Craig Anderton's ELECTRONIC GUITARIST

Phase Shifter, Part II

Last month we covered everything about the phase shifter except the phase shift module itself, so here we go.

You'll notice that I have not indicated IC package outlines in the schematic (as I normally do), because the schematic would look to confusing. I've used the symbol for an *op amp* (the little triangle), instead of its specific package outline.

The op amp is a very universal building block. It has two inputs (one called the + and the other the - input), an output, and *two* power lines. The simplest model is the 741 (see Figure 1). The two inputs connect to pins 2 and 3; the output connects to pin 6. The + power supply line connects to pin 7, and the - supply line connects to pin 4. The other pins are of no consequence to the circuit here, but don't let anything short out to them! The circuit may get very unhappy if you do.

If you want to wire up the phase shift module with 741s, you'll have to use six of them—one for each op amp. In addition to connecting up the inputs and outputs, you'll have to connect pin 7 of every 741 to the +5V point on the board, and every pin 4 to the -5V point on the board. However, this method is the least cramped, which makes assembly easier for beginners.

There is also a less cumbersome way to wire the board so you don't have to use all those 741s. A part exists called the 5558 (also called the 1558 or 4558) which contains two 741 type op amps in a single package. The pins connect up differently from the 741, though (see Figure 2). However, you can just as easily use the op amps inside a 5558 as you would the op amp inside a 741. As a bonus, you only have to use three ICs, instead of six.

The way I wired up the shifter was to use a 5558 IC for IC1A and IC1B, and a 4136 (a quad 741) for IC2A, B, C, and D. This way, I only had to use two ICs; see Figure 3 for the pinout on a 4136. The 4136 is also a lower noise unit than a 741, which is quite nice. Again, make sure that the + power supply line of all ICs goes to a place on the board called +5V, and that every - power supply line goes to a similar point marked -5V.

The CLM6000 [manufactured by Clairex; distributed through a variety of outlets] also requires some explanations. This part is called an Opto-Isolator and uses an LED, a solid-state light bulb, and a photoresistor [a resistor whose resistance varies according to the light that shines on it]. Both of these are mounted in a small, opaque package that looks like Figure 4. The dot on the outside of the CLM6000 corresponds to the cathode of the LED, as indicated in this X-ray view.

It has been my experience that about three percent of all new CLM6000 don't work after they've been installed in a circuit. This is probably because they are quite heat sensitive. Use caution when soldering, and solder fast—not so fast you get a cold joint, but as fast as possible. If the phase shifter upon completion sounds more like a wah-wah, you may be having a problem with one of the CLM6000s.

Figure 5 shows how to connect an in/out switch to take the phase shifter in and out of circuit. This type of switching is covered in more detail in my book *Electronic Projects For Musicians* if it doesn't seem clear here.

After you've wired up the phase shift module, connect point G from the board to the chassis ground lug; connect point I to terminal A of J1 using shielded cable; connect point 0 to terminal 1 of R8 (refer to last month's schematic); connect point CV to terminal 1 of R7; connect the +5V point from this month's board to the +5V point of last month's board; similarly connect up the two -5 points. This completes the phase shifter wiring.

To test, get some fresh batteries, plug into J1 and J2 with mono cords, set R9 for a moderate speed (about halfway for starters), and adjust R7 slowly and carefully until you obtain a sound that suits you. By the way, the schematic shows an optional phase/vibrato switch. If you break the signal path at this point, you hear a vibrato rather than phase shift sound. It's a neat effect, too.

This is not a simple circuit. Although I've tried to make things as simple as possible, space prevents a rigorous explanation. So if you've never built anything before, I wouldn't recommend starting on this. If it works right the first time you're okay, but if you have to troubleshoot it, look out! A little patience and knowledge can go a long way though, so if you feel up to it, work carefully, preferably with someone who knows a little about electronics. And if you don't feel up for it, save this article for later.

Parts List

R10,11,12,13	1K, 5 or 10%, ¼- or ½-watt resistor
R14,15,16, 17,18,19, 20,21,22	82K or 100K, 5 or 10%, ¼- or ½-watt resistor
R23,24,25,26, 27,28,29,30	470K, 5 or 10%, ¼- or ½-watt resistor
C8,C9	10 pF disc ceramic capacitor, 10 or more volts
C10,11,12,13	.01 uF capacitor, *mylar recommended*, 10 or more volts
C14, C15	.22 uF capacitor, 10 or more volts
IC1A, IC1B	5558 dual op amp (made by virtually all manufacturers)
IC2A,B,C, and D	4136 quad op amp (made by Raytheon)
OI1,2,3, and 4	CLM6000 opto-isolator (made by Clairex)
S1 (optional)	SPST switch
S2 (optional)	DPDT switch
Misc.	Perfboard, wire, and all parts mentioned in last month's column.

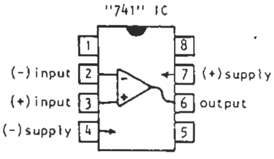


FIGURE 1.

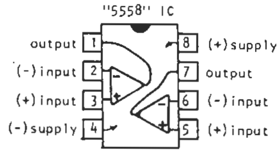


FIGURE 2.

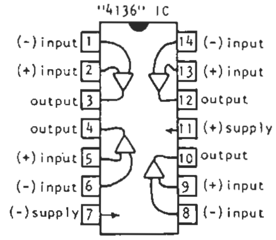


FIGURE 3.

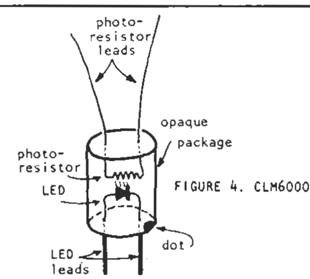


FIGURE 4. CLM6000

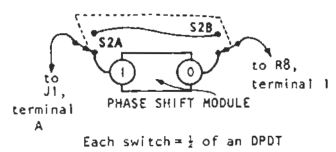
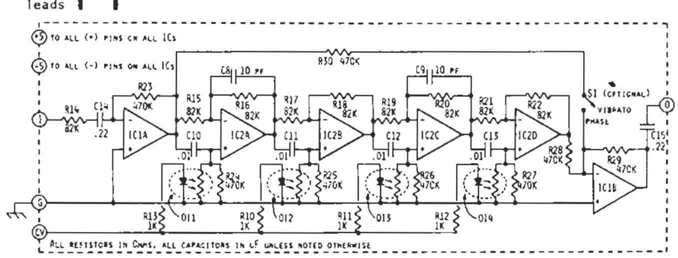


FIGURE 5. IN/OUT SWITCH

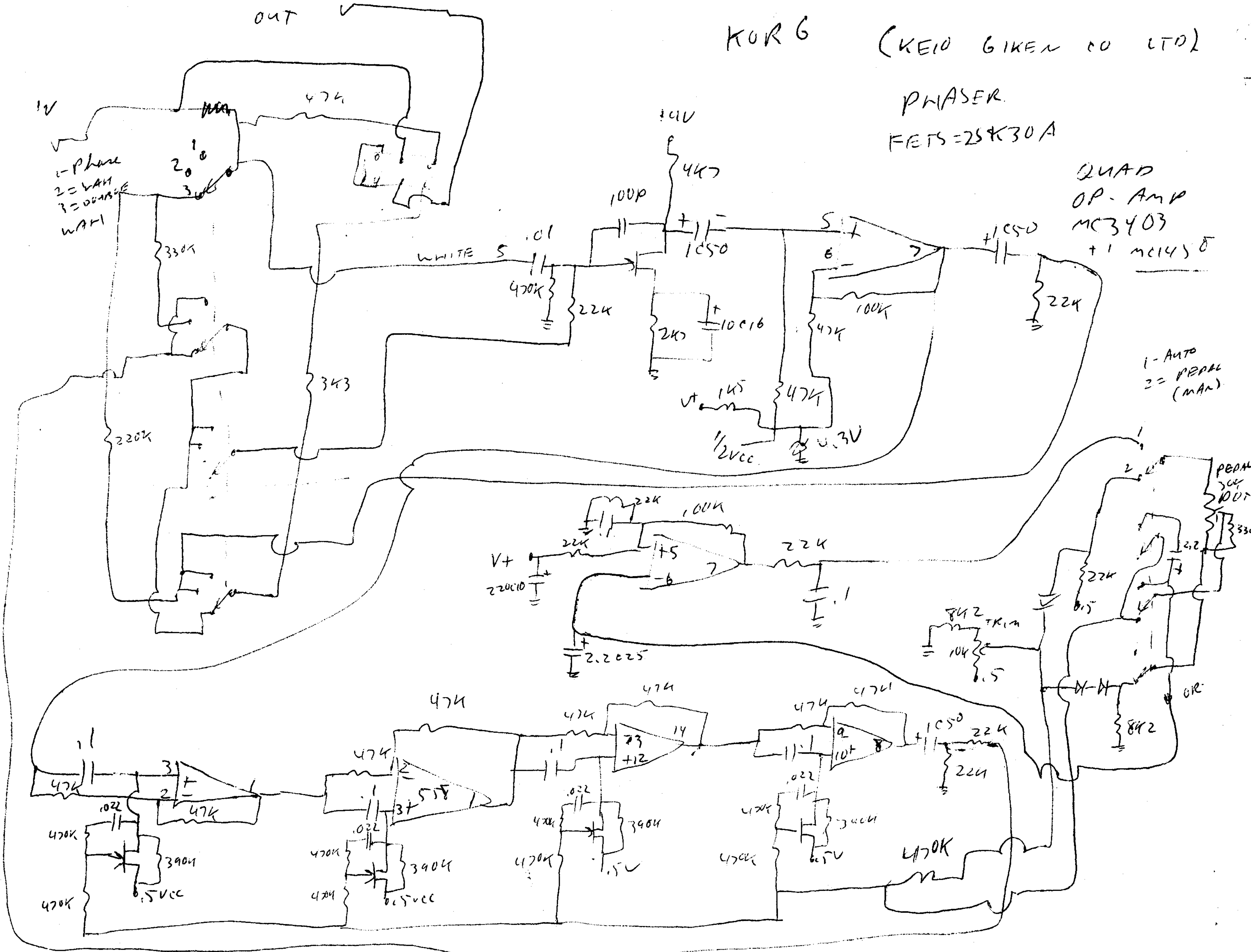


KORG (KEIO GIKEN CO LTD)

