BP200

TANDY

Handbook of

Practical Electronic Musical Novelties

by

B. B. Babani

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HANDBOOK OF PRACTICAL ELECTRONIC MUSICAL NOVELTIES

BY

B.B.BABANI

TANDY CORPORATION

Although every care is taken with the preparation of this book the publishers will not be responsible for any errors that might occur

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A KEYLESS ORGAN

YOU CAN BUILD THIS "KEYLESS ORGAN" AS A TOY, AS A GIMMICK OR AS A NOVELTY INSTRUMENT FOR USE BY A GROUP. PLAYED BY TOUCHING A STYLUS PROBE ON TO A SERIES OF GOLD CONTACTS, IT HAS A RANGE OF MORE THAN $1\frac{1}{2}$ OCTAVES, INCLUDING SHARPS AND FLATS, BUILT-IN VIBRATO, A PLEASANT WOODWIND SOUND AND A SURPRISING LEVEL OF ACOUSTIC OUTPUT.

The prototype instrument is housed in a metal case measuring $8\frac{3}{4}$" x $4\frac{1}{2}$" x 2" but the external details can be varied to suit the need - and according to the ingenuity - of the individual constructor. It can be rested on a table-top or held in the hand and is completely self-contained, though its output could readily be fed into a high powered amplifier system.

Interestingly enough, we considered presenting an instrument along these lines more than 12 months ago but the idea of using a stylus probe to play the required notes seemed to be altogether too primitive. However, we reckoned without Rolf Harris and his fascination for a commercial counterpart, as evidenced in a couple of his B.B.C. TV shows.

Apart from its novelty, the stylus approach has the practical advantage of simplicity from the constructor's point of view. There is no need to contrive a keyboard, with its properly proportioned keys, pivotting, spring return, contacts, etc. the player merely touches the tip of a stylus on to a metallic pattern which is part of a printed board - the pattern being shaped to resemble the layout of a keyboard, and hard gold plated to ensure reliability of contact.

People with a musical background can usually manager to play simple tunes straight off and, with a little practice, can manage staccato and glide effects by suitable manipulation of the stylus probe.

Like the earlier instuments, this one is "monophonic" a term which, in this context, means that it is capable of producing only one note at a time. If the probe bridges two contacts (or if two keys in the other instruments are depressed simultaneously) only the highest note will sound.

CIRCUIT DESCRIPTION:

The monophonic characteristic of an instrument such as this stems from the use of a single tone generator for the whole range of notes. The notes are provided by manipulating the time constant of the tone generator.

In this latest instrument, the single tone generator is a relaxation oscillator featuring a new three-terminal PNPN device from the General Electric Company which is called a "programmable unijunction transistor" (PUT) with the type number D13T1. The three terminals of the PUT are designated as Anode, Anode Gate, and Cathode. For the purpose of this article, it will suffice to know that the PUT is a device which can be arranged to function as a relaxation oscillator, as one of its many possible applications.

The time constant of the oscillator is determined mainly by the 0.0033uF capacitor and the series string of 21 resistors. Each note is selected by "tapping" the resistor string with a probe which is connected to the positive supply. The lowest note is selected by connecting the probe to the end of the resistor string so that the total resistance is in circuit, higher notes use less than the total resistance in the string. A sawtooth waveform is generated at the anode of the PUT. A train of pulses at the same repetition rate with positive polarity appears at the cathode, while a similar train of pulses with negative polarity appears at the Anode Gate.

In a device such as this, it is desirable that the oscillator tolerate a falling battery voltage, as in a portable radio, to ensure that the battery voltage, has a long effective service life. In practice, the PUT oscillator will function at less than half the nine volt supply with a frequency change of only a few per cent.

The range of the oscillator is from A440Hz to F-1397Hz which is one octave above the range required - from A-220Hz to F-698Hz. There are two good reasons for operating the oscillator in this fashion.

In the first place, a rough sawtooth waveform is not pleasant to listen to and requires extensive filtering before it is fed to an amplifier system. For this reason we have followed the oscillator with two NPN silicon transistors in a gated R-S flip-flop configuration to provide frequency halving and a square-wave output. The flip-flop is triggered from the Gate of the PUT. It may be thought that the flip-flop would "load" the oscillator but, in practice we found loading effects to be very slight. We tried an integrated circuit flip-flop and, while it worked well, the current drain of around 25 milliamps was obviously too high for an application such as this. This is a frequent objection where IC's are considered, the high current drain off setting their initial economy.

A further advantage of operating the oscillator over a high frequency range is that a smaller timing capacitor can be used. Since the PUT has low internal leakage, high values of resistance and a small timing capacitor can be used. This is not possible with conventional Unijunctions if reliable operation and high energy trigger pulses are required. The use of a small capacitor makes it possible to incorporate a facility not previously available on simple instruments of this type - that of variable tuning. To tune an instrument of this type so that it can be played in conjunction with other instruments requires an adjustment to the timing capacitor. Merely having a potentiometer in the resistor string will not achieve this, since all notes must be changed by the same ratio.

In this PUT oscillator an additional 200pF or so will shift all the notes down in frequency by a semitone. We used a "solid dielectric" tuning capacitor, as used in small portable transistor radios, to achieve a tuning range of a little over a semitone. This will enable the instrument to be played in conjunction with pianos, which although usually right on standard pitch can be low by as much as a semitone.

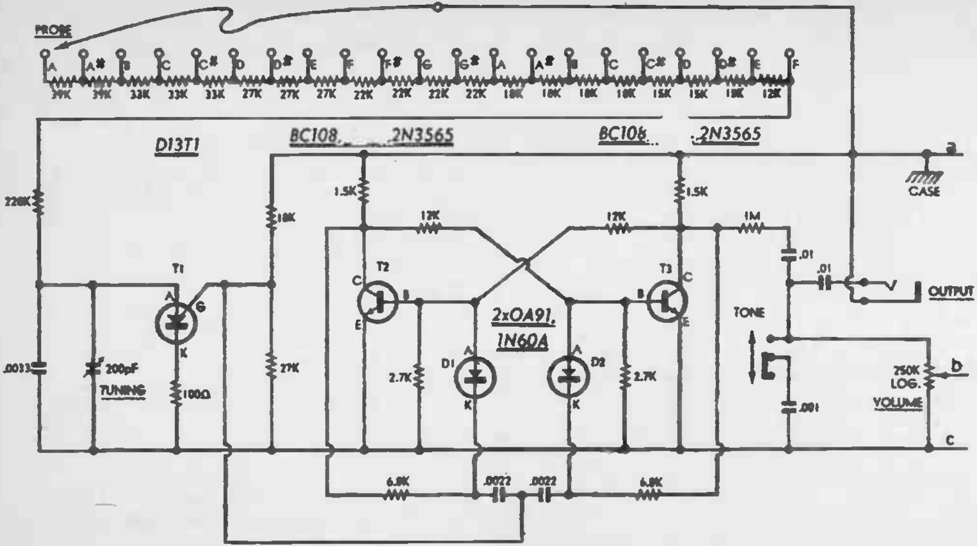
It may be thought that a better way to obtain a square-wave output would be to use a conventional multivibrator directly, with one of the cross-coupling resistors in the form of a tapped resistor string to vary the frequency. The disadvantage of this approach is that the "mark-space" ratio of the rectangular waveform tends to vary with frequency. At low frequencies, one might obtain a very close approximation to a square wave, with unity mark-space ratio, but as the frequency rises the waveform becomes more "spiky", the tonal quality changing markedly as the player moves up the scale. The system we have used with a relaxation oscillator followed by the flip-flop, provides a waveform with a constant, unity mark-space ratio regardless of the frequency.

The square wave output from the flip-flop gives a pleasant "woody" tone which, can be "softened" if desired by a simple top-cut filter. In the prototype instrument, we used a slide switch to provide a choice of two different tones. One position of the slide switch leaves the square wave unmodified while the other position connects a .001uF capacitor across the signal line to give a slight "rounding off" of the square wave and a "mellow" tone.

The output from the flip-flop is fed, via the tone switch and volume control, to a four-transistor, complementary-symmetry amplifier. The class-B output stage has low current drain at no signal and a maximum power output of over 400 milliwatts into a 15 ohm speaker with a nine volt battery.

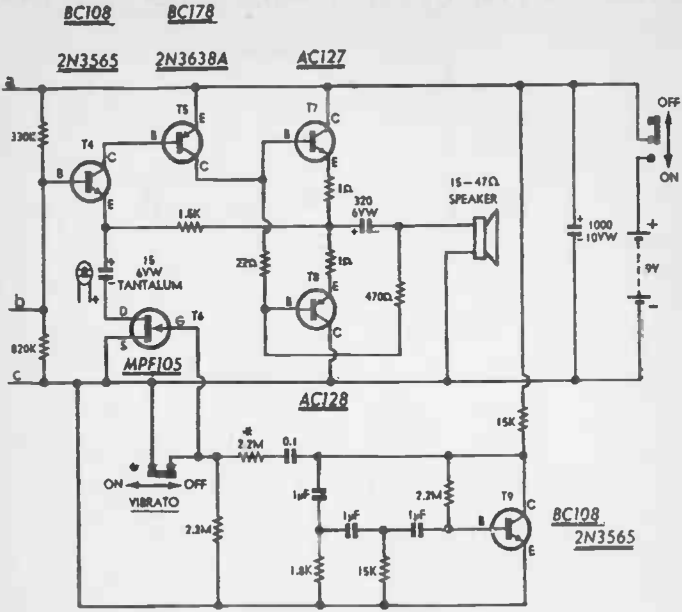
The sensitivity of the amplifier is approximately 300mV for full output, depending on components in the feedback loop, with a minimum input impedance of 200K. We used a particular 15 ohm loudspeaker but any loudspeaker with an impedance from 15 to about 47 ohms could be pressed into service. With a voice coil impedance greater than 15 ohms, power output will be reduced but so also will be the peak current drain on the battery.

The 1000uF electrolytic capacitor connected across the battery serves to limit the rise in supply impedance as the battery nears the end of its service life, minimising distortion and sustaining power output.



KEYLESS ORGAN

Connects with drawing on Page 9



KEYLESS ORGAN

Connects with drawing on Page 8

The "square wave" signal level from the flip-flop has been adjusted so that it will just drive the amplifier into clipping at the maximum setting of the volume control. This is to give a smoother volume control action and to avoid the tonal change and the undue demands on the battery which would follow if the output stage were grossly overdriven. The current drain at full "undistorted" output (i.e. just at onset of clipping) is of the order of 90 milliamps with a 15 ohm speaker. The output transistors are fitted with flag heatsinks to improve heat dissipation. With normal use they will run barely warm to the touch.

The output transistors are slightly forward biased by the 22 and 470 ohm resistors to provide a quiescent current of about 1 milliamp, which is enough to ensure freedom from corss-over distortion. The low value of quiescent current means that the usual temperature stabilisation thermistor can be dispensed with. The amplifier has very lów distortion due to the large amount of negative feedback used.

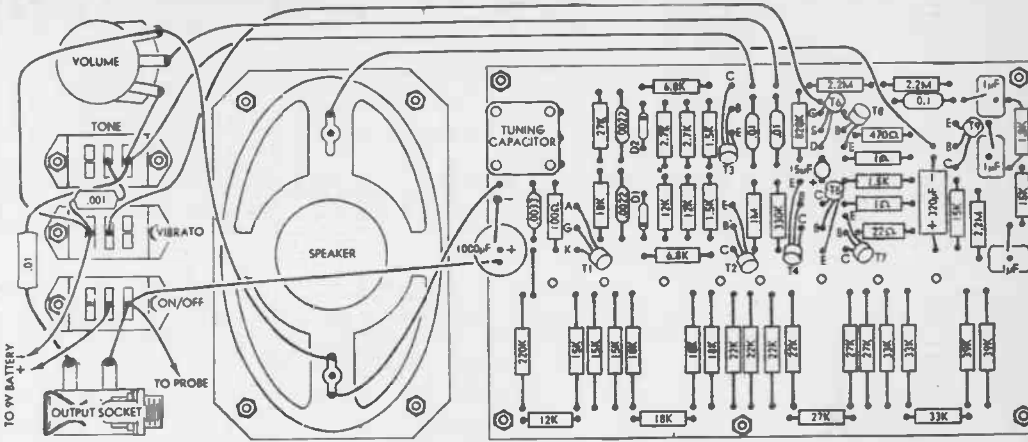
The signal from the flip-flop is also fed, via the tone switch and .01uF capacitor to an output socket for connection to an external amplifier. The level of signal available is of the order of 300mV (RMS).

The other major feature of the circuit is the vibrato, or more correctly, the tremolo or tremulant facility. Vibrato refers to a rhythmic variation in the frequency of a musical note at a rate of 5Hz to 10Hz. Tremolo refers to a variation in amplitude of a musical note at around the same rate. In a simple instrument it is hard to discern the difference between the two effects and the terms are often confused and interchanged. In our case "vibrato" (to mis-use the more common word) is achieved by modulating the negative feedback, and hence the gain, of the amplifier.

The circuit makes use of the fact that the drain-source resistance of a field-effect transistor can be modulated by a bias voltage applied to its gate electrode. We have specified either of two FETs, in order of preference. The first is a Motorola n-channel device, MPF105, while the second is the Fairchild economy p-channel device, 2N4360. The drain source resistance of these FETs is of the order of a few hundred ohms, which makes it admirably suited for use in the feedback loop of the amplifier. It is connected in series with the feedback loop via a 15uF/6VW tantulum electrolytic capacitor. The modulating voltage applied to the gate of the FET is a low-frequency sine wave obtained from a one transistor phase shift ascillator.

While either FET type is suitable, the larger parameter spreads of the 2N4360 could make satisfactory operation somewhat harder to attain.

The sine wave signal at the collector of the transistor, which has an amplitude of about 2V peak-to-peak, is applied to the gate of the FET via a blocking capacitor and a voltage divider network. The maximum modulating voltage applied to the FET gate should be less than the "pinch-off" voltage, otherwise a series of "plops" will emanate from



The above wiring diagram shows all the necessary details so that construction will be straightforward. Note that the case is connected to the positive supply rail.

the loudspeaker at the same rate as the vibrato frequency. As the "pinch-off" voltage varies for each device the voltage divider may have to be varied to obtain sufficient depth of modulation without the above effect. The total resistance of the voltage divider should not be reduced below about 4 megohms, while the value of the first divider resistor should not be decreased below about 470K.

The vibrato facility is disabled by connecting the gate of the FET to the negative supply rail, so that the FET drain-source resistance is unmodulated. Some readers may think it desirable to disconnect the supply to the phase-shift oscillator instead, to reduce battery drain, but the oscillator requires a few seconds to start.

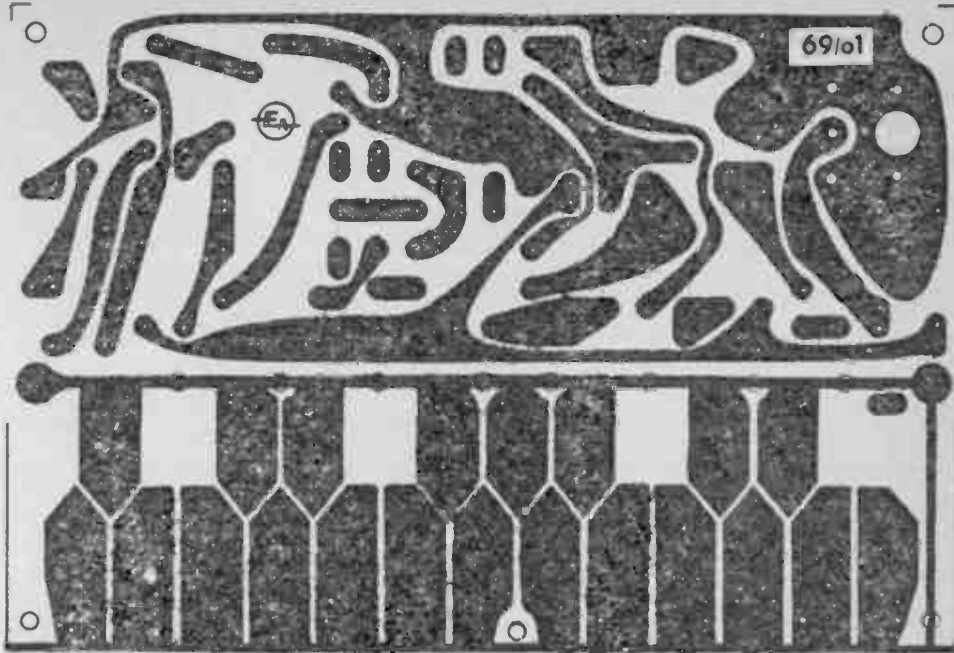
The 1uF capacitors used in the phase shift network are metallised polyester types with a tolerance of plus or minus 10 per cent. Ceramic or tantalum electrolytic capacitors are not suitable, even when matched for capacitance, since their power factor and leakage are too high for this application.

The speed of the vibrato may be varied by changing the value of the 1.8K resistor. Higher values will give lower frequencies and vice versa. The oscillator will not function reliably with values below 1K.

CONSTRUCTION: All the circuitry described above, apart from the slide switches, potentiometer and two small capacitors, is mounted on a circuit board measuring 6"x 4½" .

The copper wiring pattern of the board warrants some explanation. The actual "key" contacts are part of the copper pattern, the area of contact exposed by the case - the playing area if you like - measuring 5 3/8" x 7/8". Since copper tarnishes quickly when exposed to air, contact with the metal probe would be unreliable. For this reason we had a section of the contacts plated with gold. The area to be plated measures approximately 1" x 5½". For the gold plating operation all the contacts must be connected together but the connections must be broken before the board is ready to be used. The "natural" keys are connected together by a strip along the edge of the board; this can be removed by chamfering with a file or abrasive wheel.
The "accidental" (sharp or flat) keys are connected by a strip down the centre of the board and these connections are broken by drilling with a 1/8" drill in the apprpriate circle markings. It is not necessary to drill right through the board, as we did with the prototype. To make sure the connection has actually been broken it is a good idea to slightly chamfer the holes. Readers may notice that the above-mentioned holes in the prototype are not in a straight line but they can be so in the production boards that you will ultimately buy.

Some readers may wonder whether gold plating is justified, in view of the cost. While other metals, for example chrome, may not tarnish,



Above is the copper pattern on the printed board. Actual size of the board is 6 x 4-1/8 inches.

they would not provide as reliable an electrical contact. Chrome relies for its non-decaying properties on a tough, transparent oxide coating which protects it from the atmosphere.

The probe used to play the prototype instrument was a standard meter probe. The tip should be smooth and with a radius of at least 1/16". A minimum pressure should also be used to avoid undue marking of the gold surface. The lead for the probe is brought out through a hole in the case which should be fitted with a grommet to avoid chafing the lead insulation.

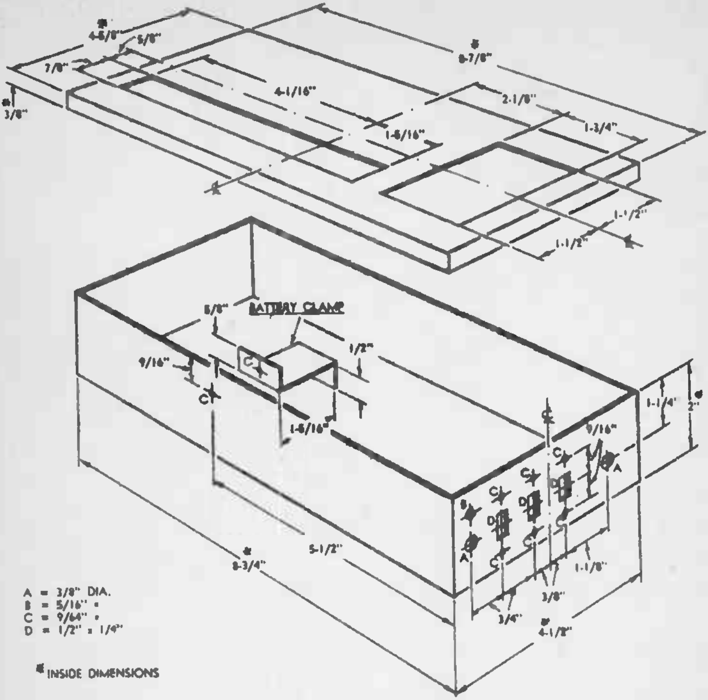
The tuning capacitor used in the prototype was a solid dielectric tuning capacitor meant for pocket radio applications. The oscillator and aerial sections give a maxiumum capacitance of 200pF when connected in parallel. Other solid dielectric tuning capacitors could passibly be used if they had around the same total capacitance, but the board may have to be modified to take them.

The tuning capacitor is fastened to the board by two small screws. A hole is drilled in the lid, in the appropriate place, for screwdriver access to the shaft of the tuning capacitor. The shaft must be fitted with a screw which is obtained with the capacitor.

All the resistors in the circuit can be of ¼ or ½ watt rating apart from the two emitter resistors for the output transistors, which are a new Philips cracked carbon type of ½ watt rating. We have used a mixture of ½ and ¼ watt resistors in the prototype. The resistors in the series string for the PUT oscillator should be high-stability types with a tolerance of not more than plus or minus 5 per cent.

The holes on the board for the FET are placed in a straight line. Because of the different lead configurations on the two FET's the source and drain connections of the 2N4360 will be reversed to that shown on the wiring board, which shows the connections for the MPF-105 FET. This is immaterial since no DC polarising voltages are present in this section of the circuit. This means that the 2N4360 can be inserted directly into the board without having to twist the source and drain leads so that they conform to the wiring and circuit diagram. The FET actually installed in the prototype in the photos is a 2N4360.

The 1uF capacitors in the phase shift oscillator are mounted vertically to save space. Use polyester capacitors with the lowest available voltage rating, in the interest of the smallest size.
The board is attached to the case lid by five 1/8" Whitworth screws and nuts, spaced off the lid by about 1/8". The cut-out for the keys measures 5 3/8" x 7/8". The board should be positioned on the lid so that it does not foul the case. Details on the metalwork diagram will show the dimensions necessary for readers to make their own case. The case can be painted a bright colour to provide a contract with the gold-plated key contacts and the expanded metal speaker grille. The colour of the prototype was a mid blue.



The above diagram shows the major details of the case.

OPTIONAL FEATURES: We outline some of the features which could be incorporated or deleted. The tuning capacitor can be deleted. The unit will still be in tune itself but could not be funted to suit other instruments. The .0033uF timing capacitor could be changed to a higher or lower value to give an octave range to suit the constructor. Operation with capacitors below 1000pF may be unreliable, however.

