averaging both when you trigger on the reference signal and when you trigger on the noisy signal itself. Is there difference? Why does triggering on the external reference signal make a difference? Explain.

d. Put the test signal into the preamplifier. Set it up single ended (ground the minus input) and observe the output signal on an oscilloscope, using both ac and dc coupling, and various gains. Note the gain settings where the signal begins to show saturation or clipping.

e. Now, connect the output of the preamplifier into the filter module. Look at the signal output of the bandpass, low-pass, and high-pass filters as you tune the frequency of the filters. Do this especially carefully for the bandpass mode, for various value of Q.

f. You are now going to create an amplitude detection system to examine this noisy signal. This system is also called a Precision Rectifier. The block diagram for such a detector is shown in Figure 45. The oscilloscope is shown to remind the user to check the output of every module, that has gain, for clipping or other distortion.

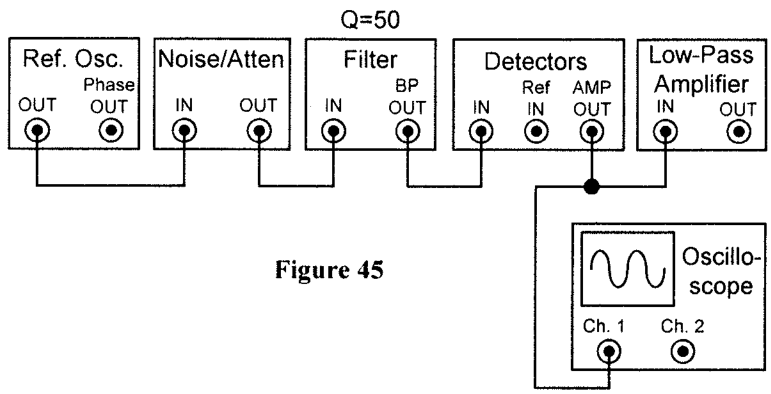

Figure 45

Measure S/N for the test signal for a variety of parameters using this experimental setup. Change the Q, the time constant, 6dB/oct and 12dB/oct, and the gain in various modules.

Measure the signal-to-noise ratio by switching off the signal. This can effectively be accomplished by the x1, x100 toggle switch on the reference oscillator.

Comparing S/N enhancement with various systems of modules will require a "standard" signal and standard overall gain, as well as signal recorder. Either some kind of a chart recorder or a printer connected to a digital storage scope will provide a record of these experiments. The data in this manual were obtained from a printout of the HP 54600B storage oscilloscope.

Set up a test S/N of about 1/30 at a frequency of 50 Hz. Tune the bandpass filter to the oscillator and set the Q to 20. Record the output of the low-pass output section either on a digital oscilloscope (using a slow sweep) or a chart recorder. Half way through the sweep, turn the signal off (divide by 100 toggle on the oscillator). The upper trace in Figure 46 was obtained that way, using 6db/oct and a 0.1 second time constant. Vary the low-pass time constants, the Q, and the gain, and observe any differences in the output. This is amplitude detection.

Now we wish to compare this amplitude detection configuration with the lock-in detection method. To make this comparison, we will set up the lock-in so that it has the same bandwidth as the amplitude detection. (See Figure 37).

At a frequency of 50 Hz and a Q of 20, what is the bandwidth of the system? Remember the bandwidth of the lock-in is given by the reciprocal of the time constant.

