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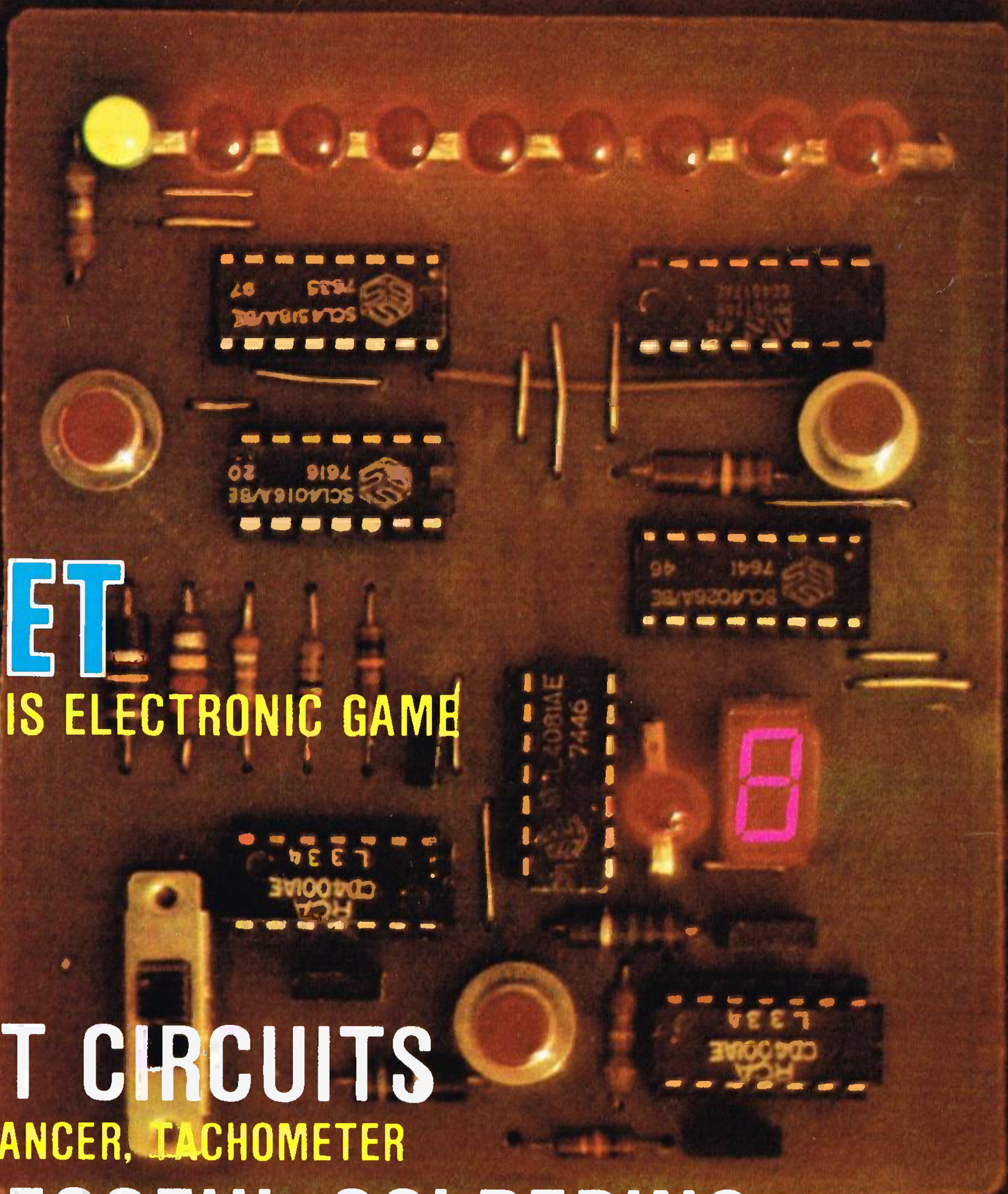

SKEET

BUILD THIS ELECTRONIC GAME

SHORT CIRCUITS

BASS ENHANCER, TACHOMETER

SUCCESSFUL SOLDERING

THE ART AND THE TOOLS

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Second Class Mail registration number 3955. Return postage guaranteed. Post Office returns to Unit 6, 25 Overlea Boulevard, Toronto, Ontario, M4H 1B1.

