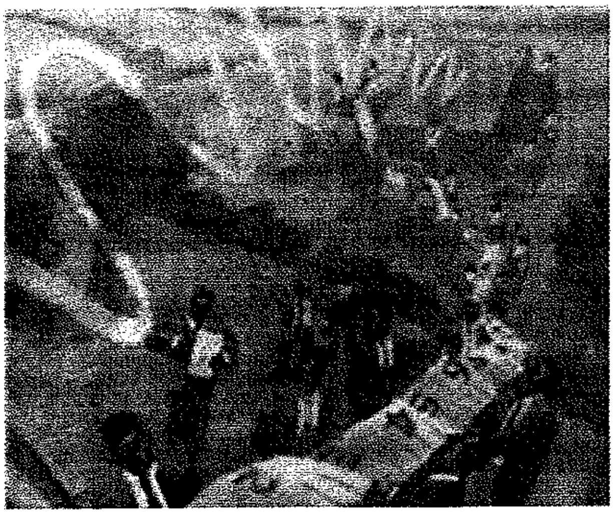
The phase-sensitive detector

Michael Slifkin and Asaf Schlesinger

believe that the phase-sensitive detector – useful for recovering small signals from noise among other things – does not get the attention it deserves. Their discussion culminates in a full design for a vector-computing phase-sensitive detector.



The phase sensitive detector, also known as lock-in amplifier, is a useful instrument that has been around for a long time. Surprisingly though, a number of electrical and electronics engineers and scientists I have encountered have no knowledge of it.

Many well-known textbooks on electronics either do not refer to it at all or give it a passing mention. There is only one book exclusively devoted to this instrument, written by Mike Meade.[1] This book is out of print and is not widely available. What follows here is a simplified account together with some historical background culminating in the design of a novel vector-computing detector.

The phase-sensitive detector, often called a lock-in amplifier or voltmeter, is an instrument for measuring a signal of a definite frequency, even when the signal is buried deep in the noise. Figures of 100dB below noise are regularly quoted. It can also be used for measuring phase.

Many engineers when asked to do the first task will think at once of using a high-Q amplifier. Unfortunately very high-Q amplifiers are difficult to build and even more difficult to keep on frequency unless kept in a constant temperature oven. If the signal shows small changes in frequency, then it becomes an impossible task using such an amplifier. The only answer is to use a phase-sensitive detector.

Although I have spoken about using a psd for measuring

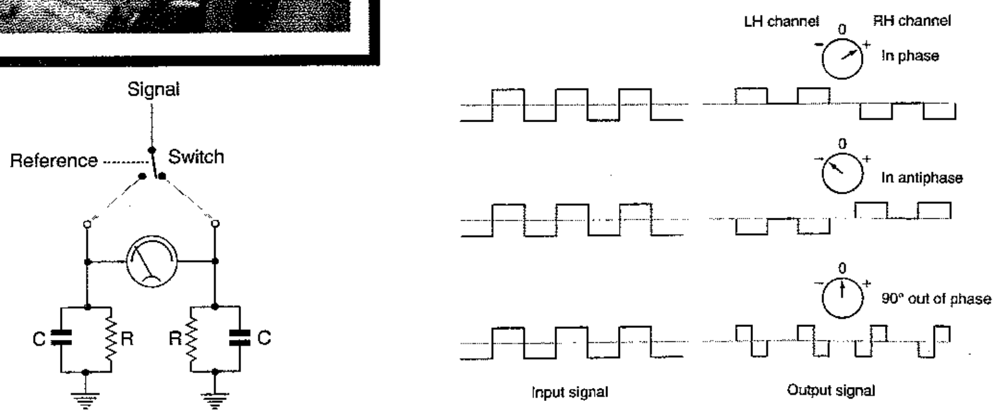


Fig. 1. In a), the basic phase-sensitive detector, the switch operated by the reference signal feeds the input signal to each of the two RC networks alternately. Signal relationships are shown in b).

signals at a specified frequency, it is just as likely to be used to measure very small dc signals. Because of the inverse dependence of noise on frequency, maximum noise occurs at the lowest frequencies. Also, there is a problem with dc amplifiers in that they are subject to zero-point drift. It is very difficult to prevent this.

On the other hand, alternating signals do not suffer in this way as the signals always adjust themselves so that they are equally positive and negative. Hence the base line doesn't shift. For this reason, it is usual to convert small dc signals to alternating signals by chopping them at some frequency.

At one time, the conversion was done using a vibrating reed relay, but these were noisy and of limited life. Nowadays electronic switching would be used.

How the psd works

Figure 1a shows a schematic diagram of a psd. Basically it is simply a switch operated by a reference signal, which has to be in the form of a rectangular wave at the signal frequency. Usually the device used to modulate the signal provides the reference.

Chopped signals will be rectangular. When the reference signal is exactly in phase with the received signal then all the positive halves are switched to one channel and the negative halves to the other channel.

At the same time, each channel has a low-pass filter which can be thought of also as an integrating circuit so that the signals are accumulated over the length of time given by the time constant of the low pass filter *RC*. These time constants can be quite long. Commercial instruments can have time constants of up to 100 seconds.

If you consider what happens to signals at some other frequency, they will have both negative and positive parts appearing in both channels so that they will tend to cancel each other out. So you can see that this switching device acts like a filter. It is possible to show mathematically that the equivalent bandwidth of this device is $1/4RC$ for a simple 6dB per octave roll-off filter.