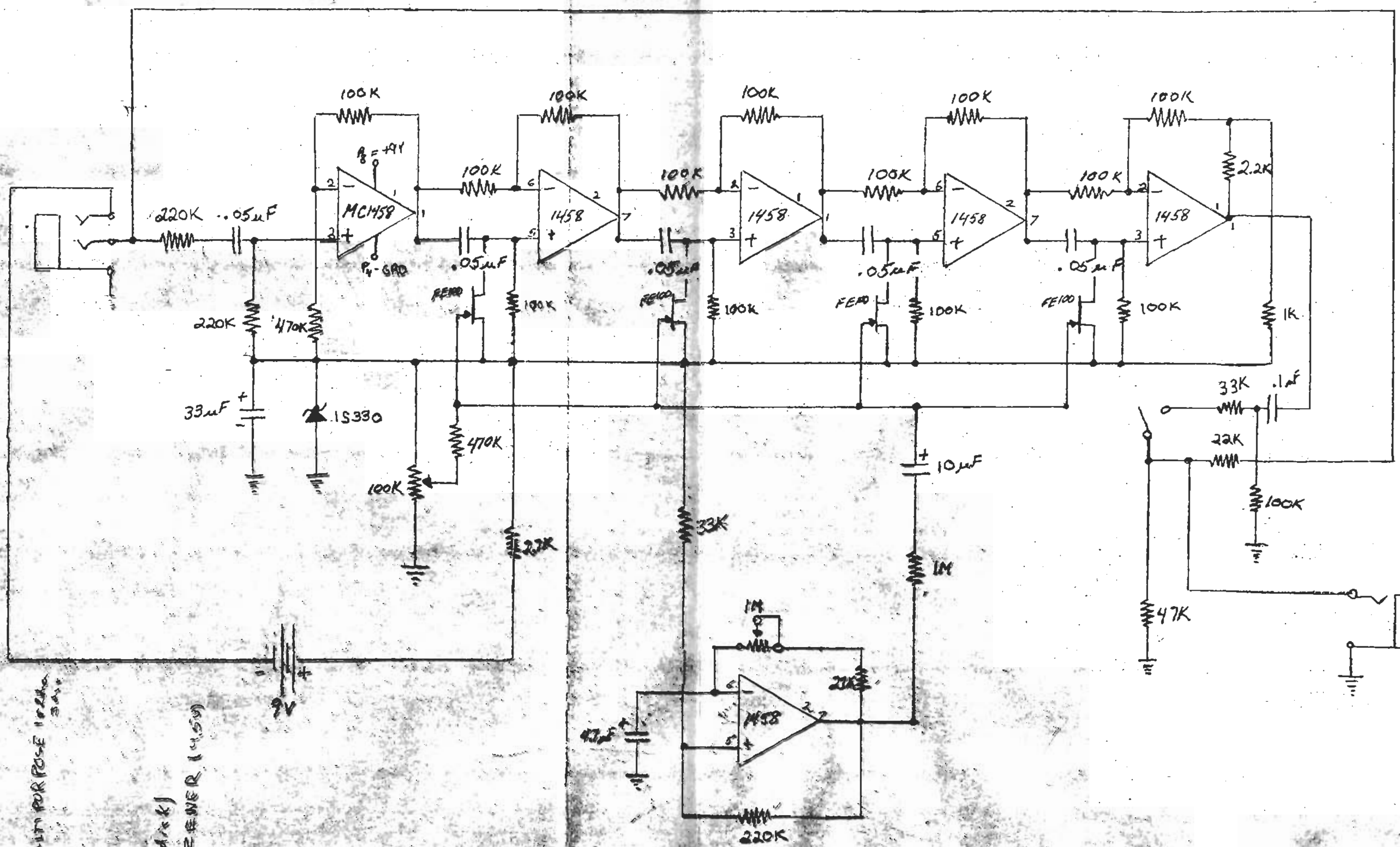
PHASER

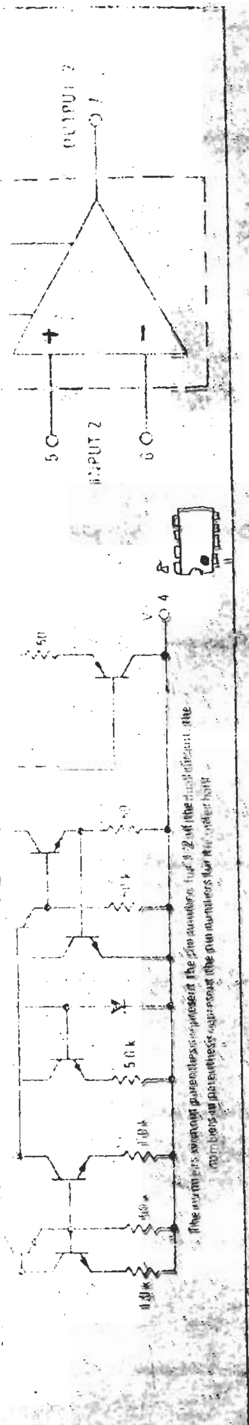
FETS=2SK30A

QUAD OP-AMP MC3403 +1 MC14558



Phase Shifter





3 = MC1458
4 FETS (ALL PURPOSE ...)
15 - 100K
3 - 220K
5 - .05µF (Cap)
1 1S330 ZENER (1459)

state of solid state

A look at an interesting circuit that adds a vibrato effect to musical passages; plus an SCR array and an ultra-stable oscillator. **KARL SAVON** SEMICONDUCTOR EDITOR

TREMOLO AND VIBRATO CIRCUITS ARE POPULAR with musicians. Some want to hook them up to their electronic organs and guitars. Others want to play tape recorders or radios through them.

The effects of the tremolo and vibrato circuits are related; they both add some form of modulation to the original music. Both modulate the original music source with a low-frequency (usually sub-audio) control signal to give a pleasing "wobulation" to the music. It is most effective when the music source is purely tonal in nature with only the attack and decay envelope shapes of specific instruments.

Tremolo is amplitude modulation. The frequency of the source is unaffected. Only its amplitude wavers, usually following the sinusoidal envelope of the modulating signal. Tremolo is relatively easy to implement with fairly simple circuitry. Lower-cost electronic instruments that include either of the two effects will invariably use tremolo—it's less expensive.

Vibrato is the more pleasing of the effects. We recognize it as the frequency modulation mastered by the professional vocalist. Inevitably it is often combined with a secondary amplitude modulation. Electronic organs sometimes go to great expense and extremes to produce vibrato mechanically. The most notable are the moving baffle and rotating speaker systems, where the sound source is physically moved. Some achieve true Doppler effects by changing the distance between

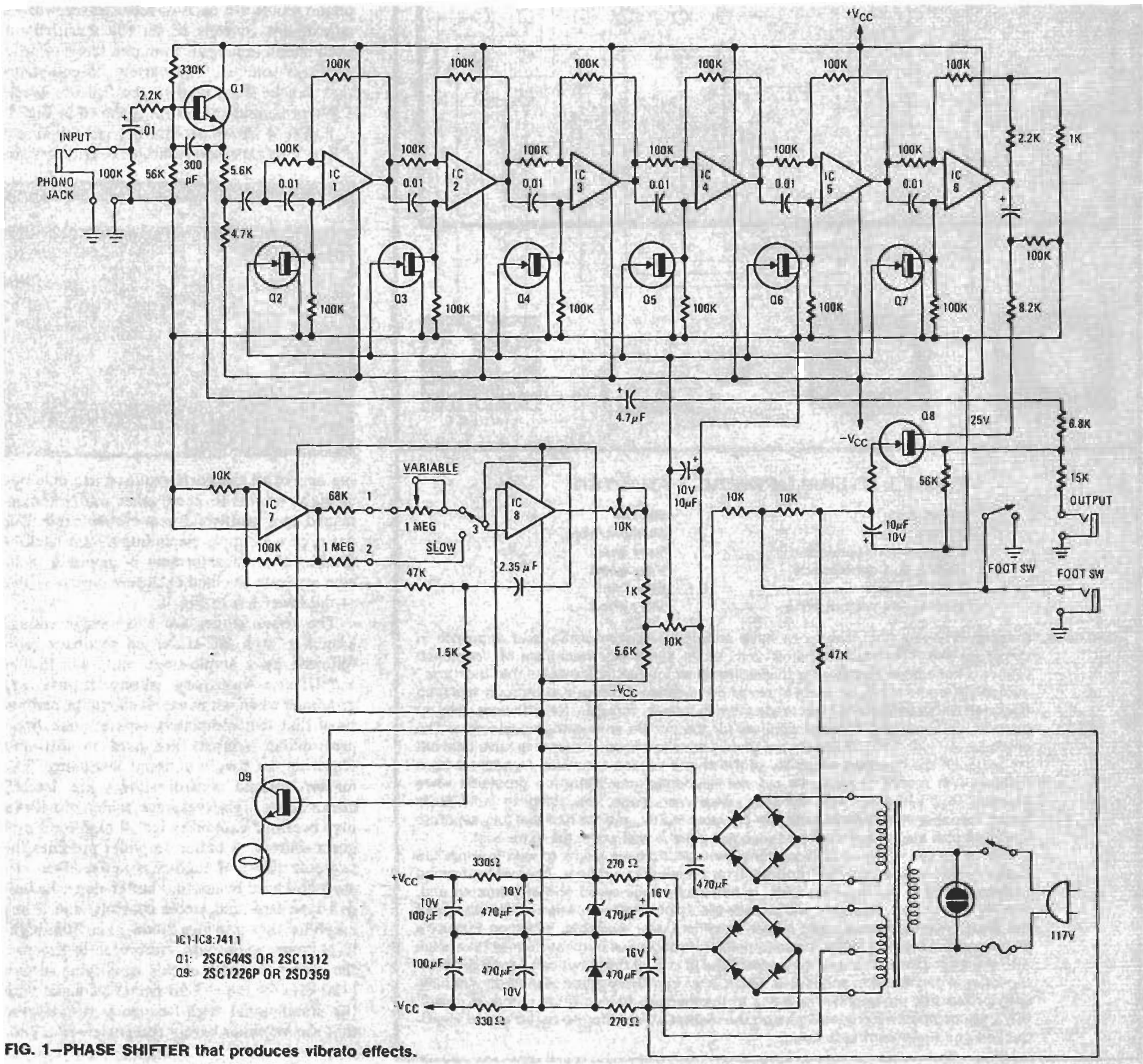


FIG. 1—PHASE SHIFTER that produces vibrato effects.

the source and the listener. At least one manufacturer uses undriven speakers as mechanical counterweights. When the modulation frequency is very low and the proper tonal quality is synthesized, the effect takes on the sound of a carillon.

Musical phase shifter

One way to produce a true vibrato is to phase-modulate the music source. Frequency is the rate of change of phase. If the phase is varied as a sine function, the frequency of the output will change at the same rate as the phase modulation with a cosine function. You may ask, "Isn't it simpler to frequency-shift the oscillator itself in the case of an electronic instrument tone generator?" Well maybe—but be careful. Whenever you design an oscillator to be frequency-shifted, its stability suffers as a rule. The act of building a frequency-shift system by its nature introduces sources of instabilities for undesirable shifts in frequency.

By phase-modulating the buffered oscillator output, the stability is unaffected. The frequency of the vibrato output will vary around the rock-stable original frequency.

This method is one that has been proven in the FM radio broadcast field. FM stations must hold their carrier frequencies to very tight standards. But they of course must find some way to modulate the carrier in frequency. Phase modulation is the answer. In this case the audio modulation signal is not just a simple low-frequency tone, but the entire audio baseband, and a conversion must be made so the effect of the phase modulation will be to produce a change in frequency proportional to the audio amplitude.

Dan Shannon of Brownsville, Texas, sent in the elegant vibrato circuit drawn in Fig. 1, an electronic phase-shifter that produces a true vibrato effect, pure frequency modulation without attendant amplitude modulation. It has eight op-amps, two bipolar transistors, and seven N-type FET's. He supplied no specifications, but the circuit does not look unduly critical and will probably work with a wide range of standard parts.

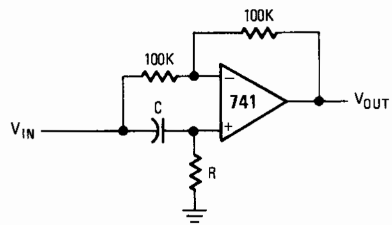


FIG. 2—A PHASE SHIFTER STAGE.

The signal from the input jack feeds the base of emitter follower Q1. The signal splits two ways, one path feeds the string of six phase-shifting 741 op-amps. A typical stage looks like the one in Fig. 2. Because of the op-amp's very high negative-feedback gain, let's make the familiar and reasonable assumption that the signal input at both the inverting and non-inverting inputs are equal in amplitude and phase. The inverting input then has a signal level equal to $\frac{1}{2}(V_{in} + V_{out})$. The non-inverting input is fed through a phase-shift network that consists of capacitor C and resistor R. This works out to $V_{in}\ (j\omega RC/(1 + j\omega RC))$, where j represents a 90° reactive phase-shift for sinewave inputs and $\omega = 2\pi f$ (f is the frequency of the input signal). Now this must equal the non-inverting input so with some algebraic juggling you end up with $V_{out} = V_{in}\ (1 - j\omega RC)/(1 + j\omega RC)$.

I've bothered to go into this much detail because this result is a very interesting circuit response. It can also be produced with a passive non-amplifying circuit, but there will be a loss of gain through it. First of all the magnitude of the gain or the signal output for the 741 phase-shifter remains fixed as the frequency changes. The stage gain is flat across the band.

These types of circuits are called all-pass networks because of their flat signal transmission.

How about phase? $\Delta\phi = \tan^{-1} 2\omega RC$. In Fig. 1, R is made up of a fixed 100K resistor and the variable resistance of an FET. At any particular frequency the phase shift is a function of R or the control voltage of the FET. Though the phase shift changes with frequency, the function is linear. The output will be delayed in time but undistorted in waveform. Cascading the six stages multiplies the phase shift and increases the phase sensitivity of the circuit.