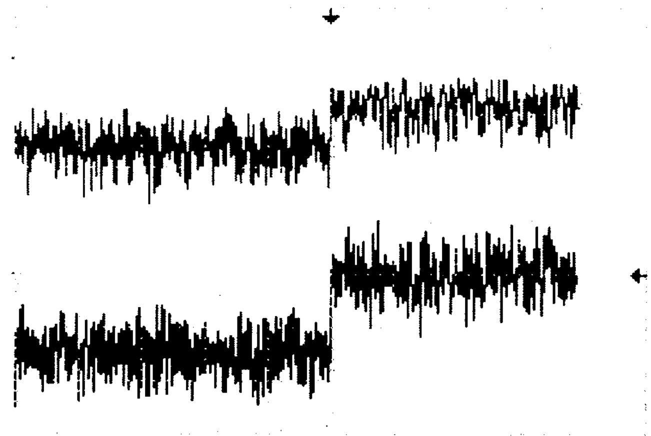

Figure 46

To adjust the phase correctly for lock-in detection, turn the noise off so that just the signal is present when setting the phase shifter. Now turn on the noise, set the low-pass output time constant to 0.1 second with a roll-off of 6db/oct. Again make a slow sweep on the scope recording the low-pass output. In the middle of the sweep, turn the signal off, as you did for amplitude detection. The lower trace in Figure 46 shows our data for lock-in detection.

Try changing the low-pass time constant, the Q, the 6dB/oct, 12dB/oct roll-off, etc., and observing the affect on the S/N ratio. Remove the bandpass filter (adjust the gain elsewhere) and observe any changes in the signal. Caution, when removing the filter and adjusting the gain, one should re-phase the lock-in. Repeating this whole series of experiments at 500Hz rather than 50Hz gives some interesting insights.

The smallest signal-to-noise that is recoverable at 50 Hz with amplitude detection is about 1/30. What is the minimum signal/noise that can be recovered with the lock-in?

What comparisons, if any, can be made between the noise voltage measured in Figure 32 and these measurements?

Lastly, examine the transient response of the system when using both 6dB/oct and 12dB/oct roll-off on the output low-pass filter. This can be done by manually switching on the X100 attenuation on the reference oscillator. Change the time constant from 1, to 3, and then to 10 seconds and compare response times.

2. LOCK-IN DETECTION IN A REAL EXPERIMENT

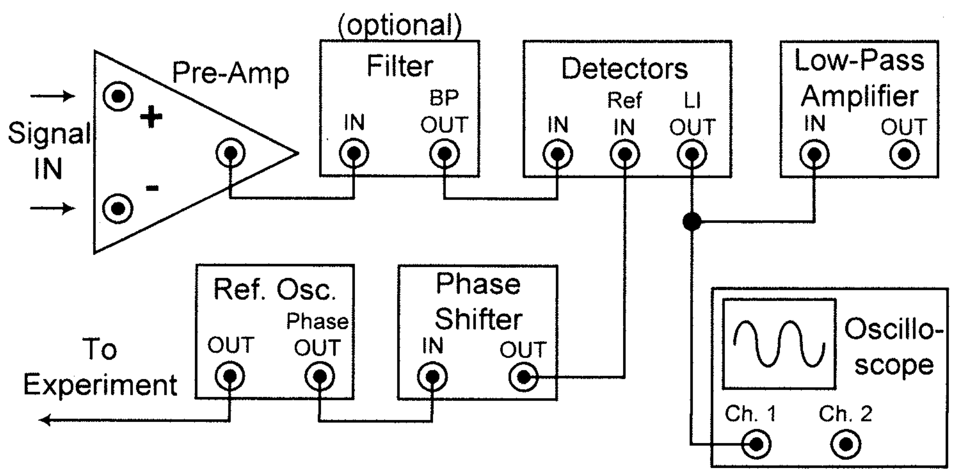

Figure 47

How are these modules configured to perform lock-in detection on a real physics experiment? Figure 47 shows the setup most generally used. The input to the preamplifier can be either single ended or differential and the filter is optional. The filter's primary role in lock-in detection is to increase the dynamic range of the lock-in detector by reducing the noise at the input of the lock-in. The reference output to the experiment provides a signal that, in some ways, codes the signal of interest from the experiment. The phase of the reference signal to the lock-in detector is adjusted to produce a maximum output, as previously discussed.

