Mono/Stereo Chorus & Flanger

JOHN M.H.BECKER PART ONE

THE majority of circuits published for musical effects units are usually only designed for mono use, and as such are ideal for the performing musician. However, there are many instances where the effects can be beneficially used with stereo prerecorded sources such as cassettes or record players. Home studio recordings can also benefit from special effects enhancement of a composite stereo mix, when facilities do not run to separate multitrack processing.

ACOUSTIC ENHANCEMENT

Solo tracks sometimes lack depth to their acoustic quality, and can sound flat and dull, particularly when close microphone recording techniques have been used. Frequently the microphone has to be kept close to the performer in order to provide sufficient signal strength, and to minimise the intrusion of background sounds. With most recording requirements it is vital to allocate separate microphones to individual performers, or groups of performers, so that with multitrack recording each track can subsequently be treated to its own mix down characteristics. Restoration of acoustic depth can then be given by several methods including echo and reverb units, of which a stereo version was published in *PE September 1984.* Greater depth can also be simulated by introducing a chorus effect.

CHORUS

Basically chorus is the sound produced by two or more performers singing or playing identical music. Naturally none of the performers, however professional, will be precisely in identical pitch, amplitude or synchronisation with the others, and consequently the sound will be characteristically fuller. It is not always possible or desirable to use several performers to produce a full sound, and other techniques are sometimes preferable. One method is to use several identical recordings played back simultaneously, though with a short time displacement between them. This though is not practical for live performances and anyway the sound is subject to the same amount of time and amplitude displacement on each track, and without pitch modification. For a better simulated chorus effect the relationships between the various time and pitch displacements should be perpetually varying. It is feasible, though not very convenient, to constantly vary the playing speeds of the same recorders to give this changing synchronisation, but modern electronics allow a simpler method to be employed, and one that can be used with both live and recorded sources.

ELECTRONIC DELAY

Several chips have become available which can be fed with a signal, and then emit it at the other end after a suitable delay. By varying the amount of delay on a regular or irregular basis, the relationship of the delayed signal to the original can be kept constantly shifting and by mixing the two together the chorus effect can be simulated. The use of such a unit will not replace the need for multi-musician groups, but an enhanced and fuller sound can be created by this simple technique.

Very usable chorus effects can be produced by using only one modulated delay circuit, though if several are used with differently varying delays a closer approximation to the true natural chorus effect can be obtained. Unfortunately in simple electronic chorus units it is not straightforward to satisfactorily provide differently varying delay rates to several delay chips. Certainly each can be driven by a separate delay controlling signal, but there is the danger of reaction between the control signals as these are produced by high frequency oscillators having their clock frequency varied by a modulating voltage. Although the clock signals are normally outside the audio range, if two or more interact then an additional composite signal can be produced that contains low frequency sub-harmonics. As the high frequency signals vary in relation to each other, the sub-harmonics can intrude into the audio spectrum, and cannot be filtered out without also losing essential information from the required audio signal. There are techniques for eliminating the harmonic interaction, but these add to the complexity and the cost. None the less, excellent results can be achieved just by using one or more delays in series all controlled by the same modulated clock frequency. If each delay is fed separately to a mixer stage and combined with the original unprocessed signal, multitracked chorusing results.

DELAY OSCILLOGRAMS

Using a pulse as the signal source, the varying delay relationships of a double tracked chorus unit can be easily seen on an oscilloscope. In the oscillograms shown in Fig. 1a to 1c the first peak is that of the original pulse and the other two are the signals emerging from two cascaded delay chips. The amplitude of the peaks is shown to be different, but it can be readily varied. The clocking frequency is being modulated so that the time delay is constantly increasing and decreasing. The photographs show the progression of the delay from its shortest to its longest. It will be seen that not only are the delayed peaks changing their time delay in relation to the original, but that they are also changing with regard to each other. In addition to the delay changes, pitch

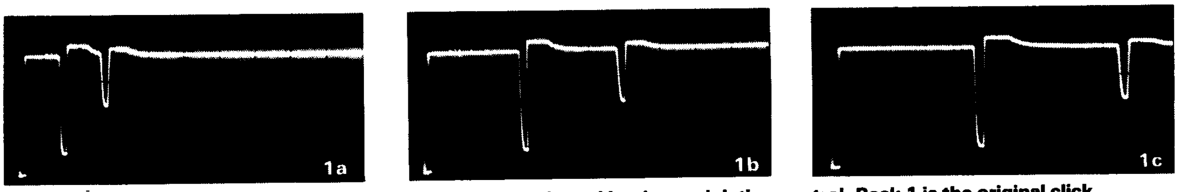


Fig. 1. Progression of pulse echo as delay is lengthened by the modulation control. Peak 1 is the original click, peak 2 is the first echo and peak 3 is the second echo

changes also occur. Consider a note of 440Hz, concert 'A'. Normally the time of each cycle remains constant at 1/440 of a second. If the time between the cycles is varied by introducing a changing delay then by definition the note is no longer the same. In effect the doppler shift principle often associated with approaching or receding fire engine sirens is being introduced. As the distance between the cycles shortens so the pitch increases, and vice versa. Thus in a modulated chorus unit not only is the synchronisation between the original and processed sounds changing, but also the frequency relationship, just as occurs with natural chorus. It will also be apparent that if pitch is being constantly varied, then vibrato is also occurring.

MODULATION

There is an optimum modulation rate that produces the most interesting and satisfying results and is associated with the consequent vibrato depth and rate. If too slow a modulation rate is given, the effect tends to sound similar to a 'wowing' record deck due to the pitch change. Too fast a modulation rate either will produce delay changes too fast to be noticed, or at its worst, will have a frequency that is within the audio spectrum, and will be heard as a low hum. Certainly this could be filtered out, but again only at the expense of the frequency response of the original. The generally accepted ideal modulation rate is about 6·5Hz and analysis of most vibrato and tremolo rates shows a rate within this region. However, slightly faster rates can also produce interesting results. The maximum depth of modulation and thus the maximum degree of pitch change can also be quantified from analysis of the recordings of professional musicians. The results show a strong tendency towards the maximum pitch change of about a quarter to half a tone of the original frequency. With a delay chip the degree of pitch change will only be the same for identical frequencies. Other frequencies present at the same time will be subject to greater or lesser degrees of pitch change. Thus for a composite signal passing through a chorus unit, true vibrato can only be approximated.