The output of the phase-shift chain is combined with a portion of the input from the emitter of Q1. FET Q8 is a transmission gate operated by a foot switch to turn the vibrato on and off as desired. Switched off, the unshifted signal path continues to provide a

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STATE OF SOLID STATE

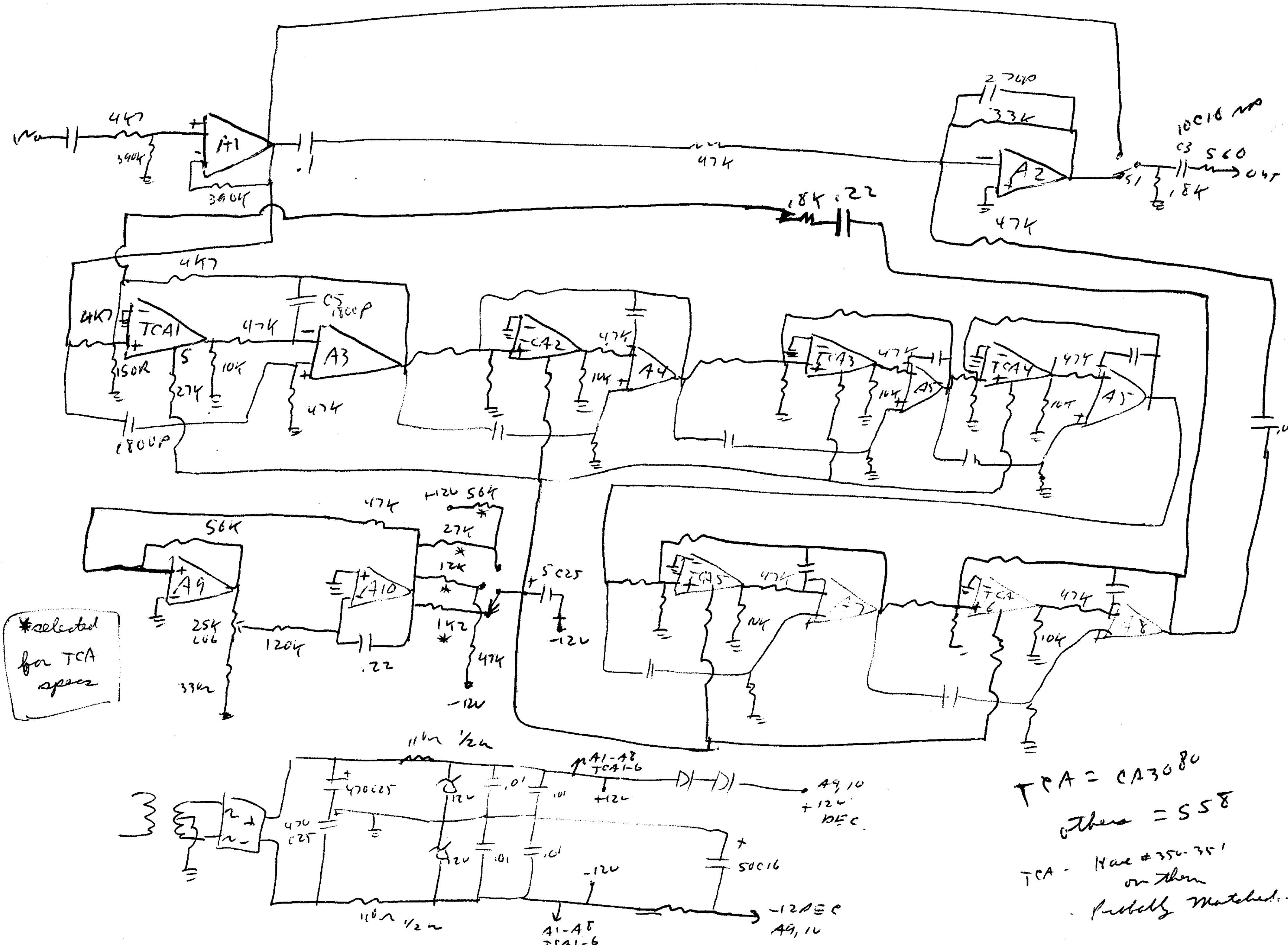
continued from page 71

vibrato-free output.

IC7 and IC8 is an oscillator circuit that controls the amount of phase shift. The output of IC8 feeds control FET's Q2 through Q7. The shift rate of the system is determined by the frequency of the oscillator, which has both a variable and fixed-rate mode. Visual indication of the vibrato rate is provided by the incandescent lamp driven by power transistor Q9 switched by the oscillator output from IC8.

Bridge rectifiers complete the circuit by providing the positive and negative voltages for the op-amps as well as the other circuitry.

Mutron Phaser · Musitronics



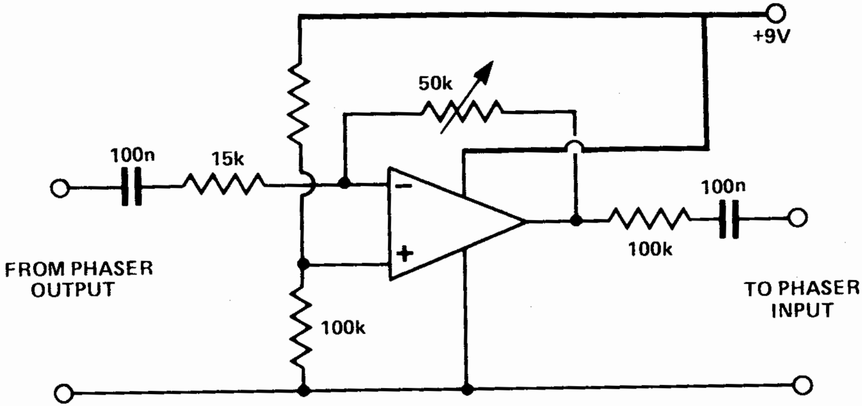
*selected for TCA specs

TCA = CA3080
others = 558

TCA - Have #350-351 on them
Probably Matched..

Phaser Mod

M. Headey



I constructed a simple variable gain op amp inverter and connected it between the output and the input.

When the feedback amp was switched into circuit the effect was dramatic. The phaser sounded much deeper.

The modification is simple enough and though can be adjusted to feedback (audio) level, sounds very good if the gain is kept down.

The circuit as shown gives very good results although you may be able to suggest some component value changes.

CCD PHASER

Astound your ears with this solid-state phaser using the latest CCD technology. Designed by David (White Noise) Vorhaus, inventor of the musical drainpipe!

PROBABLY THE MOST sought after effect in rock music is 'jet plane' sound — or phasing as it is properly called. The effect is very distinctive, and lots of firms have produced units that imitate it. The reason we say imitate is because of the way 'real' phasing is produced — which up until recently required three tape decks, a lot of skill and even more patience!

The Real McCoy

To produce phasing in a studio you record a sound onto two tapes, then replay both tapes simultaneously via a mixer onto the third machine. Because of slight variations in playback speed (usually introduced by physical slowing of a spool), the two signals shift slightly relative to each other — this produces phase differences over the entire spectrum of the sound.

This gives the 'real' phasing that musicians know and love. Too much slowing of a spool results in echo. Obviously you cannot use this technique in real time on stage, so various other ways have been devised to produce a similar effect. However, none of the imitations are as good as the real McCoy!

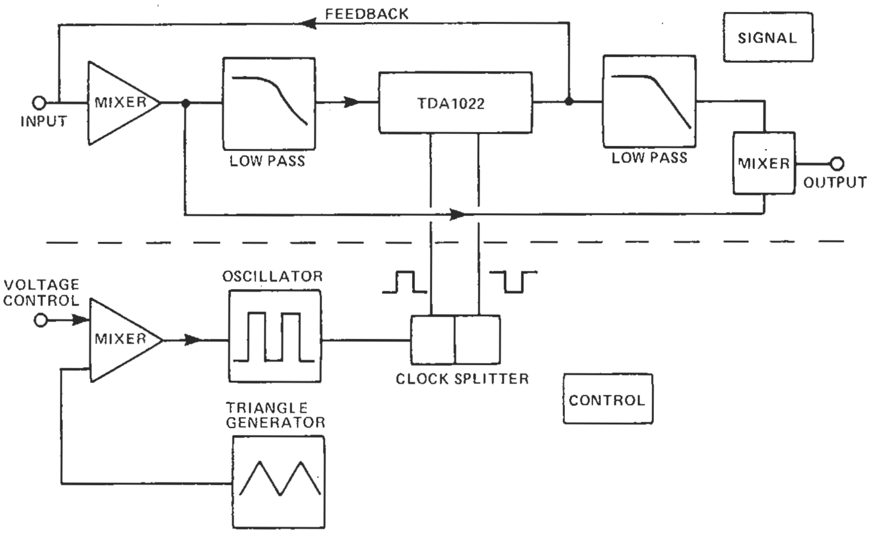


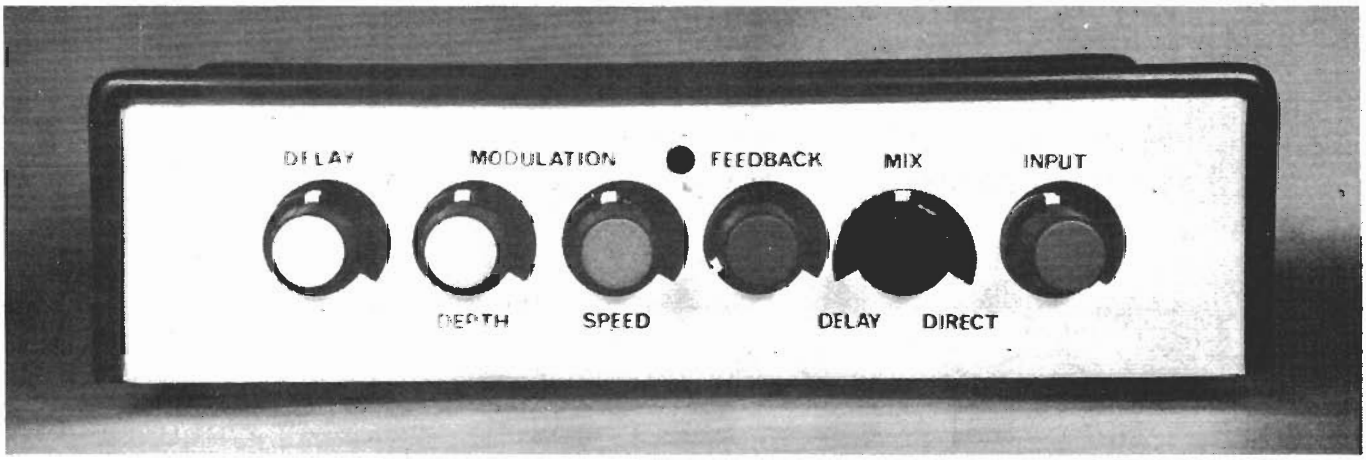
Fig 1. Block diagram of the unit, note how it can be broken into signal path and control section.

To the Rescue

With the advent of analogue delay lines came the opportunity to produce phasing in real time. By feeding the signal through a delay line and mixing the output with the undelayed input you get instant real phasing.

By adding various controls, such as input/output mix and delay length, the versatility of such a unit is increased enormously.

This phaser unit is capable of producing numerous effects — the controls permit variation of all the possible parameters. Phasing, flanging, stereo simulation are just some of the things you can do with it.