VOL. 1 NO. 7 AUGUST 1977

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Toronto, Ontario, M4H 1B1
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Published by: Electronics Today International (Canada) Ltd.

Printed by: Heritage Press, Mississauga, Ontario

News Stand Distribution: Gordon & Gotch, Toronto.

Subscription Rates: $12.00 per year, $20.00 for two years. Send to Subscription Dept., ETI Magazine, Unit 6, 25 Overlea Blvd., Toronto, Ontario, M4H 1B1.

International Editions

BRITAIN:
Electronics Today International
25-27 Oxford St., London W1R 1RF
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AUSTRALIA:
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NEWS DIGEST

I.E.E.E. CONFERENCE

The Institute of Electrical and Electronics Engineers are sponsoring the International Electrical, Electronics Conference and Exposition to be held at the Automotive Building, Exhibition Place, Toronto on Sept. 26, 27, 28.

With the theme 'The Path to New Dimensions Through Electronics' the exhibition will comprise over 300 exhibits covering the entire field of electrical and electronic interests. These range through avionics, communications, biomedical applications, filter design, custom LSI, and (of course) microprocessors and computers.

In addition to the exhibits there will be presented 134 technical papers covering the latest developments, including a special Electric Power Symposium conducted (?) by a group of visiting engineers from Poland (Yes, they have electricity too!)

For information regarding registration contact: International Electrical, Electronics Conference and Exposition, 1450 Don Mills Road, Don Mills, Ontario M3B 2X7. Phone (416) 445-7280.

The new Hewlett-Packard Model 1610A keyboard-controlled logic state analyzer is designed to efficiently test digital systems ranging from the simplest logic circuits to microprocessors and computers. A memory 32 bits wide and 64 bits deep, and as many as seven user-definable state conditions assure accuracy in locating test targets.

H.P. ANALYZERS

Anybody who has had to wrestle with an elusive hardware (or could it be software?) bug in a microprocessor or complex logic circuit will understand the value of powerful logic analyzers. One wonders if it might be an excess of neurotic bug chasing engineers at Hewlett Packard that have caused their recent interest in this field of test instrumentation. Along with the signature analyzer (see ETI-Canada June 77) two more products have recently appeared.

A new $2124 logic state analyzer, Model 1602A from Hewlett-Packard, weighs only 10 pounds and is no larger than an average-size briefcase. Yet, with F8 microprocessor control and a memory 16 bits wide and 64 words deep, it can capture 64 events — 63 words following, preceding, or surrounding the event designated as a trigger; it automatically self-tests every time it is turned on; and through its keyboard 1602A interacts with its user, pointing the way through every operation.

Specifying exactly what the state analyzer is to do is made easy by following the keyboard format: pushbuttons pick the desired logic polarity and which edge of the clock-pulse is to strobe data into memory. Other keys set the display to hexadecimal, decimal, octal or binary. Or a mixed format of binary and any other higher base can be used; that's particularly useful when displaying control lines in binary and data lines either in octal or hex. Next the trigger word is selected. Any following delay, 0 to 65,535 clock-intervals before the data-trace, may now also be specified; that's useful to work through loops, and get deep into branches. Any event that stops operations causes message codes to be displayed; definitions of these codes are inside the storage compartment on top of the analyzer.

The clip-on probe set — 16 data lines, a clock, a ground, and a qualifier — plugs into a standard edge connector on the probe. This connector is intended to mate with similar connectors at test points on new equipment, to make point-to-point probing fast and easy.

H.P. claim that significant savings are being realized by using completed 1602As to test 1602As in production.

In a more exotic vein is the new Model 1610A keyboard-controlled Logic State Analyzer a powerful, general purpose analyzer for design and troubleshooting of digital systems — from the most elementary to the most complex. Powerful triggering capability assures that the desired data is captured in digital systems ranging from basic logic circuits through microprocessor-based systems, computers, and computer systems.