A more complex two-stage low-pass filter having a 12dB per octave roll-off has half the bandwidth. Thus for a time constant of three seconds not inordinately long at all, one has an equivalent band width of $^1/_{12}$Hz, something which would be almost impossible to achieve using conventional high-Q amplifiers. It should be noted that it takes about four times the time constant for the signal to settle down after a disturbance for the simple low pass filter and eight times for the two stage low-pass filter.

Additionally, should there be some drift in the chopping frequency, the psd will still follow the signal without loss of amplitude or change of phase, more or less. This is the reason why this device is nowadays called a lock-in amplifier or voltmeter as it locks onto the signal.

Harmonic effects

You can probably see from the description that the detector will respond to odd harmonics of the signal, should it not be a pure sinusoid. However, there will be both positive and negative parts of these signals in both channels so that these odd harmonics will be attenuated.

The third harmonic will be attenuated by nine and the fifth harmonic by 25. In addition, noise at these harmonic frequencies will also leak through to the output.

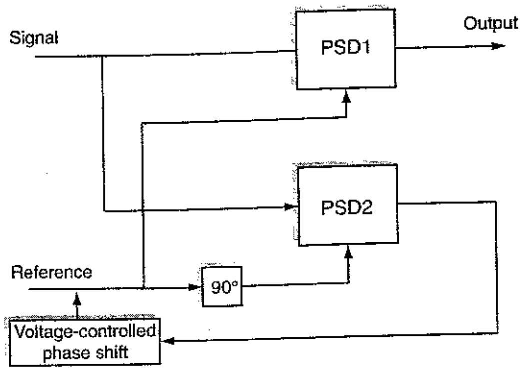


Fig. 2. Vector-tracking lock-in amplifier. Output from PSD2 feeds the voltage-controlled phase shifter via a long time constant. This shifter ensures that output from PSD2 is always zero and output from PSD1 is always maximum because of the 90° shifter.

It will be usually necessary to filter out these higher harmonics using a-low pass filter at the input. A tuned filter will not work. There is a phase shift through such a filter that varies with frequency.

Readers who read the homodyne/synchrodyne receiver article by Slifkin and Dori in the November 1998 issue will recognise that the phase-sensitive detector is simply synchronous detection using a dual channel rather than a single channel mode. One could certainly build a detector in which one channel is earthed and this would be an exact analogue of the synchronous detector.

From the simple model, you can also work out what happens if there is a phase difference between the signal and the reference. A maximum positive signal is produced when the two are in phase, as in Fig. **1b**).

When they are exactly out of phase there will be a maxi-

Chopping through history

These modulation techniques were very well understood in the forties. Unfortunately, textbooks on instrumental techniques from this period are very difficult to find.

Chopping techniques are certainly very old. Alexander Graham Bell used chopped light in his experiments on the photoacoustic effect in the 1870s.

An early instrument to measure the lifetime of phosphorescent crystals was the phosphoroscope. This consisted of two wheels mounted on a common axle. They had windows so that they could produce a square wave from a light source.

The crystal was placed in between the two wheels and the light intensity passing through measured as a function of the angle between the two wheels. This is an exact analogue to the psd used as a phase detector.

Mike Meade[1] refers to the experiments of Dicke the founder of radio astronomy in the 1940s. He wanted to measure very weak signals from outer space. His arrangement involved an antenna pointed at his subject and a dummy antenna at the same temperature. The two were switched between for many hours, integrating the signal.

Similar experiments were carried out slightly later at the Radio Astronomy department of Manchester University at Jodrell Bank. In their earliest experiments scientists there integrated the outputs over many hours by using an electrochemical cell and weighing the amount of metal deposited which was proportional to the integrated received signal.

Another technique they used slightly later was to build an up down counter. The radio signals were received in the form of small pulses, the counter would count up when connected to the real antenna and count down when connected to the dummy antenna.

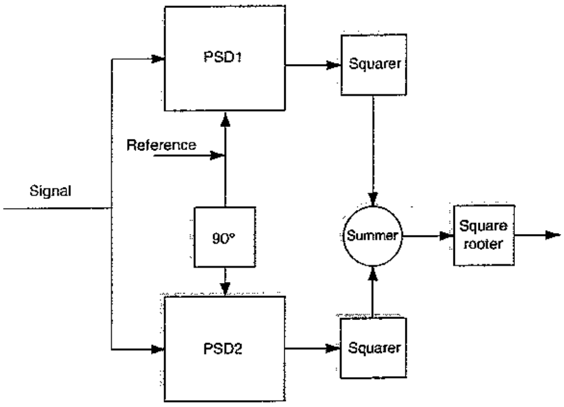


Fig. 3. Vector computing lock-in amplifier. This scheme is easier to implement, but has lower dynamic range than that of the vector tracker.

mum negative signal and when they are exactly 180° out of phase, there will no signal although there may well be some residual fluctuation due to noise.

Overall it is easy to accept that the relationship between the maximum signal V_{max} and the output signal V is given by $V = V_{max}\cos\phi$. This is why the phase is so important and why the instrument is sometimes called a phase-sensitive detector.

If we had a situation in which the input signal amplitude was always constant then the output amplitude would simply be a measure of the phase.

The phase-sensitive detector is not only useful for measuring very small signals in a noisy background. However, it can also be used to measure the time delay or phase shift of an electrical device by comparing the phase of the signal before it entered and after it left the device.