FLANGING

Flanging is an effect somewhat related to reverberation and phasing, and can be readily produced with only slight modification to a chorus unit circuit. It has similar feedback qualities to reverb, though with greater resonance, and at the same time consists of a modulated phase change relationship both to itself and to the original signal. It produces a strong tunnel-like effect with an accentuated upper frequency pitch change and under some conditions of speech or singing, this can sound like an eerie additional

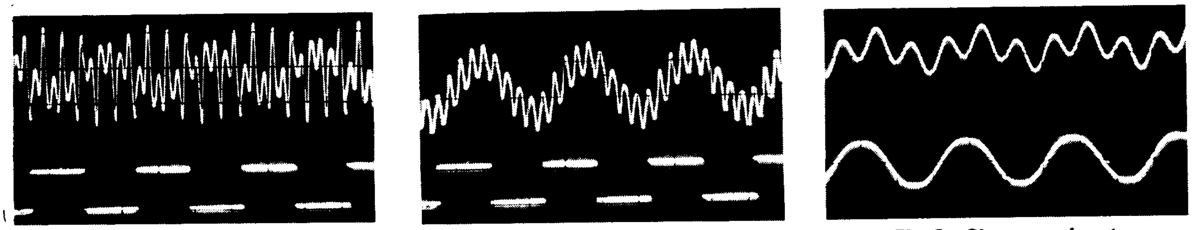


Fig. 2a. Square-wave input

Fig. 2b. Square-wave input with a different delay setting

Fig. 2c. Sine-wave input

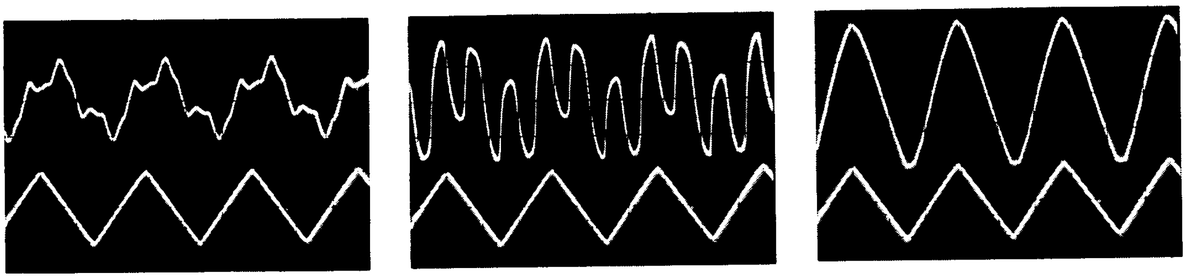


Fig. 2d to 2f. Flanging resonance as modulating sweep progresses.

Fig. 2. The bottom trace is the original and the top trace is the mixed output

background voice accompanying the performer. The effect is produced by passing the signal through the delay stage, the delay rate of which is constantly changing, though at a slower rate than for chorusing. After passing through the delay stage, the signal is fed back upon itself to the start of the delay loop. The amount of feedback is critical. If too much is given, the signal level will increase each time round the loop, resulting in perpetual howl. Whilst howl can be used if well controlled in some effects units, in this instance it is definitely undesirable. If too little feedback is given, the flanging effect does not develop. The correct amount lies within a narrow band, so that the maximum enhancement results without howl. At the correct settings and with the optimum delay times and modulation rate changes, the phase and pitch changes of the feedback loop result in repeated emphasis and de-emphasis of particular frequencies and their harmonics. The oscillograms of Fig. 2a to 2f show some of the effects visible on an oscilloscope.

The filtering occurs of course because the time delay produces a full 180° phase shift at particular frequencies, and when the shifted signal is mixed with the unshifted one, they cancel each other out. The counterpart of the cancellation is when both are in phase, and so enhance each other. (Oscillograms Fig. 3a to 3c.) The enhancement occurs at the peaks of the waveforms, in other words at their edges. By definition a flange is a raised edge, so presumably this is where the name of the effect comes from. The most noticeable flange effect is created with higher frequencies having a high harmonic content, with short delay times modulated at a slow rate. Although the effect is still produced with purer or lower frequency tones, it is less noticeable to the ear. Paradoxically a very pronounced different effect is produced by fast modulation with deep sweeping delay changes. Music then loses its tonal qualities and takes on a very deep whooshing effect, but although it is unmusical, it none the less can be used for dramatic sound changes.

CIRCUIT DESCRIPTION

The dual chorus and flanging unit to be described here has been designed to produce strong chorus and flanging effects with both mono and stereo signals. For normal stereo use each channel retains its independence with the effects being produced by separate processing circuits. With mono signals there is a choice of single or double chorusing, and single or double flanging. Stereo signals can also be combined for treating as a mixed mono signal and given the same doubly enhanced processing, though with the loss of channel separation for the processed part of the signal. A block diagram is given in Fig. 4. In essence the unit consists of two identical signal routes each providing a delay, controlled feedback, and controlled mixing producing separate composite processed signals suitable for presenting to a normal amplifier system. However, the two delay and feedback paths can be cascaded to produce enhanced chorus and flanging with identical composites appearing at both outputs.