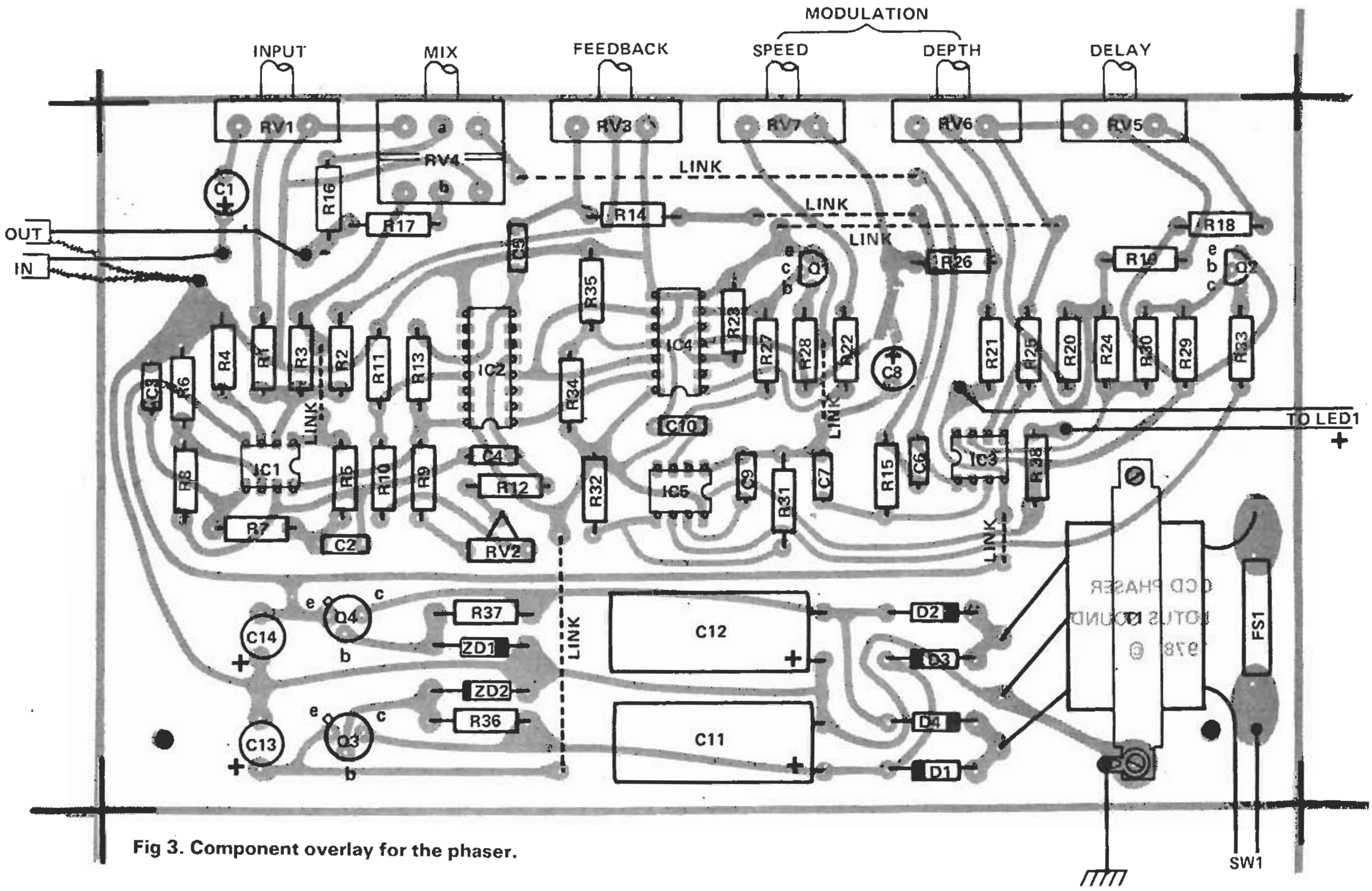
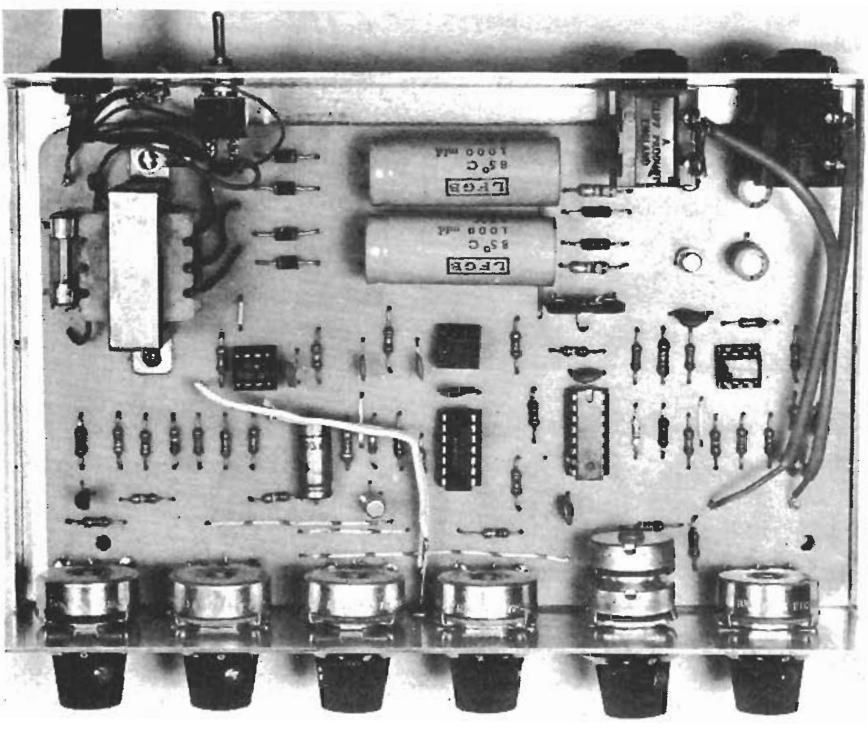


Fig 3. Component overlay for the phaser.



BUYLINES

Lotus Sound are marketing a complete kit (designer approved) see their advertisement in this issue. For those of you who would prefer to buy all the bits separately, most parts are widely available. The only difficult parts may be the PCB pots — try Electrovalue, and the delay line — Watford and Marshalls should have it.

Fig 4. Internal view of one of the prototypes, note that the control spacing has been changed on the final version.

PARTS LIST

RESISTORS (all ¼W 5%)	
R1, 4	22k
R2	180k
R3	220k
R5, 7, 20	15k
R6, 12, 14, 15, 19, 23, 24, 26	100k
R8, 18	56k
R9	2k7
R10	6k8
R11, 33, 36, 37	1k0
R13	47k
R16, 17	5k1
R21	270k
R22	470R
R25	27k
R27, 32	10k
R28	680k
R29	330R
R30	8k2
R31, 38	1k5
R34, 35	1M1
POTENTIOMETERS (all PCB mounting)	
RV1, 6	100k log
RV2	4k7 preset
RV3	100k lin
RV4a, b	10k log/antilog
RV5	50k lin
RV7	25k antilog
CAPACITORS	
C1	10u 10V tantalum
C2, 3	1n0 polyester
C4, 5	100n polyester
C6	200p ceramic
C7, 9	100p ceramic
C8	100u 12V electrolytic
C10	10n polyester
C11, 12	1000u 25V electrolytic
C13, 14	10u 16V electrolytic
SEMICONDUCTORS	
Q1	2N5172
Q2	2N3906
Q3	BC108
Q4	BC178
IC1, 3	LM1458
IC2	TDA1022
IC4	CD4013
IC5	NE566
D1-4	1N4001
2D1, 2	12V 400mW
LEP1	TIL 209
MISCELLANEOUS	
FSI	100mA 20mm + holder
SW1	DPST 240V
T1	12-0-12V 100mA

PCB, Case, 6 collet knobs, 1C sockets, 2 jack sockets, etc.

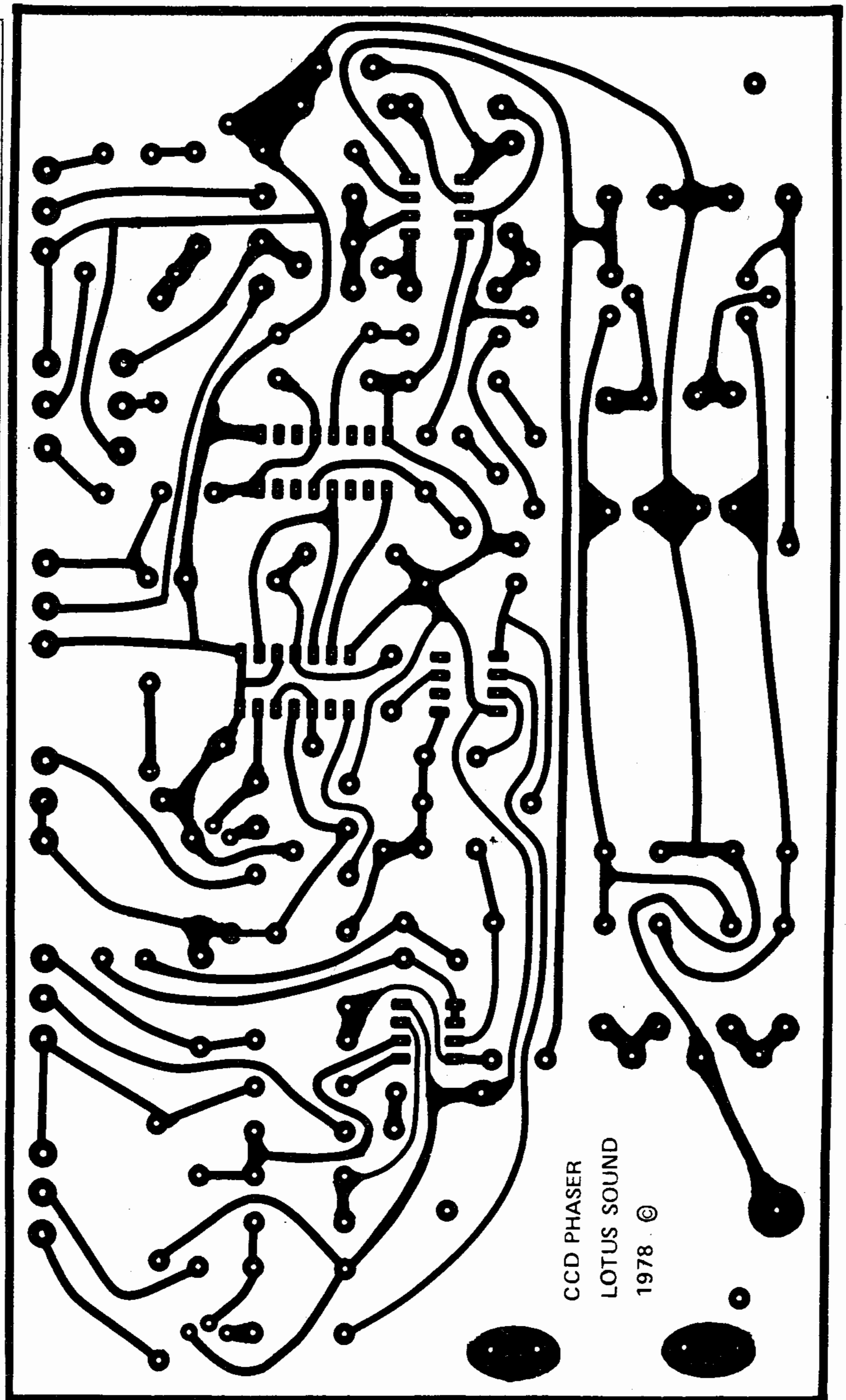
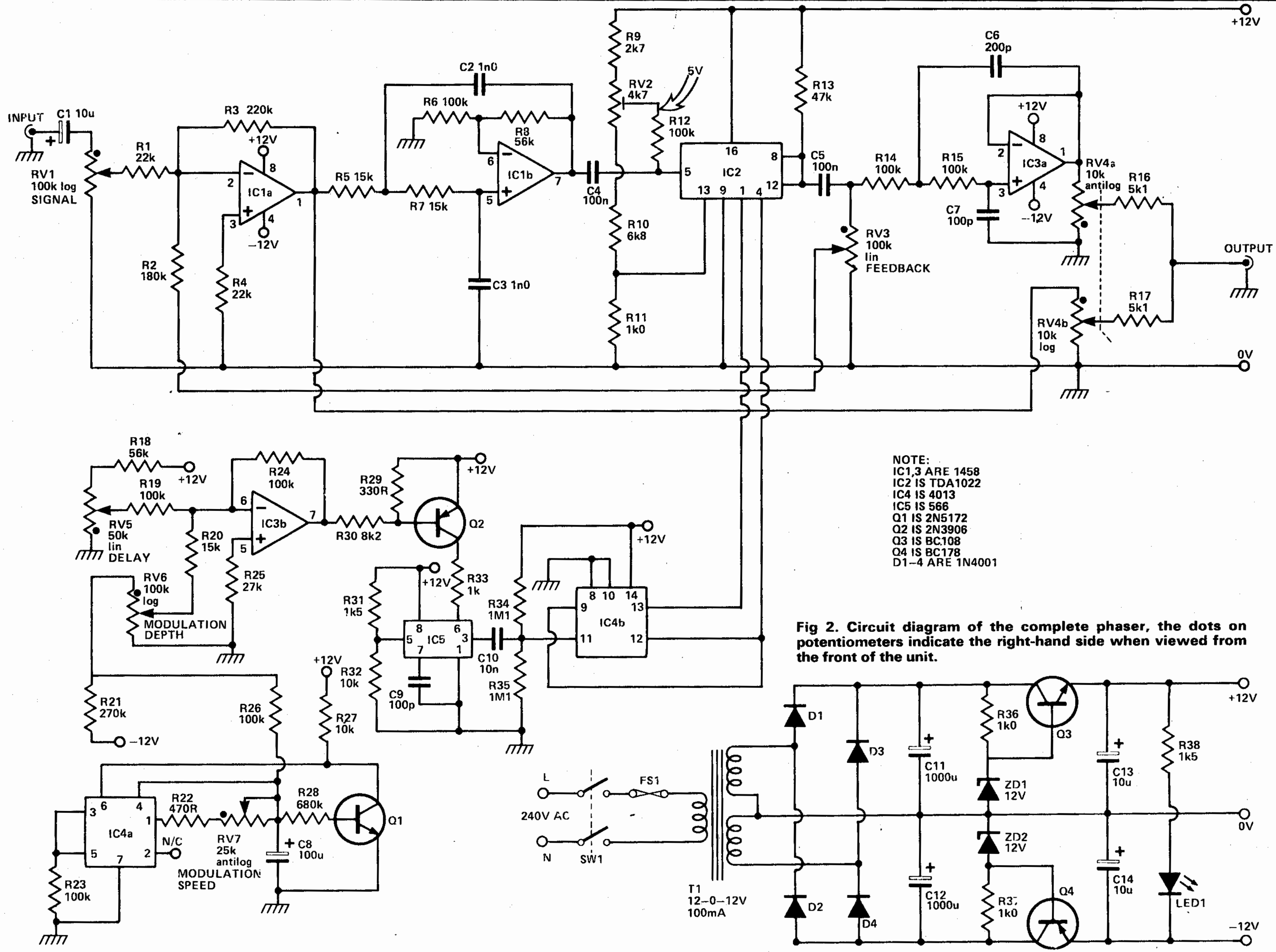


Fig 5. Full size PCB layout for the phaser (195mm x 115mm).