The vibrato section can be deleted and the source-drain resistance of the FET replaced by a resistor with a minimum nominal value of 180 ohms. This will give the amplifier an effective input sensitivity of 300mV, as mentioned earlier.

The amplifier itself, with the associated volume control, can be deleted and the signal fed to an external amplifier via the output socket. The tone switch could also be deleted.

We urge readers not to deviate from the published circuit, apart from those possible modifications listed above, unless they are experienced constructors. We hope you derive as much entertainment from this unit as we did in developing it.

SPECIFICATIONS

Pitch Range	A-220Hz to F-698Hz
Supply voltage	9 volts
Idle current	20mA
Maximum current	90mA
Power output	400mW (RMS)
Output signal to ext. amp	300mV (RMS)

PARTS LIST

1 case and lid. Inside dimensions $8\frac{3}{4}$" x $4\frac{1}{2}$" x 2"
1 4" x 2" elliptical speaker, 15 ohm
1 piece of expanded metal mesh 4" x $2\frac{3}{4}$"
1 9 volt battery
1 battery clamp
3 slide switches
1 250K (log) potentiometer
1 knob to suit
1 meter probe, with rounded point
1 jack socket
1 solid dielectric tuning capacitor

SEMICONDUCTORS

4 BC108, 2N3565 or similar silicon NPN type
1 D13T1 programmable unijunction transistor
1 MPF105 or 2N4360 FET
1 BC178, 2N3638A or similar silicon PNP type
1 AC127/128 germanium complementary matched pair (with flag heat sinks).
2 OA91, 1N60A low power diodes

RESISTORS

($\frac{1}{2}$ or $\frac{1}{4}$ watt, 5 per cent tolerance)

4 x 2.2M, 1 x 1M, 1 x 820K, 1 x 330K, 1 x 220K, 2 x 39K,
3 x 33K, 4 x 27K, 4 x 22K, 5 x 18K, 5 x 15K, 3 x 12K, 2 x 6.8K,
2 x 2.7K, 1 x 1.8K, 1 x 1.5K, 1 x 470 ohm, 1 x 100 ohm, 1 x 22 ohm,
2 x 1 ohm ($\frac{1}{2}$ watt)

CAPACITORS

1 x 1000uF/10VW electrolytic
1 x 320uF/6VW electrolytic
1 x 15uF/6VW tantulum electrolytic
3 x 1uF metallised polyester
1 x 0.1uF polyester (not ceramic)
3 x .01uF polyester or ceramic
1 x .0033uF polyester
2 x .0022uF polyester
1 x 001uF polyester

THE AUTODRUM

IN THIS ARTICLE WE INTRODUCE ANOTHER ELECTRONIC PERCUSSION INSTRUMENT WHICH WE HAVE NAMED THE AUTODRUM. AS IN THE CASE OF THE ELECTRONIC BONGOS, IT IS BASED ON THE TWIN-T OSCILLATOR. IN ADDITION TO BEING KEYED MANUALLY, IT HAS THE FACILITY OF AUTOMATIC TRIGGERING AT ANY DESIRED RATE.

Essential to almost any small musical "combo" is a rhythm section, consisting, usually of snare and bass drums and a set of high-hat cymbals. The instrument we describe in this article could conceivably replace the bulky bass drum. It is accommodated in a small diecast metal box so that it can be carried in a briefcase instead of a station wagon. As with the bass drum, it can be pedal operated but it has the additional facility that it can be automatically keyed at any desired rate.

Alternatively, the Autodrum could be used to accompany a solo piano or electronic organ. With a powerful amplifier and loudspeaker to match, it puts out a beat that anyone could follow. Lastly, if you listen to it for half an hour, as the author did when developing it, it will give you a headache!

The bass drum is synthesised by the same method as described in the article on the Electronic Bongos, referred to above. We set up an electrical analogue; a circuit which 'resonates', or produces a wave-train when an electrical impulse is appled to it, the electrical

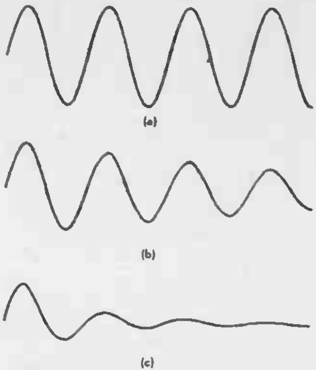
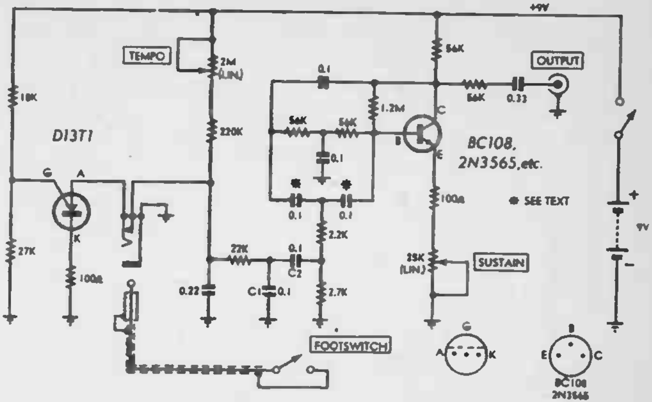


Figure 1

The above waveforms show the three modes of oscillation possible with a Twin-T oscillator. (a) is continuous; (b) is lightly damped and (c) is heavily damped.



AUTODRUM

The circuit diagram of the Autodrum. It uses a twin-T oscillator to produce the "drum" sound and a PUT pulse generator to provide the automatic repeat feature.

impulse being analogous to the physical blow applied to a drum. The circuit also requires a means of determining the damping of the resonance, so that the 'quality' of the resonance can be altered to simulate that from the acoustic instrument.

A circuit which lends itself to synthesising percussion instruments is the Twin-T oscillator, so named because of the configuration of the twin RC phase-shift networks. In this particular instrument, the oscillator is set into the "quiescent" mode, i.e. just on the point of oscillation, with the aid of a 25K potentiometer in the emitter load of the transistor. An electrical impulse applied to the junction of either of the T-networks or to the bass of the transistor will shock the circuit into brief oscillation. The degree to which the oscillation is sustained will depend on the setting of the potentiometer.

If the potentiometer is set for maximum resistance, the oscillation will be very short - only a few cycles. If the potentiometer is set so that the oscillator is just into the quiescent region, the oscillation will be quite long, probably 20 or more cycles. Beyond this again, the stage will go into full oscillation, which would be totally undesirable for the present purpose.

The waveforms shown in figure 1 illustrate the three possible modes of oscillation possible with the Twin-T oscillator. Figure 1(a) shows a continuous oscillation at constant amplitude. Figure 1(b) shows a lightly damped oscillation produced with the oscillator set just into the "quiescent" zone. Figure 1(c) shows a heavily damped oscillation produced with the potentiometer set for a very short "sustain".

The circuit may be triggered into oscillation by the same method as used in the Electronic Bongos, i.e. with the aid of touch plates connected to the junction of one of the T-networks. However, since a bass drum is usually pedal-operated, we have suggested a different method. The circuit is triggered into oscillation by a DC pulse into one of the T-networks. This can be initiated either with a pedal-operated switch or with the auxiliary triggering oscillator which we have incorporated.

The triggering circuit is a relaxation oscillator featuring a three-terminal PNPN device from General Electric which is called a "programmable unijunction transistor" (PUT) with the type number D13T1.

In actual fact, the PUT is closer in mode of operation to a thyristor than to a conventional unijunction transistor. For the purpose of this article it will suffice to know that the PUT can be arranged to function as a relaxation oscillator, as one of its many possible applications. The three terminals of the PUT are designed as Anode, Anode Gate and Cathode (A, G and K).

There is no reason why the reader should not experiment with a triggering circuit using a conventional UJT, although there may be some problems in achieving the desired low repitition rates. For our part, the other good reason for using the PUT is that it is cheaper than comparable UJTs anyway!

The time constant of the PUT oscillator, and hence the repetition rate, is determined mainly by the 0.22uF capacitor and the sum of 220K resistor and 2-megohm potentiometer (connected as a variable resistor). The 0.22uF capacitor charges exponentially toward the supply voltage via the resistors until it reaches the firing voltage of the PUT. The firing voltage of the PUT is set by the ratio of the 18K and 27K resistors and this is how the title "Programmable" arises. When the PUT fires, the 0.22uF capacitor is discharged very rapidly. It then begins to recharge and the whole sequence is repeated indefinitely. The result is a sawtooth waveform.

The repetition rate of the PUT oscillator shown ranges from 50 beats per minute to over 220 beats per minute, which should be more than adequate. The range can be increased at the high end by decreasing the 220K resistor and it can be decreased at the low end by using a 5-megohm potentiometer.

When the PUT fires it also discharges C1 via the 22K resistor. This applies a negative DC pulse to one of the T-networks of the drum oscillator via C2, so that it is shocked into oscillation. Thus the PUT oscillator is a simple way of triggering the Twin-T oscillator at a regular but adjustable rate.

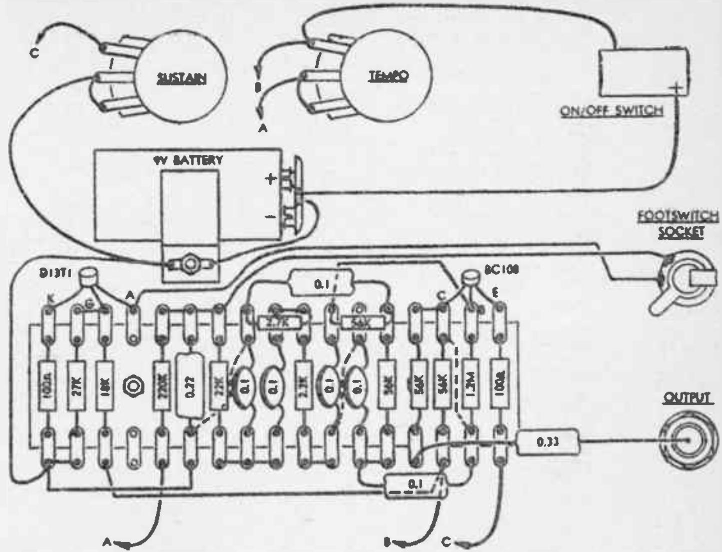
If the drummer wishes to key the Autodrum with a foot-operated switch, the PUT oscillator can be effectively disconnected from the circuit by inserting a phone jack in the socket. Using the footswitch now discharges the 0.22uF and C1 in the same way as the PUT does, so that the Autodrum can be keyed by a player instead of at the constant rate provided by the PUT oscillator.

The autodrum could be arranged to simulate the Tom-tom by changing the 0.1uF capacitors on the circuit diagram marked with an asterisk to 0.047uF. Other variations are possible - just vary the capacitors.

The Autodrum can be used with almost any amplifier, provided it does not overload the amplifiers input. The more powerful the amplifier, the better; the same can be said about the size and power handling capability of the loudspeaker. For the best results, it should be used with a high power guitar amplifier and matching loudspeaker system.

The output signal from the Autodrum is approximately 0.5 volts RMS, which is enough to drive almost any amplifier to full power using the "pick up" or "auxiliary" input. Do not attempt to feed it into a "microphone" or low-level "guitar" jack, as it will most likely overload the input stage.

DIAGRAM SHOWS COMPONENTS AND WIRING



The assembly of the Autodrum will be a straightforward process if this wiring diagram is followed closely.

A diecast metal box measuring approximately 4 5/8" x 3 5/8" x 2" is used to house the components. All of the components, with the exception of the potentiometers, are mounted on a 17-lug section of tagboard. Layout is not critical, but we suggest that beginners use the wiring diagram supplied to simplify construction and minimise errors.

When drilling the diecast box, use a sharp drill at high speed and a low rate of 'feed' (i.e. do not put too much pressure on the drill). Preferably, use a drill stand. If care is not taken when drilling, the box may be cracked.

The battery is a small 9 volt type. The current drain will depend on the settling of the Tempo control but it will always be less than 0.5mA so that the battery should have a very long life. If the constructor desires, the circuit can be operated at voltages below 9 volts, although we would suggest a minimum of 6 volts to ensure adequate output signal.

We have left the details of the footswitch to individual constructors. It must be reliable and rugged. Several ideas are suggested. The footswitch from an electric sewing machine could possibly be adapted. Alternatively, a unit could be built using a small magnet and reed switch; this would have the very obvious advantage of being virtually free from contact troubles.

The stop-start facility could also be operated by footswitch instead of using the push on/push off switch that we used in the prototype.

The Autodrum could even be triggered by using the hand to interrupt a light beam which actuates an LDR/relay combination, this would introduce an element of "mystery" into its operation though it will also introduce a substantial degree of complication.

When assembly is completed, connnect the Autodrum to an amplifier and loudspeaker. Switch on and turn the sustain control fully clockwise. The drum oscillator should be operating continuously at about 60Hz. Now rotate the sustain control anticlockwise until the oscillator just stops. Varying the Tempo control will now control the rate at which the Autodrum beats. Critical adjustment of the Sustain control will make the Autodrum sound like Bass drum or a kettle drum.

Finally, a word to avid experimenters on how the basic circuit described here could be expanded to make a simple rhythm generator. Two Twin-T oscillators would be needed, each one having an associated relaxation oscillator to trigger it. One relaxation oscillator would be run from the output of the other relaxation oscillator, so that it acts as a frequency divider. One relaxation oscillator would provide the basic beat while the other provides the accented beat, on a different note.

PARTS LIST

1 Diecast Utility box, 4 5/8 x 3 5/8 x 2 inches
1 Push on - push off switch
1 Phone jack socket
1 17-lug tagboard
1 Output socket
1 9V battery and connector to suit
1 25K (lin) potentiometer
1 2M (lin) potentiometer
1 BC108, 2N3565 or equivalent silicon NPN transistor
1 D13T1 programmable unijunction transistor

CAPACITORS

(Voltage ratings higher than 9V will suffice)
1 x 0.33uF, 1 x 0.22uF, 6 x 0.1uF, all ceramic or polyester

RESISTORS
(½ or ¼ watt rating)
1 x 1.2M, 1 x 220K, 4 x 56K, 1 x 27K, 1 x 22K, 1 x 18K,
1 x 2.7K, 1 x 2.2K, 2 x 100 ohms

SUNDRIES

2 knobs, footswitch (see text), screws, nuts, battery clamp, hookup wire, solder etc.

ELECTRONIC BONGOS

OVER THE PAST FEW YEARS THERE HAS BEEN GREAT INTEREST IN ELECTRONIC MUSICAL INSTRUMENTS, RANGING FROM MERE TOYS TO LARGE AND EXPENSIVE RECITAL UNITS. IN THIS ARTICLE WE DESCRIBE A DEVICE WHICH CAN SIMULATE BONGOS. THE CIRCUIT CAN, IN FACT, BE ADAPTED TO SIMULATE OTHER PERCUSSION INSTRUMENTS.

The electronic organ has undoubtedly held pride of place as the best known electronic instrument, although there have been electronic pianos and other instruments like the "Theremin", which involve a sound and technique all their own.

During the past few years, devices have appeared which simulate the sound of the common percussion instruments and which can automatically reproduce a wide variety of dance rhythms. The earliest example were perched alongside organs and pianos within easy reach of the player. More recent units, vastly reduced in size by solid-state circuitry, are being built right into the basic instrument.

With the advent of musical combos in which all the instruments are electronically amplified, interest in "synthetic" percussion sources has received a further boost. In this article we introduce one of the circuits used for simulating percussion instruments.

A percussion instrument may broadly be defined as one which is played by being struck, whether it be by hand or with drumsticks and such like. While this definition would seemingly include pianos and other such instruments, the definition has been limited by usage to the more elementary type of instrument.
Physical objects which are struck forcibly tend to vibrate in a resonant manner. The frequency, waveform, amplitude and duration of the vibration (or vibrations) will depend on the shape and the nature of the material.

If the object is a tuning fork it will vibrate at virtually a single frequency and the waveform will closely approximate that of a sinusoid (i.e., sine or cosine function). The vibration will be sustained for a relatively long time.

In the case of a bronze bell, the vibration will be more complex, in that it is made up of a fundamental frequency and a number of other frequencies which one would hope to be musically related to the fundamental. The bell's vibration will be large in amplitude and will also continue for some time.

On the other hand if the object being struck was a large block of concrete the vibration would be small (relative to the force of the blow) and of very short duration. Its vibration would be muffled or "damped" by the nature of the material. For all practical purposes we could say that the tuning fork and bell are resonant bodies, while the concrete block is non-resonant.

To be musically significent, percussion instruments need to be resonant bodies whether they be bells, blocks or the stretched skin of a kettle-drum.

In setting out to synthesize a percussion instrument, the starting point is an electronic analogy. A circuit which resonates when an electrical impulse is applied to it, the electrical impulse being analogous to a physical blow. A necessary adjunct is a means of determining the frequency and the damping of the resonance, so that we can alter the "quality" of the sound, as reproduced, so that it will simulate that from the acoustic instrument.

Referring now to the circuit diagram, the basic configuration we have suggested is a Twin-T or Parallel-T oscillator. The name arises from the configuration of the two RC phase-shift networks associated with each oscillator. While we have used the Twin-T configuration here to achieve a particular result, it has a wide range of possible applications in electronic musical instruments.

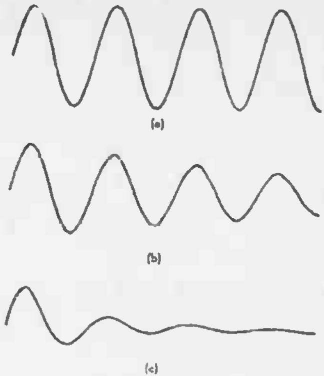
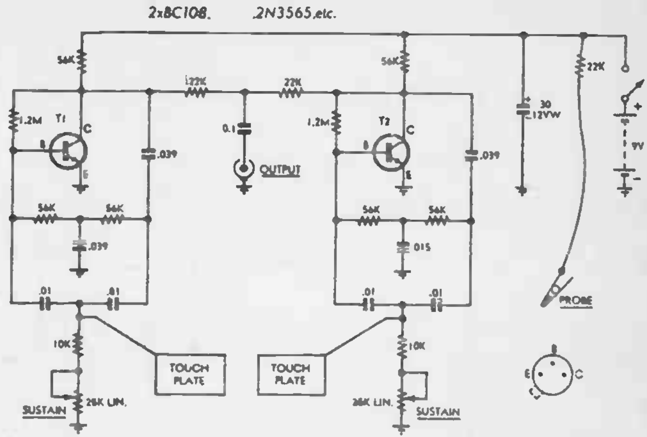


Figure 1

The above waveforms show the three modes of oscillation possible with a Twin-T oscillator. (a) is continuous; (b) is lightly damped and (c) is heavily damped.



ELECTRONIC BONGOS

The circuit diagram of the Electronic Bongos. Two Twin-T oscillators are used. The name "Twin-T" arises from the configuration of the two RC phase-shift networks associated with each oscillator.

As the resistance of the 25K potentiometer in one of the T-networks approaches its minimum resistance setting, the circuit will begin to oscillate continuously. The output wave form taken from the collector will be substantially sinusoidal and the frequency can be set anywhere in the audio range by selection of the other components in the T-networks. While it could be used as a continuous tone source, other modes of operation are possible.

For example, if the DC supply rail is connected via a decoupling network consisting of a suitable resistor and capacitor, the circuit can be arranged to have a "sustain" feature. When the supply is connected (by an organ key for example) the capacitor would rapidly charge to the supply potential and allow the oscillator to start. When the supply connection is broken, the charge stored in the capacitor would allow the oscillator to continue operating with a gradual decrease in amplitude, giving a "sustain" effect. The frequency changes only very slightly with decreasing voltage.

As we have already indicated, increasing the resistance of the 25K potentiometer will cause the amplitude to decrease to the point where the oscillator stops altogether. At this point, the oscillator is said to be quiescent. An electrical impulse applied to the junction of either of the T-networks or to the base of the transistor will shock the circuit into brief oscillation and the degree which the oscillation is sustained will depend on the setting of the potentiometer. If it is set for maximum resistance, the oscillation will be very short - only a few cycles. If the potentiometer is set so that the oscillator is just into the quiescent region (i. e. just on the point of continuous oscillation) the oscillation will probably last for twenty or more cycles. Obviously enough, the potentiometer can be used to vary the damping of the oscillator.

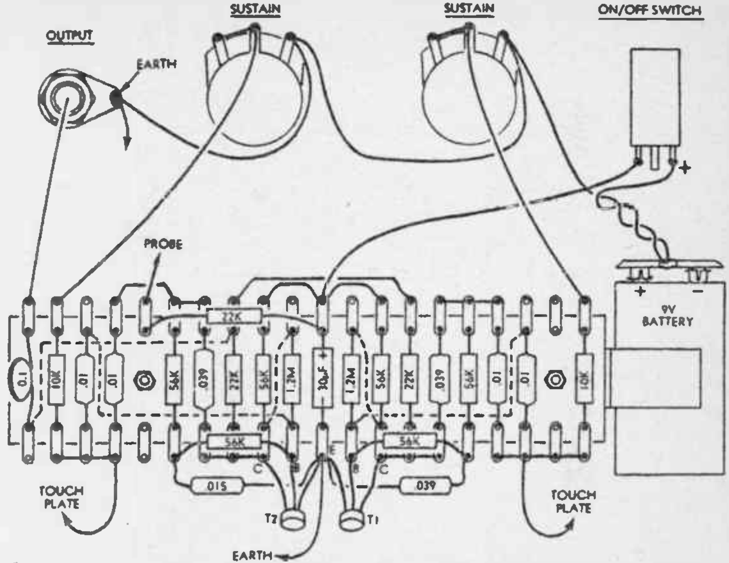
Figure 1 shows the modes of oscillation possible. 1(a) shows a continuous oscillation at constant amplitude. 1(b) shows a lightly damped oscillation and 1(c) shows a heavily damped oscillation. Note that the waveform remains essentially sinusoidal.

To synthesize Bongos then, as we have in the particular instrument featured, we have used two Twin-T oscillators in the ringing mode. Other percussion instruments can be similarly simulated by changing the values of the capacitors in the T-networks. For example, a bass drum can be simulated by using larger capacitors to lower the fundamental frequency of oscillation. Bass drums being what they are, of course, only one oscillator would normally be involved.

For a bongo set, two Twin-T oscillators are needed. One is tuned to around 280Hz and the other to around 400Hz. Of course, constructors may build units with more than two oscillators, if they so desire. Tuning is simply a matter of selecting the capacitors in the T-networks.