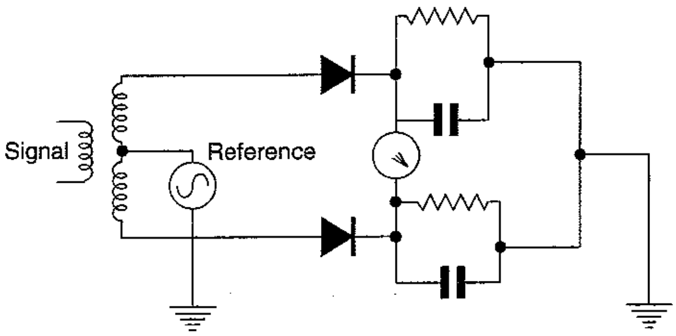


Fig. 4. Diode-operated phase-sensitive detector. The early versions performed well, but their frequency response was limited due to the inductance of the transformers.

How you would build an accurate phase shifter to work over a range of frequencies will not be dealt with here.

Measure between the zeros

It is much more accurate to measure the phase between the zeros or the signal rather than the difference between maxima of the signals. A cosine/sine waveform has its greatest change passing through the zero. At its maximum, it has its minimum change. A cosine/sine waveform is almost flat at the peak. The relationship between the phase shift and time is given by the relationship $\arctan\theta = 2\pi ft$.

In general, measurement of phase is not that accurate. An alternative way of using the phase-sensitive detector is to measure the amplitude of the signal as a function of the frequency with an input signal having the same phase as the reference signal.

It can be shown mathematically that the output of the system under test will be a sigmoidally shaped curve – also called a logistic curve – having the form,

$$V = \frac{k2\pi ft}{1 + 4\pi^2 f^2 t^2}$$

where V is the signal, f is the chopping frequency, t is the delay time and k is some instrumental constant. The value of t can then be found by a variety of methods including plotting $1/V$ against $(2\pi f)^2$ which gives a straight line with a slope of t/k and an intercept of $1/kt$.

A more sophisticated method would be to use a curve fitting program with a computer which would give the value of t directly. These types of methods are much more accurate than a simple measurement of phase.

For a versatile instrument, one requires that one can obtain automatically a maximum signal, the in-phase signal, or the out-of-phase signal. There are several ways of doing this. Of those, the vector tracking method seems to be the one now used commercially.

There is however another method – namely the vector computing method. While inferior because of its limited dynamic range (dynamic range is explained in the panel), it is easier to implement. We present a prototype later that we have designed and built.

Note that any of these methods requires the use of two detectors.

Vector tracking

A schematic of the vector tracking method is shown in **Fig. 2**. The device works as follows.

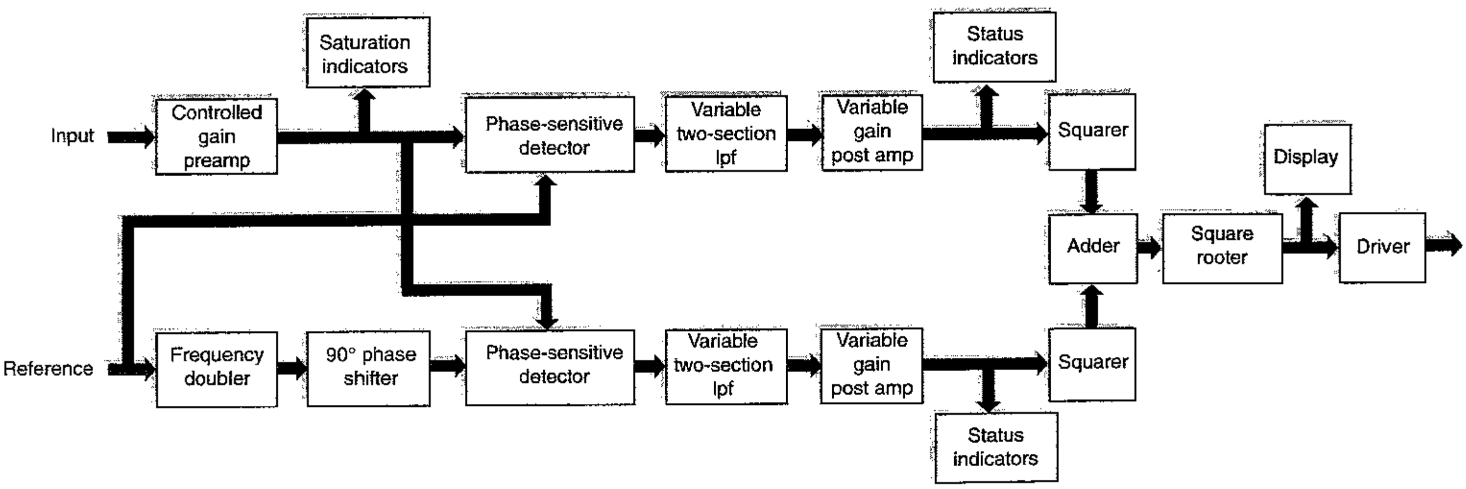


Fig. 5. Block diagram of the vector-computing phase-sensitive detector – the subject of the main circuit, Fig. 6.