NOTE:
IC1,3 ARE 1458
IC2 IS TDA1022
IC4 IS 4013
IC5 IS 566
Q1 IS 2N5172
Q2 IS 2N3906
Q3 IS BC108
Q4 IS BC178
D1–4 ARE 1N4001

Fig 2. Circuit diagram of the complete phaser, the dots on potentiometers indicate the right-hand side when viewed from the front of the unit.

HOW IT WORKS

The heart of the unit is a 512 stage CCD (Charge Coupled Device) type TDA1022. This particular IC was the subject of our May 1977 data sheet, and the theory of CCDs was covered in the September 1977 edition. Reference should be made to these articles for a description of the TDA 1022 operation. However, even though the TDA1022 is the heart, the rest of the circuitry is the body and will be described in detail.

Figure 1 shows a simplified block diagram of the whole unit, as can be seen, the circuit can be divided into two sections — signal path and control circuitry.

SIGNAL PATH

First the signal path starting with RV1 which is a straightforward 100 k logarithmic level control. From its wiper the signal passes into IC1a, which is connected as an inverting amplifier with a gain of ten (set by the ratio of R1, R2) RV3 (feedback) is also connected to IC1a — the reason will be explained further on.

The output of IC1a is fed into IC1b and also to RV4b (direct level). IC1b is connected as a second order Butterworth low pass filter, with its upper —3dB point at 10kHz and a gain of approximately 4 dB in its passband. There are two reaons for this configuration. Firstly, if the input to the delay line has a frequency greater than half the delay line's clock frequency, the result is distortion. The delay line will operate with clock frequencies as low as 5 kHz — as this would limit input frequencies to below 2.5 kHz a tradeoff has to be made. The clock (described later in the control section) works in the range of 5 kHz to 400 kHz, but as the most useful effects are above 20 kHz, 10 kHz was chosen as the input cut-off frequency.

The 4dB gain is required because the delay line has a typical loss of 4dB — if the gain is introduced before the CCD the signal to noise ratio at the output is improved by 4dB.

The input of the delay line is pin 5 which IC1b feeds via C4. The resistor chain R9, RV2, R10, R11 is to hold pin 13 approximately 1V above OV which produces maximum dynamic range in the delay line. RV2 is used to set the DC voltage at pin 5 for class A operation, which minimises distortion. Pins 1 and 4 of the CCD are its clock inputs, which must be 180° out of phase. R13 loads the output, as the line likes a nice standard load to ensure consistent operation. The output feeds via C5 to RV3 (feedback) and IC3a.

The feedback control (RV3) is to enable recirculation of the delayed signal output fed back to the input, via R2. The output filter is IC3a, which is similar to the IC1b filter, in that it is again a second order Butterworth low pass filter, but has unity gain and a —3 dB point of 20 kHz. The cut-off frequency is chosen to eliminate any clock frequency that may be present in the output from the delay line, and hence prevent HF overload of any subsequent equipment.

The output control RV4a,b, enables the user to mix from delayed signal to normal signal, the output from the twin control is resistively mixed by R16, 17. A log/antilog control is used to give a smooth transition with no 'dead band' in the centre of rotation.

CONTROL CIRCUITRY

All of the second section has one purpose — to alter the clock frequency, and hence the delay time, of the CCD. IC4b is a D flip-flop which is wired to give the required two phase input to the delay line. Pin 11 is the clock input to IC4b, this is fed a stream of pulses from IC5 via C10. IC5 is a 566 voltage controlled oscillator, except it is wired as a current controlled oscillator! Pin 5 (the voltage input) is held at 10.5 volts by R31, 32 and pin 6 is fed a variable current provided by Q2. With the values shown the 566 will oscillate over the range 10 kHz to 800 kHz, which produces a clock frequency (after the divide by two of IC4b) of 5 kHz to 400 kHz.

The current injected by Q2 into the 566 is dependent on the voltage from IC3b, fed to its base. This voltage is controlled in two ways. Firstly from the delay control (RV5), the 56 k resistor R18 is to ensure that the control is useful over most of its travel — otherwise the 566 (IC5) could stop oscillating when RV5 was at its positive end.

IC4a and Q1 also control the frequency of the 566 via RV6 (modulation depth) and RV7 (modulation speed). They are connected up as a triangle generator, the frequency being controlled by RV7. The timing function is dependent on the rate of charge (and discharge) of C8, which is directly controlled by R22 and RV7. The triangle waveform produced is mixed with the voltage from RV5 via RV6 and hence changes the voltage at Q2 base — and therefore the delay time.

Construction. 1, 2, 3 . . .

Install the seven links and six terminal pins first. Follow with all the resistors and capacitors — double check polarities on the electrolytic capacitors. Soldercon sockets or moulded sockets should be used for the 4013 and TDA1022, and for the hell of it the other three ICs — ever tried to unsolder one? Put them in now anyway.

Cut the control spindles to length **before** you mount them.

The six front panel controls can now be mounted on the board one at a time, and after careful alignment soldered in place. If you don't use the specified PCB mounting controls — you have to be accurate and cunning, or you'll end up with a dog's hind leg (just like the original prototype — yes folks, even ETI can make mistakes!).

After the control pots comes the transformer, fuse holder and preset RV2, followed by the diodes and transistors.

Insert the two standoffs into the base of the box, then spindles first (not women and children) place the board into the box — it should click into place (if you drilled your holes in the right place).

Nearly there now, fit the mains switch, jack sockets and mains cable. Make sure you wire the live mains lead into the fused side of the transformer.

The LED, nuts and knobs, ICs and you've done it . . . light the blue touch paper and . . .

Set all the controls fully anti-clockwise **except** for the delay control (on the left) which should be fully clockwise. Feed in an audio signal (preferably a sine wave) and put the output through an amplifier.

Rotate the level control (on the extreme right) clockwise, until the signal comes through at the same level it is going in. If all is well rotate the mix control clockwise — as you do so the sound should 'phase'. With the mix control fully clockwise, adjust RV2 for either 5V at its wiper, or symetrical clipping of the output when viewed on an oscilloscope with 2V5 peak to peak going into the delay line.

When RV2 has been set the delay line is operating at its optimum bias point.

The delay control can now be checked, turning it anticlockwise should alter the output signal and near the end of its range a whistle should break through — and the signal should deteriorate into a very 'crunchy' sound, but not disappear completely. If it does stop increase R18 until it appears again.

The modulation control comes into effect when turned clockwise. Modulation speed is increased by clockwise rotation of the speed control. Make sure the delay control is set clockwise initially. With the mix control halfway a regular 'phasing' will occur as the modulation depth is increased, faster as the modulation speed is increased.

Now you can play with a real signal — white noise is particularly nice to feed into the system (fuzz guitar has a lot going for it as well). ETI

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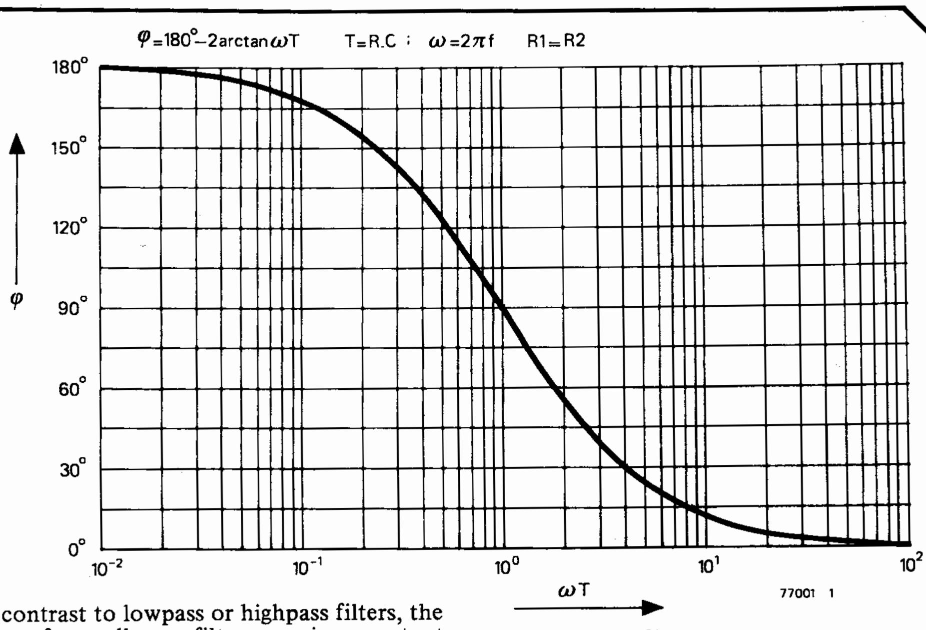
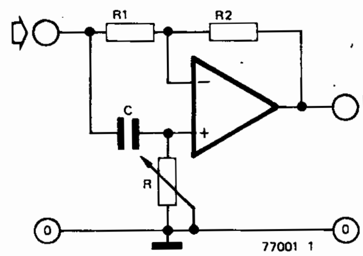
phaser

1





In contrast to lowpass or highpass filters, the gain of an all-pass filter remains constant over the range of frequencies at which it is used, but it does introduce a frequency-dependent phase shift. A number of all-pass filters may be cascaded to produce a phasing unit for use in electronic music. The phasing effect is produced by phase-shifting a signal and then summing the original and phase-shifted versions of the signal.

Figure 1 shows the basic circuit of a first order all-pass filter. The phase-shift is dependent on the relative values of R and C and on the input frequency. At low frequencies C has a very high impedance and the circuit simply functions as an inverting amplifier, so the phase-shift is 180°. At high frequencies the impedance of C is low and the circuit functions as a non-inverting amplifier with zero phase-shift.