The electrical impulse to shock the oscillators into the "ringing" mode is obtained by touching the "touch plates" connected to the junction of the T-networks containing the potentiometers. In some cases, the "stray"

DIAGRAM SHOWS COMPONENTS AND WIRING



The assembly of the Bongos is a straight-forward process if this wiring diagram is followed closely.

hum field will be strong enough for the oscillators to be triggered when the plates are touched. (i.e. the hand is used to momentarily inject hum into the circuit). In other cases, there may be no mains wiring in the vicinity and so we provided a lead connected to the positive supply rail, via a 22K resistor.

This is held in one hand or connected to the player's metallic watch band, if this proves more convenient. In this way, a DC pulse is injected into the circuit via the 22K resistor and the player's skin resistance, when the respective plates are touched. The 22K resistor prevents the battery from being discharged rapidly if the lead is inadvertently shorted to the case with the unit switched on. The 22K resistor is not shown in the photograph of the prototype, as it was inserted after the photographs were taken.

Perhaps, in passing, we should point out that there are several other applications in electronic musical instruments where this circuit can be used to advantage. One of the most notable is that of active filter, which can be used for waveshaping in electronic organs. Most of the tone generators in electronic organs produce square waves but, to produce flute or tibia voices, the square wave must be filtered to almost a pure sine wave. This would be difficult with passive filtering but the active filter can be fed with any waveform (into the base of the transistor) and will produce fairly clean sinusoidal wave forms near its resonant frequency.

While we have specified the use of high-gain NPN silicon transistors there is no reason why high-gain PNP silicon transistors cannot be substituted if they are on hand. The only modification required to the circuit is to reverse the polarity of the supply voltage and the electrolytic capacitor.

Note that the prospective constructor is not limited to the method of con struction that we have used. For example, a unit could be built up featuring miniature bongo drums made out of wooden egg-cups; the touch plates would be the 'skins' of the bongos. The method we have used is described in the following paragraphs.

A diecast box measuring $6\frac{3}{4}$ x $4\frac{3}{4}$ x 1/8inches is used to house the com ponents. This is large enough, if need be, to house more than two oscillators. All of the components, with the exception of the two potentiometers are mounted on a 20-lug section of tag strip. While layout is not critical, we would advise novice constructors to follow that shown in the wiring diagram and photograph.

When drilling the diecast box, use a sharp drill at high speed and a low rate of "feed" (i.e. do not put too much pressure on the drill). Preferably, use a drill stand. If care is not taken when drilling, the box may crack.

The touch plates we used were made of light-gauge aluminium and measured 3 x $2\frac{1}{2}$ inches. They were mounted using three 1 inch insulating pillars per plate. Each plate is connected into circuit via lead which is soldered to a lug under one of the plate-securing screws, underneath the plate.

While we have used roundhead screws to secure the touchplates in the prototype, constructors may wish to use countersunk screws to lessen the possibility of snagged fingers when the unit is played exuberantly.

The lead connected to the supply was brought out through a hole in the case lid. The lead is fitted with an alligator clip, to be connected to the player's watch band. However, this may prove awkward in practice and it may be more convenient to solder the lead to a hand-grip made of tinplate or other metal.

The battery is a small nine-volt type. Since the current drain is considerably less than 1 milliamp the battery life should be long - almost equal to the shelf life.

The Electronic Bongos can be used with just about any amplifier, provided they do not overload the amplifier input. The more powerful the amplifier, the batter; the same can be said about the size of the loudspeaker. For best results, it should be used with a high power guitar amplifier and matching loudspeaker system. The output signal from the Electronic Bongos is approximately 0.5 volts RMS which is enough to drive almost any amplifier to full power using the "pickup" or "auxiliary" input.

When assembly is completed, connect the Bongos to an amplifier and loudspeaker. Switch on and turn both potentiometers fully clockwise. Both oscillators should be operating. If not, turn off and check for mistakes in wiring. Now set each potentiometer so that its associated oscillator is just "on the verge" of oscillation. Hold the probe in one hand and tap each of the touch plates with a finger. Each oscillator will emit a sharp "bong" when the appropriate touch plate is touched. It is surprising how similar it is to play to a conventional set of Bongos.

2 ADDITIONAL IDEAS FOR THE ELECTRONIC BONGOS

The first concerns the hand-held probe to which is connected the positive supply line. As an alternative, a third touch plate is installed between the existing two plates. The third plate is connected to the positive supply and the 22K probe resistor may be omitted. In playing the Bongos, the thumb of one hand would rest permanently on the centre touch plate while the fingers tapped the oscillator plates in the normal way.

The second modification was suggested by a reader during a telephone conversation and unfortunately, he did no give his name so that we could acknowledge it. His suggestion involves the use of a stereo amplifier and associated loudspeaker systems. Instead of mixing the two oscillator signals together via 22K resistors, each oscillator signal is coupled individually via a 22K resistor and 0.1uF capacitor to each channel of the stereo amplifier. In this way, the subjective level of each "drum" can be adjusted for equal intensity with the aid of the balance control.

If the amplifier had separate tone controls for each channel, this would enable further adjustment to obtain the best effect from each "drum".

PARTS LIST

1 diecast box and lid. Approximate dimensions $6\frac{3}{4}$" x $4\frac{3}{4}$" x 2 1/8"
1 20-lug tagstrip
1 SPST toggle switch
2 25K (Lin) potentiometers
1 output socket
2 BC108, 2N3565 or similar silicon NPN transistors.

CAPACITORS

(Voltage ratings higher than 9V will suffice)

1 x 30uF/12VW electrolytic
1 x 0.1uF, 3 x .039uF, 1 x .015uF,
4 x .01uF, all ceramic or polyester.

RESISTORS

($\frac{1}{2}$ or $\frac{1}{4}$ watt rating)

2 x 1.2M, 6 x 56K, 3 x 22K, 2 x 10K.

SUNDRIES

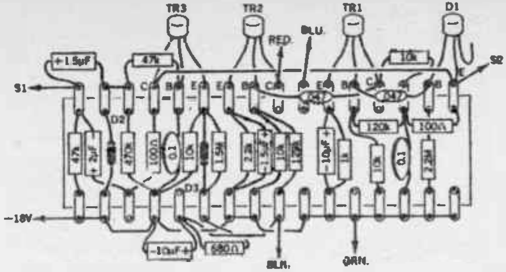
2 knobs, 6 insulating pillars, aluminium for touch plates, screws, nuts, battery clamp, hook-up wire, alligator clip, solder etc.

AN ELECTRONIC CRASH CYMBAL

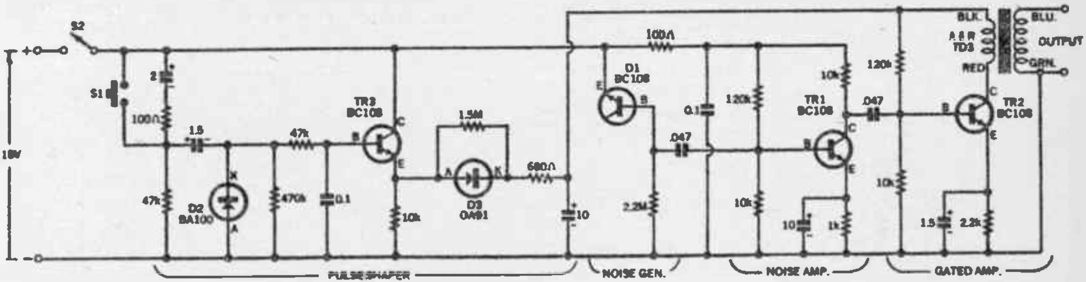
The crash cymbal, a large brass disc resting on a central pivot, is a logical choice for such an effect and, in fact, has been a popular choice of dance bands and similar groups for many years. Struck with a drum stick it creates a resounding crash, with just a hint of metallic tone.

But it isn't quite so easy for an organist, or a guitarist, or other solo player, to create such an effect. A real crash cymbal is reasonably bulky, while there is some doubt whether a solo player could always arm himself with a drumstick in time to create the desired effect.

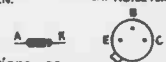
With our electronic cymbal the musician need only press a button to imitate the crash cymbal effect. If one were to run leads to a remote foot pedal the unit could be triggered without missing a beat.



TRANSFORMER MOUNTED BELOW BOARD



CRASH CYMBAL



The operation of the circuit will be most easily understood if the various sections, as marked, are considered separately. The function of each section is explained in the text. Note the BC108 used as a noise diode.

Another advantage is the compact size of our unit. An amplifier must complement this unit but most guitarists and organists employ an amplifier as part of their equipment.

Before describing the circuit in detail, it may help to discuss the sound effect we wish to simulate and the means employed to do so.

At the moment the cymbal is struck, the sound intensity is maximum, decreasing to zero over a short period. The majority of the sound generated is very similar to white noise, but there is a slight ringing effect.

A white noise generator forms the basis of our unit. The noise generator output is fed to the input of a gated amplifier; an amplifier which is normally off until a suitable pulse occurs to turn it on.

In our case, we use another transistor stage to generate a suitably shaped pulse. Closing a switch in the input circuit of this stage creates a pulse in its output stage which charges a capacitor. This capacitor functions as the supply rail for the gated amplifier, switching the amplifier on at full gain when the capacitor is fully charged and reducing the gain to zero over a brief period as the capacitor discharges.

The output from the gated amplifier is transformer coupled to the output terminals. This transformer gives the output a very slight ringing effect to make the cymbal sound more realistic.

We can now discuss the circuit in detail. As mentioned earlier, the noise generator is the basis of our unit. This consists of a reverse biased BC108 base to emitter junction (D1) in series with a 2.2M resistor across the supply (see 'Noise Gen' on the circuit diagram). To increase the noise output a conventional amplifier is included (TR1).

The output from the noise amplifier is coupled to the base of the gated amplifier via a .047uF capacitor. Normally the gated amplifier is non-operative as there is no collector voltage or base bias, thus there is no output to the external amplifier.

The gated amplifier is controlled by the pulse shaping circuit. When the push button S1 is pressed, the fully supply voltage is applied across the 47K resistor. The 1.5uF electrolytic connected to the junction of this 47K and the S1 contact begins to charge rapidly. Since the base emitter junction of TR3 is part of the charge path. TR3 is biased into conduction for as long as this capacitor is charging.

When TR3 conducts the majority of emitter current flows through the OA91 diode in series with the 680 ohm resistor, charging the 10uF electrolytic. Eventually, the voltage across this capacitor reaches the potential at TR3 emitter (minus a small drop across the OA91 diode) and the OA91 biases off. At this time the capacitor is charged to about three-quarters of the supply rail voltage, but is isolated from the charging circuit by the reversed biased diode. This capacitor becomes the supply for the gated amplifier collector and bias network. The 1.5M resistor shunting the diode is to allow the capacitor to completely discharge after the gated amplifier has completed its cycle.

Several components in the pulse shaping circuit are included to eliminate "plops" as the switch is pressed or released. The plop as the switch is pressed is eliminated easily by virtue of the .1uF capacitor from TR3 base to the common rail. This capacitor limits the fast rise time of the triggering pulse.

Any plop as the button is released is most noticeable if the 1.5uF electrolytic has not fully discharged. The network across S1 and the BA100 diode are included to eliminate this problem.

As already suggested, the gated amplifier (TR2) derives its collector and bias voltage from the 100uF capacitor. As the capacitor discharges through TR2 the collector and base voltages decrease gradually to zero. Thus initial output from this stage is high, decreasing in a short period to zero. Unless the button is pressed again there is no further output.

Output from the gated amplifier is transformer coupled to introduce an amoutn of ringing into the signal. We used a driver transformer and considered the resonance was suitable, but it can be modified by connecting various values of capacitance across the primary or secondary windings. Values between .002 and 047uF across the primary winding would provide a good starting point.

The reader should now be familiar with the circuit operation and ready to begin construction. We wired our unit on a miniature tag board with 18 pairs of tags. The output transformer and mounting pillars should be fastened to the underside of the board first, as their positions dictate the location of other components and the need to insulate some component pigtails.

The wiring of the noise generator should be straightforward, there being only three components including the coupling capacitor. Next the noise amplifier can be wired. There is only one component for this section under the board, a 0.1uF bypass capacitor. The only pigtails which need insulating are those on the coupling capacitors.

The gated amplifier should cause no problems. There are no small components under the board. The emitter bypass electrolytic pigtails should be insulated. The transformer leads are colour coded as shown on the wiring diagram. With this section wired check your work so far against the circuit and the wiring diagram.

In the pulse shaping and trigger circuit the 100 ohm resistor in series with a 2.2uF electolytic across S1 is under the board. The pigtails of both these components are insulated.

Finally fit all interconnecting wires as shown on the wiring diagram, and connect the output cable and switches. If the reader intends to use batteries as the source of supply, two 9 volt type 216 batteries in series would suffice. A red wire from one battery clip and a black wire from the other are soldered together and the join insulated. The remaining red wire goes to the on-off switch while the black wire is soldered to the tagboard common rail.

We mounted our unit in a die-cast aluminium alloy case. The batteries fit under the board and it is advisable to insulate (with tape) any tags which might short to the battery cases.

The metal box is not essential, unless the device is to be used as an external, add-on unit, with a variety of instruments. If it is to be a permanent part of, say, an organ, the terminal board can be mounted in some convenient place inside the cabinet, along with the other electronic gear.

Before testing the unit, check your work against the wiring diagram and circuit. To test, connect the output lead to the input of an amplifier with reasonably flat response, (our unit gave approximately 10 volts PP initial output which should rive any amplifier).

When the push button is pressed the cymbal effect should be heard. The duration of the output can be varied by changing the value of the 10uF electrolytic in the emitter circuit of the pulse shaper, more capacitance giving a longer effect.

The only other adjustment which should be necessary is the tuning of the output transformer if you are not happy with the tone of the small ringing content in the output. This was discussed earlier in the text.

External to the unit proper is the switch used to initiate the sound. While this does not make any great demands electrically, its mechanical characteristics are important. In particular, it must be easy to operate and prominent enough to ensure that the user can find it without fumbling. This rules out some types of press button switches which are both small and stiff to operate. What is needed is a key or lever comparable with the keys or stops of an organ.

PARTS LIST

Metal case (optional) $4\frac{3}{4}$" x $2\frac{1}{2}$" x $1\frac{1}{2}$"
Transistor amplifier driver transformer

TRANSISTORS

4 BC108 or equivalents

DIODES

BA100, OA91

CAPACITORS

2 10uF 16VW electrolytic
1 2.2uF 25VW electrolytic
2 1.5uF 33VW tantalum
2 0.1uF 100VW polyester
2 .047uF 100VW polyester

RESISTORS (½ W)

1 2.2M	4 10K
1 1.5M	2.2K
1 470K	1 1K
2 120K	1 680 ohms
2 47K	2 100 ohms

2 9 volt batteries. 216 or similar
2 battery clips to suit above
Miniature tag board, (16 pair of tags)
Miniature toggle switch (on-off)
Suitable triggering switch (see text)

TREBLE BOOST PREAMPLIFIER FOR YOUR GUITAR

FOLLOWING THE NUMEROUS REQUESTS FROM READERS FOR SUCH A UNIT, WE ARE PRESENTING A TREBLE BOOTS PREAMPLIFIER FOR USE WITH ELECTRIC GUITARS. BEING SELF CONTAINED AND INDEPENDENT OF ANY EXTERNAL POWER SUPPLY THE UNIT MAY BE USED WITH ANY GUITAR AMPLIFIER SYSTEM TO PROVIDE UP TO 26dB BOOST AT 3KHz WITH REFERENCE TO 300Hz.

In a "pop" music group, more often than not, the balance of instruments is given over to electric guitare. Essentially, there are two distinct types used, bass and rhythm, with the latter played either as a a simple rhythm accompaniment or featured as a lead instrument.

As a lead instrument playing melody, it is usual to expect that a rhythm guitar will be provided with more than normal treble emphasis. Indeed there seems to be no limit to the amount of treble boost required by some lead guitarists. Nevertheless, a discreet application of treble boost is desirable as a highlight for the melody section and to add interest generally.

Quite often the required boost is within range of the amplifiers normal tone controls together with local tone controls on the guitar. However, the controls associated with the guitar are purely passive networks providing limited bass and treble cut only. As such, when used, they tend to reduce the effective guitar output and diminish, to some extent, the average level of sound.

Usually, reduced guitar output can be accommodated by increasing amplifier gain. But a situation will probably arise, perhaps with a borrowed amplifier or new guitar, where the system lacks sufficient treble boost and/or adequate gain with full bass cut. And, inevitably there will be an occasion when a large amount of boost is required for some special effect; more than can be provided by both the amplifier and guitar tone controls together.

Apart from fitting an external treble preamp, which is the topic of this article, it may be possible with some guitar amplifiers to make a slight modification to add a limited amount of treble boost. The technique consists of adding a treble pass network to the feedback circuit.

However, we stress that modification to amplifiers should not be made unless the reader is reasonably confident and that, further, only a limited amount of boost may be obtained in this way. In essence, the modification enables the negative feedback to be reduced, via a suitable switch, at treble frequencies, so increasing the amplifier's gain.

In most valve amplifiers the feedback network is fairly simple, consisting of a series resistor from the output-transformer secondary back to a cathode resistor in one of the stages. A simple series RC network to shunt the cathode resistor will then provide the required treble boost.

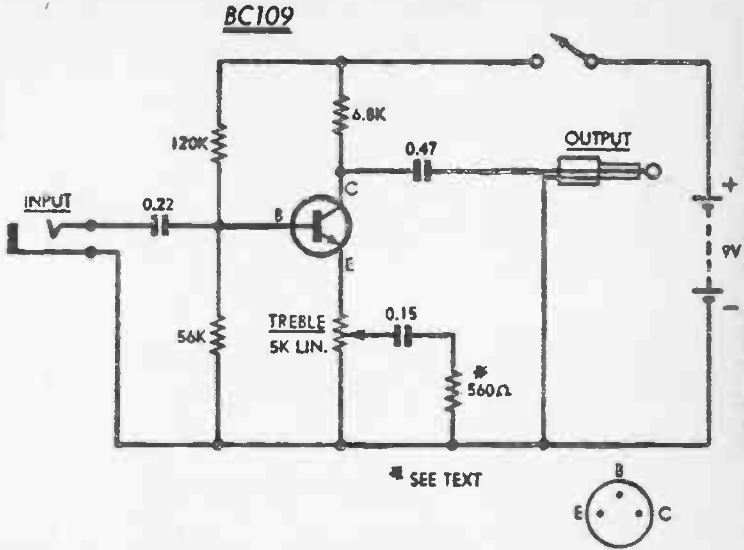
Nevertheless, it would be unwise to aim for more than about 6 or 8dB by this means. As even the better amplifiers do not have any more than about 16db of negative feedback in the main loop, reductions of more than the suggested amount could result in stability and distortion problems.

Just to put all these dBs in perspective, for those who may not be proficient in their use, we most often use them to relate changes in voltage or power to actual loudness changes as heard by the car. About the smallest perceptible change for a healthy ear is 3dB, any smaller change would not be heard. However, the smallest significant change in sound level is around 6dB; a voltage change of two times, or a power change of four times.

However, it takes a change of 10dB (10 times the power) to produce a subjective impression of "twice as loud". (This is well worth keeping in mind when comparing the power output specifications of various amplifiers). From this it can be appreciated that the 20dB of boost provided by this treble boost amplifier will produce an audible increase of approximately four times. This is quite a large amount, when compared with a typical domestic amplifier which may provide around to 12 to 16dB available boost at 10KHz.

The externally fitted treble boost preamplifier described here will, as we said, provide 20dB boost at 3KHz with reference to 300Hz. However, with a minor alteration an extra 6dB can be obtained, making the total boost 26dB.

A circuit diagram for the boost pre-amplifier. Note that the omission of the 560 ohm resistor will provide extra boost making a total of 26dB.



TREBLE BOOST PREAMPLIFIER FOR GUITARS

The preamplifier is quite simple in both construction and operation, consisting of a single transistor stage with a facility for adjusting the amount of treble boost. A lead from the guitar plugs into the preamp's input jack, while an output lead from the preamp connects to the guitar amplifier. An on/off switch is incorporated in the boost control for greatest convenience.

Continuous control of treble boost is provided by a 5K linear potentiometer incorporated in the circuit as shown. Its function in providing the large amount of treble is quite simple.

At bass frequencies the potentiometer, which is in the transistor emitter circuit, introduces a large amount of degenerative feedback maintaining a stage gain of only slightly more than unity. However, as the frequency increases the emitter resistance is progressively shunted by an impedance, comprising a 560 ohm resistor and 0.15uf capacitor, to a point at about 3KHz where the gain is 20dB, and limited only by the 560 ohm resistor.

As the bypass tapping point is moved down the resistor, when the potentiometer is rotated, the maximum treble gain is reduced to some intermediate value. Thus, the treble boost effect can be varied over a wide range to meet all situational requirements.

In restricting the maximum gain to 20dB at 3KHz, our aim was to prevent any overload and consequent distortion in the guitar amplifier input stages. However, the chance of over load occurring will depend upon the guitar output and its high frequency harmonic content, and the maximum allowable input to the amplifier.

Depending upon playing technique and position of the guitar pickup heads the output waveform will vary from something essentially sinoidal to one which comprises mainly consonant harmonics. But, the amplitudes of harmonics in the region of 3KHz will be far smaller than the fundamental and lower harmonics. Provided that the amplifier has a reasonable overload margin the 560 ohm resistor could be omitted, thus increasing the treble gain by a further 6dB.

Whether to include the resistor or reduce it in value is a matter for individual decision, depending on the particular requirement and equipment used. Conceivably, the 560 ohm resistor could even be increased if less than 20dB treble gain is required.

As presented, the preamplifier operates from a 9-volt supply which, for convenience, can be a small battery with press-fit terminals. The supply is switched via a switch on the treble potentiometer but, if the unit is accidentally left on occasionally, it will have little effect on the battery life as the preamp current drain is very modest.

Construction of the preamplifier is very straightforward and should pre sent little difficulty. We built the prototype in small aluminium box, it measures 4¼" x 2 3/8" x 1½".

The components were wired on a small section of miniature resistor panel mounted over the battery at one end of the box. A bracket fashioned from a small piece of aluminium serves to retain the battery, and connection is made via a standard pressfit connector.

Output from the preamplifier is taken via a length of shielded microphone cable which should be fitted with a connector to suit the amplifier with which it is to be used. The input connection is made via a standard jack-socket, these being most frequently used in guitar hook-ups. The socket was mounted in the end of the box, opposite the wiring board and battery assembly and the potentiometer was mounted centrally between the two.

The housing was then completed by the addition of the lid, which was secured by two self-tapping screws. A knob fitted to the treble control potentiometer and an indicating arrow inscribed with black drawing ink and laquered completes the unit.