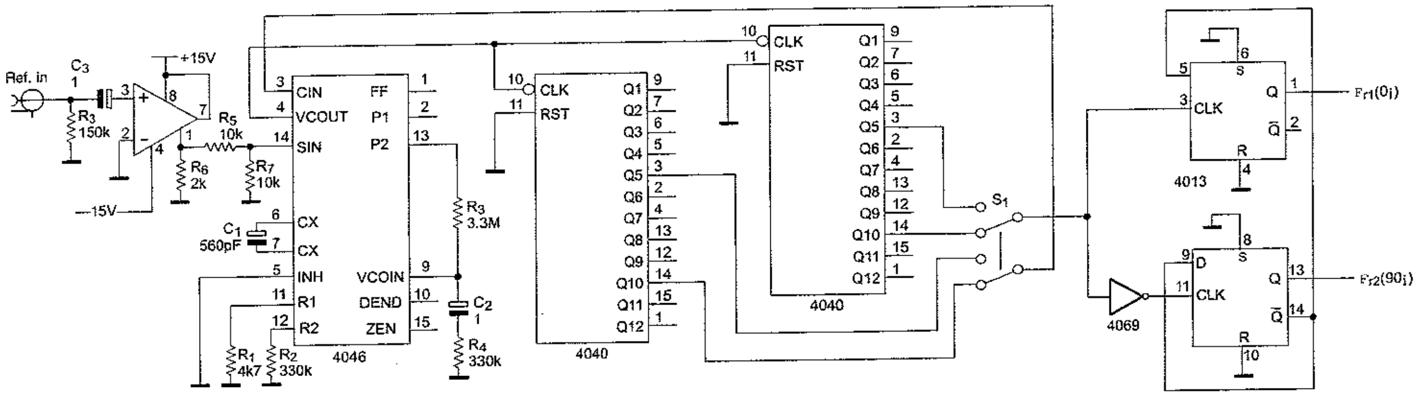


Fig. 6a). Input section and digital 90° phase shifter of the phase-sensitive detector.

Output from the lower detector is fed back using a long time constant to control a voltage controlled phase shifter. It is fed back in such a way that the output from this channel is always zero and hence because of the 90° phase shifter the upper channel is always a maximum.

This arrangement enables good dynamic reserve to be obtained. Amongst other possible methods is the vector computing method this is illustrated in **Fig. 3**. If the original signal is $V_{max}\cos\theta$ then the 90° phase shifted signal is $V_{max}\sin\theta$. When the two signals are first squared and then added, we obtain V_{max}^2 as $\cos^2+\sin^2=1$ and thereafter V_{max} following square rooting.

This is a relatively simple technique and easy to implement using modern integrated circuitry. It suffers the problem of lower dynamic range than that of the vector tracking method. This is because the vector tracking method obtains the signal from one psd only at optimum signal to noise ratio. The second detector channel merely serves to condition the reference channel.

By contrast, the vector computing method obtains the signal from both channels which in itself will give increased noise. In addition the signals will not be at optimum signal-to-noise ratio in both channels and the extra computing circuits all combine to degrade the signal to noise ratio

A more complex version is the heterodyne psd. In this technique, the detector works at a high fixed frequency and the incoming signal is heterodyned with the incoming signal to the psd frequency.

This has the big advantage that the odd harmonics are now so high that they lie outside the range of the instrument. There remains only the fundamental signal and the noise at the fundamental frequency at the output. In addition, AC amplification is more efficient at frequencies higher than a few hundred hertz so that this technique is doubly useful with low frequency signals.

Some modern lock-in amplifiers incorporate a microprocessor and can be linked directly to a computer and operated from it.

Finally, dsp lock-in amplifiers are now coming on to the market. With these instruments, the incoming signals are converted to digital signals using a-to-d converters and then are mathematically processed to give the required signals. Their manufacturers claim that they are superior to analogue lock-in amplifiers.

The earliest phase-sensitive detectors

The earliest form of phase sensitive detection involved mechanical switching using either vibrating reeds or synchronous electric motors with a cam opening and closing microswitches.

The first electronic phase-sensitive detector dates to the early fifties and consisted of a centre-tapped transformer and diodes as in **Fig. 4**. These performed quite well but were limited in frequency because of the inductance of the transformers. In addition in order to get identical channels, it was essential to use very high grade centre tapped transformers.

The next generation used transistor switching. With the introduction of integrated circuits, it was now possible to obtain switching with gain.

It can be shown mathematically that the phase-sensitive detector action is simply a multiplication between the reference and the signal. Modern practice is to use high-grade IC multipliers or high specification IC mixers.

The first commercial lock-in amplifiers were built as part of nuclear magnetic resonance apparatuses and were not sold as stand alone instruments. One made by JEOL in the early fifties had plug-in peak tuned filters at the front end, although this would appear to be completely contrary to the phase-sensitive detector philosophy.

Other lock-in amplifiers of a slightly later period, such as those made by PAR, also used high-Q filters at the front end.

The first instruments sold as general purpose lock-in amplifiers in the UK were probably the ones by Brookdeal in the early sixties. These used individual transistors and provided

Integrated circuits involved

IC_1	4046
IC_2	4040
IC_3	4040
IC_4	4069
IC_5	4013
IC_9	OP27
IC_{11}	AD534
IC_{12}	AD534
IC_{14}	AD534
IC_{15}	LM311
IC_{16}	AD524
IC_{19-26}	OP27

Dynamic reserve

All the forms of synchronous switching that we have described in this article will work adequately – particularly at low frequencies.