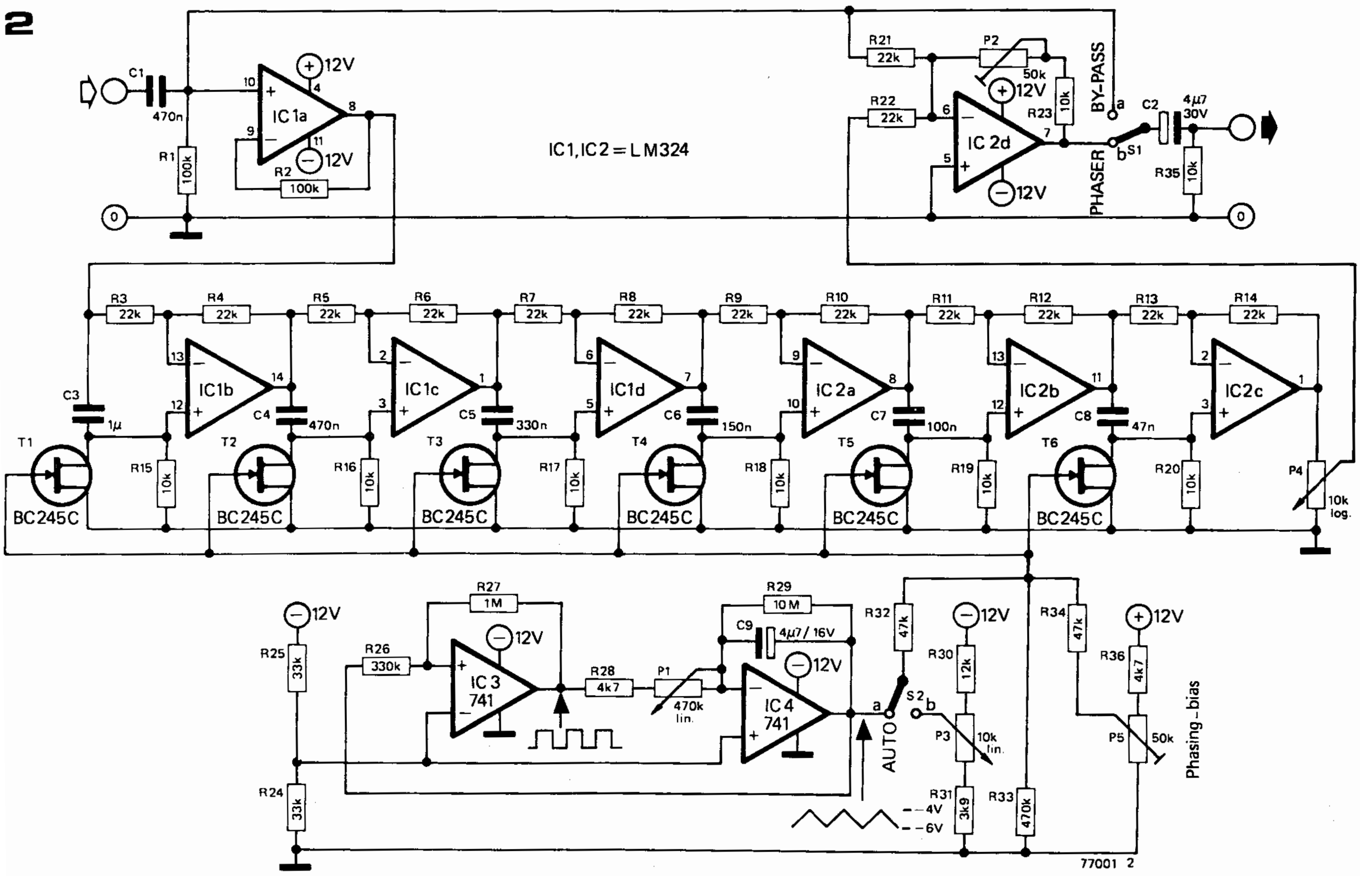
The gain of the filter depends on the relative values of R1 and R2. In this case R1 and R2 are chosen equal so that the gain is unity.

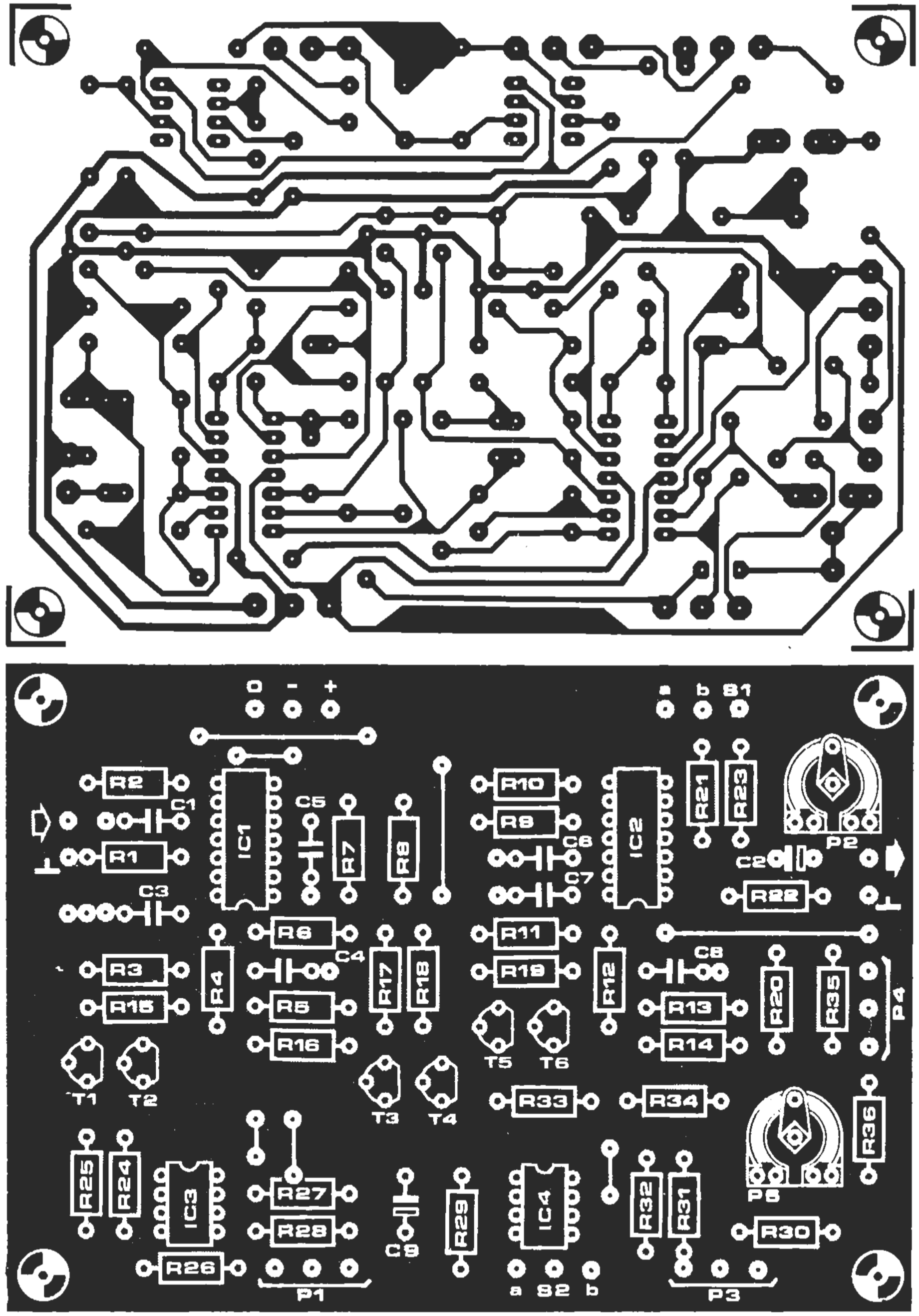
The graph shows the phase-shift v. frequency curve for the filter.

The complete circuit of a phasing unit using all-pass filters is shown in figure 2. Six all-pass filter stages are cascaded, so the total phase-shift at low frequencies can be up to 1080°! The use of a total of ten op-amps in the circuit may seem rather excessive, but as eight of these are LM324 quad op-amps the total package count is only four ICs.

IC1a functions as a unity gain input buffer

2

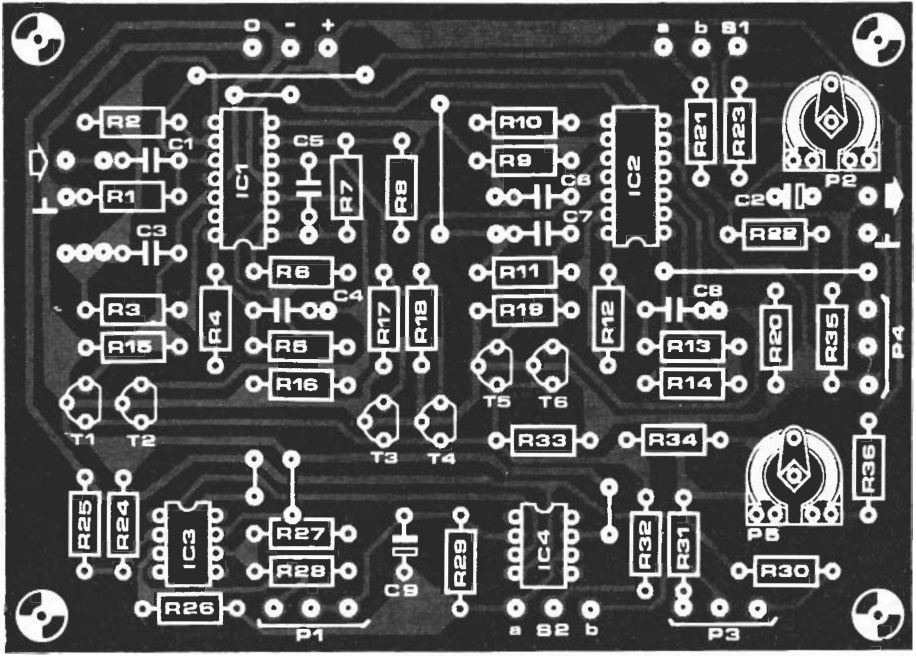




and IC1b to IC2c are the six filter stages. The direct and phase-shifted signals are summed by IC2d. The proportion of phase-shifted signal and hence the depth of phasing can be adjusted by means of P4.

The degree of phase-shift at a particular frequency can be varied by FETs T1 to T6, which function as voltage controlled resistors. By varying the gate voltage the drain-source resistance can be increased or decreased, thus altering the effective value of 'R' in each all-pass filter and hence varying the phase-shift. This may be controlled either manually by means of P3 or may be swept up and down automatically by the output of the triangular wave generator consisting of IC3 and IC4. As the gate voltage of the FETs must always be negative the output of this oscillator swings between −2 and −6 V.

The oscillator frequency may be varied by means of P1, and the best phasing effect occurs at frequencies between 0.5 Hz and 1 Hz. At higher frequencies (around 4 Hz) the phasing effect is lost but a vibrato effect is obtained instead.



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G. Knapienski and F. Mitschke

phasing

Nowadays there are a great number of electronic sound effects. A favourite methods of producing unusual effect is 'Phasing' and in this article this is accomplished, somewhat unusually, by using a 'path filter'. This is a cheap alternative to the already well-known, charge-coupled analogue shift register or 'bucket brigade' memory.

Phasing occurs when a portion of a signal is delayed and then mixed with the original signal. In the middle of the audio spectrum delays of less than about 100 μs will produce no noticeable effect, whilst delays greater than about 30 ms will produce a distinct echo. A delay between these limits will give the required 'phasing' effect.
Of course, a fixed delay time will not have the same effect on signals of all frequencies. If, for example, a 1 kHz signal is delayed by exactly 1 ms and mixed at equal amplitude with the original, then the result will be a signal with twice the amplitude of the original, since the delayed signal has in fact been phase-shifted by 360°. For a 500 Hz signal, however, the situation is quite different. Here a 1 ms delay corresponds to a 180° phase shift, so if the delayed signal is mixed with the original signal the two will cancel, resulting in no signal. This cancellation will occur for all frequencies for which the delay time is an odd number of half-periods. For example with a delay time of 1 ms and a 1.5 kHz signal, the delayed signal is phase shifted by 540°, or 3 half cycles. At 2.5 kHz the delayed signal is phase shifted by 5 half-cycles.
As with the 1 kHz signal, all signals for which the delay time is an even number of half periods have their amplitude doubled. This is true for 2 kHz, 3 kHz, 4 kHz etc. The result is a series of peaks and nulls throughout the spectrum, as shown in figure 1. A circuit that produces this type of response is known as a 'comb filter', because of the unusual shape of the response curve.