3. BRIDGE CIRCUITS

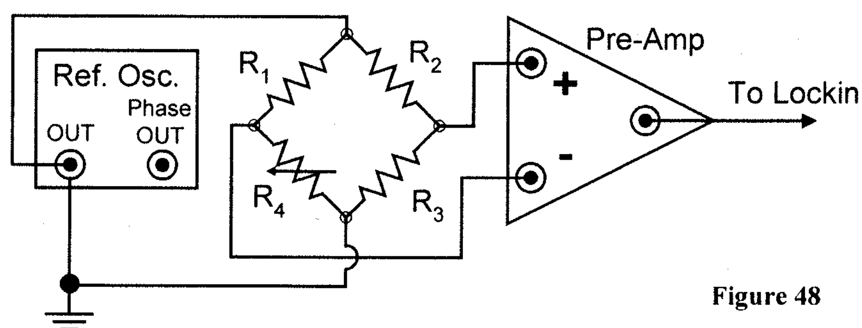

Figure 48

Bridge configurations in electronic circuits are an extremely important design for many applications. The Wheatstone bridge was the first one invented but now, a wide variety of both ac and dc bridges are now available. We will examine only the Wheatstone bridge, but use lock-in detection to increase its sensitivity. The circuit is shown in Figure 48. The beauty of this device is that it compares resistors to make an absolute measurement of resistance, rather than passing a known current through an unknown resistor, and measuring an absolute voltage across the unknown resistor. That method requires an absolute measurement of both current and voltage. In a bridge, one set of resistors has to be known. This is because the output voltage, at balance, is zero. Zero voltage reference is always available. (A short circuit, not considering contact potentials).

In lock-in detection mode, the bridge ac current is provided by the reference oscillator. $R_4 = R_3$ and $R_2 = R_1$ for balance and the signal at the output should be zero.

Suppose we wish to measure the temperature coefficient of resistance of one of the components of the bridge, say R_3. Construct a bridge with the following:

$R_1 = R_2 = 10K$

$R_4 = 9.5$ K + 1 K Potentiometer

R_3 is treated as the unknown

Setup the bridge using 2 V pp output from the reference oscillator. Connect the bridge output to the preamplifier in DIFFERENTIAL MODE. *Do not connect the output ground of the oscillator to the input of the preamplifier.* This will create unwanted low-level signals.

With the bridge out of balance, adjust the phase of the reference to give a maximum output. Now balance the bridge with the 1K potentiometer. Use an appropriate low frequency modulation, like 200 Hz.

Now place your hand on R_3 and change its temperature. Wait about 1-2 minutes. readjust the 1 K pot to balance the bridge. Determine the change in resistance for the change in temperature $\frac{\Delta R}{\Delta T}$. (Estimate the temperature change you might expect) Make the same measurements for several types of resistors – carbon composition, carbon film, metal film, wire wound.

In some cases the change may be so small that you will have to estimate the resistance change from the output signal from the lock-in.

How small a variation in R can you measure?

How does this compare with a measurement where the lock-in has been replaced by a digital voltmeter, even a digital voltmeter of high sensitivity?

4. MEASURING LIGHT

Lock-in detection is particularly useful when the signal is plagued by large amounts of background interference. TeachSpin devised a simple experiment students can perform which simulates this kind of real experimental situation.

The apparatus consists of a light emitting diode (LED) and a photodiode detector. The LED can be 100% amplitude modulated by connecting it to the TTL square wave output of the reference oscillator. The photo diode (with its internal resistor) connects directly to the input of the preamplifier. The system is set up in room light, so that the light from the diode is only a small fraction of the light on the photodiode.

Using the preamplifier as a dc amplifier and directly connecting its output into the low-pass output, determine how far the detector can be placed from the light source and still observe a signal from the diode. The diode can either be connected to a 5 volt dc source or to the TTL output at some high frequency (400 Hz) where it is only on 50% of the time.

Now repeat these experiments using both amplitude and lock-in detection. See how much farther away from the source the detector can be placed and still register the light from the diode. It should be impressive. Choose a modulation frequency that is far away from the interference frequencies of the background light. It is worthwhile to study the frequency characteristics of the background light first, using the bandpass filter as a spectrum analyzer. These measurements should make it clear what modulation frequency is appropriate for these experiments.

It is also interesting to compare sine and square wave modulation of the LEDs. Why do they give different results?