BASIC SIGNAL ROUTING

The complete circuit diagram of the Chorus and Flanger unit is shown in Fig. 5. A mono signal, or one half of a stereo pair, is brought into the initial buffer stage IC1a. Two inputs are provided to this stage, one via R1, the other bypassing R1. The route via R1 is more suited to higher level signals from a preamplified source up to about 1·5V r.m.s. The stage gain here is nominally at unity, that is, the same level comes out from IC1a as goes in. The low input bypasses R1, and the gain is then about 10. This input is more suited to sources producing an output up to about 150mV r.m.s., but for retention of the best signal to noise characteristics any input signal should be capable of producing an output from IC1a in the region of 1 to 1·5V r.m.s. and so ideally should be pre-amplified. From IC1a the signal is split, one path going direct to the output mixer stage IC1d, and the other to the filter and mixer stage IC1b. To achieve greater clarity of the processed signal, slight pre-emphasis of the treble regions is given by the parallel path of R5 and C2. The gain of stage IC1b is set by the relationships of R5 to R8, but higher frequencies bypass R5 via C2 and so are given greater gain. Lower frequency signals are forced to pass through R5 as they cannot get through C2, and so only achieve normal gain. The low pass characteristics of IC1b are set by C3 and C33, and limit the very high frequencies which could otherwise produce distortion within the delay stage IC2. Normal waveform balancing through IC2 is trimmed by applying an optimum d.c. bias to its input via VR3. At the correct setting of VR3, equal emphasis is given to both phases of the a.c. signal passing through. IC2 is a superb delay line chip that consists of 1536 separate delay stages within its small 8-pin d.i.l. format. It is a bucket brigade type analogue delay unit that passes a sampled amplitude charge from stage to stage at a rate determined by a controlling clock frequency. The time taken for a sample charge to pass through can be calculated from the formula: time = (delay stages/clock frequency)/2000 where the frequency is in hertz and the answer in milliseconds. The stream of sampled charges emerge from two outputs at the end of the delay, are summed at VR1 and are typically at the same level as they went in. Because the signal has been sampled, a residual trace of the clocking frequency is also contained in the output. This is partially cancelled out by adjustment of VR1. The remainder is filtered out in the stage

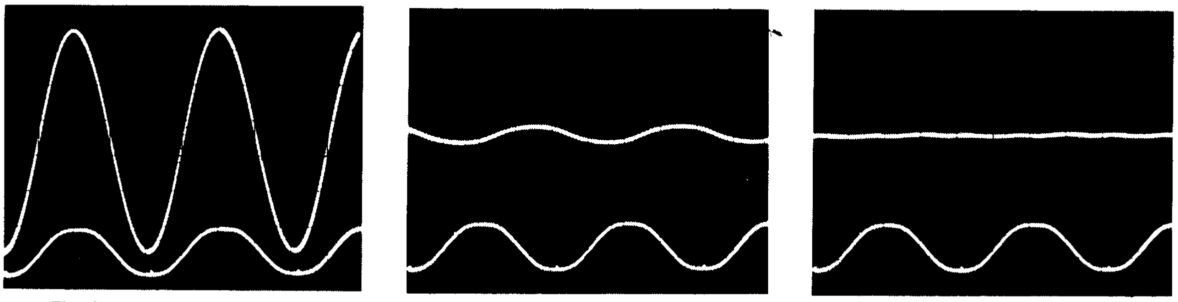


Fig. 3a. In-phase (accentuated) **Fig. 3b. Nearly out of phase** **Fig. 3c. Out of phase**

Fig. 3. Phase enhancement and cancellation. Top trace combined output, bottom trace original signal

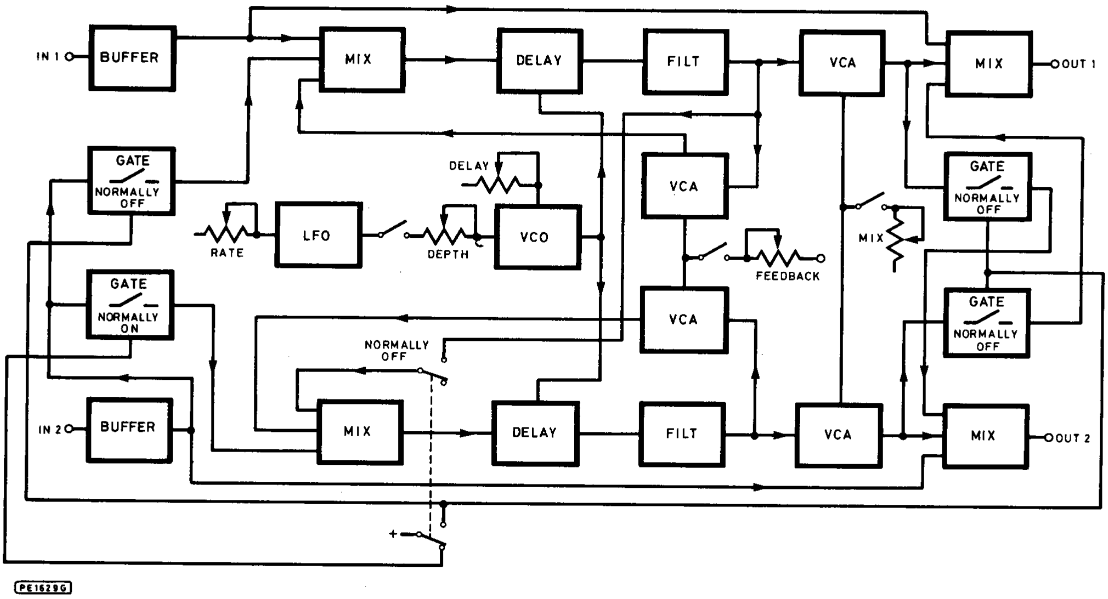


Fig. 4. Block diagram of the Chorus and Flanger unit

COMPONENTS . . .

Resistors

R1, R28	91k (2 off)
R2, R29, R73, R75, R82, R83	10k (6 off)
R3, R4, R6, R7, R8, R9, R10, R14, R15, R16, R17, R21, R22, R23, R24, R25, R26, R30, R31, R33, R34, R35, R36, R37, R41, R42, R43, R44, R45, R49, R50, R51, R52, R56, R57, R66, R71, R72, R74, R80	100k (40 off)
R5, R32, R63	390k (3 off)
R11, R12, R13, R38, R39, R40, R54, R55	47k (8 off)
R18, R19, R46, R47, R59, R60, R61, R65, R67, R69, R70	1k (11 off)
R20, R48	200k (2 off)
R27, R53, R62	470 (3 off)
R58, R64	180k (2 off)
R68	4k7
R76, R78	2k (2 off)
R77	82k
R79	300k
R81	1M2

All resistors $\frac{1}{4}$W 5% high stab. carbon

Capacitors

C1, C4, C8, C9, C13, C18, C20, C21, C23	1µ 63V elect. (9 off)
C2, C10	4700p polystyrene (2 off)
C3, C12, C25	56p polystyrene (3 off)
C5, C6, C15, C16	330p polystyrene (4 off)
C7, C17	180p polystyrene (2 off)
C11, C33, C34, C35, C36	100p polystyrene (5 off)
C14	4µ7 63V elect.
C19	470µ 6V elect.
C22, C26	22µ 10V elect. (2 off)
C24	470p polystyrene
C27, C28	470µ 10V elect. (2 off)
C29, C30, C31, C32	100n polyester (4 off)