PARTS LIST

1 Aluminium box 4¼" x 2 3/8" x 1½"
1 9V battery and connector
1 5-lug length of miniature resistor panel
1 Standard jack socket and plug
1 Knob

TRANSISTORS

1 BC109 or equivalents

RESISTORS

1 5K linear potentiometer with switch
1 120K, 1 x 56K, 1 x 6.8K 1 x 560 ohms (All ½W 5%)

CAPACITORS

1 0.47uF low voltage plastic or ceramic
1 0.22uF low voltage plastic
1 0.15uF low voltage plastic

A PREAMPLIFIER FOR ELECTIC GUITARS

WITH THIS SIMPLE LITTLE PREAMPLIFIER, IT IS POSSIBLE TO PLAY AN ORDINARY ELECTIC GUITAR THROUGH ANY AMPLIFIER THAT HAS ENOUGH GAIN TO WORK FROM A CRYSTAL OR CERAMIC PICKUP. WITH IT, THE BUDDING GUITARIST CAN GAIN ALL THE PRACTICE HE NEEDS, WITHOUT HAVING TO SPEND A LOT OF MONEY INITIALLY ON A SEPARATE AMPLIFIER AND LOUDSPEAKER SYSTEM.

In fact, the preamplifier came into being because of a situation involving one of our staff members. A young relative had come across an electric guitar which was available cheaply and which he felt he might be able to lean to play.

However, being rather short of money, he was quite unwilling to become involved also in the purchase of a guitar amplifier, on the off-chance that he could use it. What he did have on hand was an ordinary record playing amplifier, with ample power for guitar practice, but not enough gain to work from the 20-off milli-volts of signal available from a typical instrument.

Would it not be possible to make up a small preamplifier which would bring the signal from the guitar up to a suitable level to drive the amplifier?

On the assumption that others might possibly be faced with a similar dilemma, we decided to build up a preamplifier for the job. The requirements were set down as follows:

* The preamplifier would have to be self-contained and independent of the power mains or the power supply in the basic amplifier. This meant that it would have to be transistorised and work from a small battery, housed inside the case.

* Input impedance would preferably have to be 50K or higher.

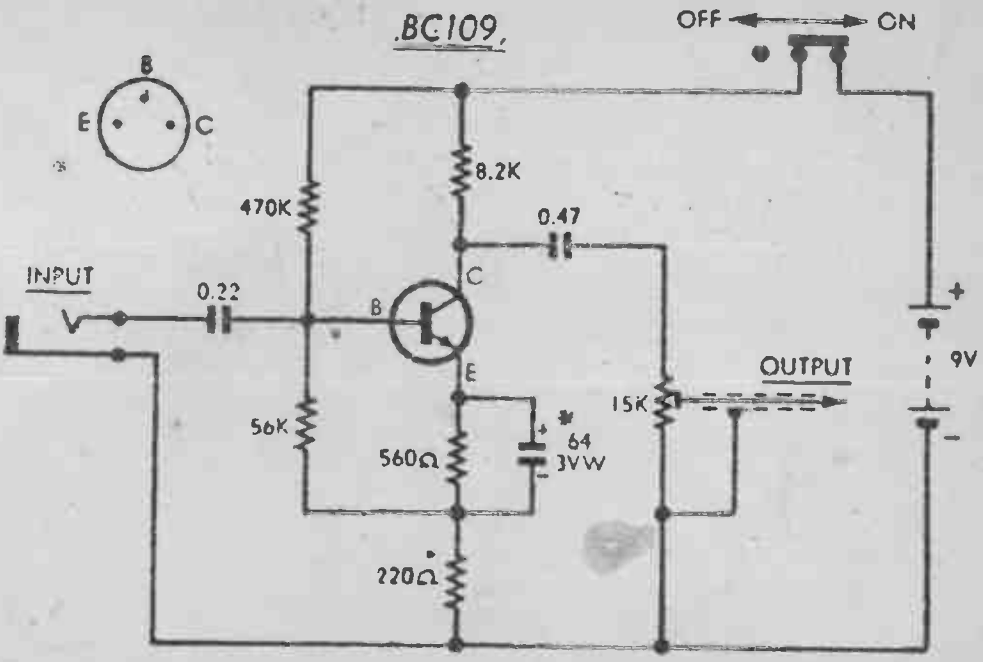
* To boost the output from a guitar to the approximate level in a ceramic or crystal pickup channel, the preamplifier would need to have a gain of from 10 to 20 times. It would also have to be capable of accepting the full likely signal from the bass string, without overloading.

* The preamplifier would not need provision for tone compensation, vibrato or fuzz, at least in its present form. Its role would simply be to allow the would-be musician to learn the fundamentals without becoming involved in effects. Ordinary volume and tone control functions would be available from both the guitar and the basic amplifier.

A circuit which meets these requirements and which, at the same time, is simple and inexpensive, is used. It involves a single transistor connected in a common emitter configuration, with degenerative feed-back in the emitter circuit and a "boot-strapped" bias divider to secure increased input impedance. The intrinsic voltage gain is about 18 times, assuming the use of a transistor with fairly high beta (i.e. current gain).

Output could be taken directly from the 0.47uF capacitor coupling to the transistor collector but we have suggested the provision of a tab pot, which allows the gain of the preamplifier to be varied to suit individual requirements. If the input circuit of the basic amplifier is prone to overload, or if the gain of the system overall is high enough to make volume control settings critical, the tab preset pot. can be turned back as necessary.

As presented, the preamplifier operates from a 9-volt supply which, for convenience, can be a small battery with press-fit terminals. The supply voltage largely determines the signal voltage swing at the collector of


BC109
OFF
ON
B
E
C
8.2K
470K
0.47
INPUT
0.22
B
C
E
+
9V
-
OUTPUT
15K
56K
560Ω
*
64
3VW
220Ω

*SEE TEXT

the transistor and, in the circuit as shown, the maximum RMS output voltage is a little over 2V. This, in turn, means that the input voltage should not exceed say, 100mV, otherwise appreciable distortion will result.

However, the intrinsic gain of the preamplifier stage may be decreased to six times by simply removing the emitter bypass capacitor, providing relatively distortion-free amplification for signals peaking up to 300mV. It is most unlikely that signals anything like this will be encountered from normal instruments, however, If they were, there would be no need for this preamplifier; such a guitar could feed straight into a pair of pickup terminals!

We built the prototype preamplifier in a small aluminium box measuring 4" x 2 1/8" x 1 5/8" inches. The components were wired on a small section of miniature resistor panel. While the tab-mounting potentiomete was secured to the bottom of the box, with a hole provided for access to the screwdriver-slot adjustment.

A bracket fashioned from scrap aluminium serves to retain the 9-volt battery, while a small slider type switch serves for the off-on function. The current drain is only about 0.5 milliamp and it will not matter much if anyone using the preamplifier forgets on occasions to switch it off. Signal input to the preamplifier is via a standard phone socket, which seems to be the most usual connector for electric guitars. Output is via a length of shielded microphone cable; this should logically be terminated in a connector to suit the amplifier with which it is to be used.

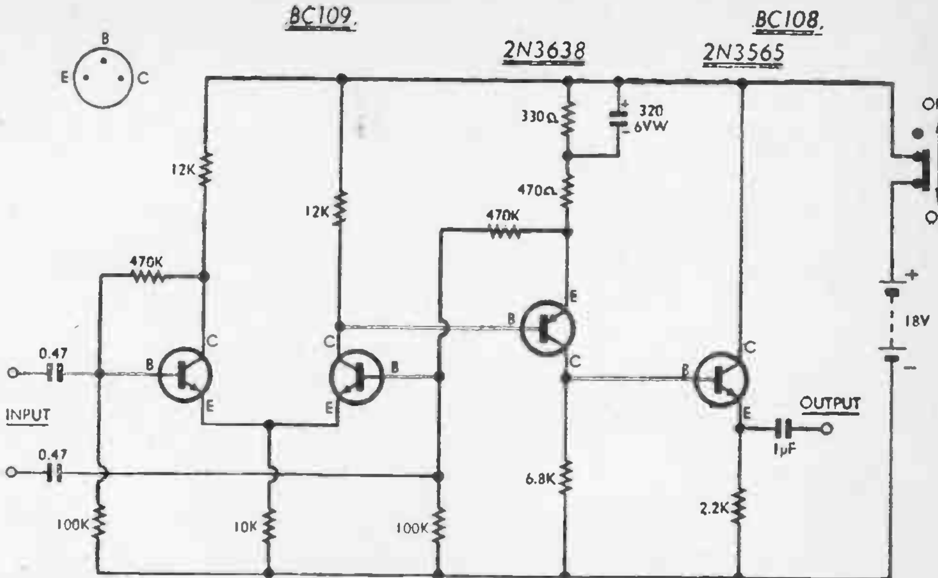
In the case of stereo record players, connection to both channels is desirable in order to secure the full available power output. In some cases connection to both channels will be available via a mono "tape input" socket; in other cases, output from the preamplifier will have to be bridged to both inputs. Naturally, the braid of the cable must go to the earthy side of the amplifier input system.

Tested with typical amplifier systems and a solid body guitar, the preamplifier gave ample output for practice in the home with the guitar and amplifier volume controls about half on. There was no sign of overload with the bottom string plucked hard, and enough gain to exploit novelty effects with the top string.

Needless to say, the young would-be musician was delighted - though we didn't call back to register the reaction of other members of the household.

A DIFFERENTIAL AMPLIFIER

While experimenting with the guitar preamplifier just described, we took the opportunity of mocking up a type of stage which we have occasionally been asked for - a differential amplifier capable of accepting in-



This circuit may be of value to the occasional reader who may desire to use microphones or other devices with balanced output in conjunction with equipment having unbalanced input.

put from a circuit of which both sides are active and balanced with respect to earth. This kind of system is commonly employed in professional audio equipment.

As the name implies, a differential amplifier amplifies the difference between two input signals. In addition, it can provide a degree of rejection to signals which are common to both inputs - that is, a signal which appears at both of the balanced inputs with the same phase and amplitude. The ability to reject such signals is termed common mode rejection.

The advantage of a balanced input providing common mode rejection is that it offers a degree of hum immunity, this being important where long connecting cables are necessary. Discrimination against hum is achieved by virtue of the fact that a hum signal will usually appear at both inputs in the same phase and so be cancelled by the differential action. The wanted signal from a microphone or other source is amplified in the normal manner.

In professional circles, balanced inputs are commonly fed to transformers which fulfil necessary functions such as : impedance step-up, impedance step-down, balance to balance, or balance to unbalance. Because such transformers have to be above criticism in terms of frequency response and distortion and effectively shielded against hum pickup, they are usually quite expensive.

There is therefore some incentive to devise an active stage using transistors which will accept balanced input and provide unbalanced output (i.e. one side earthed), and with some scope for meeting the needs of impedance and gain. The circuit herewith may prove useful on this score.

In the circuit as shown, the degree of common mode rejection is determined mainly by the size of the resistor common to the emitters of the input transistors. For complete rejection of signals common to both inputs, the resistor wouldl have to be infinitely large. This is not practical, although a high order of rejection can be achieved by using an extra transistor as an emitter load with constant current characteristics.

For our purpose, however, we have simply used a 10K common emitter resistor in the interests of simplicity and economy. As presented, the preamplifier provides 30dB of common mode rejection.

Directly coupled to the second transistor in the differential pair is a PNP common emitter stage providing additional gain, making the total about 220 times. Thus the input sensitivity will be a little more than 1mV for a nominal output of 250mV.

Although differential amplifiers typically have low input impedances (approximately 2K for the present preamplifier) noise output tends to be high than for a single-ended stage of similar voltage gain. However, the signal to noise ratio of the preamp, including the single ended PNP stage, is quite satisfactory at 48dB.

As an option, we have provided an emitter follower stage for low output impedance. This stage can either be included or omitted depending upon the load requirements. For load impedances of greater than, say, 50K the emitter follower may be omitted, the load simply being coupled to the collector of the PNP transistor via a 0.47uF capacitor.

Otherwise, the direct coupled emitter follower should be retained, providing a low output impedance sufficient to accommodate loads of less than 1K. For lower load impedances a larger coupling capacitor is required to maintain adequate low frequency response.

The frequency response of the preamplifier is in fact 3dB down at 45 KHz and at the low end, 30Hz. The total current drain of the preamp as presented is 5mA but, without the emitter follower stage, the drain is reduced to about 1.7mA.

BEGINNER'S ORGAN

It is not possible, with an organ as simple as this, to play chords. If more than one note is pressed only the higher frequency one will sound. More elaborate organs which are capable of playing chords, have either an oscillator for every note, or use a system of frequency division.

We have given this organ a range of approximately one and a half octaves, from B to F. This is the maximum range which we felt was safe to suggest, taking into account the spread of transistors. It may be possible to extend the range, depending on the beta of your individual transistor but, sooner or later, the circuit will refuse to operate.

For our keyboard we evolved a simple but effective method of making keys from tinplate, salvaged from a discarded food tin if necessary. We did this rather than encourage our readers to embark on a search for proper keys discarded from old instruments. While the latter make a very nice job, they can be very hard to find. If our readers waited until suitable keys turned up, they might never finish the project.

The oscillator is mounted on a piece of Veroboard, and a drawing of the layout is shown.

The individual resistors in this network determine the frequency of the oscillator. By selecting each value correctly, we are able to tune the oscillator to the notes of the musical scale.

The values of the resistors in this network may appear a little strange to the beginner. Some are standard values which can be purchased directly - these are known as preferred values - while others are nothing like any value you will find in a shop. These latter values can be obtained by connecting two preferred value resistors in series.

To help the builder identify the individual resistors as they will be supplied we are reproducing the standard resistor colour code, which should be self explanatory. Also, having identified the individual resistors, it will be necessary to determine how the various values are paired in order to produce the odd values already discussed. To simplify this we have prepared a layout diagram showing the complete resistor configuration.

Naturally, the job of providing suitable resistor values would be made far easier if we used small variable resistors - "preset pots" - instead of fixed one. But these are rather expensive - over three times the price of fixed resistors. However, there is nothing to stop you using them, if you feel that way inclined - and can afford it.

If you use preset pots. the layout and construction we used will not be suitable, because there will not be enough room for the pots. Some alternative arrangement will have to be found.

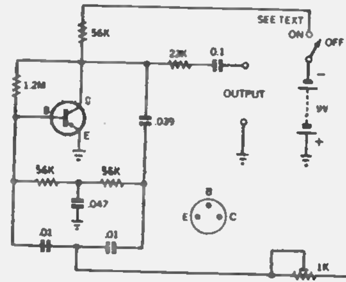
For these reasons we imagine most readers will settle for fixed resistors. The values we have nominated should bring each note very close to the correct frequency, but a final check can be made by comparing them against their equivalent on a piano or organ. For maximum accuracy use only good quality (preferably new) five percent resistors. Should it transpire that one or two notes need some small correction, this can be taken care of quite easily. We will have more to say about this later on.

We used a piece of "no copper Veroboard" on which to mount the resistors and keys.This differs from conventional Veroboard in that it has no copper strips on it. Normally, this board is supplied with metal pins which may be inserted in the holes, as required, and used as anchor points for components.

The layout we have shown is for a piece of 0.15" pitch Veroboard. Fit all the eyelets into place before attaching any keys or resistors. To fasten the eyelets, rest the underside on a firm flat surface and tap the other side gently with a hammer. This will roll the edge of the eyelet and secure it firmly in the board.

Contacts for the white keys were made from drawing pins. These are pushed through the holes and the ends bent over with a small hammer. With all the eyelets and drawing pins fitted, the board should appear as in the accompanying photograph. Now turn the board over and fit the resistors and associated wiring according to the layout diagram.

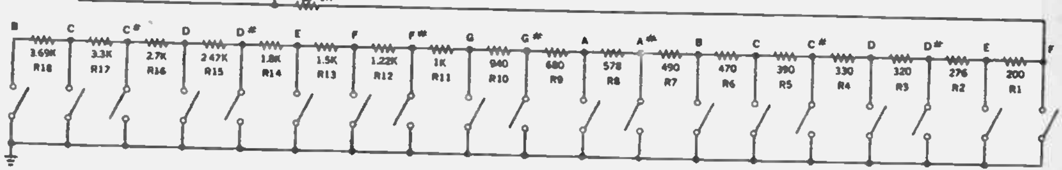
At this stage the keyboard network can be connected temporarily to the oscillator and a rough check made of the accuracy of the notes. Connect the common side of the oscillator to the common side of the keyboard. Connect a short length of hookup wire to the free end of the 1K pre-set pot, and initially, connect this to the common side of the



Resistor Combinations

R1 — 2 x 100
R2 — 220, 56
R3 — 220, 100
R4 — 330
R5 — 390
R6 — 470
R7 — 270, 220
R8 — 560, 18
R9 — 680
R10 — 2 x 470
R 11 — 1K
R12 — 1K, 220
R13 — 1.5K
R14 — 1.8K
R15 — 2.2K, 270
R16 — 2.7K
R17 — 3.3K
R18 — 3.3K, 390

Circuit diagram shows exact resistance needed for each key. List at left shows which preferred values to connect in series to provide the exact values required.



oscillator. The oscillator should function and the pot should be set to give a frequency equal to the second F above middle C.

Disconnect the length of hookup wire from the common side and connect it in turn to each of the tappings on the keyboard resistor network. It should produce a series of notes very close to what is required but, if some are slightly out, do not worry unduly. They can be corrected later. Note that the pitch standard we used is A equals 440Hz.

The board is now ready to be fitted with keys. The white keys are fitted first to the row of eyelets down the middle of the board. Then the black keys are placed on the row along the edge.

The keys themselves can be cut from an old "tin can". Make sure, however, that the metal is not tarnished in any way, and that the tin is not one of the "corrugated" variety. You can cut the keys, (using our diagrams as a guide) with a pair of tinsnips, or old scrissors. But don't be tempted to use good scissors to cut the tin.

We suggest that you cut the corners of the keys, particularly the white keys. It is very easy to cathh one's finger under the white keys and suffer a painful cut from the sharp edges. Due to their position, the black keys do not present so much of a risk.

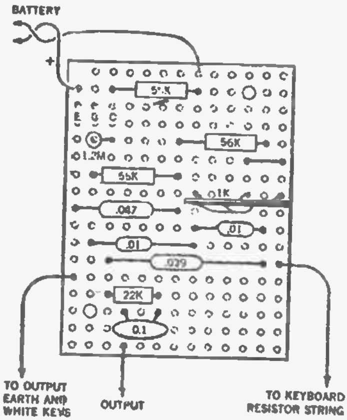
To fit the keys, tin the eyelets and both sides of the short ends of the keys. With the key held in a pair of pliers, put it in the correct position, and melt the solder with a reasonably hot iron. When it is molten, take the iron away, and hold the key in position until the solder cools. Make sure that none of the keys fouls its neighbours.

The manner in which the black and white keys are connected into the circuit is a little unusual and calls for an explanation. The white keys are permanently connected to the common side of the circuit (indicated by the earth symbol) and make momentary contact with the active side as they are depressed. The black keys are permanently connected to the active parts of the resistor network, and make momentary contact with the common side as they are depressed. The common side is, in fact, the rear of the white contacts.

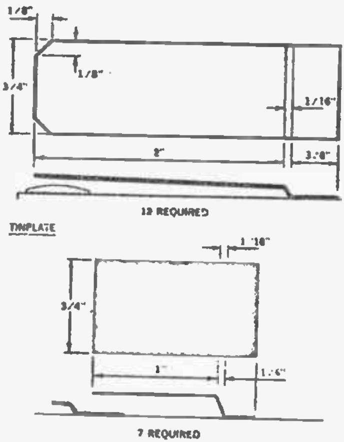
Electrically, it matters little which way round the keys are connected; either way will complete the appropriate circuit. However, the arrangement does simplify layout and saves a lot of space.

On the other hand it does create a risk of hum being introduced into the system, by way of the black keys. Being connected to the active side of the circuit they will be prone to hum as the fingers touch them - in the same manner as an amplifier will hum when the active input terminal is touched. Fortunately, the insulation provided by the black paint is sufficient to prevent this from happening.

The keys are best painted after the board is assembled. We used a spray pack of glossyblack, with paper and masking tape to mask the



Component layout for the audio oscillator. Note the changes from the layout published last month.



Layout and dimensions for keys.

remainder of the board. Alternatively, the paint may be brushed on if a tin of paint is already on hand.

To tune the organ to some other instrument, vary the 1K preset pot until the top note corresponds to the 2nd F above middle C. If the instrument is tuned to the standard A = 440Hz scale, the frequency of the F note should be approximately 698Hz.

Assuming reasonably accurate resistors the rest of the notes should be very close to their correct frequencies, certainly close enough for most purposes. If one or two notes are too far off, proceed as follows. If a note is too high, the appropriate resistor should be decreased in value. If it is to low, the resistor should be increased. The easiest way to decrease resistance is to connect another resistor in parallel with the one which is too high. As a rough rule of thumb, a resistor 10 times larger than the existing one will give a combined value which has been reduced by 10%. A shunt value 20 times larger will give a reduction of 5%.

To increase resistance, connet additional resistance in series. In this case a resistor having 1/10 the value of the existing one will increase the total by 10%; a value of 1/20 by 5%.

Incidentally, the stability of these additional resistors is not so important as that of the main ones. Since they will account for only a small percentage of the total resistance, typical variations due to age etc. will have only a negligible effect on the total value.

Because the resistor network for the keyboard is a simple series arrangement, changing the value of any one resistor will have some effect on other notes. In order to keep this effect to a minimum, always make the first correction to the highest note which requires it, then progress down the scale as required.

We mounted the various sections of the organ on a Masonite baseboard measuring 11½" x 6". Four rubber feet were fitted to the underside to prevent assembly screws etc. from scratching any surface on which it might be placed.

One problem which we encountered is that of dirty contacts. With such a simple keyboard as this, dirt invariably gets onto the contacts, thus preventing reliable contact being made. We noticed this particularly after a humid day.

One way to overcome this problem is to use one of the commercially available switch and contact cleaners. We found that a short spray of this onto the contacts kep them operating for a long time. If, after a few weeks you find that one or more of the notes is not reliable, all you need do is give it a quick spray. Naturally, your environment would have a lot to do with this, as would the period between uses.

If you put the organ into a case, ther would be a lot less chance of dust contaminating the contacts. If you do not use a box, some sort of cover over the "keyboard" when not in use would help.

We used two spring terminals for the output connections. To attach these terminals it will be necessary to counterbore the mounting hole, on the underside of the baseboard, to about ¼" to accommodate the terminal nuts. As an alternative, the terminals could be replaced by a length of shielded lead, possibly terminated in a plug appropriate to the amplifier input with which it is to be used.