What distinguishes a good lock-in amplifier from a poor one is the dynamic reserve. This is defined in various ways. However, a useful definition is given by Mike Mead[1] as the maximum allowed peak to peak level of asynchronous input divided by the peak to peak value of the full-scale synchronous signal. In other words, it is a measure of the worst-case signal-to-noise ratio that can be applied to the input of the instrument to give a full-scale signal at the output.

The better the dynamic reserve the better the instrument at measuring a signal buried in noise. This is a function of the quality of the amplification filtering and inherent noise in the instrument.

Another factor that is clearly related is the input overload level, i.e. that signal which will cause the instrument to overload or go into a non-linear mode. Most commercial instruments have some means of indicating when the overload point is reached, since output above this point is no longer reliable.

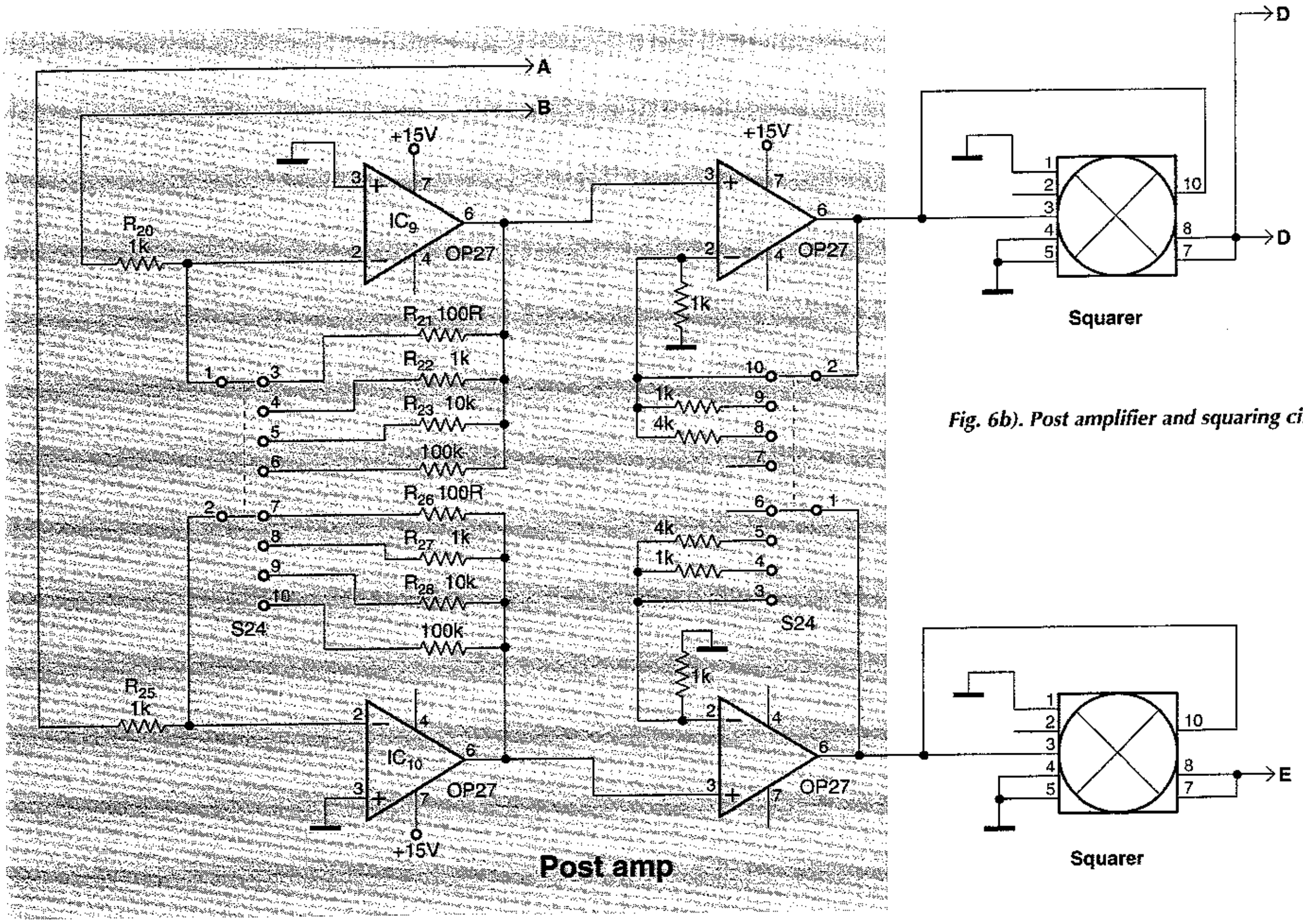


Fig. 6b). Post amplifier and squaring circuits.

no means of phase shifting.

Some lock-in amplifiers have a built-in reference oscillator so that one can for example obtain noise spectra by measuring the output as a function of the reference frequency. Nowadays there is a wide selection of lock-in amplifiers with all kinds of features. Dynamic reserves for such instruments are quoted as being in the range 100 to 130dB.

Implementing the psd concept

In this final section, we describe a computing vector lock-in amplifier using modern integrated circuitry. The instrument has an input that handles anywhere between a few microvolts and a few volts. Its frequency range is 10Hz to 3kHz and its linearity 2%.