5. LOCK-IN DETECTION OF MAGNETIC RESONANCE

Imagine we have a resonance signal with a magnetic field dependant absorption signal that looks like Figure 49. Although the resonance change of power to the detector is quite obvious in this diagram, in many magnetic resonance experiments the change in power is a very small fraction of the total power incident on the detector. The combination of the detector noise, amplifier noise, and noise in the incident power, tends to make this small signal difficult to measure. How can the lock-in help?

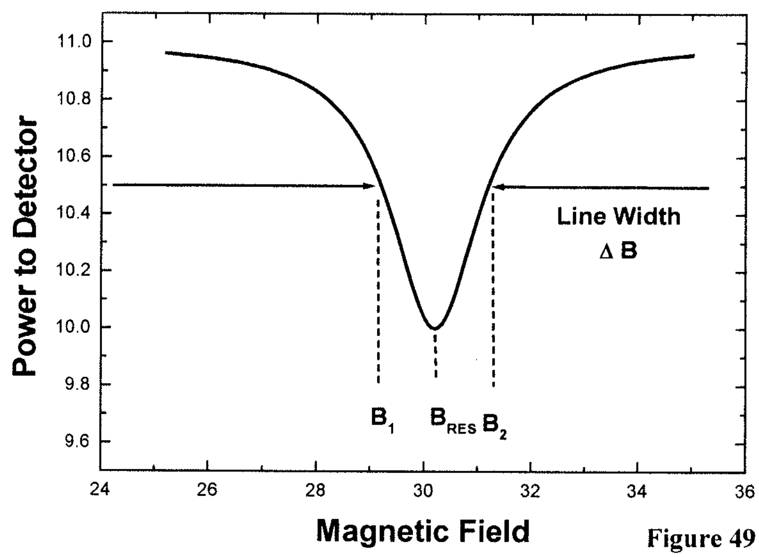

Figure 49

To begin with, we must "code" the resonance signal. This is usually done by modulating the magnetic field with an ac current, so that the following is true.

$$B_{mod} < Linewidth\ \Delta B \quad and \quad B = B_0\,(Sweep) + B_{mod} \sin 2\pi f_{mod}\, t$$

The resonance is searched for by slowly sweeping the magnetic field with its superimposed ac modulation field. Suppose the slow sweep reached the value of $B_1 + B_{\mathrm{mod}} \sin 2\pi f_{\mathrm{mod}} t$, see Figure 50. The magnetic field is modulated at a frequency f_{mod} and one can see from the diagram that the power to the detector is also modulated at f_{mod} with an amplitude that depends on the slope of the magnetic resonance absorption, $\left(\Delta P/_{\Delta B}\right)$. At the value $B_{res,}$ the slope is zero, and the power to the detector is not modulated. At B_2 one obtains the same Δ Power as at B_1, but it is 180° out of phase with respect to the signal at B_1. One should note that Δ Power is also zero far away from the center of the resonance.

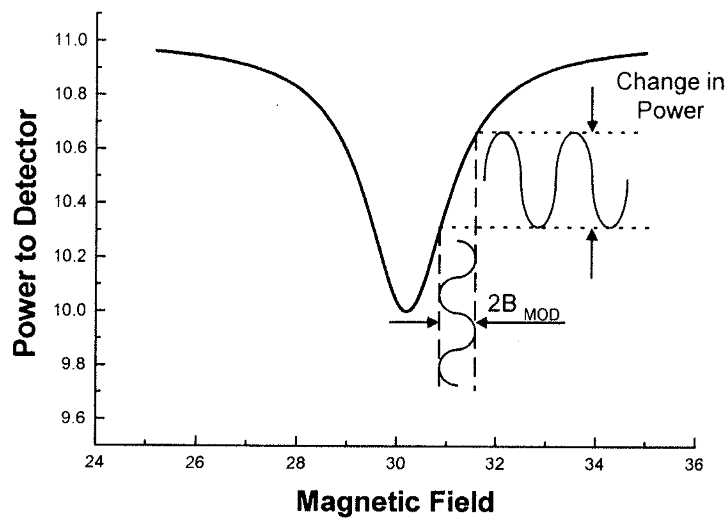

Figure 50

As long as $B_{\mathrm{mod}} < \Delta B\ Linewidth$ the Δ Power is proportional to the slope of the resonance line, and oscillates at the modulation frequency.