We have not used a battery switch on our organ. This was mainly for economy - it is a simple matter to remove the battery connector to cut off the supply. We have left plenty of room between the battery and the oscillator board to include a switch if desired. Simply break either battery lead and place the switch in series.

The battery bracket can be made from a strip of the same tinplate as used to make the keys. Simply bend the tinplate around the battery, with a flange either side for fastening to the baseboard. It may be held with either small woodscrews, for a wooden baseboard, or small nuts and bolts for a Masonite one.

PARTS LIST

RESISTORS (all 5% 1/2 watt)

1 x 18 ohms	1 x 680 ohms
1 x 56 ohms	2 x 1K
3 x 100 ohms	1 x 1.5K
4 x 220 ohms	1 x 1.8K
2 x 270 ohms	1 x 2.2K
1 x 330 ohms	1 x 2.7K
2 x 390 ohms	2 x 3.3K
3 x 470 ohms	1 x 22K
1 x 560 ohms	3 x 56K
1 x 1.2 Megohms	

CAPACITORS (Low voltage, plastic or ceramic).

1 x 0.1uF	2 x .01uF
1 x .039uF	1 x .047uF

1 piece of "No Copper" Veroboard, type no. 4505, 3¾" x 9 7/8".
1 piece of ordinary Veroboard, 0.1" pitch approx. 1¾" x 1½".
1 piece of Masonite or similar, approx. 11½" x 6½"
approx. 60 "Zephyr" matrix board eyelets, type no. 280. 2 spring terminals.
1 9 volt battery and connector
4 rubber feet
scraps of tinplate, nuts, bolts etc.

ORGAN TREMULANT VIBRATO

THIS ARTICLE DISCUSSES TREMULANT AND VIBRATO EFFECTS, AS APPLIED TO ELECTRONIC ORGANS AND GOES ON TO EXPLAIN THE POTENTIAL ADVANTAGES OF THE LESS WELL-KNOWN PHASE MODULATION. A FULLY DEVELOPED PHASE MODULATION UNIT IS PRESENTED.

To add variety and interest to sound, music makers commonly adopt the trick of modulating - or varying in periodic fashion - either the frequency or loudness of the sound, generally at the rate of a few cycles per second.

Singers achieve a modulating or pulsating effect by suitable muscular control over breath, vocal chords and mouth configuration. Tone, amplitude and frequency are all likely to be modulated and, depending on the vocal expertise of the singer, the result can be variously pleasing or displeasing, interesting or painfully monotonous. It has been suggested that, in extreme cases, singers may modulate frequency or pitch by about 6 per cent, which is equivalent to plus and minus one complete semitone.

Violinists commonly vary or modulate the frequency of notes by rocking the fingers of the left hand, which are being used to "stop" the strings. "Pop" guitarists achieve a similar result by means of a lever, which allows the tension of the strings to be varied in a periodic manner. As a general rule, the frequency variation applied to instruments is less than the extreme frigure quoted for the human voice, and 3 per cent or a half-semitone is accepted as a realistic limit figure.

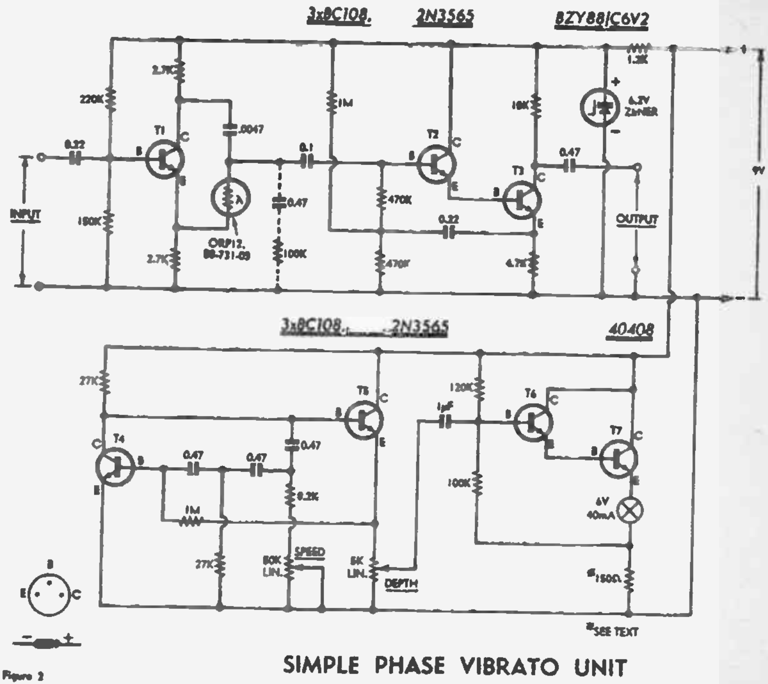
The technique of periodically varying the frequency of a musical note is commonly described as "vibrato".

Vibrato rates usually lie within the range 5 to 8 cycles per second (i.e. 5 to 8 Hertz) with 7Hz probably the most commonly used figure. Rates outside this range are sometimes used but only for rather special effects

In acoustic organs, it is not easy to modulate the frequency produced by the pipes and an alternative approach has long been used of varying the pressure of air to the pipes in a periodic fashion, usually by means of small, mechanically driven supplementary bellows. Varying the air pressure has a minor effect on frequency but its main effect is to vary the loudness of the sound produced.

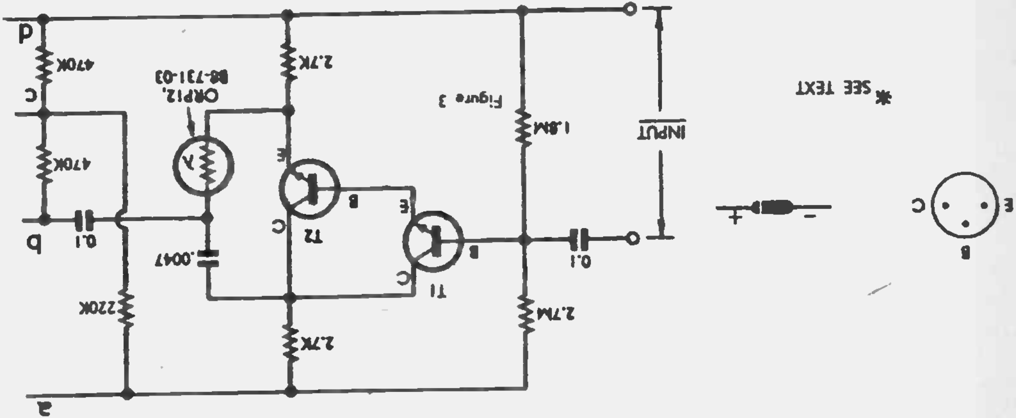
The technique of modulating the loudness of notes is commonly referred to as "tremulant". As with vibrato rate, tremulant rate usually falls in the range 5-8Hz with 7Hz again the most usual figure.

A circuit diagram for the simple phase vibrato unit is shown at right. Main sections of the circuit are the lamp oscillator/amplifier section and signal modulating section containing the LDR.

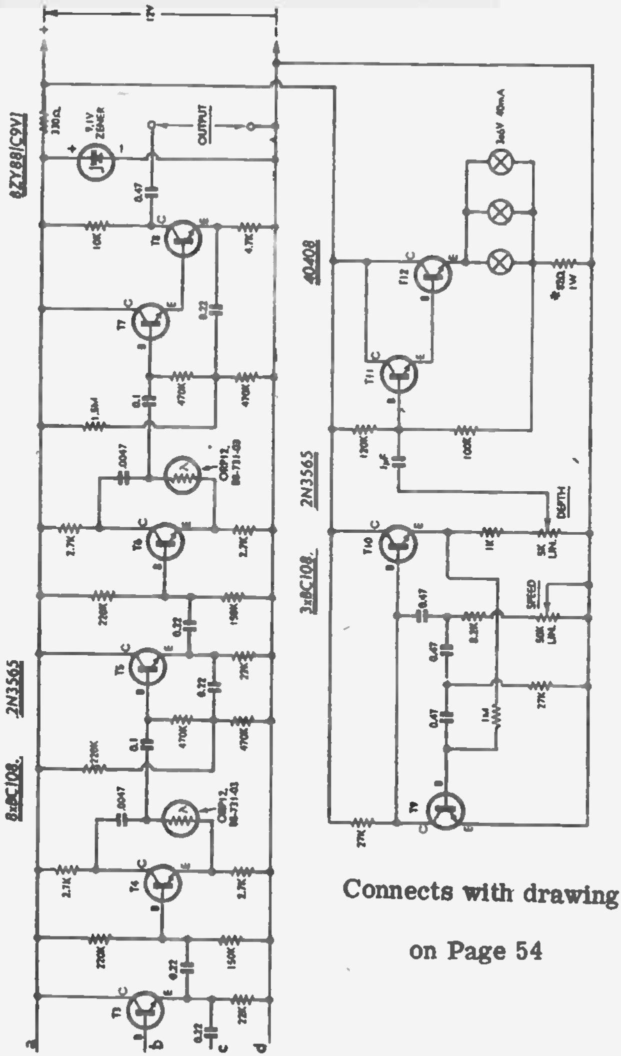


SIMPLE PHASE VIBRATO UNIT

PHASE VIBRATO UNIT



Connects with drawing on Page 55



Connects with drawing

on Page 54

Vibrato or tremulant effects are exploited heavily in modern electronic (or electric) organs, partly from necessity, partly by reason of opportunity. Let's explain this:

When a complex chord is played on an acoustic organ, the sound as heard is, the resultant of many independent pipes, speaking simultaneously. For practical reasons, organ pipes are never perfectly in tune and the constantly changing phase relationships, along with the spatial separation of the pipes, emphasis the complexity of the chord and adds to its aural interest.

Electronic organs, for the most part, are designed within limits of cost and complexity which dictate the use of a much smaller number of sound source generators. In many cases the generators are highly interdependent, which makes for ease and accuracy of tuning, but the phase relationships are fixed and locked rather than random and variable. And, of course, the sounds are usually propagated through common loudspeaker channels.

Compared with a pipe instrument, the chord structures from an electronic organ tend to lack complexity and aural interest; or, to borrow a phrase, the unembellished sound tends to be "bland". .

This being so, the designers of electronic organs have been forced to adopt all kinds of measures to render the sound less bland - or more complex. They - and the players - have come to rely on vibrato and/or tremulant effects to a much greater extent then is traditional or necessary in pipe instruments. So much for necessity!

As far as opportunity is concerned, designers of electronic organs can draw upon a very wide range of techniques, either to simulate traditional organ effects or to achieve others which have no obvious counterpart in acoustic instruments.

A tremulant effect can be achieved by causing the gain or amplification of a stange in the amplifier to vary at the desired tremulant rate. Or, again, by introducing into the signal path a resistive element whose value varies in a cyclic manner. In current practice, this commonly takes the form of an LDR (Light Dependent Resistor) illuminated by a small, rapidly flashing lamp.

More significent, however, is the fact that an electronic organ designer can outdo his acoustic counterpart by providing full vibrato. This can be achieved by suitably varying the operating conditions of valve (or to a lesser extent transistor) oscillators or by varying the reactive quantities in the frequency determining circuits.

Should it be considered desirable, there is no special problem about providing part tremulant, part vibrato, making either peculiar to individual organ voices, varying modulation depth and speed or even providing different rates simultaneously for special effects.

It has been stated fairly frequently that there is not a great deal of difference aurally between tremulant and vibrato and this, is in fact, a conclusion one might easily reach, based on observation of single tones or even simple chords. In both cases, the main aural impression is a rapid fluctuation in the loudness of the sound.

With tremulant this is exactly what one would expect to hear, because the signal is, in fact, being modulated, in terms of loudness or amplitude.

With vibrato or frequency modulation, the effect has to be explained on a completely different basis.

Imagine that a pure, unvarying tone is being propagated into a listening area. What the listener hears is a mixture of direct sound from the loudspeaker, with that reflected from walls, ceiling, structures etc. The subjective loudness at the listening position is a random resultant of how the multi-reflections add or cancel in terms of amplitude and phase but, provided everything remains fixed, so will the level of sound as heard.

But now change the frequency of the tone by a small amount. Immediately the pattern of reflections and standing waves in the listening area changes, as also will the amplitude and phase resultants at the listener's ears. An inevitable effect will be a change in the apparent loudness of the signal, in addition to the change in frequency.

In fact, there is good reason to believe that the average listener may react more to the loudness change than to the frequency change!

Evidence of this is provided by the fairly common complaint that the vibrato (not tremulant) in certain electronic organs can scarcely be heard when the player is listening to his own music only through earphones. On the loudspeaker, the vibrato may be quite heavy; on phones it is scarcely discernible.

Almost certainly, the explanation is that there is no pattern of reflections and standing waves between the transducer and the listener's ears and no cyclic variation in this pattern as the frequency is modulated. The listener is left only with pure frequency modulation, to which - surprisingly - he is less sensitive than he might otherwise have believed.

At first glance, this seems to confirm the idea that, in a typical listening environment, there is little to choose between tremulant and vibrato; that vibrato only has a clear impact on the listener when it is transformed by the acoustic environment into a loudness modulation.

For single or simple tones this is not far from the truth but the contention breaks down completely when the reasoning is applied to complete chord structures and harmonically rich sounds, as commonly produced by electronic organs.

Assume a given chord structure, complete with harmonics, being played at fixed level in a fixed listening situation. The listener hears each component of the total sound, not just as a direct radiation from the loudspeaker but as a resultant of the direct sound and its multiple reflections. And, ovviously enough, the pattern of reflections will be different for every single component of the total sound.

Now imagine that the frequency of all the oscillators responsible for the particular chord structure is shifted slightly. Immediately the standing wave patterns in the listening area will be affected - but no uniformly. At the listening position some components of the sound will increase in level by various amounts, some will decrease by various amounts and others will not be affected much at all.

A simple shift may not produce a very obvious aural effect but a constantly changing pattern certainly does and here the vital difference emerges, between pure tremulant and pure vibrato, as applied to an electronic organ.

With pure tremulant, simple tones or complex tones alike appear merely to be modulated in loudness.

With pure vibrato, all tones exhibit the expected tremulant effect, but the complex structures tend also to separate out as each individual frequency component behaves in its own individual fashion.

Additionally, in a live environment, new instantaneous frequency components are heard simultaneously with the echo of what has preceded them in time.

In short, frequency modulation tends to impart an individual characteristic tones and overtones which, aesthetically, have been locked more than they should have been. Thus, vibrato adds an order of complexity and a potential aural interest to chord structures, which is not provided by tremulant.

Is vibrato or frequency modulation, then, the obvious and best choice for electronic organs? Not necessarily, and for a number of reasons.

The first is the very practical reason that it cannot easily be achieved when the tone generators are toothed wheels (as in most Hammond organs), reeds (as in early electronic Wurlitzers) or some forms of transistor oscillator, which are more inclined to "stay put" than be swung in frequency.

A second reason is that, if the oscillators which supply the manuals also supply the pedal clavier by frequency division, vibrato intended for the melody and accompaniment is imposed on the petal notes. This is usually reckoned to be musically undesirable because the vibrato frequency becomes too great a percentage of the music tone frequency.

A third reason is voiced by some but not admitted by others. This is that the percentage of frequency swing which might be regarded as musically desirable or appropriate for 8ft tones in the melody region, may be excessive for notes in both the lower and upper region.

In the lower register, heavy vibrato is judged by some to be unnecessary and unpleasant, for reasons which differ only in degree from those which apply to the pedal notes.

In the case of high notes and overtones, the interplay of numerous rather wildly deviating frequency components can produce resultants which impart an unpleasant quality to the final sound. Nasal, unnatural, off-key are some of the adjectives used to descrive this situation.

Out of these circumstances, of both choice and necessity, has emerged another technique - that of modulating the phase of the organ tones at some convenient point within the amplifier chain. Advancing and retarding the phase of the signals in a periodic fashion has a rather similar effect to varying the frequency.

The method has the advantage that it can be applied to individual preamplifier chains so that in more complex instruments, it can operate on the manuals but not the pedals, on individual manuals or even on particular voices. Such flexibility involves additional circuitry but this is not such a problem in these days of transistors and ICs.

An additional and interesting characteristic of most practical phase modulation circuits is that they tend to produce maxdumum phase shift in a frequency region, which can be selected, with a diminishing amount of phase shift towards the lower and higher frequencies.

This means that the aural effects expected of a vibrato system can be achieved for frequencies in the melody region of the keyboard, with a diminishing amount of modulation in the lower register (particularly the pedals) and for the higher order overtones.

Phase modulation therefore has a strong appeal to the designer who finds tremulant inadequate but who is unable or disinclined to apply vibrato to the basic tone generators. Or again, to the designer who wishes to provide vibrato selectively for individual manuals and voices, which are derived from a common oscillator/divider system.

In general terms, the amount of pure vibrato effect which can be obtained by not-unduly-complex phase shifting systems is less than can be obtained by direct frequency modulation of those basic generators which are amenable to the treatment.

It is therefore substantially true to say that frequency modulation of the basic generators can produce a more dramatic vibrato more easily, than can phase modulation within the amplifier chain. For the reasons already explained, however, frequency modulation can produce a greater array of undesirable effects an phase modulation may therefore be described as more modest but "safer" in its application to a wide variety of voices.

Phase modulation can be achieved in a number of ways, some of them mechanical, as by rotating capacitors or capacitive switches, rotating transformer elements or rotating loudspeakers. Electrical systems include sequential sampling of the 180-degree displaced signals across a push-pull signal source and/or deriving signal from a R/C phase-shifting network of which one or more elements are varying in a cyclic fashion.

In applying these systems a major problem is to achieve phase modulation without simultaneously modulating the amplitude. It is all too easy to end up with a system which is more akin to tremulant than vibrato.

Two practical phase modulating vibrato circuits follow. The first is a relatively simple apprach, providing only a limited frequency shift effect. The second circuit, while more complex, is capable of providing a substantial phase shift, resulting in an entirely acceptable vibrato characteristic.

The basic phase modulating circuitry is shown, in simplified form, in figure 1. In essence, it consists of a single stage phase splitter providing two signal sources, one at the collector and the other at the emitter, out of phase by 180 degrees. The modulated output is taken from the junction of a capacitor and resistor which are connected in series from collector to emitter.

The circuit differs from some earlier configuration in that the output from a network is strung between two active signal sources, not between one such source and earth.

Looking back from the output terminal there are two impedances, one the frequency conscious reactance of C and the other a variable resistance R, connected to the independent out-of-phase signal sources. Operation of the circuit may be understood by considering the limiting values of R infinity and zero.

If R has an infinitely large value, signal output will come from the collector, with a phase angle of - 180 degrees with respect to the input, via the reactance of C. On the other hand, if R is zero, the output will come from the emitter with a phase angle of zero. The relatively high reactance of C compared with the zero resistance will have negligible shunting effect,

For intermediate values of R, the output phase will swing between the limits of - 180 and zero degrees. There will be no reaction on signal amplitude - provided the junction point is not significantly loaded by the external following circuitry.

To explain in detail this variable phase constant amplitude characteristic would require resort ot a vector, or more correctly, a phasor diagram. Those who understand such diagrams should have no special difficulty in working out what goes on. For those without as much background it may be sufficient to suggest a vector resultant of constant length, rotating from zero phase angle to - 180 degrees and back again, with its tip describing a half-circle.

Ideally, the value of R should vary between zero and infinity for maximum modulation but, in practice, we must be content with a swing between two finite limits. This practical limitation modifies the modulator's basic characteristics, fortunately to advantage in certain respects.

A resistance variation between zero and ininity would theoretically provide a full 180 degrees modulation over the whole frequency range but, in practice, the finite resistance variation compared with the reactance of C results in modulation of less than 180 degrees peaking at one particular frequency. At frequencies above and below this point the phase deviation tapers off gradually.

While the reduction of maximum phase deviation from 180 degrees is a definite disadvantage, the progressive reduction of modualtion for frequencies towards either end of the musical range may be regarded as a desirable effect, for reasons explained earlier in this article.

It is also interesting to note that while constant amplitude is a natural characteristic of the circuit, an amplitude or tremulant effect can be obtained by deliberately introducing some loading between the signal output point and earth, as shown dotted. This extra loading forms a voltage divider with resistor R and variations in the latter then affect the amplitude of signal reaching the following stage.

For application as an organ vibrato system the value of R must be made to vary electronically at a rate of between 5 and 8 Hertz. To accomplish this, a light dependent resistor or LDR is used and its resistance is varied by a source of modulated light.

Ideally, the variation of resistance should be equal in both directions about a centre value, resulting in symmetrical modulation of phase angle about a reference point. Unfortunately, the resistance characteristic of the LDR-lamp combination is by no means linear; it is, in fact, rather exponential in shape.

While in theory it would be possible to obtain a linerar resistance characteristic by driving the LDR's light source from an asymmetrical waveform, the generation of such a complementary waveform would be a rather difficult task. Added to this is the variation in resistance characteristic between LDRs of the same type, requiring individual compensation.

Accepting that the LDR's associated incandescent lamp must be driven with a symmetrical waveform, we are consequently restricted to a relatively small resistance swing to obtain reasonable modulation linearity. In the circuits presented here the modulating waveform is sinoidal with the lamp(s) driven from a class A amplifier consisting of two transistor stages.

Essentially, the amplifier is a compounded emitter follower, comprising transistors T6 and T7, as shown in the circuit of figure 2. Sufficient current amplification to drive up to three 6V 40mA lamps is available, with the addition of negative current feedback to stabilise their operating points and raise the input impedance of the driving stage.

The quiescent operating point of the lamp should be such as to produce an LDR resistance of about 18K. This can be adjusted by means of the 150 ohm resistor in series with the lamp. Depending upon the exact mechanical set-up and proximity of LDR and lamp, the above condition will be met with approximately 1.2 volts across the lamp.

It is important that the LDR resistance be set between 15 and 20K in the quiescent condition, otherwise there will be a displacement in the frequency of maximum phase deviation, which should be approximately around centre keyboard.

Modulating signal for the unity-voltage-gain lamp amplifier is generated by a phase-shift oscillator, T4. As readers will no doubt recall this oscillator depends upon the inherent phase shift in a resistance/capacitance network. By adjusting one of the resistance arms, in this case the 8.2K resistor and 50K potentiometer, the oscillator frequency can be varied.

An emitter follower (T5) functioning as an impedance buffer then delivers the signal to the lamp amplifier via a 1uF plastic capacitor. Note that this capacitor must be a plastic or other non-polarised type; electrolytics are not suitable. With the 5K potentiometer in the emitter of T5 the signal level can be adjusted, thus providing a modulation depth or deviation control.

For simplicity we used potentiometers for both the speed and depth control in the prototype oscillator/amplifier combination. However, some may wish to provide pre-set speed and depth controls, usually in the form of rocker switches, on an organ console and these may be arranged in the present circuit if desired.