With the 1610A keyboard, the user can trace events in as many as 32 channels at rates up to 10 MHz, selecting only the particular occurrences, coincidences, or logical sequences that are of interest, with results displayed in a well organized format on the CRT screen. A memory 32 bits wide and 64 bits deep can be commanded to capture everything that went on for 63 clock-periods after the trace point of interest, or for 63 periods before; or the trace point may be selected to be in the center of a trace. Not only can the instrument trace and display logic states, it can also measure absolute or relative time intervals between events, it can count events, it has a graph mode for an overview of all 64 words in memory, and it can produce documentation. Price is $11,210.

DIG THEM DIGITS

Newport Laboratories Inc. of Santa Ana, California has announced their new digital counter model 6130 which is panel mounted and may be programmed from the rear connector for frequency, frequency ratio, time interval, period, period average, totalize and stop watch. Sixteen selectable gate times are also available from the rear connector.

The full scale count is 99999 bright 13mm (½ inch) LED digits. Parallel BCD outputs are buffered and gated. Analog to digital isolation of 350 Volts by an optoisolator is standard. The case is a DIN standard with a panel cutout requirement of (92 x 45)mm.

The model 6130 may be set up to read directly in engineering units by using the optional predivider which provides pre-division by any integer from 2 through 12. The internal crystal may be replaced with an optional high stability time base or by user-supplied external time base. Prices start at $300 for 1-9.

For further information, contact the distrubutor: National Electrolab Ltd., 1536 Columbia Street, North Vancouver, B.C. V7J 1A4, Telephone: (604) 985-0511.

NEW NAKAMICHIS

Nakamichi's characteristic attention to faithful reproduction is evident in their latest line. This includes power amp 420 with THD at 1 kHz of less than 0.0008%, at any power level below clipping. These low distortion levels are achieved without heavy feedback, in fact even with feedback removed from the amp it exhibited a THD of only 0.1% at full power! The FTC power rating for the 420 is 50 watts per channel minimum "rms", both channels driven at 8 ohms, 5-20,000 Hz, with less than 0.02% THD. The amplifier, however, "clips" very cleanly, creating the impression of far greater power reserve. List price $545.00.

The 630 FM tuner employs a dual gate MOSFET front end, followed by an i.f. with ceramic filters, and an advanced phase locked loop multiplex demodulator. Type B Dolby is included. Some of the circuitry of this unit is shared with the 410 pre-amp, claimed to have outstanding specs. Harmonic distortion below 0.001% and phono noise 80 dB below a 1mV reference level (IHF-A). The pre-amp also features rumble and tonearm resonance filters, and fully defeatable tone controls. List prices: 630 at $995.00 and 410 at $510.00.

MORE FROM MITSUBISHI

Mitsubishi's line of audio equipment features some technical ideas that might tickle the listener's fancy . . . and pocketbook.

Integrated Logic Control Turntable DP-EC1 is the name for their frequency generator control direct drive record spinner. Very automatic control is accomplished through feather touch switches, assorted ICs and an optical sensor system which guides the stylus to the lead grooves — $839.95.

Separate power supplies is the approach taken by Mitsubishi to reduce crosstalk in their pre-amp (DA-P10 $399.95), 100W/ch (DA-A10 $549.95), and 150W/ch (DA-A15 $799.95) power amps. The power amps have a transient response which may be judged by the slew rate of 15 V/usec. Special heatsinks and transformers exposed to the air are used. (ie. the unit does not have a fancy wooden box)

Complementing the above units are a tuner, power meter, and speakers.

CLEVER KLIPSCH

From the Klipschorn people comes the MCM 1900 Loudspeaker System, comprised of three units to handle low, medium, and high frequency segments. With a combined weight of about 175 kg (that's over 380 lb!) and price tag of $3850 each, this system is definitely for heavy duty applications. It is intended for bi-amplification, ie. seperate low and high frequency amps, and comprises two 15" low frequency drivers, one 1.9" throat compression driver and five 2" x 5" horn piezoelectric tweeters. All the drivers are matched to the air by means of horns, although the bass horn is a folded one. The system is quite efficient as speakers go, being able to produce, for example, about 30 acoustic watts with 150 electrical watts into the bass driver, and this at less than 3% modulation distortion.