Although this instrument performs quite well, it cannot be compared to the state-of-the-art vector tracking lock-in amplifiers. However it is relatively easy to build and certainly far cheaper than the commercial instruments. A block diagram is shown in **Fig. 5** and the circuit diagram in **Fig. 6**.

If you want to build a single-channel phase-sensitive detector, you can easily adapt the circuit diagram. It would then be preferable to include a manual phase shifter in the reference channel so that the output can be maximised.

The main ICs are manufactured by Analog Devices. There are three identical computing elements incorporating the *AD534* four-quadrant multiplier. These are used as squarers and square rooter.

The same element could also be used as the psd but the *AD630* is cheaper and intended specifically for use as a high-grade mixer. It is simply a two-quadrant multiplier.

The precision instrumentation amplifier, *AD524* is also taken from the Analog Devices range. It has a low offset

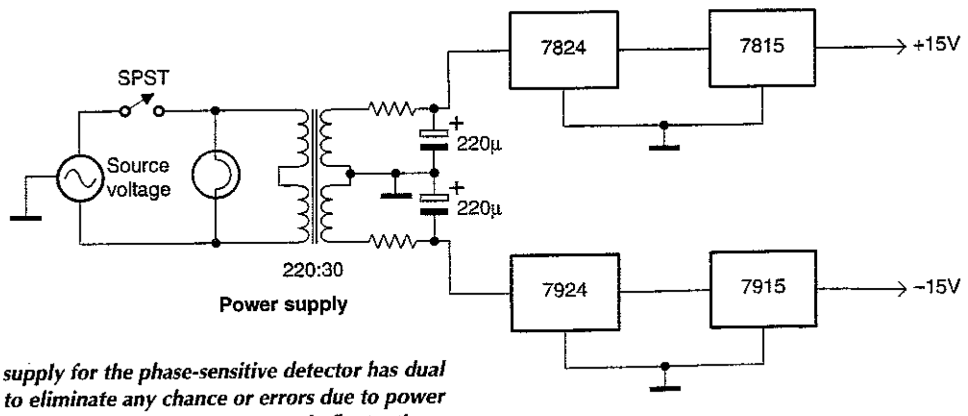


Fig. 6c). Power supply for the phase-sensitive detector has dual regulators on each rail to eliminate any chance or errors due to power supply fluctuations.

voltage. In addition, it can be configured to give automatic zero offset with some additional circuitry. We avoided doing this because of the increased complexity of the circuit.

Although this is an analogue psd, we have chosen a digital technique for the 90° phase shifter, as this is by the far the best available. The reference is frequency doubled and then passed to the CLK input of two D-type flip flops, one input being inverted.

The D-type flip-flops are configured as divide-by-two circuits. Thus, the Q outputs are at the original frequency but one is shifted by 90° relative to the other. This works very well over a wide frequency range.

Frequency doubling is obtained from a *4046* digital phase lock loop with a *4040* divide by two ripple counter in the feedback loop. This generates a frequency exactly twice that of the input frequency.

An explanation of the phase lock loop is given in the article on the synchrodyne/homodyne receiver mentioned earlier. One could replace this digital phase lock loop with an analogue phase-locked loop such at the *NE565*.

The low-pass filters are active devices to give idealised behaviour. We have used second order Chebyshev filters to give decreased bandwidth. You may prefer to use a simple *RC* filter depending on your particular application.

Some might prefer to sacrifice bandwidth for settling time and simplicity. We have provided time constants of 0.3, 1, 3 and 10 seconds.

There is both pre amplification and post amplification. Pre-amplification has fixed values of 1, 10, 100 and 1000. Post-detection amplification is 1, 2.5 and 5.

We have included saturation indicators for the preamplifiers. When these are fully on then the amplifiers are overloaded and should not be trusted.

However, the instrument is usable during low-frequency flashing of the leds. We have also included high and low detectors for the phase sensitive detector itself.

The signal presented to the psd should be neither too high nor too low. The high and low level indicators come on when these conditions are violated. Again, low-frequency flashing of these indicators is no problem.

There is an output driver should you want to observe the signal on for example a chart recorder. This can be very useful. It is much easier to get an estimate of noisy signals by examining the trace on a chart recorder than by trying to read a varying digital display.

We have not included the output display in our circuit diagrams as there are many alternatives around. You could even use a digital voltmeter.