Turning now to the actual modulating stage (T1) in the circuit of figure 2, it will be seen that there is very little difference between it and the basic stage of figure 1. The collector and emitter load resistors are relatively small in value, providing sources of constant voltage. An ORP12 or LDR03 or RYP25 or LDR in conjunction with a .0047uF capacitor, may be used in the series network between collector and emitter.

Output from the modulator is coupled via a 0.1uF capacitor to a compounded stage having an input impedance of several megohms. As already mentioned, high impedance is vital, as any loading of the modulator will introduce amplitude modulation, i.e. tremulant.

This following stage, comprising transistors T2 and T3, employs "boot-strapping" of bias resistors to obtain the very high input impedance. At the same time it provides a voltage gain of two, off-setting a small loss in the modulator.

A maximum phase swing of about 120 degrees is available from the single modulating stage, as presented. This may be considered quite sufficient for a classical organ situation, particularly as it is possible to augment the phase modulation with a judicious level of amplitude modulation.

This may be introduced by loading the output of the modulator stage with a capacitively coupled resistor. A suggested value of 100K in series with a 0.47uF capacitor is shown in dotted form. For increased amplitude modulation the resistor may be reduced, but not to less than 47K. Incidentally, the circuit can be made to provide amplitude modulation only by simply removing the .0047uF capacitor.

The power required for the circuit of figure 2 is an approximate 9 volts at about 18mA. The most convenient source would probably be a small mains supply or rectifier and filter powered from a heater line. Alternatively, a small commercial "Battery Saver" supply may be considered.

With the signal frequency circuits operating from an effective 6 volts the signal level which the unit can cope with is strictly limited. It should be inserted at an early point in the organ amplifier system, where the maximum signal level is not likely to exceed 0.5V RMS.

For those requiring a more dramatic vibrato effect, as for popular organ music, the more elaborate circuit of figure 3 is a natural and consequent progression from the simpler phase modulating system just described. In essence, the enlarged vibrato system is simply three cascaded modulating stages with the phase shift in each being additive, and giving a maximum of 360 degrees at a centre frequency.

We did look at the effect which could be had with two stages, giving about 240 degrees, but ultimately decided in favour of the three-stage system as being more appropriate to the requirement.

The comments relating to the basic modulation circuitry apply equally to this circuit but there is one additional point which must be mentioned. It is important to have the quiescent resistance of each LDR about the same, otherwise the frequencies of maximum phase shift will not coincide. In practice, it will be satisfactory if they are within the limits of 15 and 20K

In this more developed vibrato system, the input impedance to the unit has been increased to over 1 megohm to accommodate all likely applications, including its possible use with valve circuitry. This requires the use of an extra transistor in the first modulator, connected Darlington fashion. If an input impedance of 70K is considered sufficient, however, the simpler input circuit may be retained.

The first modulator, be it high or medium input impedance, is followed by a bootstrapped emitter follower (T3) again to avoid loading the modulator output. These three transistors, T1, T2 and T3 thus form the first modulating unit. This is followed by a-second unit comprising transistors T4 and T5.

A signal which has been modulated by up to 240 degrees is available at the output of T4. Constructors desiring to use only two stages of modulation, could follow T4 with the compounded output buffer used in the previous circuit of figure 2. This same output buffer, providing two times voltage gain, is also used in the present circuit (T7 and T8) following the third and last modulating stage, T6

In a sense, the various units used in the three-stage vibrato can be considered as building blocks. By appropriate interconnection, various requirements of phase deviation and input impedance can be satisfied.

One other contribution to the flexibility of the unit has been made by increasing the specified supply voltage to 12. Allowing for decoupling, this permits the use of a 9V zener and supply for the signal frequency circuits and allows them to cope with up to 1V RMS at the input, without overloading the output stage. In these circumstances, the unit can be introduced into any convenient portion of the organ preamplifier circuitry, where the maximum anticipated signal level lies below 1V RMS.

The decoupling just mentioned is an inbuilt precaution against modulation of the signal circuit supply by the varying lamp current, with consequent and possibly deleterious amplitude effects. Its importance would vary with the impedance of the supply actually provided.

The current required was measured at 54 milliamps.

Whatever the form of the supply, and despite the in-built zener diodes, reasonable filtering must be provided. Since the vibrato units operate ahead of the main power amplifer, undue ripple on their supply will inevitably produce a hum problem.

In terms of their construction, both vibrato units are similar and relatively simple. To a large degree the style of construction and layout can be altered to suit the requirements of particular applications. For the prototype units, we found that a length of Veroboard served very well as a construction basis, enabling a workmanlike appearance to be achieved.

As will be apparent from the accompanying photographs, the wiring method is to simply push the component leads through the board and solder them, point to point, on the underside. For prototype construction this is a very easy and quick method and most home-builders will probably rate it as perfectly satisfactory for the finished article as well.

Alternative methods might include the use of the special metal pins for use with Veroboard, miniature resistor panel or the more conventional tag-panel wiring. The primary requirements are correct electrical connections and well soldered joints.

The placement of components in the prototype units follows as far as possible the circuit progression: There should be little difficulty in recognising the various sections.

The layout sequency for the small unit begins at one end with the largest lamp-driving transistor followed by an associated current amplifying transistor. Adjacent to this pair are the two transistors and associated components in the phase shift oscillator. Separated from this section by the zener diode are the single modulating and output stages.

The LDR and lamp are assembled in a $1\frac{1}{4}$" length of tubing which can be cut from any convenient material, provided that it excludes extraneous light. We found that the aluminium cans from some discarded electrolytics made very satisfactory housing. Incidentally, their internal diameter was 9/16" comfortably enclosing the LDR. The lamp was inserted in the other end through a rubber grommet and pushed against the LDR encapsulation. The complete assemblies were then simply wired to the component board.

Although there are more components in the larger vibrato unit and the layout is a little more compact, the various sections can be seen quite clearly. Again the lamps' driving amplifier and phase shift oscillator are at one end of the board. Next to these sections, separated by a zener diode, is the output stage and then the last LDR/lamp complement.

Between the middle and last LDR/lamp complements is the second impedance buffer stage comprising transistors T5 and T6. Next there is the middle can followed by the first buffer stage (T3 and T4) with the first modulating stage (T1 and T2) at the end.

The LDR/lamp cans in this unit were not wired down on the board. Instead, the rigidity provided by two bus bars supplying the lamps together with the LDR pigtails was sufficient to securely retain the cans in position.

Connection of the various function controls and input/output signal leads were made directly to the back of the component board. If desired, the heavier shielded leads could be tied down with light gauge tinned copper wire or form-tying plastic.

A GUITAR PREAMP WITH FULLY SOLID-STATE VIBRATO

HERE IS A FULLY SOLID STATE PREAMPLIFIER AND VIBRATO UNIT FOR USE WITH ELECTRIC GUITARS. COMPLETELY SELF CONTAINED AND POWERED FROM A SMALL 9V BATTERY, THE UNIT EMPLOYS A NOVEL VIBRATO SYSTEM WHICH DOES NOT INCLUDE A FRAGILE AND INEFFICIENT FLASHING LAMP.

Apart from the normal tone controls found on most hi-fi amplifiers, the usual amplifier system used for practice will not embody any of the "frills", including vibrato and "fuzz", often found on a guitar amplifier.

This has prompted us to design a preamplifier which still performs the same basic job of providing the required extra gain, but which at the same time provides a vibrato facility. Not only can such a preamplifier be used with a practice amplifier but it may also be used with a suitable power amplifier to make up a guitar amplifier system complete with a vibrato facility.

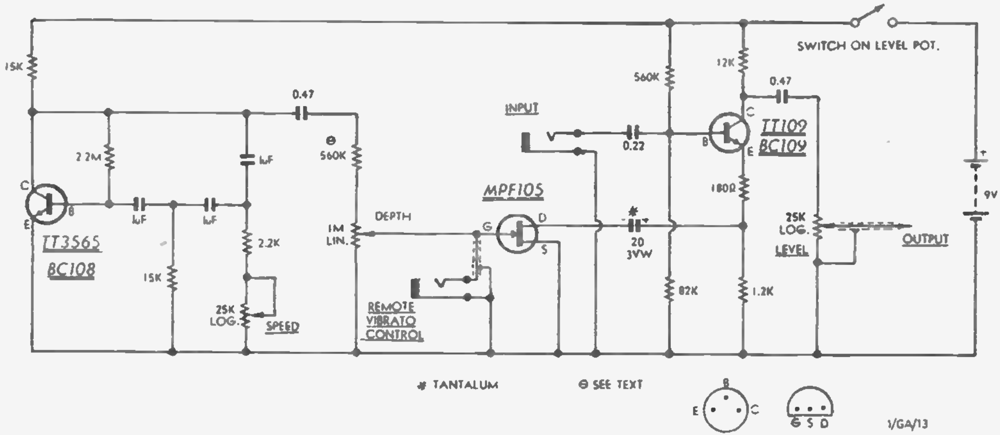
Those already adequately equipped with guitar amplifiers and speaker systems may also be interested in using the design as an extra outboard vibrato unit, which could prove useful where one amplifier is being used with two guitars. In such a situation it may not always be desired to apply the vibrato effect to both guitars simultaneously. Using one, or possibly two of the outboard units, vibrato could be applied to each guitar individually.

In any vibrato system, it is essential that the low frequency vibrato modulation signal does not itself appear at the amplifier output, to be fed to the loudspeaker. If it does, it will cause the loudspeaker to "pump" in and out - sometimes resulting in an audible distortion, and in any case increasing the risk of speaker damage when a high power amplifier is used.

Up to the present time, the vibrato systems which we have described have all used the fairly popular optical modulating system, consisting of a light dependent resistive signal divider and a flashing lamp, to overcome the problem of speaker pumping. While this system proved very effective it was, nevertheless, rather clumsy in both a physical and electrical sense. Furthermore, it was not very compact, this latter shortcoming being accentuated when transistorised circuitry was used.

However, the most significent disadvantage of the optical vibrato system is the power consumed by the incandescent lamp which is necessary in a transistorised version. In a valve circuit where higher voltages are involved it is possible to use a neon lamp which perhaps paradoxically, consumes much less power.

It was this last factor, more than any other, which prompted us to consider an alternate system of modulation. The method which we have used takes advantage of the variable drain-source resistance characteristic of a junction field effect transistor to modulate the guitar signal. As such, it is completely electronic and has no flashing lights or other clumsy hardware.



GUITAR PREAMPLIFIER AND VIBRATO

For low signal levels of either polarity applied between the drain and source a FET behaves as a linear resistor which can be varied by a bias applied to its gate. If a low-frequency alternating bias is applied to the gate then the resulting slow alternation in drain-source resistance can be used to modulate a signal in a similar manner to the previously used light dependent resistor.

Compared with the light dependent resistor, the FET resistance is very low, of the order of a few hundred ohms. However, the low resistance is a distinct advantage in transistorised circuitry, making possible a more convenient means of modulation. Reference to the circuit will show how modulation is accomplished.

The vibrato circuit consists essentially of the same amplifier configuration used in the preamp circuit described last month. However, it will be seen that there are two resistors in series in the emitter circuit of the BC109 transistor.

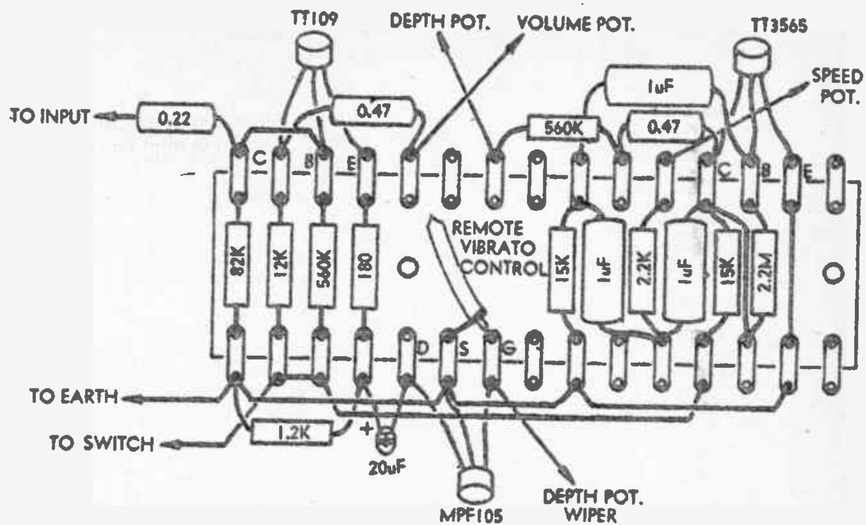
The FET is connected across the lower 1.2K resistor via a 20uF tantalum electrolytic capacitor. It is by this means that the signal is modulated with the vibrato frequency. So far as the signal is concerned, degenerative feedback in the preamp emitter is provided by the 180 ohm resistor in series with the parallel combination of the 1.2K resistor and the FET resistance, about 580 ohms.

Because the preamp stage gain is an inverse function of the degenerative emitter feedback, it can be varied by the FET resistance in the emitter circuit. Hence, by applying an alternating bias to the FET's gate the gain of the transistor preamplifier will be modulated at the vibrato frequency.

The vibrato modulating frequency, which is roughly sinusoidal, is generated by a one-transistor phase shift oscillator. The oscillator is essentially a voltage amplifier with a phase shifting network from output back to input. The network effectvely reverses the phase of the output to make it in phase with the input, a condition which is sufficient to ensure continuous oscillation. By making one of the resistors in the phase-shift network adjustable in value, the oscillator frequency may be varied. This allows adjustment of vibrato "speed".

The 1uF capacitors used in the phase shift network are specified as plastic types with a 10 per cent tolerance. Ceramic or electolytic capacitors should not be used, otherwise the oscillator may not function as a result of excessive capacitor losses.

The signal at the collector of the oscillator, which is about 2V peak to peak, is applied to the gate of the FET, through an attenuating network consisting of a 560K resistor in series with a 1M linear potentiometer. The pot thus functions as a vibrato "depth" control, with maximum depth corresponding to the highest modulating signal amplitude applied to the FET gate. When the pot is turned right back, zero signal is applied to the gate, and the vibrato is effectively "off".



A wiring diagram of the miniature resistor panel is shown above; this will assist with the placement of most of the components used in the preamp.

The maximum modulating voltage applied to the gate should be some what less than the pinch-off voltage of the FET, otherwise a prominent "plop" will be heard in the absence of signal.

Because the pinch-off voltage varies from device to device, facility for setting the maximum gate voltage has been provided. This takes the form of a resistor in series with the depth control with a value set at 560K in the prototype.

With the depth control in the maximum position, the resistor value should be adjusted to a point just above that which produces the "plopping" should in the speaker. With most FETs the resistor can be reduced in value from the 560K shown, but in the few cases of devices with low pinch-off voltage it will be found necessary to increase the resistor value.

If readers desire to switch the vibrato in and out by means of a remote foot switch, it should be arranged to short the FETs gate to its source, as shown in the circuit diagram. The switch could alternatively be arranged to disable the oscillator, but it was found with the prototype that the oscillator required a few seconds to re-start.

While we did not include the remote facility as a permanent facility in the prototype, we did make a temporary connection. A non-shorting type jack-socket should be used for connection of the remote shorting switch. Also, it is necessary to use shielded cable to connect to the switch, so as to avoid injecting hum at the gate of the FET, and subsequent hum modulation of the signal. The inner conductor of the shielded cable should connect to the gate, and the shield to the source.

The gain of the preamplifier stage, without vibrato modulation, is about 20 times with a maximum undistorted output of about 2V RMS or 5.6V peak to peak. This means, then, that the signal should not be more than approximately 280mV peak to peak.

We actually measured the preamp output when fed from a typical Hawaiian steel guitar and found it to be no more than 200mV peak to peak with maximum volume control setting. Thus there would appear to be a reasonable overload margin of some 80mV peak to peak.

In the prototype, we included an output level control on the panel, together with the speed and depth controls. However, the level control could take the form of a pre-set tab mounting pot as with the preamp described last month, or it could be discarded in the interests of simplicity, and the level control on the guitar used instead.

If the level control is omitted an alternate on/off switch may have to be provided. However, with a current drain of only 0.7mA for the complete unit it could almost be left on continuously with a probable battery life of not much less than normal shelf-life.

For this version of the preamp we again used a small aluminium

We used a tantalum electrolytic capacitor to couple the FET into the preamp emitter, because it has extremely low current leakage. Significant leakage in the coupling capacitor would modulate the DC conditions of the preamplifier stage to cause "pumping" and severe distortion with larger input signals. Furthermore, tantalum electrolytic capacitors apparently have no tendency toward drying out and subsequent loss of capacitance, as occasionally occurs with other electro lytic capacitors.

The length of miniature tag board was supported by two 1 1/8" countersunk screws, using nuts to space the board. It should be spaced from the bottom of the box just sufficiently to clear the battery, otherwise the components projecting from the panel will foul with the potentiometers in the lid.

The input connection to the preamp is made via a standard jack socket while a captive length of shielded cable fitted with a standard jack plug is used for the output. The shielded cable is passed through a grommet and knotted to prevent undue strain on the electrical terminations.

The lid which, completes the unit, is held in place by two small countersunk screws in the sides of the box. The lettering on the top panel of the box was applied to the prototype using black indian ink and a drawing pen and lettering stencil. The lettering was then lightly sprayed with clear lacquer to render it permanent.

PARTS LIST

HARDWARE

1 Aluminium box, 5"x 2¼" x 2¼"
1 9V battery
1 15-lug length of miniature tag board
1 Standard jack socket and plug

POTENTIOMETERS

1 25K log, with switch
1 25K log.
1 1M lin.

TRANSISTORS

1 BC109 etc.
1 BC108 etc.
1 MPF105 Motorola FET

CAPACITORS

1 20uF 3VW electrolytic
3 1uF low voltage plastic
2 0.47 low voltage plastic
1 0.22 low voltage plastic

RESISTORS

2 560K, 1 x 82K, 1 x 12K, 2 x 15K, 1 x 2.2K,
1 x 2K, 1 x 180 ohms.

utility box, which measured 5" x 2¼" x 2¼".. The three controls were mounted in the U-shaped lid with sufficient lead length to facilitate easy lid removal for battery replacement. The battery was held to the bottom of the box with a clamp made from some scrap aluminium.

Most of the components were mounted on a 15-lug strip of miniature tag board. The board is divided roughly into two halves, one end for the oscillator components and the other for the preamp, with the FET about in the middle.

THE MUSICOLOUR II

THE MUSICOLOUR IS A DEVICE FOR OBTAINING A MUSICAL KALEIDOSCOPE FROM YOUR FAVOURITE RECORDS. IT AUDIO-MODULATES COLOURED 240-VOLT LAMPS TO GIVE VIBRANT AND COLOURFUL DISPLAYS. IT CAN BE DRIVEN BY A STEREO AMPLIFIER OR FROM TAPE RECORDERS, GUITAR AND ORGAN AMPLIFIERS.

So called "psychedelic" light shows have become very popular in clubs, discotheques, dance halls and the home. Musicolours and units similar have been incorporated in commercially built stereograms and jukeboxes.
The Musicolour is a light modulating device of the type often referred to in American electronics magazines as "Colour Organs" or "Light Organs".

Basically, the Musicolour splits an incoming audio signal into three frequency bands, hereafter referred to as the high, medium and low channels. The signal derived from each channel is used to control a Triac - a semiconductor device which varies the AC power fed to lamps. The power supplied to the lamps then becomes proportional to the amplitude of the derived signal.

The variety and colour of the light displays available is almost limitless and indeed, the effects are very hard to describe in print. Some ideas for building suitable displays are given at the end of this article.

The average drive signal required by the first Musicolour was about 1 watt, at an impedance level of about 8 ohms. This meant that the unit could not be merely connected in parallel with one of the loudspeakers in a domestic stereo system, for example, because for correct operation the volume delivered by the loudspeakers was unreasonably loud. Thus, a separate, low-powered amplifier was required to drive the Musicolour so that the signal delivered to it became independent of the amplitude delivered to the loudspeakers.

The revised Musicolour does not require a separate amplifier - it can be driven from the headphone socket provided on most stereo amplifiers and tape recorders. It is about 100 times more sensitive than the original circuit.

Another major advantage of the new circuit is that the separation between the three channels is much greater and there is virtually no interaction between the various controls.

Since the circuit uses Triacs, the full mains voltage is present in many sections and indeed, depending on how the power point from which it is operated is wired, the whole circuit board will have 240 volts applied to it - no section of the circuit is earthed. This means that the incoming audio signal must be completely isolated - for safety's sake - from the Musicolour circuit.

The method of isolation used is a small mains transformer, 240V to 12.6V working backwards. That is, the audio is applied to the 12.6 volt winding and is stepped up in the 240V winding. The prime reason for using a mains transformer instead of a better quality audio type, is that, apart from providing a suitable turns ratio at a modest cost, the transformers specified have very high insulation between windings. They conform to the same specifications laid down for transformers used in battery chargers, model train controllers and similar "appliance" applications.

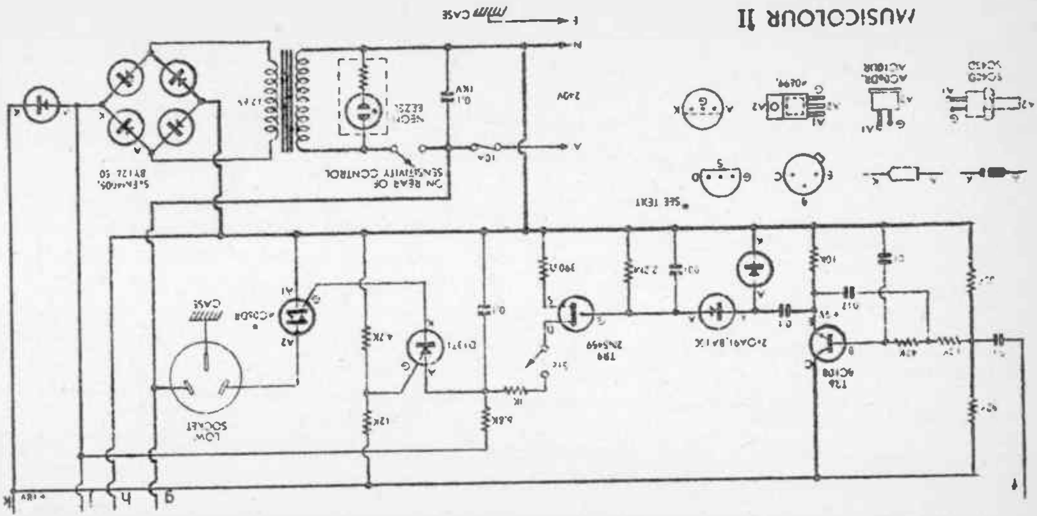
After passing through the step-up transformer, the audio signal is fed, via the sensitivity control potentiometer, to a voltage amplifier stage, Tr1 which has a gain of 10 times. In turn, the amplified signal passes to an emitter-follower stage which has, as its emiter load, three 2K potentiometers connected in parallel. These three potentiometers control the signal level applied to the high, medium and low channels.