Practical Realisation

Early attempts at phasing often used tape recorders running slightly out of synchronism, but this entails a number of difficulties, not least of which being that the sound is not 'live', and consequently the musician cannot adjust the sound during the performance.
There are numerous methods of achieving 'live' phasing.
Electrical delay lines are impractical for the relatively long delay times required. Electromechanical delay lines can be used to give the required delay, but their delay times are fixed by their mechanical dimensions. All-pass LC or RC phase-shift networks may also be used, but these have the disadvantage that the phase shift cannot easily be varied over a wide range. An obvious solution would be to use an analogue shift register such as the TCA590, but these devices are rather expensive.
A cheap alternative is the path filter, the principle of which is shown in figure 2. S1–S7 are closed and opened successively at a high rate, i.e. S1 is closed, then S2 is closed while S1 is opened, then S3 is closed while S2 is opened and so on. This cycle is repeated continuously. When a particular switch is closed, the associated capacitor can charge from the input voltage through the input resistor R. The voltage on each capacitor is dependent on the time constant RC (which is fixed if all the capacitors have the same value) the time for which each switch remains closed (which is also fixed) and the instantaneous level of the input signal.
It is therefore apparent that after a cycle of the switch sequence the voltages on the capacitors are a sample replica of the input waveform during that period (albeit slightly distorted due to the non-linear charging of the capacitors).
If successive cycles of the input waveform and the switching cycle occur in the same phase relationship, then the voltage on each capacitor will eventually become equal to the input voltage at a particular point along the waveform. No further charging of the capacitors will occur, and the input signal will be available at the output. This is true for the frequency at which one cycle of the input frequency is equal to the switching cycle time, and also for multiples of that frequency.
At other frequencies the signal is heavily attenuated. Consider what happens when a half-cycle of the input waveform is equal to the switch cycle time. Imagine that on the positive half-cycle the peak of the input waveform is stored on C4 in figure 2. During the negative half-cycle S4 will be closed at the trough of the waveform. The net voltage on C4 will be zero. This is true for the other capacitors, so the output signal is zero. This will also occur at all frequencies

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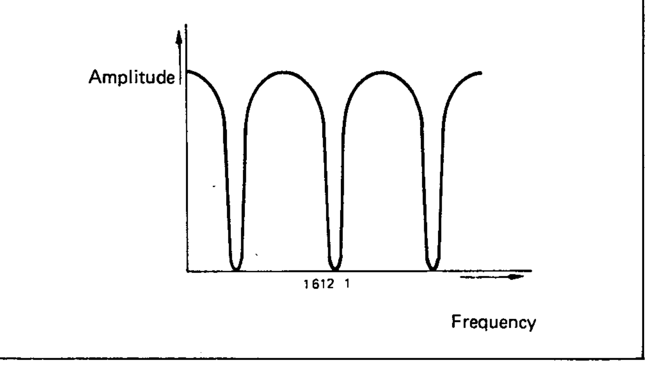

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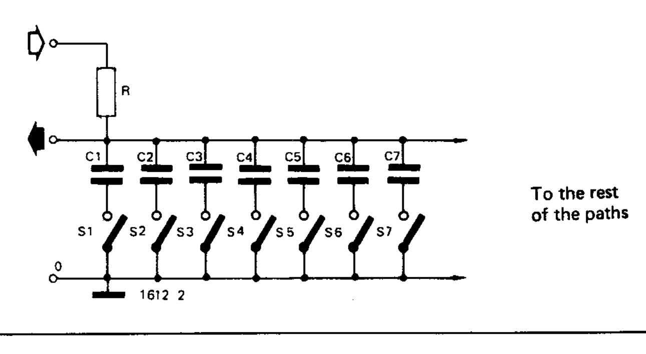
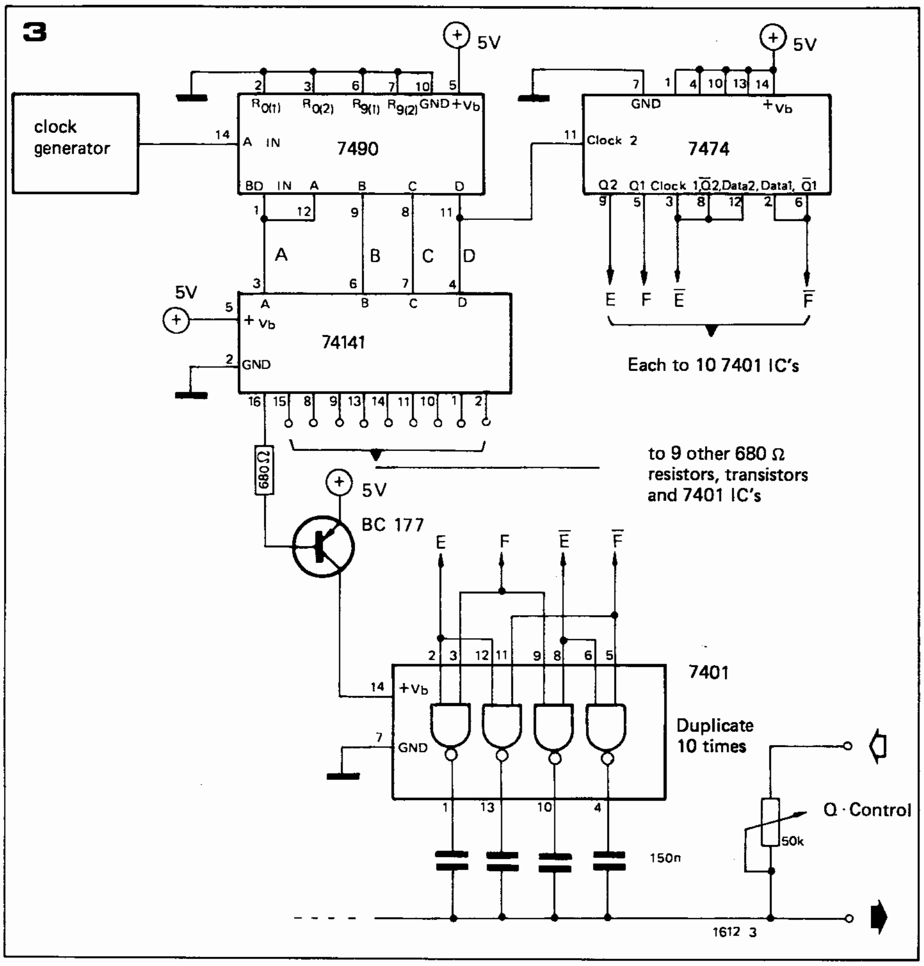


where an odd number of half-cycles is equal to the switch cycle time..
In practice, of course, the switching is accomplished electronically, for example by a ring counter. The result is a comb filter whose rejection frequencies can be varied by varying the clock frequency of the ring counter. The Q-factor can be altered by the single input resistor, R. Distortion of the output signal may be reduced by increasing the number of 'paths', i.e. the number of capacitors.
A practical realisation of a 40-path filter is shown in figure 3. A 7490 decade counter and a 7474 dual D-flip-flop form a divide-by-40 counter. The outputs of the 7474 are decoded by ten 7401 packages, each of which switches four capacitors, making 40 in all. The outputs of the 7490 are decoded by a 74141 BCD-to-decimal decoder/driver and used to switch the supplies to the 7401's via PNP transistors. The capacitors are thus arranged in a 4 x 10 matrix, and are switched as follows:
At the start of a cycle the outputs of the 7474 are all '0' so the capacitors connected to pin 4 of each 7401 are switched in sequence as the 7490 counts from 0 to 9 and the supplies to each 7401 package are switched in turn. When the count reaches 10 output E of the 7474 becomes '1'.
The capacitors connected to pin 13 of the 7401's are switched as the 7490 counts the second decade, and so on. The Q-control is provided by the 50 k potentiometer. The signal source must have a low output impedance and the output of the filter must be connected to a high impedance load.

Applications

This filter has a very narrow bandwidth, with the Q-control at maximum typically less than a semitone. Various effects can be obtained with the circuit. If a narrow pulse waveform is fed in, chimes or percussion effects can be produced at the output, depending on the control frequency. Aircraft noises and other engine noises can also be simulated by filtering out harmonics of complex tones.
The phasing effect occurs when a clock frequency is used which is higher than the upper limit of the audio spectrum (say 20 to 100 kHz). The Q-control must be set in a fairly high position.
The path filter may also be used with an electronic organ or synthesizer, to produce strange effects. A particularly unusual sound can be obtained by feeding the clock input of the filter from the signal outputs of an electronic organ (squarewave outputs from dividers, before filtering) and by feeding a noise signal into the signal input. The results are, to say the least, unlike any organ in existence.
The circuit as described does have its limitations. It will not operate effectively below the frequency whose half-cycle is the same length as the counter cycle. Lowering the clock frequency to compensate for this introduces problems



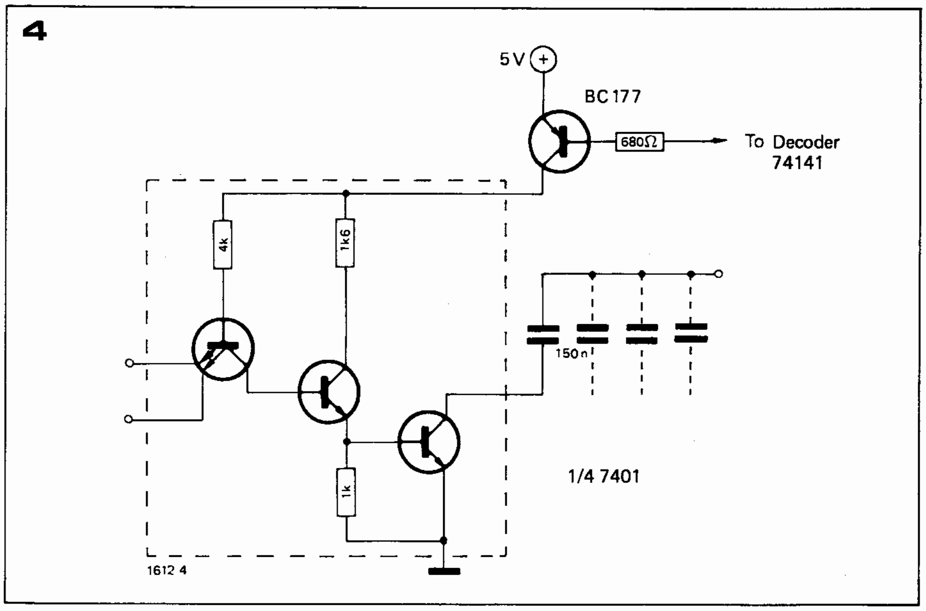
with noise due to the clock frequency and the switching of the supplies to the 7401's. This is aggravated by differences in the characteristics of the 7401's and of the transistors. Increasing the number of stages so that a higher clock frequency may be used will overcome many of these problems.

Figure 1. Frequency response of a comb filter. Frequencies phase-shifted by odd multiples of 180° are almost completely rejected.

Figure 2. Principle of a path filter. All capacitors have the same value and the number of capacitors may be optionally increased almost indefinitely.

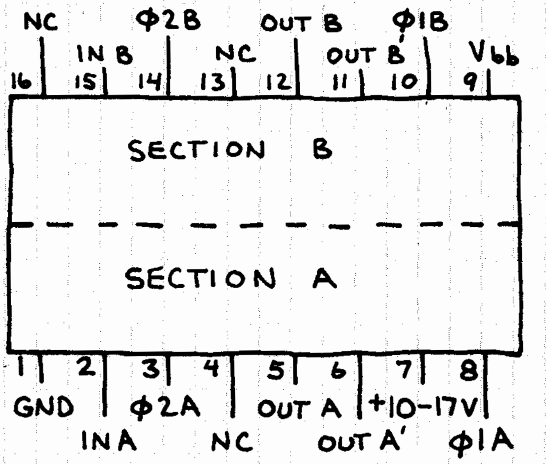
Figure 3. Circuit for a practical path filter. The 7401 and the associated supply switching transistor are duplicated 10 times. Each 7401 is connected to the outputs of the 7474 as shown.

Figure 4. Showing the internal circuitry of one gate in a 7401 package, and how each capacitor is connected.



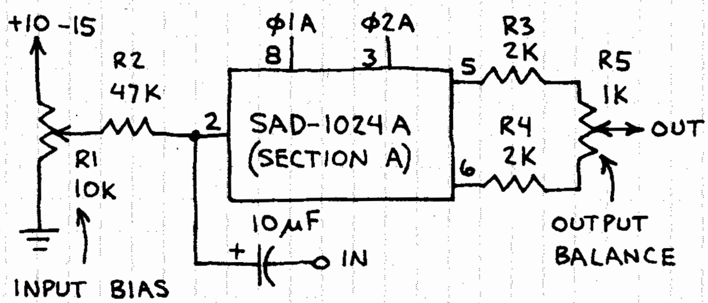
DUAL ANALOG DELAY LINE SAD-1024A

CONTAINS TWO INDEPENDENT 512 STAGE SERIAL ANALOG DELAY (SAD) LINES (ALSO CALLED ANALOG SHIFT REGISTERS). OK TO USE EACH 512 STAGE SAD SEPARATELY OR IN SERIES. ANALOG DELAYS OF UP TO ½ SECOND CAN BE ACHIEVED. A 2-PHASE CLOCK IS REQUIRED TO DRIVE INPUTS $\phi 1$ AND $\phi 2$. INPUT DATA RIDES THROUGH THE SAD ON ALTERNATING CLOCK PULSES AND APPEAR AT THE TWO OUTPUTS AFTER PASSING THROUGH ALL 512 STAGES. CONNECT V_{bb} TO V_{DD} (PIN 7) OR, FOR OPTIMUM RESULTS, TO 1 VOLT BELOW V_{DD}. THIS CHIP CAN BE TRICKY TO USE SINCE SEVERAL EXTERNAL ADJUSTMENTS ARE REQUIRED. CIRCUITS ON THIS PAGE EXPLAIN OPERATING REQUIREMENTS WHILE A COMPLETE CIRCUIT IS SHOWN ON FACING PAGE.