This technique of magnetic field modulation has "coded" the magnetic resonance signal and made it possible to use lock-in detection. *The output of the lock-in detector is a derivative of the absorption signal as shown in Figure 51.*

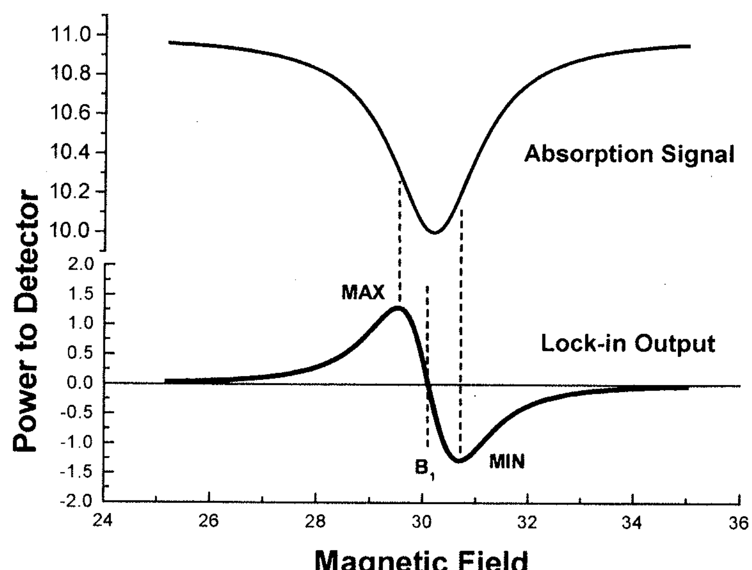

Figure 51

There is one more essential condition that must be satisfied. The lock-in output signal must always be in "quasi-equilibrium" during the sweep. Remember, we are sweeping the magnetic field through this resonance. But the lock-in is averaging the instantaneous signal that is fed into the output low-pass filter. The sweep rate must be slow enough so that the output of the low-pass filter is always in quasi-equilibrium. If the sweep rate is too fast, the output will be distorted.

A good rule of thumb is as follows: The time it takes to sweep from the derivative maximum to derivative minimum should be about ten times the time constant of the low-pass filter in the lock-in;

$$t_{sweep} \text{ (max to min)} = 10 \text{ (time constant)}$$

This is a conservative estimate.

Larger modulation amplitude can, in fact, produce a larger signal for the lock-in, but it will distort the signal. Sometimes, large modulation amplitudes are used to search for an unknown signal, but to reproduce the line shape accurately; the modulation amplitude must always be small compared to the linewidth.

A much more comprehensive discussion of these techniques can be found in Poole; see references.

6. DC LAB AMPLIFIER

Sometimes it is useful in the research laboratory to have a dc amplifier that will signal average over variable time constants. The final stage low-pass output, in conjunction with the filter and the preamplifier, can be configured to create such an amplifier. The preamplifier could be used for differential input signals. The filter could be used for an even sharper "roll-off" of the low-pass filter (18 or 24dB/oct). The maximum time constant of the low-pass filter is about 300ms (3Hz).