Frequency splitting for the three channels is accomplished by a combination of active high and low pass filters. The filters consist of a simple second order filter (i.e. double RC network) combined with an emitter-follower stage to provide positive feedback. The positive feedback greatly increases the attenuation factor of the filter. The theory of operation is beyond the scope of this article but it will suffice to know that each filter has a gain of approximately 0.9 over its bandpass and an attenuation slope approaching 18dB/octave beyond the turnover point.

A high pass filter, using Tr3, is ued for the high channel - it passes only those signals above 2KHz. Similarly, a low pass filter, using Tr6, is used for the low channel - it passes only those signals below 300Hz. The medium channel uses a high pass filter, using Tr4, followed by a low pass filter, Tr5. The bandpass of the combined filter is from 300Hz to 2KHz. While the frequency allocation for each of the channels is certainly not an even division of the audible spectrum, it is a logical one in terms of music and from the viewpoint of obtaining the best visual display from most music signals.

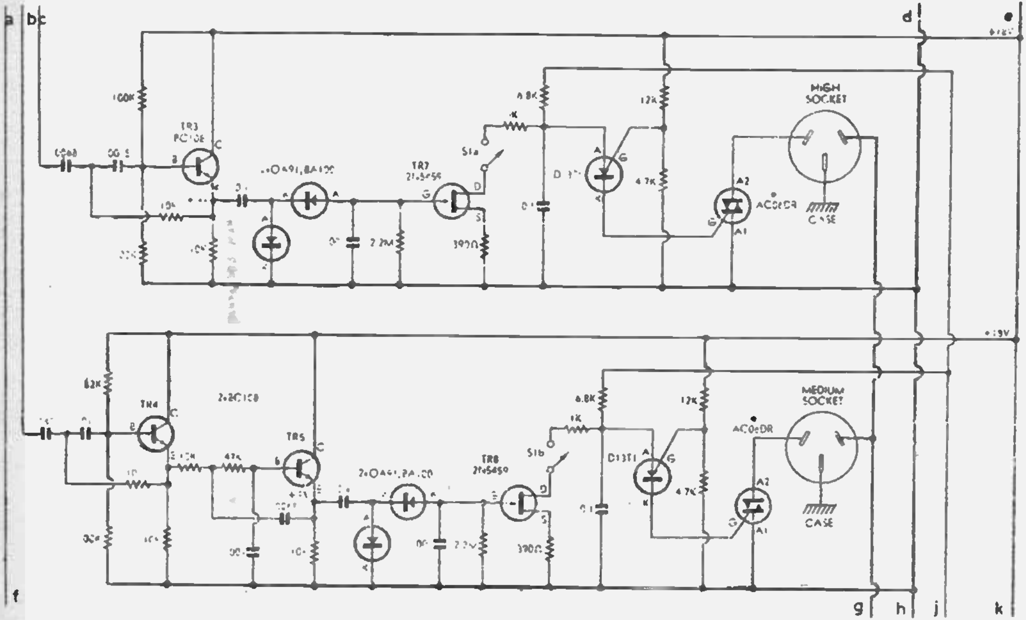
As the signals from each of three channels are eventually used to control a Triac in each channel, it is now appropriate to briefly describe the operation of Triacs.

In essence, the Triac is a bi-directional switch which after being triggered into conduction, stays "on" until the supply voltage decreases to zero or reverses in polarity, when it turns off and can be switched on again. Used with AC, a Triac can be triggered into conduction at any point on either half cycle by a low voltage signal of either polar-

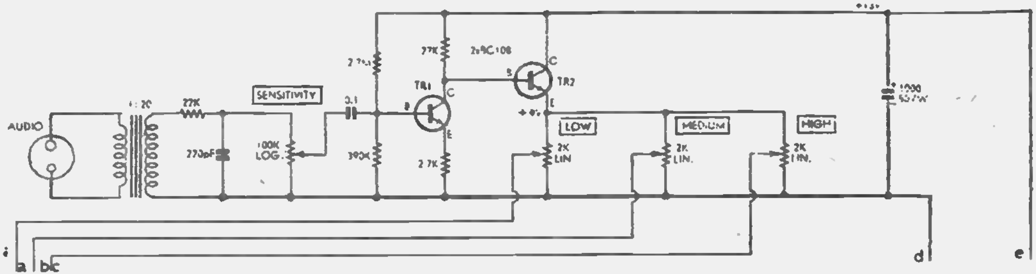


Connects with drawing on Page 75

Connects with drawing on Page 76



Connects with drawing on Page 74



Connects with drawing on Page 75

ty applied between the gate electrode and terminal 1 (anode 1). Note that, since the Triac is a bi-directional device it has no anode or cathode as such. The two end terminals are normally referred to as "anode 1" and "anode 2" or "terminal 1" and "terminal 2".

As the Triac is a switching device which is enither fully conducting or "off", the only means by which it can be used to obtain variable control of power is to use it as a very rapid switch which closes for variable periods of time during each half-cycle of the AC voltage wave-form - by adjusting the instant during the half-cycle when it triggers into conduction.

The method of Triac triggering we have used is called "phase control" - it involves applying to the gate electrode a sharp pulse of current whose phase, relative to the AC waveform, can be varied. The pulses are generated by a programmable unijunction transistor or PUT, the GE D13T1.

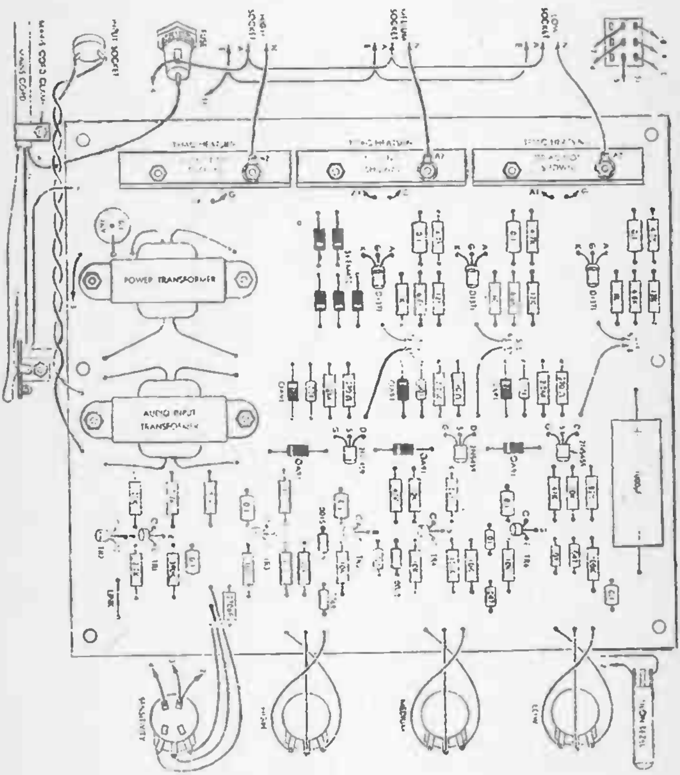
Briefly, the mode of operation of the three PUT pulse generators (one per channel) is as follows:

A 0.1uF capacitor, one side of which is connected to the anode of the PUT is charged via a 6.8K resistor from the 12.6V, full-wave rectified but unsmoothed, DC supply rail. When the capacitor voltage rises above the PUT anode gate voltage which is provided by a voltage divider consisting of 12K and a 4.7K resistors, the capacitor is discharged by the PUT and a sharp ulse is delivered to the gate of the Triac, switching it into conduction. In this way, the PUT delivers one pulse per AC half-cycle to the Triac. If the pulse is early in each AC half-cycle, the Triac delivers a large amount of power to the lamps, and vice versa.

We control the phase of the PUT-generated pulses by utilising a field-effect transistor, FET. See Tr7, Tr8 and Tr9. The FETs are used as variable resistors which shunt each 0.1uF capacitor and thus control its rate of charge. The drain-source resistance of the FET, in conjunction with 1K and 390-ohm resistors forms the variable element. The two resistors protect each FET against excessive dissipation and alter its gate voltage/resistance characteristic.
If a large negative voltage (i.e. approx. -10V) is applied between gate and source of the FET, its drain - source resistance becomes very large, many megohms in fact, so that the 0.1uF can charge rapidly and hence a gate pulse is delivered to the Triac early in each half-cycle. Conversely, if the voltage applied between gate and source is almost zero, its drain source resistance is of the order of several hundred ohms so that the 0.1uF is prevented from charging.

Thus, by altering the gate voltage of the FET, we can directly control the phase of the pulse delivered by the PUT to the Triac. This is done by a half-wave "voltage doubler" rectifier which develops a negative voltage in direct proportion to the audio signal delivered by each channel. Each rectifier consists of two low-power diodes and a filter network consisting of a .001uF capacitor and 2.2M resistor shunting it.

Before going any further with the details, we can sum up the operation of the Musicolour as follows : (1) Audio signal from an amplifier or other source is "stepped up" in the isolating transformer; (2) The signal is further amplified and then fed to the frequency splitting filters for the three channels ; (3) The signal from each of the three channels is rectified to develop a negative DC voltage which is used to control the drain-source resistance of FETs; (4) The FETs in turn,



The complete wiring diagram of the Musicolour showing all circuit connections. The Standby switch may be omitted by substituting three wire links on the board control the PUT pulse generators which trigger the Triacs into conduction. If the Triacs are triggered into conduction early in each AC half-cycle the lamps will be bright, and vice versa.

Returning now to the operation of the PUT stages, the reader will notice, if he refers to the circuit diagram, that the voltage divider for the anode-gate of the PUT is connected to the filtered DC voltage rail which supplies active filter circuitry, instead of the unsmoothed DC rail, as might be expected. By using the smooth DC rail for the anode-gate supply, there is no interaction between channels caused by spikes from one Triac being triggered.

Returning now to the operation of the PUT stages, the reader will notice, if he refers to the circuit diagram, that the voltage dividers for the anode-gate of each PUT are connected to the smoothed DC supply rail, instead of the unsmoothed DC rail, as might be expected. By using the smoothed DC rail, spurious triggering of the PUTs and Triacs, due to spikes impressed on the AC and unsmoothed DC supplies, is eliminated.

Referring again to the circuit diagram, readers may notice the three-pole switch, S1, associated with the drain circuit of each FET. This is a standby switch - in the closed position the lights are modulated by the incoming audio signals; in the open position, the FET resistance is removed from the PUT circuits and the lamps run at full brilliance. This feature is handy at parties and dances where the room would be plunged into darkness if the programme stops. More about this later on in the article.

In conventional Triac trigger circuitry, the pulse generated by the PUT is normally coupled to the gate or the Triac via a pulse transformer. This is made necessary because the low voltage unsmoothed DC for the PUT circuit is usually derived, for reasons of economy, directly from the mains via a voltage divider and bridge rectifier.

Pulse transformers could have been used in the Musicolour but we would need three, and since commercially made pulse transformers are quite expensive, each would have to be handwound by the constructor. Frankly, we were not happy with the possibility of poorly wound pulse transformers causing damage to the circuitry and possible safety hazards. We decided to eliminate pulse transformers from the design. The cost savings referred to above are small, anyway. Hence in the new design the low voltage rails are supplied by a transformer of the same type as used for the input coupling capacitor. A bridge rectifier supplies the unsmoothed DC and an additional power diode and 1000uF/50VW electrolytic capacitor provides the smoothed DC of about 18 volts for the transistor circuitry.

As with other circuits using phase-controlled Triacs, the Musicolour tends to generate electrical interference which affects the broadcast and shortwave radio bands. This is reduced by the 0.1uF/1KV capacitor connected directly across the mains input.

Two features of the circuit diagram remain to be discussed, the power switch and the RC low pass filter circuit across the audio input coupling transformer. The power switch on the rear of the sensitivity control switches the power to the low-voltage power transformer only and not the main load current which can be as high as 10 amps, according to the lamp load. This means that the power switch need not have a heavy rating - there is no point in having a heavy duty power switch if the Triacs will do the job just as well. If no pwoer is supplied to the low-voltage circuitry then the Triacs remain in the non-conducting condition.

The RC filter circuit across the secondary of the audio input coupling transformer attenuates signals above about 10KHz so that the following circuitry does not react to "hash" which is mutually coupled from the power transformer to the audio transformer.

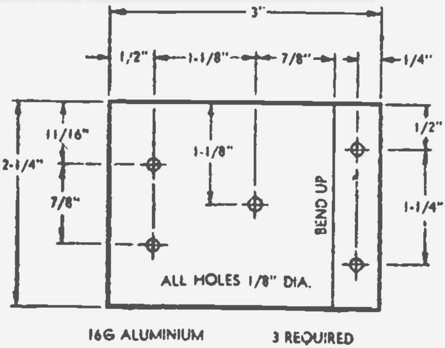
Having described the main features of the circuit diagram, a few paragraphs on "driving the Musicolour" are appropriate. Firstly, the input circuit, i.e. the transformer primary is arranged so that it presents a load of around 250 ohms or higher to the driving source. This means that it can be easily driven from the stereo headphone socket on most stereo amplifiers and tape recorders. An advantage of driving it this was is that, if the headphone plug is wired so that both channels are connected together, signals from both channels are automatically fed to the Musicolour.

Alternatively, the Musicolour can be driven directly from across the loudspeaker outlet of any amplifier capable of about 100 milliwatts or more - in fact, it can be driven directly from a pocket transistor radio. At the other end of the scale, if it is to be driven from a high power electronic organ or guitar amplifier, the audio signal from the speaker outlet should be coupled in via a 470-ohm resistor. Without this, the sensitivity control has to be used almost at "zero setting", where it is not very progressive and difficult to set easily.

As a third possibility, if the amplifier has a high-level output (say 500 mV) with an output impedance of 300 ohms or less, which is not affected by the amplifier volume control, this may be used to drive the Musicolour independently of the volume delivered to the loudspeakers. Failing this, the tape-recorder outlet of the amplifier could be provided with a low output impedance emitter-follower stage to perform the same function.

Each cahnnel can control up to 1,000 watts of incandescent lamp load. Note that fluorescent lamp loads must not be used. Note also that, although each channel can handle 1,000 watts, it is unlikely that the system will be called upon to handle more than a total of 2400 watts.

This limitation is not the Musicolour's but the power outlet from which it will be operated. Most mains outlets have a maximum current rating of 13 amps which automatically limits the total load to 3100 watts. This rating should not be exceeded. Anyone who has used a 2400W radiator will know how hot the wall socket can become.



Dimensions for the Triac heatsinks.

The prototype Musicolour was assembled on a chassis, U-shaped, measuring 9¾" x 7 7/8" x 3 1/8", and has a wrap-over cover. These chassis are available from suppliers at a reasonable price.

Three 3-pin mains sockets are mounted on the rear panel for the load of each channel. A two-pin socket is used for the audio input signal. Also accommodated on the rear panel is the fuseholder and the 3-pole standby switch, which is optional.

All of the circuit components, with the exception of the potentiometers, are mounted on a wiring board measuring 8" wide x 7" deep.

Each Triac has a separate headsink, which is mounted on the board. This is necessary to maintain the 1000W/channel rating because the headsinks are no longer shared and the additional "Standby" feature means that all lamps can be run at full power which means maximum dissipation in the Triacs. With the heatsinks specified, the 1000W/channel rating holds up to ambient temperatures of 40° C(100°F). If ambient temperatures rise above this, the load should be reduced, e.g. if the temperature rises to 110 degrees F, the load per channel should be reduced to 800 watts.

The board and heatsinks are compatible with "plastic pack" Triacs such as the 40669 from RCA or the SC141D or SC146D from General Electric. Note that for these devices the centre lug for A2 must be clipped short before they can be inserted into the board. The load connection to A2 is made directly from the output socket to a solder lug on each heatsink. The Motorola plastic pack Triac, MAC-11-6, may be used in a similar manner but its leads must be bent slightly to the narrower lead spacing of the other Triacs.

Each L-shaped heatsink is made from a piece of 16-gauge aluminium, 3" long x $2\frac{1}{4}$" wide, bent so that it has a $\frac{1}{2}$" flange. The heatsink should be drilled to take mounting clip or the single mounting screw of a plastic pack Triac. Heatsink dimensions are shown in an accompanying diagram.

The transformers should be spaced off the board by the thickness of two washers to avoid applying undue stress to the board when they are tightened down.

A suitable order of assembly will make construction easier. Proceed as follows: First, mount all the small components such as semiconductors, capacitors and so on. Do not bend the pigtails too close to the bodies of the components, otherwise they may fracture. Next, connect suitable lengths of wire to those points on the board that are wired to the potentiometers, standby switch and so on. If you wish to delete the standby switch, three links should be wired into the board instead.

When mounting the transformers and L-shaped heatsinks for the Triacs, ensure that the board has been drilled correctly so that the attaching screws are well clear of the copper pattern. All screws should be fitted so that the nuts are on the component side of the board. It is a good idea to install lockwashers on all components mounted on the board, for reliability.

Having checked the board carefully for wiring errors, compoents may be installed in the chassis. Rubber feet are secured with a screw and nut, the nut being held in the foot itself. Potentiometer shafts should be cut to suit the knobs used. The mains cord is passed through a grommeted hole in the rear of the chassis and anchored by a clamp underneath the fuseholder. The active wire goes to the fuseholder whole the neutral and earth wires are terminated on a three-way tagstrip, as shown in the wiring diagram. The earth wire connects to the "foot" terminal of the tagstrip, so that it is connected directly to chassis. When terminating the mains cord, the earth wire should be left with a loop of slack, as shown in the wiring diagram, so that if the cord is strained to the limit, the earth wire is the last to break.

Proper earthing of the chassis is the most essential step in the construction of the Musicolour. If it is not properly earthed a wiring mistake or component failure could make the chassis "live" and lethal!

Care is particularly necessary where the equipment is to be used in a public situation, in association with a public address system, musical instrument amplifiers, coloured spotlights, festoon lighting, etc. In these circumstances, the Musicolour unit itself should be checked by a qualified electrician, along with the lighting fixtures to be connected to it.

The holes in the chassis for the wires to the output sockets should be fitted with grommets to avoid chafing of the cable insulation. Note also that the wires to the output sockets should have the same current rating as the mains power cord, which itself should have a rating to suit a 2400W load.

Having installed all the mains wiring, the board may now be mounted. It is mounted using 1/8" screws and nuts, with two nuts used to space the board at least ¼" from the chassis. The connections from the board to the rest of the wiring may now be made. Note that neither side of the input wiring is connected to chassis, to avoid earth loop problems.

The pilot light is a neon bezel with a current limiting resistor incorporated. If a neon bezel without a limiting resistor is on hand a resistor of 150K should be connected in series with it. The neon bezel we used is moulded in red plastic.

The standby switch used was a miniature type three-pole, double throw unit, used as a single-throw switch.

Before the unit is connected to the display lamps and power applied, several checks should be made. First and most important, check that there is a direct connection between the earth pin of the mains plug and the chassis. Also, check that there is high resistance (e.g. several megohms) between the heatsinks of the Triacs and the neutral line of the mains. There should be high resistance between both sides of the mains and the chassis. These checks should be made with a multimeter.

In operation, it will be found that there is an optimum setting for sensitivity controls for the particular programme in use. If the signal level is too high, the lamps will tend to glow continuously. If the signal level is too low, the lamps will be extinguished for most of the time. A little experimenting with controls will produce the most varied display for each programme. It will also be noticed that the low channel is not as sensitive as the other two - this is quite normal and is mainly due to the characteristics of the input transformer.

Finally, if you are one of those unlucky readers whose Musicolour does not function, here are a few points on trouble-shooting. Remember though, that this procedure can be extremely hazardous because the full mains voltage is present in the circuit. If you do not have a multimeter and/or do not feel confident about your ability to cure a fault in the device, leave it strictly along. Take it, along with this article, to a competent serviceman.

Trouble-shooting can be made less hazardous if the mains active an neutral lines are exactly as shown in the circuit diagram, i.e. with the common line of the circuitry tied to the neutral line. This can be verified with a multimeter : Measure the voltage with multimeter switched to a high AC range, between the negative terminal of the 1000uF capacitor and chassis. If it is zero, okay. If it is 240 volts, swap the active and neutral leads on fuseholder and tagstrip. Remember though, that while most of the circuitry is now at chassis potential, the full mains voltage is applied to the neon pilot lamp, to the power switch terminals, to the fuseholder and if a load is connected, to the three Triac heatsinks.

Trouble-shooting should begin by ensuring that there are no wiring mistakes or incorrectly connected components. If not, start with the Triacs. First, with no audio signal applied, switch the standby switch to either of its positions. In one position, all lamps should be at full brilliance; in the other, extinguished. If a lamp is alight in both conditions short the gate of the appropriate Triac to its A1 terminal; if the lamp is still alight the Triac is faulty.

If the lamp stays extinguished in both positions of the Standby switch, the Triac or PUT may be faulty. The Triac may be tested by disconnecting the gate electrode and connecting a 1K resistor between A2 of the Triac and gate; the lamp should light. If not the Triac is faulty. If the lamp does light, the Triac is okay and the PUT stage is malfunctioning.

With the standby switch set for the Modulation mode, audio signal applied and sensitivity controls suitably adjusted, all lamps should be capable of being modulated. If not, the FETs can be checked for faults. If a lamp is partially alight with no signal applied, a cure may be effected by reducing the 390 ohm resistor. If the lamp will not light with signal applied, the FET may be short circuited. This can be checked by counnecting a 9V battery across the 2.2M resistor, negative to gate. The lamp should light; if not, the FET is faulty.

If the FET is okay and the lamp still does not light, the diodes in the voltage-doubler network should be checked, in situ, with an ohmmeter. They should be about 300 ohms in the forward direction and several megohms in the reverse direction. If they are leaky, replace. Each of the filter stages can be checked for correct operation by measuring the voltage at the emitter of each transistor. This should be within 1 volt of half the supply voltage (i.e. about 9V). DC voltages should be measured with respect to the negative terminal of the 1000uF capacitor. An operational transistor will have 0.6 volts drop from base to emitter. The preamplifier stage can be checked in a similar manner - i.e. 9V at emitter of Tr2.

These checks will not find every fault but they should at least identify the stage where a fault is occurring. Again, remember that mains voltage is applied to the circuit, so absolute care is essential when working on the unit.