A useful addition would be a frequency meter in the reference channel. ■

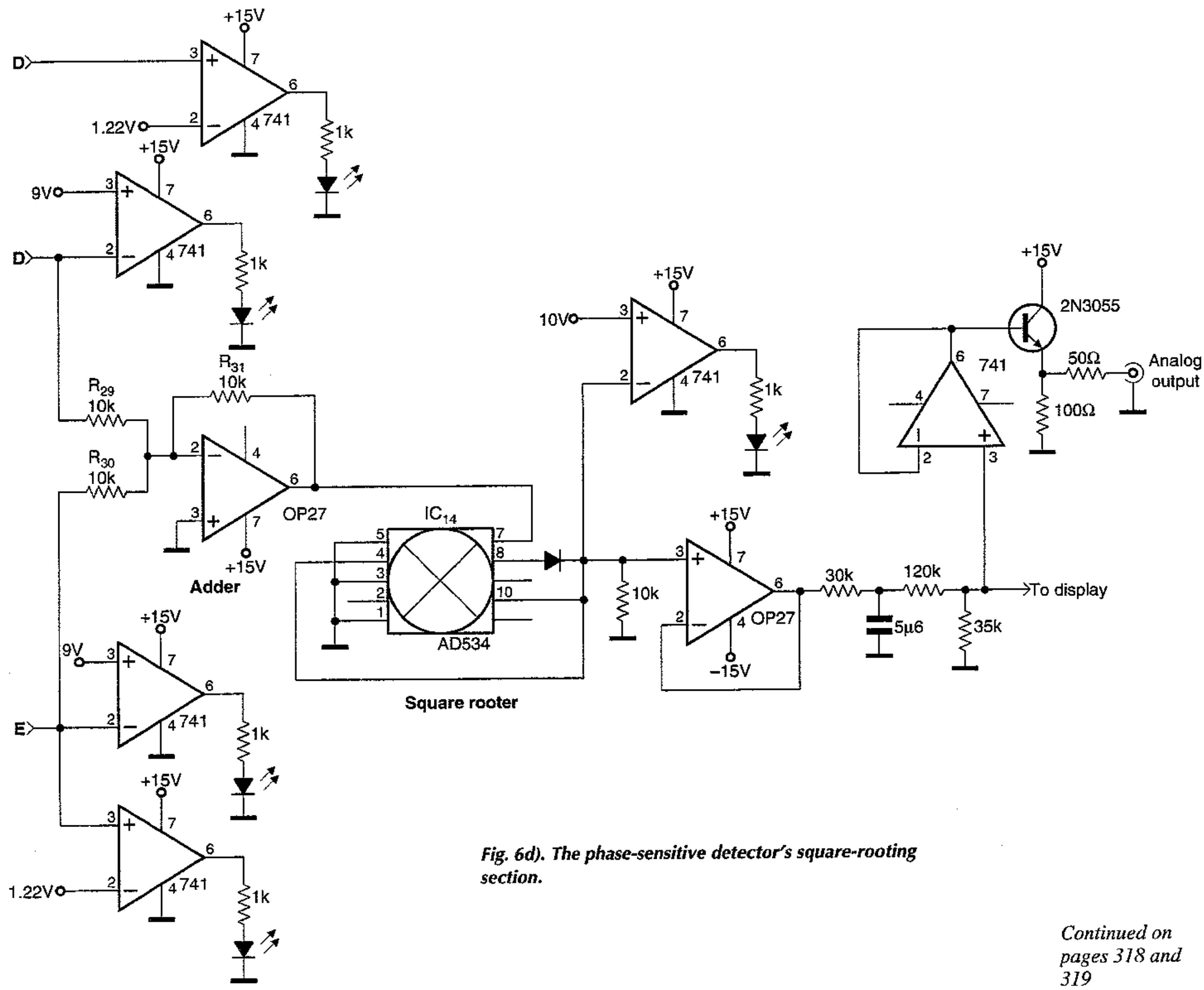


Fig. 6d). The phase-sensitive detector's square-rooting section.

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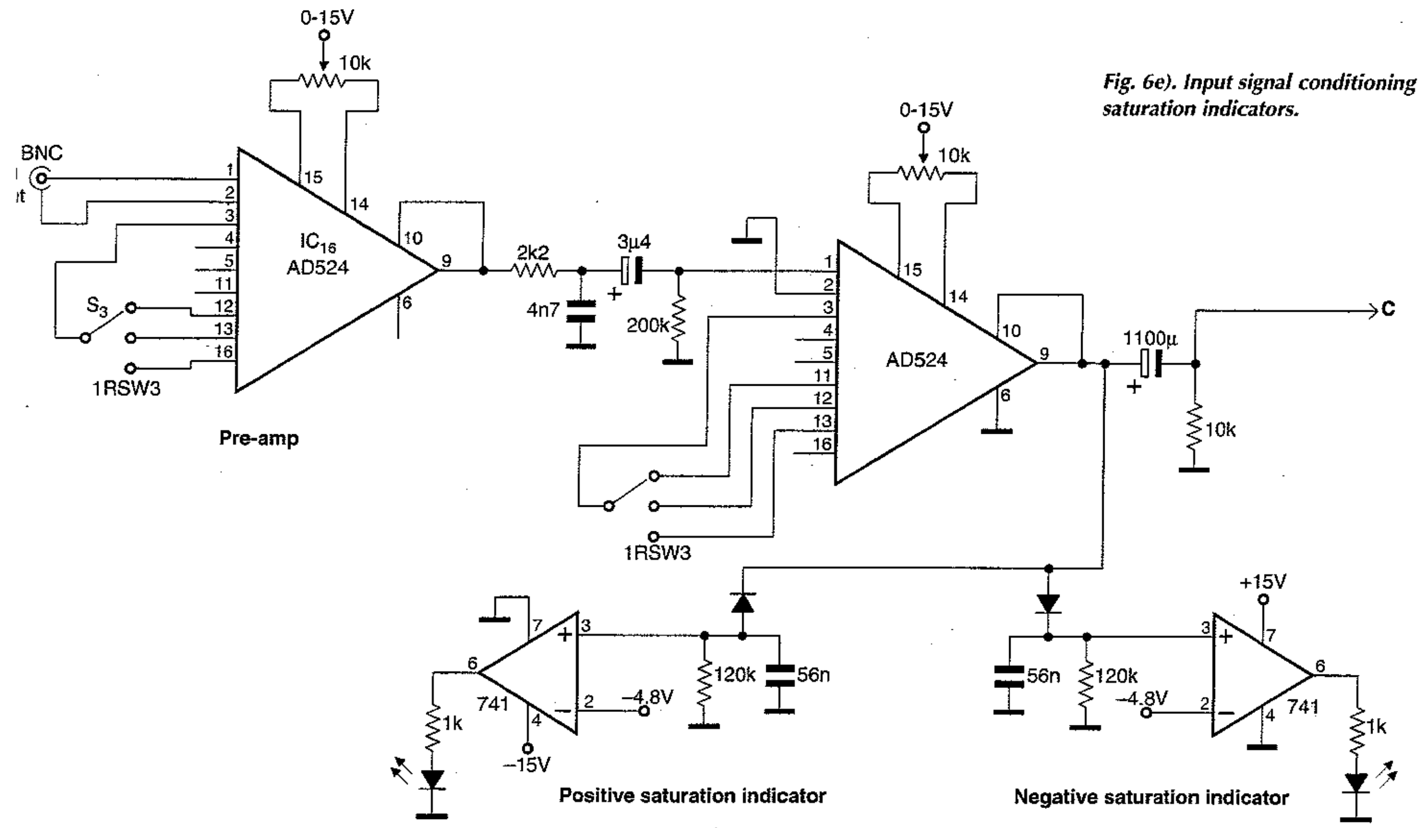