Potentiometers

VR1, VR2	4k7 skeleton pre-set (2 off)
VR3	10k skeleton pre-set
VR4	1M skeleton pre-set
VR5, VR6, VR8, VR9	100k lin (4 off)
VR7	1M lin

Semiconductors

IC1, IC4	324 (2 off)
IC2, IC3	TDA 1097 (2 off)
IC5, IC6	LM13600 (2 off)
IC7	4066
IC8	TL082
IC9	4046
IC10	4013
IC11	4011B

Miscellaneous

PP3 battery clips (2 off)
P.c.b. clips (8 off)
Knobs (5 off)
Stereo jack socket (3 off)
8-pin d.i.l. socket (3 off)
14-pin d.i.l. socket (5 off)
16-pin d.i.l. socket (3 off)
S2, S3, S4 s.p.d.t. (3 off)
S1, S5 d.p.d.t. (2 off)
Case BK 3
P.c.b.s 235A, 235B
4 rubber stick-on feet
Wire, solder

Constructor's Note

A complete kit of parts for the Chorus and Flanger unit is available from Phonosonics, 8 Finucane Drive, Orpington, Kent BR5 4ED. Price £55.66 excluding VAT and p&p. Plus £1.00 p&p £8.34 VAT.

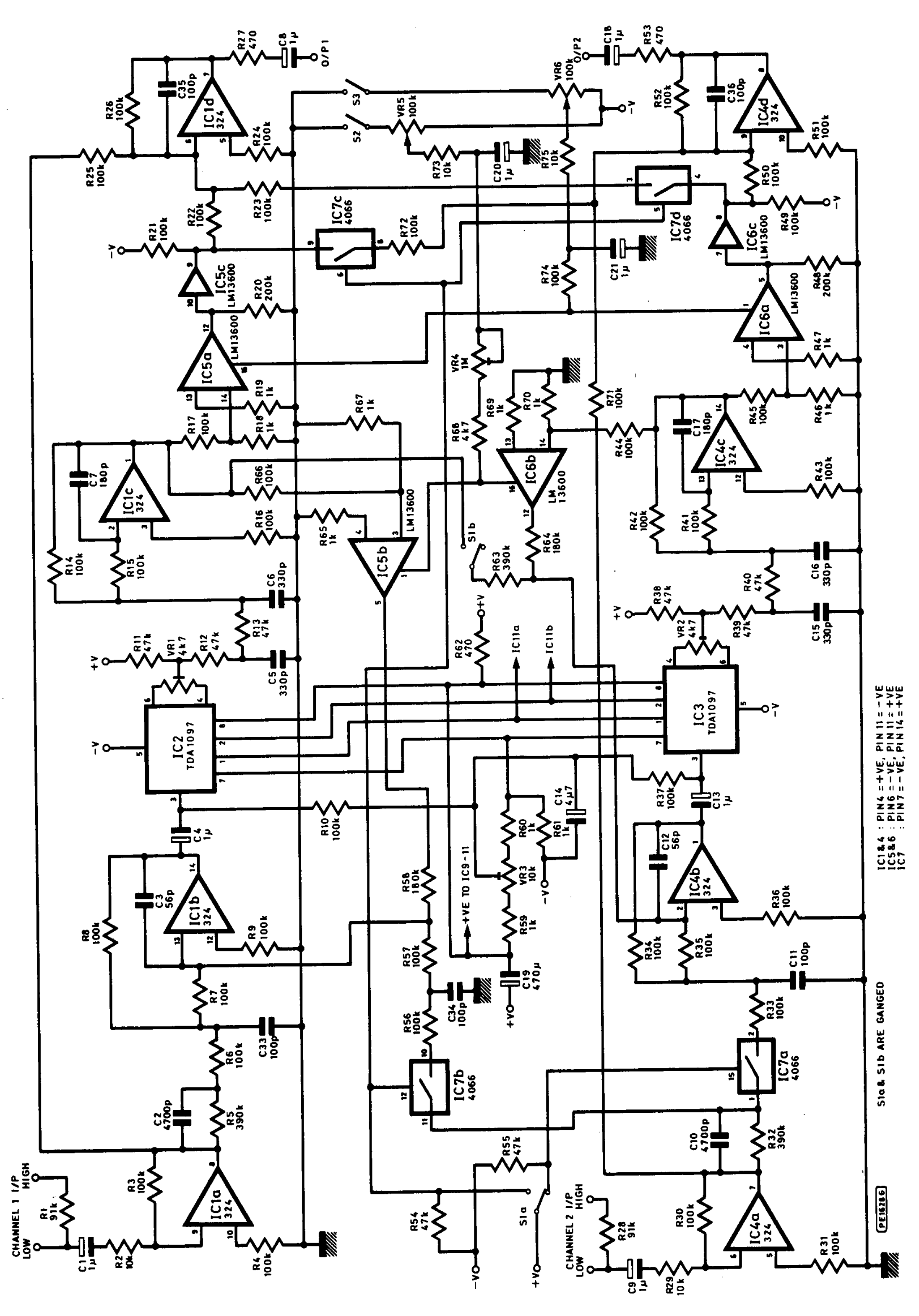


Fig. 5. Circuit diagram of the Chorus and Flanger unit

around IC1c. C5, C6 and C7 perform the cleanup job, leaving only the required delayed audio signal to pass.

VCAs

IC1c is followed by two transconductance amplifiers, IC5a and IC5b. They both conduct a signal through in response to a current on their control nodes. This is derived from the voltage seen at the wiper of the controlling potentiometers VR5 and VR6, and is in proportion to the resistance between the wiper and the control node. IC5b is the VCA controlling the amount of signal fed back to the start of the delay loop at the mixer-filter IC1b. With VR5 at its negative end, IC5b cannot conduct but as VR5 is rotated towards the ground end, so conductance increases. The maximum conductance available is deliberately higher than actually required, so that the precise feedback level can be tightly adjusted by VR4. Switch S2 allows for the feedback to be switched in and out as required. With S2 open the wiper of VR5 is automatically held at a negative level, so that IC5b ceases to conduct. The presence of C20 gives a slight fade in and fade out effect when S2 is used. IC5a governs the amount of processed signal that is allowed to pass through to the final mixer IC1d via the high impedance buffer IC5c. VR6 controls the amount of conductance, S3 permits the effect to be switched in and out, and C21 creates a similar fade slope to that produced by C20. The maximum signal strength arriving from IC5c is set at a little above the original signal level arriving on the bypass route via R25. The combined processed and unprocessed signals pass through the unity gain mixer IC1d, and thence can be fed to a normal amplifier system. Although IC1d contains output short circuit limiting, R27 is included as an additional current limiter so that using a mono jack plug in the stereo socket will not over-tax the endurance of the i.c. The value of R27 is too low to significantly affect the output level.