As noted previously, the possibilities for displays are endless and are limited only by the reader's imagination. The ideas outlined here are only a guide and we will be interested to hear from readers who have thought up other ideas.

Most of the displays can be built around 25-watt or 40-watt coloured bulbs. These are available from Philips and other manufacturers in colours such as red, yellow, green and blue. It is interesting to note that the blue lamp will not appear nearly as bright as the red and yellow types. This is because the eye is less sensitive to the blue end of the spectrum, and tungsten filaments emit most of their light in the red and yellow region of the spectrum. This means that a blue filter shops most of the light. In general then, the power needed for the blue lamps will be two or three times that needed for red and yellow lamps.

The dispays should be arranged so that the lamps are not viewed directly. Looking directly at bright lights is tiring, to say the least. The basic materials needed to make interesting patterns are crinkled aluminium foil and frosted, fluted or patterned glass.

The simplest possible display is to mount three or more coloured lamps on a board and place them behind a stereo system cabinet so that they light the wall behind it. We suggest red for the low channel, green for the medium channel and blue for the high channel.

Another idea is to mount a number of lamps in a row along a board, place frosted glass in front of them and mount the whole display on top of the stereogram, organ or in the particular "interest point" in the room. Lights can be placed inside a cabinet, with crinkled aluminium foil behind them, and frosted glass in front. The result is a portable, completely enclosed display.

One of the most obvious tricks would be to modulate strings of "Christmas tree" lights. These could be strung around the house for the most novel Christmas decorations in your district.

For higher power displays, on stage for musical groups or in discotheques, coloured spotlights will be required. While you can buy your own spotlights and use gelatin filters to colour them to taste, coloured spotlights are marketed by Philips Electrical Limited and available from any trade houses which specialise in lighting. The lamps are in the Philips Comptalux range and are available in red, yellow, green and blue. The remarks we made above about the brightness of different colours stil hold for these lamps.

Many interesting displays could be obtained with these spot lamps aimed against walls, using beam splitting mirrors and rotating mirror balls. As we stated before, your imagination is the only limit.

PARTS LIST

1 Chassis with overall dimensions $9\frac{3}{4}$" x 7 7/8" x 3 1/8"
1 Metal cover with dimensions to suit chassis
1 Front panel3 Aluminium heatsinks (see text)
1 3-pole double-throw miniature switch
1 Cartridge fuseholder with 10-amp fuse
2 Power transformers
1 Polarised 2-pin socket
4 Knobs
1 Neon pilot lamp assembly
3 3-pin mains sockets

SEMICONDUCTORS

3 Triacs: ACO6DR, SC141D, 40669, MAC11-6 or similar
3 n-channel FETs : 2N5459
3 PUTs : D13T1
6 NPN silicon transistors: BC 108, 2N3565 or similar
5 Silicon power diodes : BY126/50, or similar
6 Signal diodes : BA100, 0A91 or similar

RESISTORS ($\frac{1}{2}$ or $\frac{1}{4}$ watt rating)

1 x 100K (log) potentiometer with switch
3 x 2K (Lin) potentiometers
1 x 2.7M, 3 x 2.2M, 1 x 390K, 4 x 100K, 2 x 82K,
2 x 47K, 1 x 27K, 1 x 22K, 3 x 12K, 8 x 10K, 3 x 6.8K,
3 x 4.7K, 1 x 2.7K, 3 x 1K, 3 x 390 ohms

CAPACITORS

1 x 1000uF/50VW electrolytic
1 0.1uF/1KV paper or mixed dielectric
8 0.1uF/50VW polyester
2 0.047uF polyester
2 0.01uF/VW polyester
2 .0068uF/50VW polyester or polystyrene
1 .0015uF/50VW polyester or polystyrene
4 .001uF/50VW polyester or polystyrene
1 270pF/100VW polystyrene or ceramic

AN ELECTRONIC STEAM WHISTLE

Our "Electronic Steam Whistle" was developed with the model railway enthusiast in mind. It might also serve as a sound effects simulator for amateur theatricals. Featuring single and dual horns, with background steam, it will add realism to your model setup. A separate "Noise" outlet is also provided to simulate steam alone.

We considered it essential that all the factors affecting the final sound should be capable of being varied over a wide range; preferably a wider range than appeared to be strictly necessary. This would permit the builder to vary any or all of these factors, as he thought fit, in order to achieve what he considered the best imitation of the particular whistle he had in mind.

If we consider the type of sound a steam whistle makes, we will be in a better position to decide how we can imitate it electronically. For a start, when the steam valve is opened, the sound intensity appears to rise to a constant level over a short period. In other words, it has a certain rise time. Also, during this rise time, the frequency of the whistle falls slightly.

Now consider the effect as the steam valve is closed. The steam in the whistle will take a short period to escape (as the pressure drops), thus the sound intensity fades over a short period. In other words the whistle has a certain decay time. Another feature peculiar to whistles is the background sound of the steam or air which operates them. As mentioned earlier, our device also features dual horns. This type of sound is usually produced by air horns. The two horns having a frequency ratio of about 1.5 to 1.

We must now design circuitry which will imitate the above features. To avoid confusion, we shall forget the dual horns for the present. To produce the single whistle, we require an oscillator. A simple phase shift oscillator would appear to be suitable. (Tr1 and associated components). There are no inductances required, and the frequency can be varied easily by changing resistance or capacitance.

To simulate the steam, a logical choice is a white noise generator. This is merely a reverse biased diode, arranged so that avalanche takes place, producing a substantial amount of noise. We use a reverse biased BC108 (D3) emitter to base junction in this mode.

Initially, we tried mixing the output from the noise generator directly with the oscillator output, but we were not happy about the level of noise available. It may have been sufficient in some cases, but it was a marginal condition. Accordingly, we fitted the noise amplifier (TR3) which provides more than sufficient noise for any likely requirement.

The signals from the oscillator and noise generator must be mixed in the correct ratio before they are fed to the main amplifier, and they must only reach the input to this amplifier when we wish to initaite the whistle. Also a means to vary the rise and decay times of the signal must be devised. All these functions are conveniently performed in the gated amplifier stage (TR4).

The whistle and noise signals are mixed at the input to the gated amplifier. The ratio of these signals is adjusted by resistance in series with the coupling capacitors from these stages.

The gated amplifier is normally gated off by a voltage divider network consisting of the 4.7K in the emitter circuit and the 47K to the positive rail. To gate the amplifier on, in the simplest case, it would be sufficient to shunt the emitter resistor with another resistor of suitable value. However, we can provide the required attack and decay times by adding suitable time constant circuits which control the rate at which the stage is gated on or off.

The time constant circuits consist of the 1K and 2.2K resistors and the two 10uF electrolytic capacitors in the emitter circuit of TR4. The total resistance of 3.2K is that required to gate the amplifier on, but the rate at which this can happen is determined by the rate at which the capacitors can be charged or discharged through their associated resistors. The 10uF capacitor across the 4.7K emitter resistor also functions as a conventional by-pass to maintain the amplifier gain.

When the test button is pressed, the lower leg of the voltage divider to TR4 emitter becomes approximately 1.9K ohms. However, before the stage is biased on, the emitter bypass capacitor must discharge through the emitter resistor in parallel with the gating resistance, and the other electrolytic must discharge via the 1K resistor.

Thus the output of the gated amplifier rises to a constant level over a short period, as the electrolytics discharge. We have thus introduced the necessary rise time. This rise time can be increased by increasing the capacitance of either electrolytic, or the gating resistance, and vice versa for a decrease. Note: If the gating resistance is increased too much, the stage may not bias on at all, thus it is preferable to vary only the capacitance if it is desired to change the rise time.

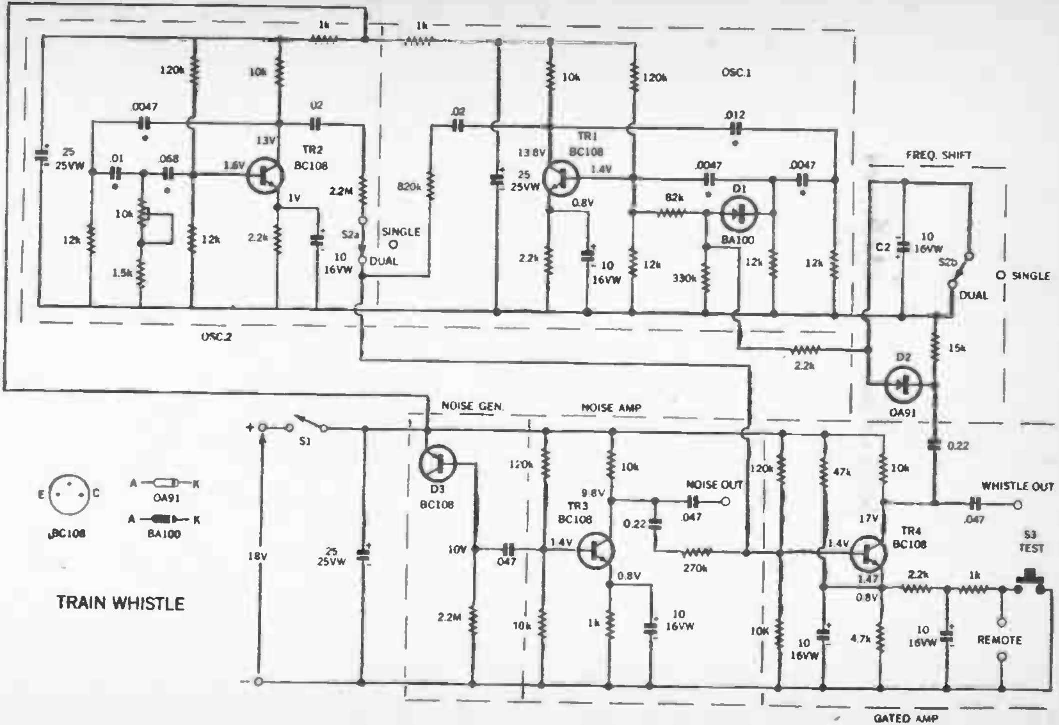
These electrolytics also provide the decay time. When the test button is released, the emitter bypass capacitor charges via the 47K, while the other electrolytic charges through the 2.2K before the gated amplifier is biased off. Thus the output from this amplifier fades over a short period. The rise and decay time adjustments interact, thus a compromise must be accepted, but more about this later.

Earlier we mentioned that a steam whistle appears to lower pitch during the rise time. The method we used to achieve this effect is as follows: The gated amplifier output signal is fed via a 0.22uF capacitor to a rectifier (D2). The DC thus produced is proportional to the output signal amplitude and, therefore, increases during the rise time. However, C2 (DC reservoir) must charge before a constant negative potential exists, thus the value of C2 controls the rate at which this potential increases.

In OSC1 a BA100 (D1) is connected in series with an 82K from TR1 base. Normally, D1 is biased on, thus the 0.0047 is shunted by 82K. Under these conditions the oscillator frequency is approximately 780Hz. If the DC control voltage is now applied to the anode of D1 via a 2.2k, this diode will be biased off. The effect of the 82K as a discharge path now becomes negligible, thus the overall phase shift of the network increases, and the frequency settles at approximately 750Hz.

OSC2 was added to provide the dual horn feature. In this mode, SW2b disables the DC control voltage, thus OSC1 operates at approximately 750Hz continuously. Meantime, SW2a switches the 2.2M in series with the OSC2 coupling capacitor to the gated amplifier base. Now when the test button is pressed, we hear the dual tone plus noise. The rise and decay effects still occur, but there is no frequency shift. The final frequency of our second oscillator was found to be approximately 490Hz for the most realistic dual sound.

The 10k preset pot in one leg of the OSC2 phase shift network was included to allow the frequency ratio between oscillators to be finely adjusted, but more about this when we discuss final testing.


OSC.1
OSC.2
TR1
BC108
TR2
BC108
FREQ. SHIFT
SINGLE
DUAL
S2a
S2b
C2
D1
BA100
D2
OA91
NOISE GEN.
NOISE AMP
D3
BC108
TR3
BC108
TR4
BC108
NOISE OUT
WHISTLE OUT
S1
S3
TEST
REMOTE
18V
GATED AMP
BC108
OA91
BA100
TRAIN WHISTLE

Construction should not present any serious problems for the average reader with some previous experience. While the finished product may look a little complicated, it must be remembered that it is a collection of separate sections. To make things easier, we have described the construction on a section by section basis, with details of how to test each section before preceeding to the next one.

We constructed the two oscillators on a piece of tag board, 17 pairs of terminals long. The noise generator, noise amplifier, frequency shift network and gated amplifier are built on a second length, 19 pairs long. We suggest the oscillator board be wired first but, if the dual horn facility is not required immediately, the OSC2 section may be omitted for the present. To allow the tag boards to fit into the small case we chose, it was necessary to limit the number of tags on each board. This made it necessary to place some components on the underside of each board.

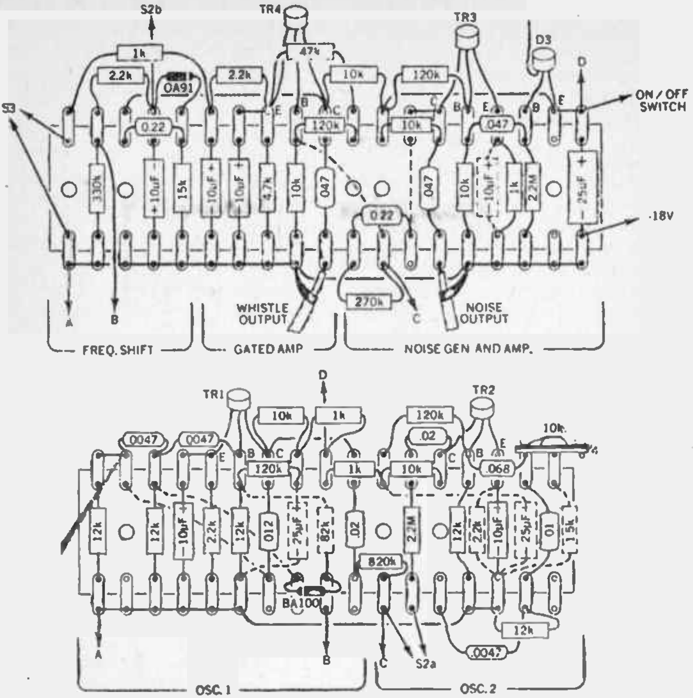
Two threaded pillars hold the tag boards apart, while another two secure the whole assembly in the case. We cut the heads off two 1/8 Whit. screws and used these as studs to sandwich the oscillator board between the pillars. Take care that no components are placed where the pillars are to be located.

If one follows the circuit and wiring diagrams, wiring OSC1 should be relatively straightforward. There are only two components under the board the 82K and the 25uF electrolytic. Where a capacitor or resistor is soldered to tags which have at least one other tag between them, the pigtails of that component should be insulated with spaghetti, to eliminate any possible shorts. Also insulate the pigtails of those components under the board. With this section wired, check it against the OSC1 section of the circuit, and the wiring diagram.

At this stage, OSC1 can, and should, be tested. Testing will call for an amplifier or, better still, an oscilloscope. As a last resort a high impedance headphone will suffice. Connect an 18V positive battery terminal to the free end of the 1K resistor and the negative terminal to the negative rail. Connect the 820K resistor to the amplifier, oscilloscope, or headphone, either of which should indicate a sine wave output. It must be realised however, that much of the realism is still missing. Do not make a final judgment until the noise (steam), attack and decay, and frequency shift have been added.

OSC2 occupies the rest of this board. The same procedure as that described for OSC1 applies to this section. The only extra component is the 10K preset pot, although the phase shift capacitor values have been changed to lower the frequency. One of these capacitors (.068uF) is situated under the board, along with the 25uF electrolytic, and a 1.5K and a 2.2K resistor.

To complete this board, interconnect the oscillator negative rails with a short length of wire as shown on the wiring diagram.



Wiring diagrams of the two boards. The function of each section is clearly shown.

OSC2 can be tested in the same manner as OSC1, but Rv1 should be varied during the test, to ensure that the pitch of OSC2 does vary noticeably with this adjustment.

To begin the second board we can wire the noise generator. This section has only three components, so it should not take long to complete it. The collector of the BC108 is not used, so simply bend this lead up out of the way.

The noise amplifier is logically situated next to the noise generator. The 10uF emitter bypass is the only component under the board. The positive rail for this section is via a wire to the emitter of the noise generator transistor. Once again the sections should be checked against the circuit before testing.

To test the noise generator and amplifier, connect the positive 18V terminal to the emitter of the noise generator transistor (used as a diode) and the negative terminal to the negative rail. Connect the free end of the .047uF capacitor (labelled Noise Output) to the input of your amplifier or oscilloscope. If there is no output, place a fairly large value capacitor in series with the amplifier input lead and check if there is output from the noise generator at the amplifier base.

The rest of this board accommodates the gated amplifier and frequency shift network. The 1K gating resistor mentioned earlier is the only component under the board. Make sure that the negative end of C1 (refer to frequency shift network) connects to the OA91 anode. A short length of wire from the gated amp collector to the 0.22uF capacitor (shown on the wiring diagram), and another length to connect the negative rail, completes this board. Check the complete section against the circuit and wiring diagram.

The gated amplifier can be tested using the amplifier or oscilloscope, as for earlier tests. Until the test button is pressed, there should be no output from the gated amplifier, although there will be signal at the base of TR4 continuously. The emitter voltage under these conditions should be approximately 1.47V. When the test button is pressed, there should be output, and the emitter voltage should be approximately 0.8V.

If the output is observed on a CRO, there should be some distortion obvious, one half cycle being more rounded than the other. This is deliberate, and adds a little to the realism of the sound. The background noise (steam) should also be obvious as "grass" superimposed on the waveform. If only an amplifier is available, simply check that the stage gates on and off correctly, and that the level of noise is adequate.

The rise and decay of the output signal is relatively fast, so if there is need to prove that these effects are present, remove the two electolytics from the emitter circuit of TR4. If the gate is now opened there should be a noticable difference in the attack and decay characteristic.

Four wires interconnect the two boards. Make these long enough to allow future access to the boards should it be necessary. If the two boards are fastened one above the other (with the oscillator board on the bottom) using the threaded pillars, the assembly should sit rigidly while the flying leads are soldered. Make these leads long enough to allow the boards to be removed from the case while the unit is operating, to allow for service or adjustment. Connection from the boards to the whistle and noise outlet sockets is by shielded cable.

While we have mounted the complete unit in a metal box and some readers may elect to do the same, this is not essential. Other readers may prefer to wire the boards directly into their existing model railway electrical system, with leads running directly to the various ancillary devices.

Those who make their own boxes should be particularly careful in positioning the mounting holes for the boards. Any serious error here may locate the terminals on the boards too close to the metal side of the box, with consequent risk of short-circuits.

The train whistle should now be complete, apart from final testing and adjustment. As in the case of the initial tests, an 18V battery is required, conveniently made up from two nine volt batteries as specified ind the parts list. Output from the whistle is approximately 500mV PP across 50K ohms, so the amplifier it is to drive need not be particularly sensitive. If the amplifier is very senstive, it would be advisable to place a 50K preset pot across the whistle output, and adjust the output to a level which does not overload the input stage of the amplifier. The noise output socket should also give approximately 500mV across a 50K ohm load.

Connect the whistle output to the input of the amplifier via a suitable lead. Switch to single horn, and press the test button. The whistle pitch should fall as this putton is pressed. The rise and decay effects may not be so noticeable, as they are much faster, but if they are eliminated the difference will be obvious. The whistle should be accompanied by background "steam".

If the whistle to steam ratio is not suitable, the 820K (whistle) or the 270K (noise) resistors connected to TR1 base can be changed. More resistance for less noise or whistle and vice versa. The oscillator pitch can be vaired by changing any of the capacitors in the OSC1 phase shift network. These are marked with asterisks. More capacitance lowers the pitch and vice versa. If the pitch or whistle to noise ratio is altered it must be appreciated that OSC2 pitch or output must also be altered.

If C2 is increased, the rate at which OSC1 changes pitch will be decreased. If the 2.2K connecting to C2 is increased, the amount by which it changes pitch will be decreased.

As mentioned earlier, the rise and decay time settings interact to some extent. If both the electrolytic capacitors in the TR4 emitter circuit are increased in value by the same amount, the rise and decay times will increase together, and the ratio between these times will remain relatively constant. Likewise if both capacitors are lowered in value the rise time and the decay times will decrease.

If the reader desires to vary the ratio between rise and decay times, the TR4 emitter bypass electrolytic will mainly affect decay time. More capacitance for more decay time and vice versa. Likewase the 10uR electrolytic connected to the junction of the 1K and the 2.2K will mainly affect the rise time.

If OSC2 has been included, switch to dual horn and short the remote socket so that the whistle is on constantly. The frequency of OSC2 can be varied broadly by changing the capacitors marked with asterisks, as for OSC1, but this should only be necessary if OSC1 pitch has been altered. For fine frequency adjustment use Rv1. The dual horn sound can now be adjusted to suit your impression of how it should sound by adjusting the frequency of OSC2. Normally, this frequency will be correct when there is minimum low frequency beat discernible in the output. The output of OSC2 can be varied by changing the 2.2M in series with the .022uF coupler, but again this should only be necessary if OSC1 output has been changed.

The unit may be triggered remotely by any suitable means, using the remote socket. It could be triggered by placing a reed switch between the rails, and a permanent magnet under the loco. This arrangement would require a suitable holding circuit to maintain the whistle for a short period after the pulse occurs.

PARTS LIST

1 Metal box 5¼" (135mm) x 3 1/8" (79mm) x 2 1/8" (54mm)
2 Belling Lee chassis sockets, L604/S
2 Belling Lee plugs L734/P
1 2 pin speaker socket and plug to match
1 Miniature toggle switch
1 Miniature toggle switch, 2 pole 2 way
1 Miniature push button switch, normally off.
1 length miniature tag board, 17 pairs of tags
1 length miniature tag board, 19 pairs of tags
2 9v batteries
2 Clip connectors to suit above.

RESISTORS (½ watt)		CAPACITORS
2 2.2M	1 15K	Electrolytics
1 820K	6 12K	3 25uF 25VW, 6 10uF 16VW
1 330K	6 10K	Miniature Polyesters (100VW)
1 150K	1 4.7K	2 0.22uF, 3.047uF, 2.022uF, 1 .01uF,
1 270K	4 2.2K	3 .0047uF
4 120K	1 1.5K	Polyesters (160VW)
1 82K	4 1K	1.068uF, 1 .012uF
1 47K		
1 10K preset miniature pot.		

SEMICONDUCTORS

5 BC108 transistors, or equivalent
1 BA100 silicon diode
1 OA91 diode

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