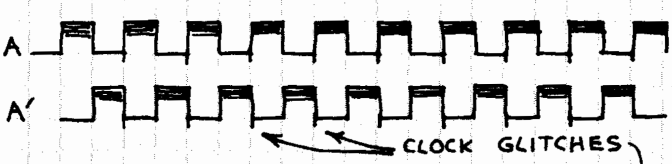


CAUTION: THIS NMOS CHIP IS VULNERABLE TO DAMAGE FROM STATIC DISCHARGE! FOLLOW CMOS HANDLING PROCEDURES.

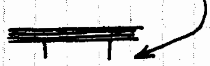
SAD IN/OUT CONTROLS



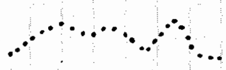
ADJUST R1 (INPUT BIAS) FOR OPTIMUM AUDIO OUTPUT. OUTPUTS APPEAR LIKE THIS ON A SCOPE:



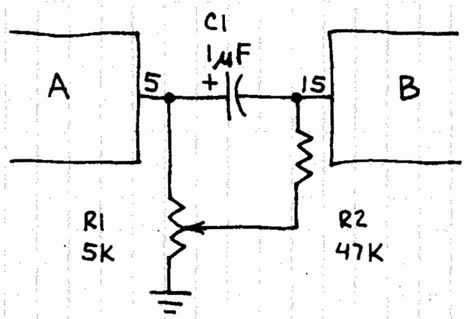
SUMMED OUTPUTS (A+A'):



SET SCOPE TO VISUALIZE INPUT SIGNAL (COMPRESSING CLOCK RATE):

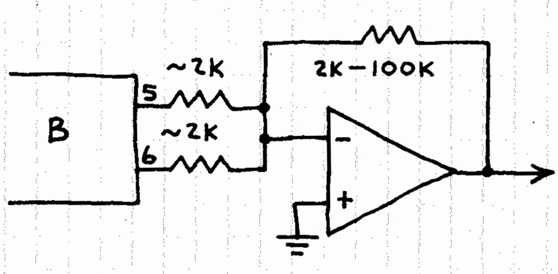


SERIAL OPERATION



R1 CONTROLS BIAS TO SECTION B. NOTE THAT ONLY ONE OUTPUT OF A IS CONNECTED TO INPUT OF B.

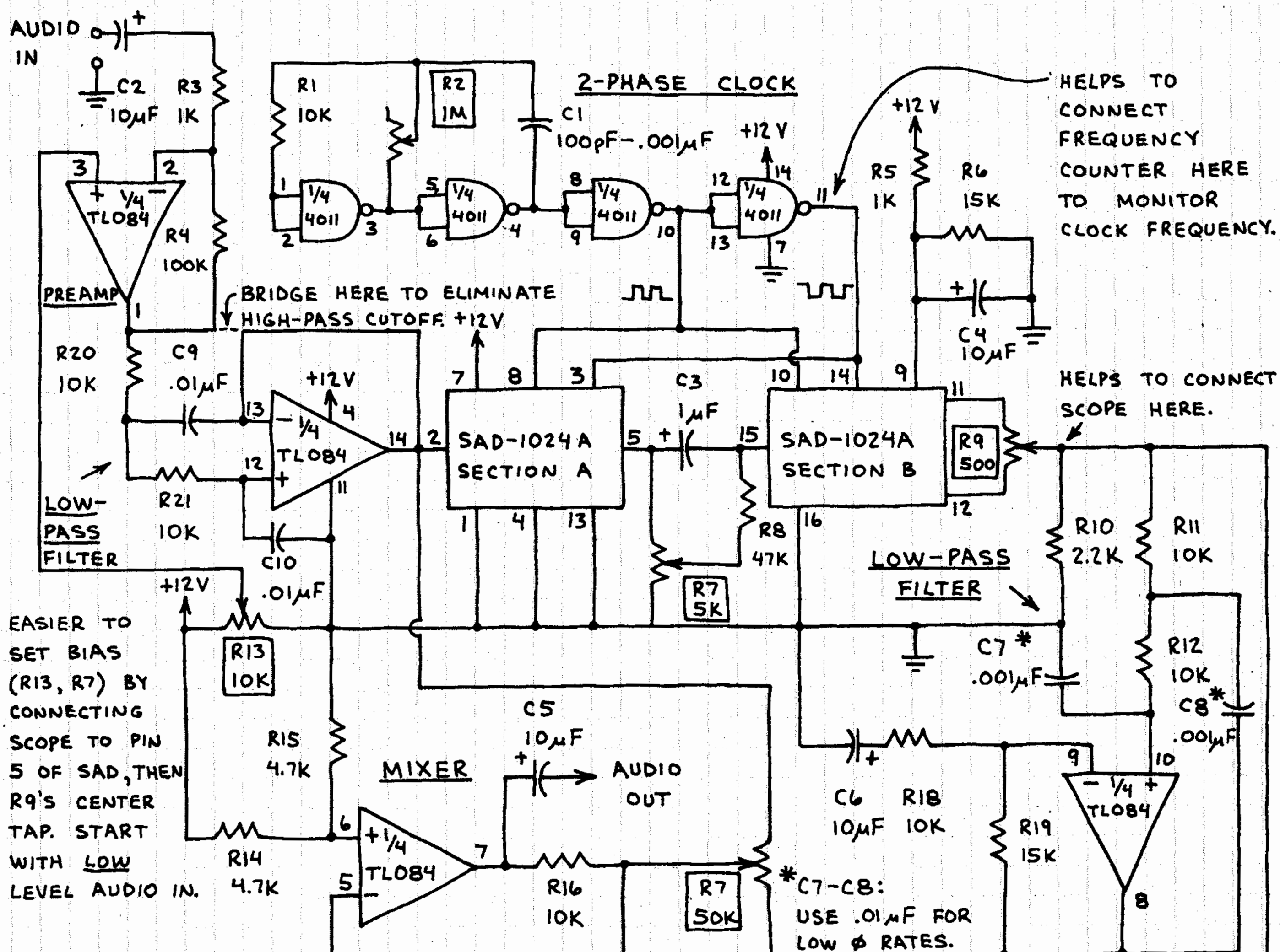
OUTPUT SUMMER



ANY OP-AMP CAN BE USED, BUT LOW NOISE FET INPUT TYPES ARE BEST.

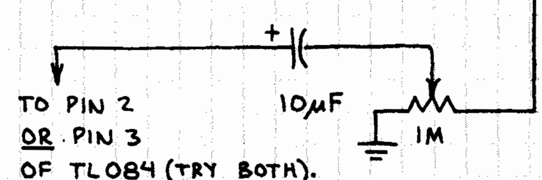
DUAL ANALOG DELAY LINE (CONTINUED) SAD-1024A

ADJUSTABLE FLANGER OR PHASER

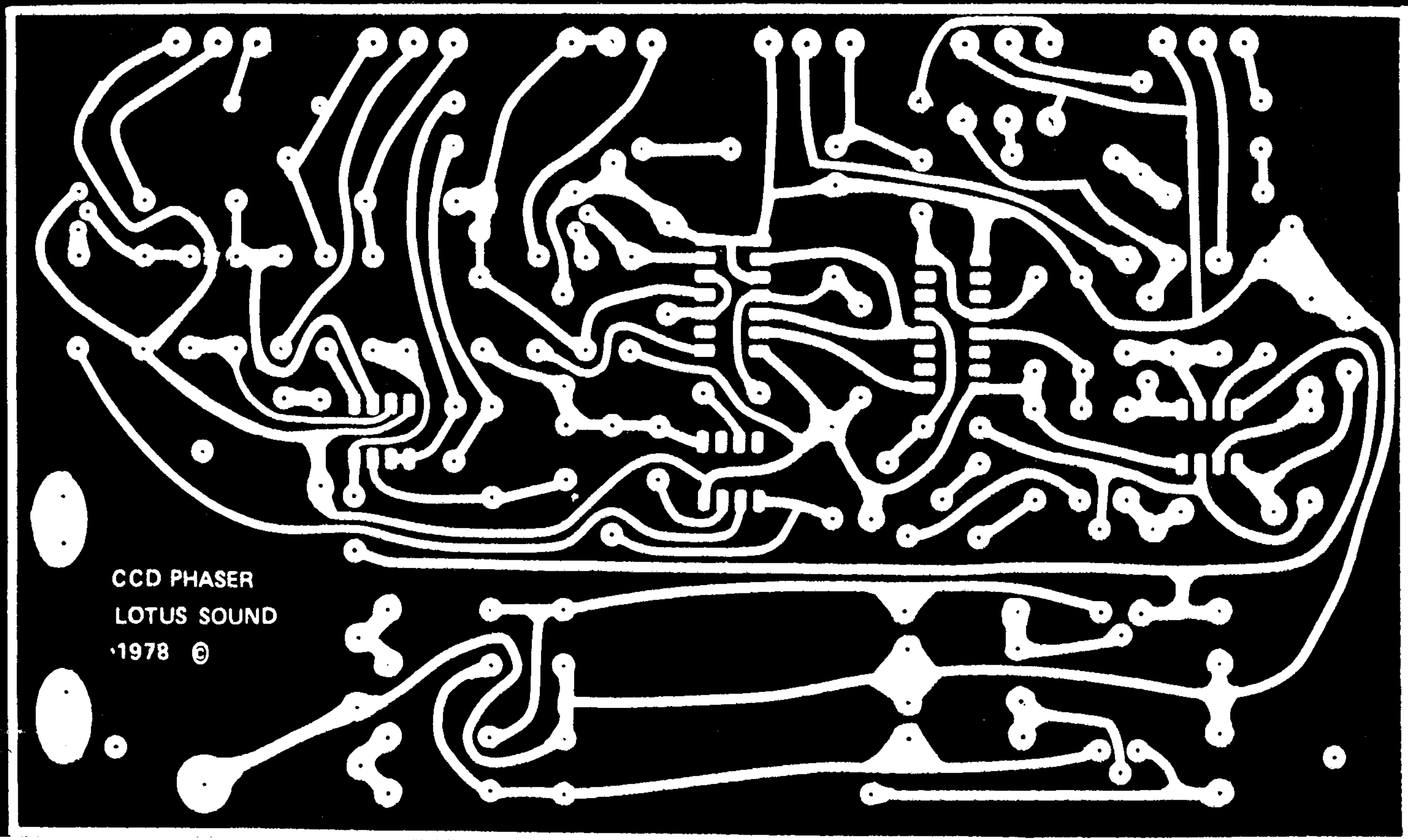


ADJUST CIRCUIT FOR DESIRED EFFECT BY CONNECTING TRANSISTOR RADIO TO AUDIO INPUT. TUNE RADIO TO A TALK SHOW FOR BEST RESULTS. R13 AND R7 CONTROL BIAS TO SECTIONS A AND B OF THE SAD. R9 BALANCES THE SAD OUTPUTS. R2 CONTROLS THE CLOCK RATE. R17 IS THE MAIN BALANCE CONTROL. IT CONTROLS THE RELATIVE AMPLITUDES OF THE ORIGINAL AND DELAYED SIGNAL APPLIED TO THE MIXER. CONNECT THE OUTPUT TO A POWER AMPLIFIER. YOU MUST ADJUST BIAS CONTROLS PROPERLY FOR BEST RESULTS. SET R2 FOR LOW FREQUENCIES (3-8KHz) FOR SINGLE ECHO. USE HIGHER CLOCK FREQUENCIES (20-100 KHz) FOR HOLLOW, SWISHY SOUNDS. NOTE: THIS CIRCUIT IS NOT FOR BEGINNERS.

REVERBERATOR



ADD THIS FEEDBACK CIRCUIT FOR UNUSUAL REVERBERATION EFFECTS. SLOW CLOCK FREQUENCIES GIVE MOST STRIKING REVERBERATIONS. TRY 5-20 KHz. FASTER CLOCK (20-100 KHz) AND CAREFUL ADJUSTMENT GIVES ROBOT-LIKE SOUND USED IN SOME SCIENCE FICTION MOVIES.


CCD PHASER
LOTUS SOUND
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