CHANNEL TWO

Up to this point both channels are essentially identical. The equivalent channel 2 stages consist of the input buffer IC4a, filter mixer IC4b, delay stage IC3, filter IC4c, feedback VCA IC6b, effect level VCA IC6a, buffer IC6c and final output mixer stage IC4d. The control nodes of IC5b and IC6b are directly coupled so that VR5 controls the feedback of both channels simultaneously. Likewise the control nodes of IC5a and IC6a are coupled to the effect level control VR6.

ENHANCED MODE

In the normal mono and stereo modes, each channel can give weight to the amount of effect passed through and fed back. The circuit has been arranged though so that the effect created by channel one can be added to the effect from channel two. Switch S1 in conjunction with the quadruple electronic gate IC7 performs the switchover to the dual emphasis route. With S1 off IC7b–d are held closed to signals by the negative voltage from R54, whilst IC7a allows signals through due to the positive voltage from S1a thus allowing signals to pass from IC4a to IC4b. Switching on S1 reverses the polarity seen at the control nodes of the gates. IC7a then prevents the output of IC4a from reaching IC4b. Instead any signal from IC4a is routed now via IC7b to join up with the signal from IC1a of channel one at IC1b, so presenting a mono composite to the input of IC2. The delay occurs as before, passing through IC4b and IC5a as desired. However, from IC1c the first delay signal is additionally routed via S1b across to the start of the second delay loop at IC4b. The signal is then given further delay and feedback treatment via IC3, IC4c, and IC6b. The signal coming from IC6c then contains double the delay and feedback enhancement to that coming from IC5c. Each of these signals is now cross switched to the opposite output mixers IC1d and IC4d via IC7c and IC7d respectively. The mono doubly enriched chorus or flanging effect is thus present at both outputs. Obviously the stereo processed signal has lost its channel separation identity, though in both channels the bypass signals arriving at the outputs retain their separate identities. The original stereo image is thus preserved but can be overlaid with the mono doubly enhanced composite. If a mono only signal is input to the unit via IC1a, the output at IC1d will consist of the original plus the first and second delays. The output at IC4d will contain only the doubly processed signal without the original. Remember though that with this variety of feedback and enhancement available, signal levels are going to be modified and that IC2 and IC3 prefer a maximum input level of no more than 1·5V r.m.s. On each feedback path when the phase shift is positive the level of the feedback signal is added to the original signal, the total thus being about twice the original. If both delay paths are cascaded, then the levels of these signals are also added, resulting in a further increase in the final output level. When switching across to the dual emphasis route with a signal present on both inputs, the amplitude of these two signals is also added. Therefore before switching between single and double enhancement modes with stereo signals it is preferable to reduce VR6 first.

NEXT MONTH: Clock circuit, p.s.u., construction and setting up.

Mono/Stereo Chorus & Flanger

JOHN M.H.BECKER PART TWO

THE clock signal that causes the delay chips to sample and transfer their charges from stage to stage is produced by IC9 (Fig. 6). This is a standard linear voltage controlled oscillator chip that produces a squarewave output the frequency of which is related to the value of C24, the current through VR9, and the voltage present on pin 9. The single output from IC9 needs to be split into two opposing phases as required by the delay chips. If a normal phase split were to be given then the opposing edges of the antiphase square waves would coincide. This overlap is prone to causing system noise from the delay chip outputs even though the TDA1097 is basically a low noise device, capable of a 77dB signal-to-noise ratio at a 100kHz clock frequency, though this degrades slightly with lower clock rates. The overlap on the edges of the clock is eliminated by the flip flop stage IC10 in conjunction with the NAND gates IC11a–b. C25 and R83 slow down the mutual triggering of the flip flop and gates, resulting in a twin phase output having a short delay between the respective squarewave edges. Oscillograms Fig. 7a to 7c show the 'with' and 'without' effect of the overlap elimination.

Varying the voltage applied to pin 9 of IC9 varies the clock frequency. For the automatic modulation of the clock a constantly varying voltage is produced by the low speed triangle wave oscillator around IC8a–b, and having a frequency governed by the resistance of VR7 and the value of C22. (Oscillogram Fig. 8.) Decreasing either increases the output frequency. The modulation can be switched in and out by S4, and the level varied from nil to full by the depth control VR8. C23 slightly rounds off the triangle peaks at faster modulation speeds. The modulating frequency range is controllable between about 50 milliseconds and 30 seconds, the clock frequency range is between about 12kHz and 100kHz. For a single delay chip the delay time range is thus about 64ms to 7·68ms, cascading two delays doubles the delay times. With the modulating oscillator switched out of circuit the unit can of course be used as a standard reverb or short-echo unit, though these effects will not be so pronounced as those obtainable with the *September 1984 PE Echo-Reverb* unit.

POWER SUPPLY

The unit has been designed to operate from two 9 volt batteries producing +9V/0V/–9V, and drawing between 13mA and 20mA, depending on the clock oscillator rate. IC2 and IC3 though do not like a total voltage drop across them in excess of 16V, which also means that controlling voltages must not exceed this either. The positive voltage delivered to IC2, 3, 9, 10 and 11 is thus reduced to a more suitable level by the drop across the resistor R62 in the delay line bias divider network. The voltage at R62 is within limits with all i.c.s in circuit, but may rise if any of the said 5 are not in their sockets when power is applied. IC9–11 will not mind, but IC2 and 3 may object. The unit may be operated from a stabilised power supply if preferred. The acceptable range is from +5V/0V/–5V to +9V/0V/–9V. If it is necessary to run from a power supply greater than +9V/0V/–9V then two voltage regulator devices should be inserted between the power supply and the unit as shown in Fig. 9. The voltage drop across the regulators must be greater than 2V, and R62 may be replaced by a link wire.