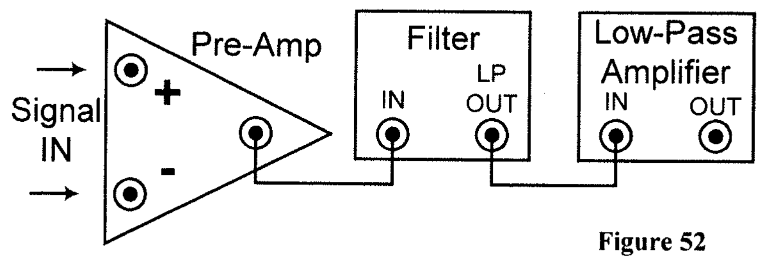

Figure 52

7. SQUARE WAVE TO SINE WAVE CONVERTER

The phase shifter in the SPLIA1-A will ***only*** work with an external modulation source, if the source is a sine wave. It will not work with a square wave or TTL signal. Without the phase shifter, lock-in detection is not possible. A solution to this problem is to use the filter section to extract the fundamental sine wave signal from the square wave. This is done in a configuration of the modules as shown in Figure 53. The optional attenuator should be used if the signal from the modulation source is too large for the filter (remember that the filter with a Q of 20 has a voltage gain of 20). If the signal is a bipolor square wave, then the low-pass filter could be used since this would give higher attenuation of the high frequency harmonics. With TTL type signals, the bandpass output should be used to remove the dc offset.

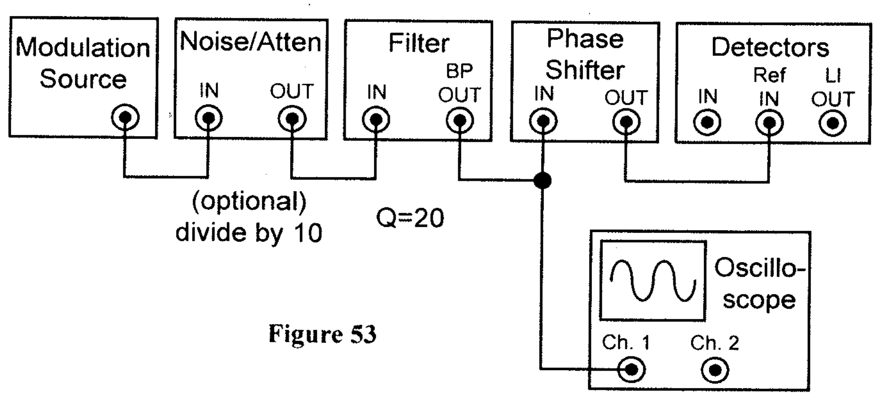

Figure 53

8. AMPLITUDE DETECTOR / PRECISION RECTIFIER

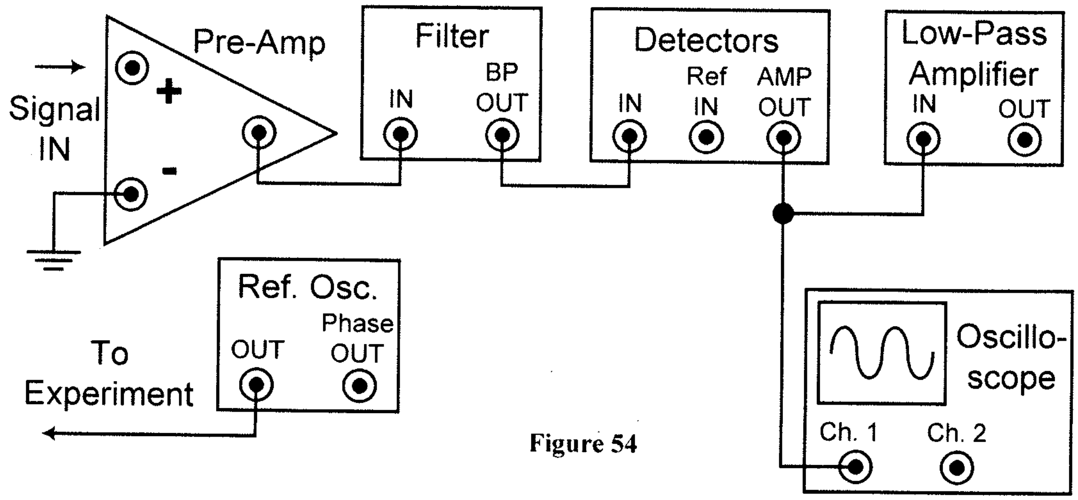

Figure 54

There may be a situation where the experimenter wishes to use the SPLIA1-A just as an amplitude detector or an rms voltmeter without any phase sensitive detection. That can be done by using the modules configured as shown in Figure 54.

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