Fig. 6e). Input signal conditioning and saturation indicators.

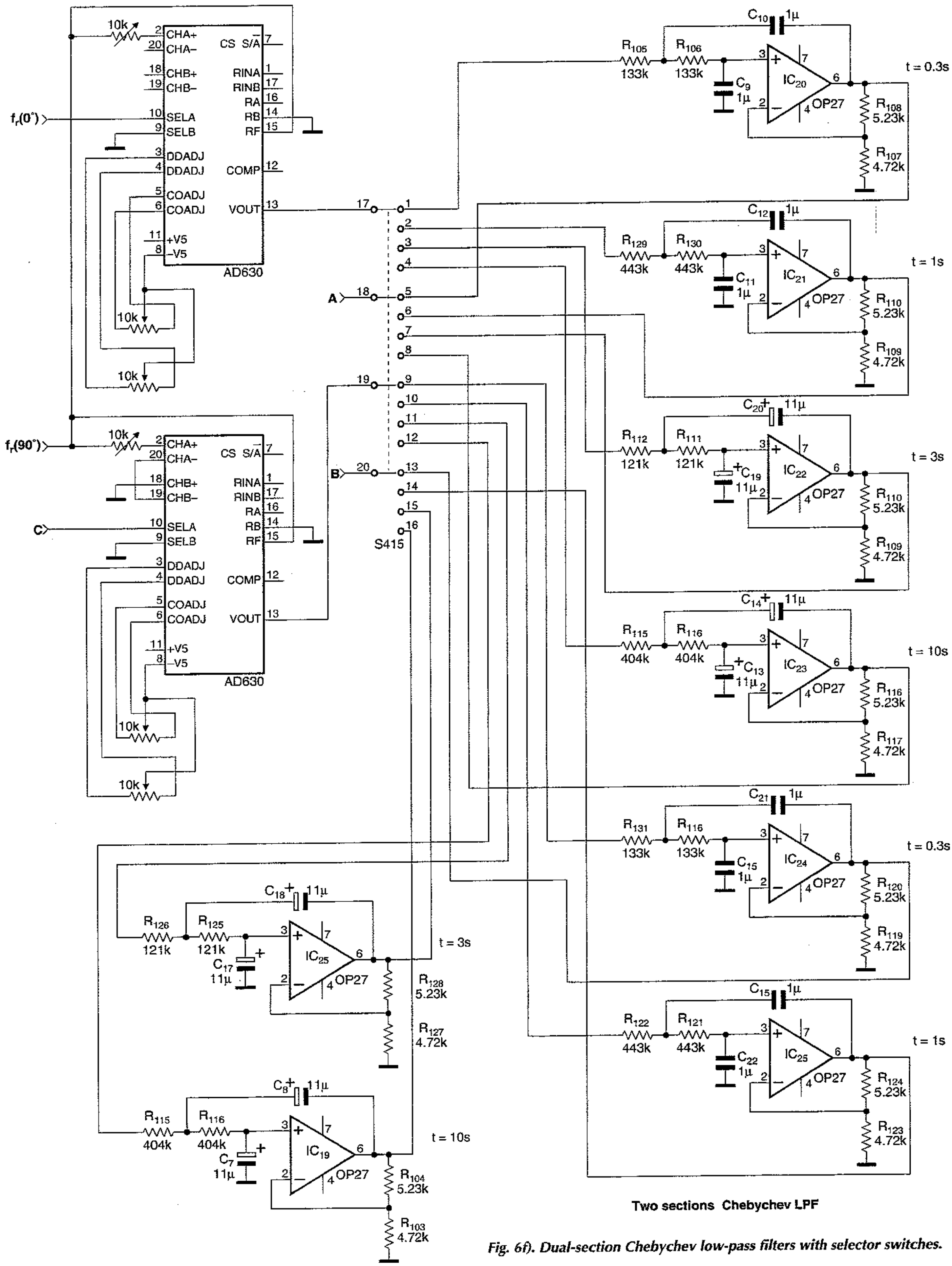

Fig. 6f). Dual-section Chebychev low-pass filters with selector switches.