CONSTRUCTION

The component layouts for both boards are shown in Figs. 10 and 11. The short link wires on the p.c.b.s can be made from resistor cut-off leads shaped to the correct spacing with thin nosed pliers. Sockets should be used with all i.c.s. The wiring diagram for the unit is shown in Fig. 13. Bring the

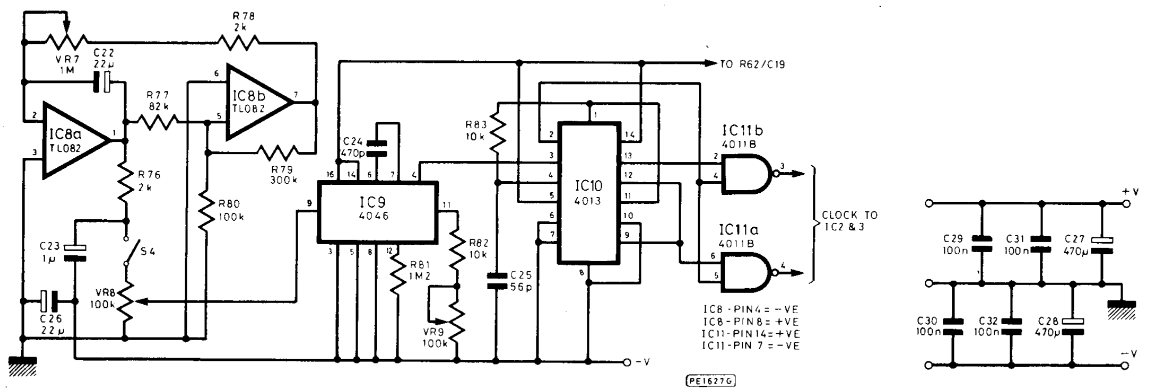


Fig. 6. Circuit diagram of the Clock Circuit

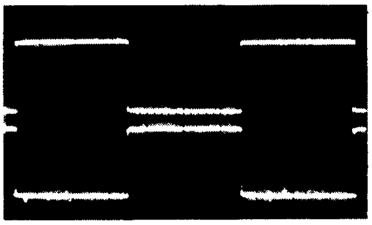


Fig. 7a. Usual appearance of two square-waves without overlap removal

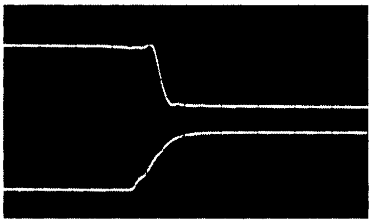


Fig. 7b. Close up of overlap

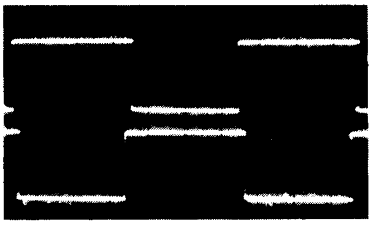


Fig. 7c. Accentuated overlap removal as used in unit

Fig. 7. Clock edge overlap of two anti-phase square-waves

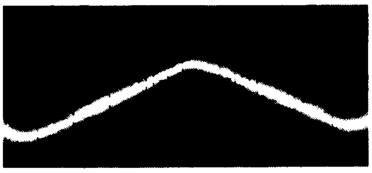


Fig. 8. Modulation oscillator waveform

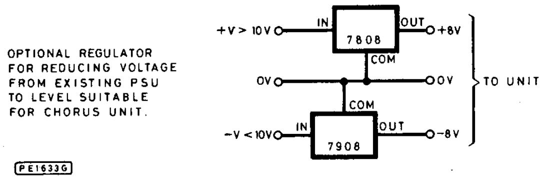


Fig. 9. Optional regulator circuit

connecting wires neatly around the edges of the p.c.b.s to the controls. The clock leads to IC2 and IC3 should be brought forward past C19, turn left at the front panel, then along to the small p.c.b., turn right and connect. Do not take them on what appears to be the shorter route across the main p.c.b. as this would direct them across some parts that might pick up any stray radiation signal. Unless you have the eyes of an eagle, thoroughly check all the p.c.b. joins with a magnifying glass. Only after all checking has been done should the i.c.s be inserted into their sockets, remembering that IC2, 3, 7, 9, 10 and 11 are MOS devices and require the normal handling precautions. The main point being to keep yourself and equipment free of static electricity by touching a grounded source before handling them.

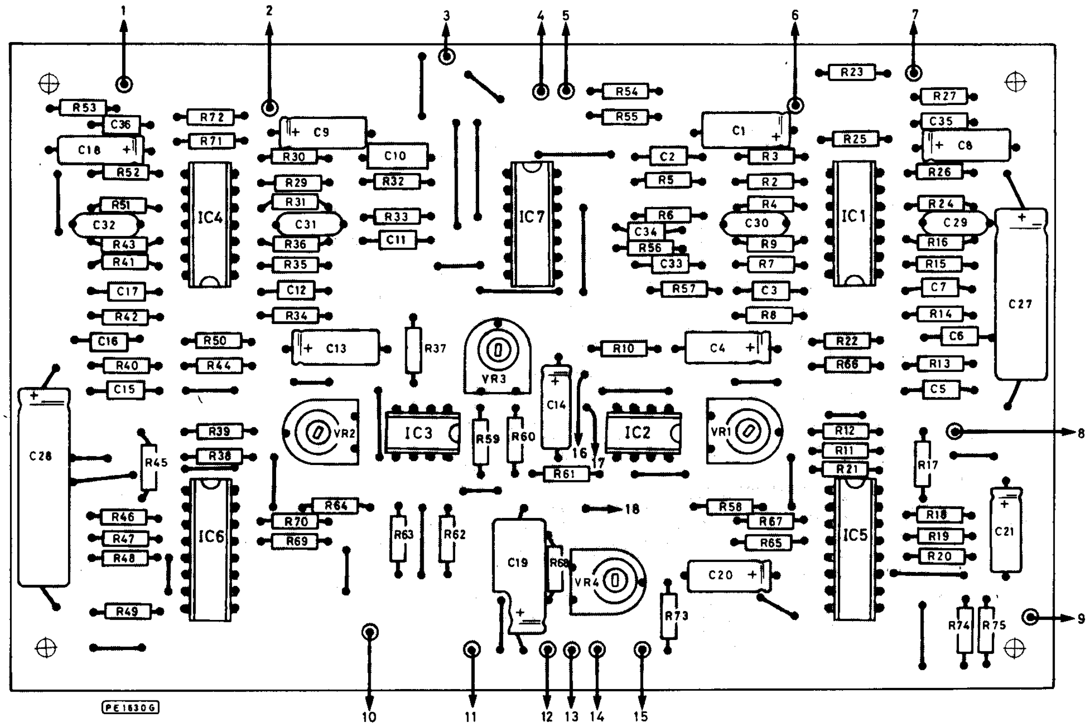


Fig. 10. Component layout of the Main Board

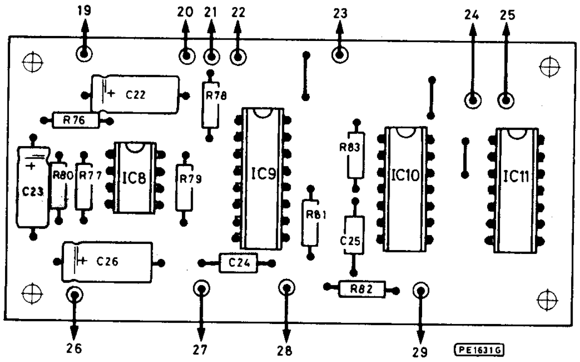


Fig. 11. Component layout for the Clock Board

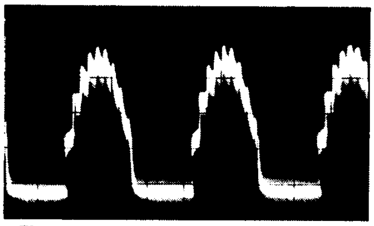


Fig. 12a. Sine-wave with VR3 unbalanced but VR1 correct

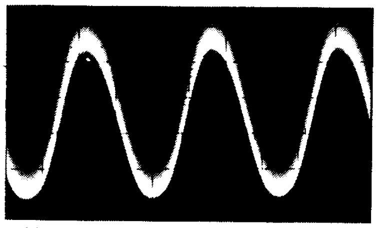


Fig. 12b. Sine-wave with both VR1 and VR3 correct

Fig. 12. Traces seen at the wiper of VR1

SETTING UP

This is quite straightforward and specialised equipment is not needed. First, VR1 to VR3 midway, VR4 max resistance (anticlockwise), VR5 and VR6 min, VR7 to VR9 max, S1 to S4 off. Plug in a music signal from a prerecorded source into the X1 socket. Check that the output level reaching the main amplifier used is the same as the original. Switch on S3 enabling the VCA, and bringing up VR6 a change in amplitude and tonal quality should increase. Rotating the clock oscillator speed control VR9 to its maximum resistance will slow down the delay and emphasise the double tracking effect. This will be more apparent with staccato sounds rather than mellow drawn out notes. Adjust VR3 around its midway point until minimum waveform distortion is heard, which will also coincide with the best delay effect. If an oscilloscope is available, the waveform balance will be obvious when monitoring the output at VR1 and VR2 in the presence of a strong input signal. (Oscillograms Figs. 12a and 12b.)

Switch on S4 bringing in the sweep modulator. Varying VR8 will vary the modulation depth, and VR7 will vary the modulation rate. Switch off S4, reset VR9 to slowest clock speed, VR6 to maximum level, switch on S2 for feedback enabling. Slowly bring up VR5 and a hollowness to the signal should come in. Maximise VR5 and carefully reduce the resistance of VR4 until the circuit almost goes into full feedback howl. If howl occurs, sharply

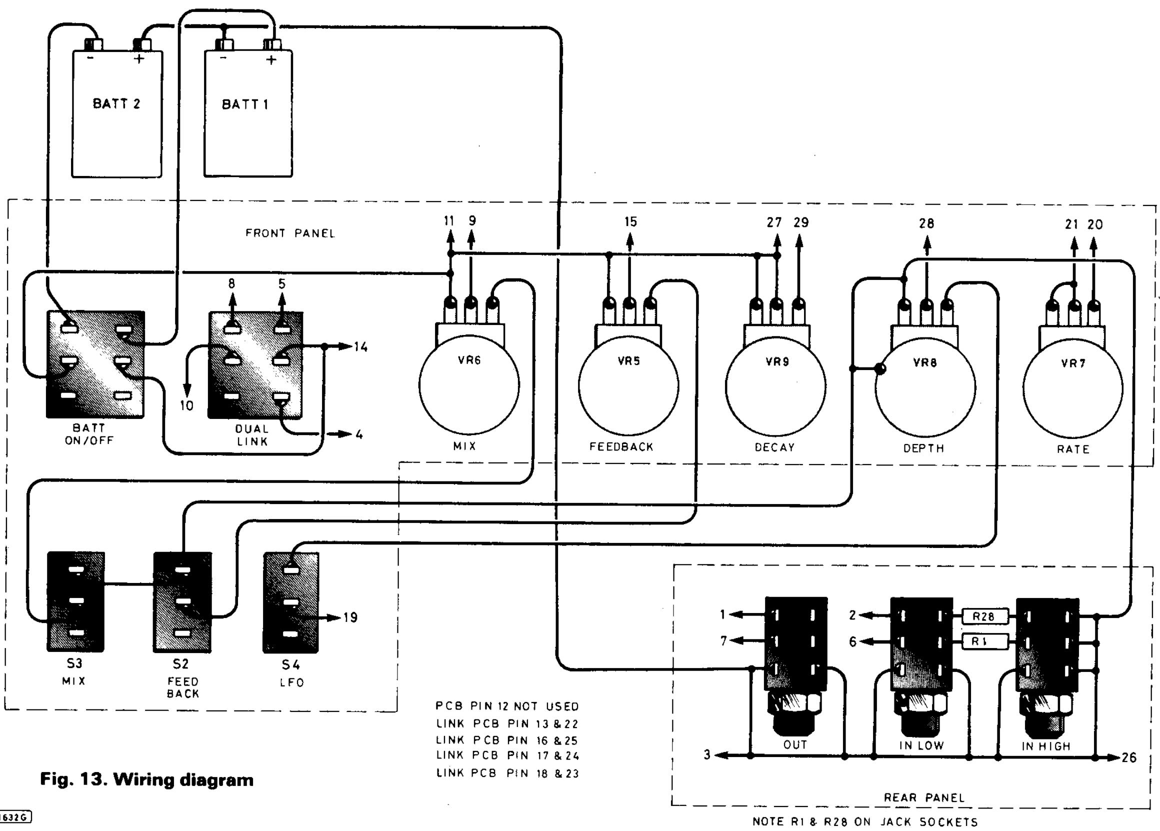


Fig. 13. Wiring diagram

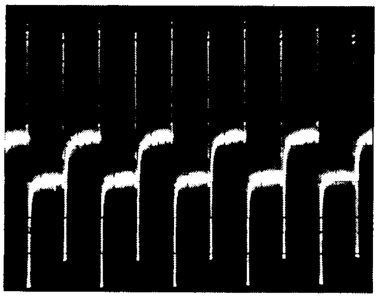


Fig. 14a. Clock residual with VR1 unbalanced (no signal)

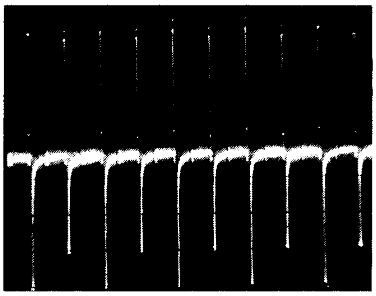


Fig. 14b. Clock residual with VR1 balanced (no signal)

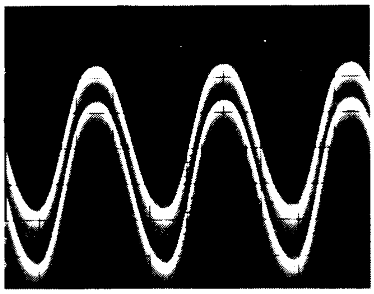


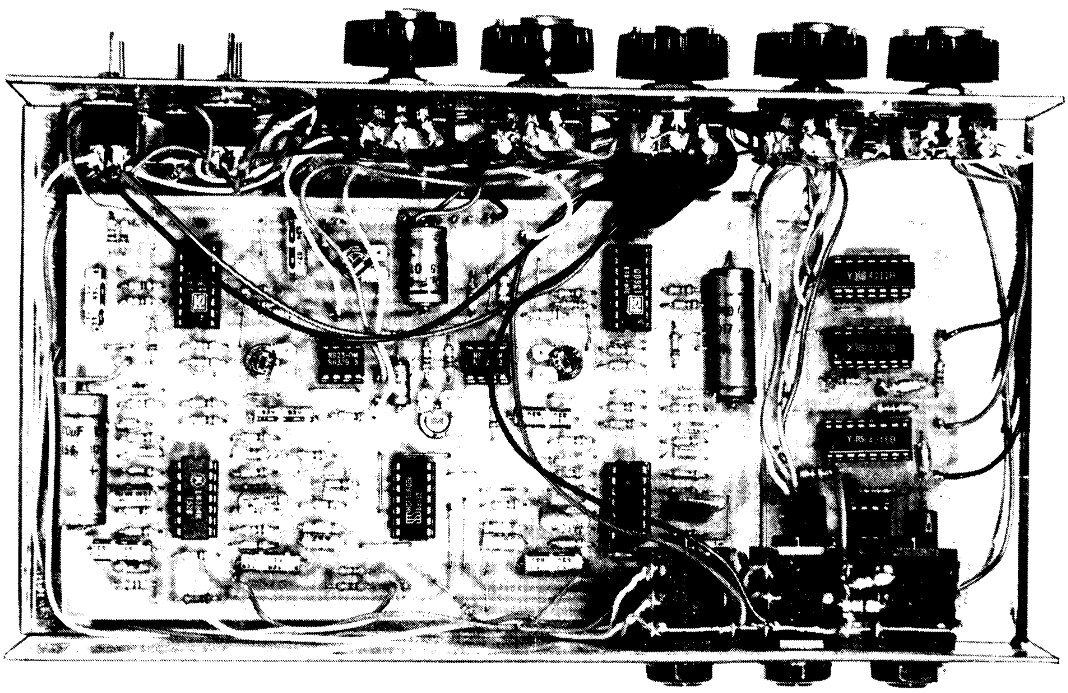
Fig. 14c. Sine-wave signal with VR1 unbalanced but VR3 correct

Fig. 14. Traces seen at the wiper of VR1

reduce VR4 and start again. Aim for the closest possible to the howl point. Howl is more likely to occur with strong bass notes. Switch on S1 to couple the two delay circuits in series and so produce the double emphasis. If necessary back off VR4 slightly as the increase in level may kick the circuit into howl again. If an oscilloscope is not available VR1 and VR2 should be left midway and ignored, otherwise adjust them for the best balance point of the residual clock frequency in the absence of an input signal. (Oscillograms Figs. 14a to 14c.) Switch on S4 and experiment with the various settings until familiar with the control options available, if necessary readjusting VR3 or VR4.

USE

There are no restrictions to the type of signal fed in provided that the amplitude is less than the distortion level, and that the type of music lends itself to enhancement within the factors discussed earlier and summarised below. It will soon become obvious which type of music requires which particular control setting for the best effect. This is a matter of personal preference, but the author feels that as with any effects unit, moderation is the keyword. Certainly overemphasis of an effect is dramatic, but it is easier to become tired of an over dramatic effect than one which produces a discrete change. In general terms music having a high harmonic content, but otherwise of a simple nature, will benefit most. Mellow or full orchestral sounds will not show the same degree of change. In the first case there is insufficient harmonic information available in the signal for the effect to fully develop. In the second case, the sound is already so full that the effect will probably be lost amongst the tonal complexity unless the original sound is full of spiky waveforms. The harsher sounds of voice, drums, harmonically rich synthesiser and organ music produce excellent effects as the waveforms involved are complex. Pure sine tones and muted waveforms, especially in the lower octaves, will be less apparent. For the chorusing effect a slower clock oscillator speed is preferable as the delay time is greater, for flanging, faster clock speeds are better as the phase shift occurs then at a more marked rate and spacing. ★



Photograph illustrating the internal details of the Chorus and Flanger Unit