Nakamichi, Klipsch, and Mitsubishi products are available from: Superior Electronics Inc., Consumer Products Division, 1330 TransCanada Hwy. S., Montreal, Quebec, H9P 1H8 (514) 683-6331.

CATALOGUE LOG

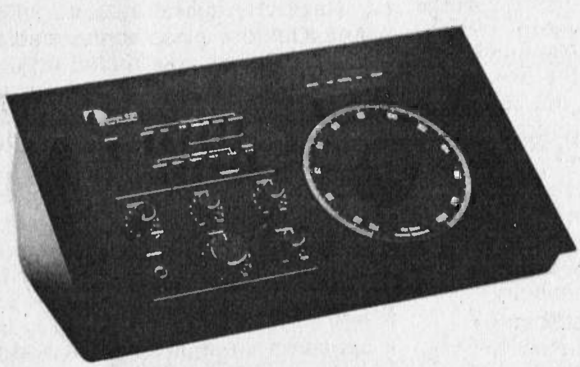
A couple of catalogues have come our way from John Fortin covering E & L Instruments' extensive line of solderless breadboarding aids, books and useful basic circuit modules for the experimenter.

They also have the 'MMD-1' Mini Micro Designer an 8080 A based educational and development system, including RAM, ROM, keyboard and power supply, available in kit form at $550 or assembled for $830.

Contact: John Fortin Company Limited, 660 King Edward St., Winnipeg, Manitoba, R3H 0P2, or branch office, 980 Alness St., Unit 32, Downsview, Ontario M3J 2S2.

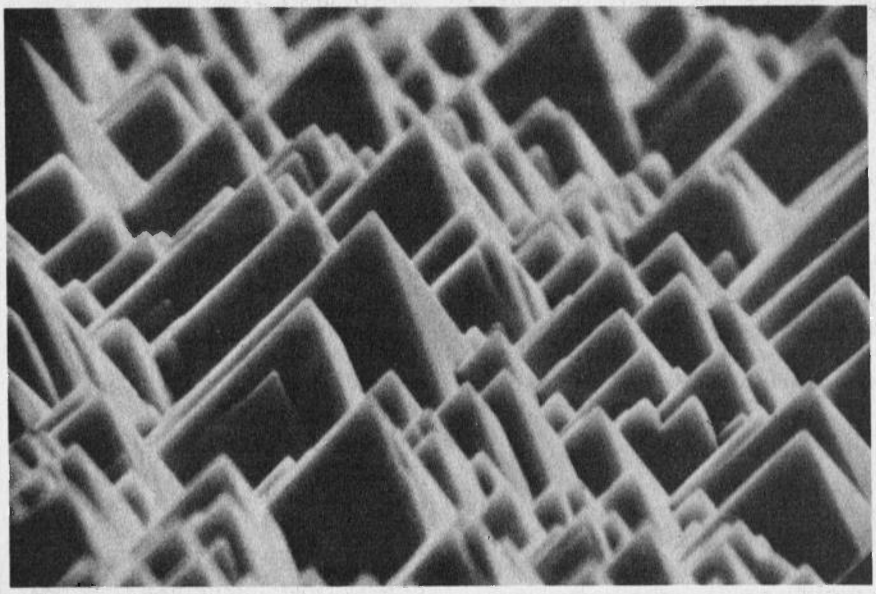
KILOWATT CONCRETE?

The resistance movement in the USSR has taken a new turn, as electroconductive concrete is being applied to high volume, large energy disipation resistors for electrical systems. Bet you never thought you'd be needing a shovel to build that big power amp project!



Unusual wedge shape of Nakamichi 630 FM tuner.

Mitsubishi DA-A15 power amplifier.



Textured surface of Motorola's solar cell greatly improves conversion efficiency by repeatedly intercepting reflected light rays that would otherwise escape from the wafer.

SOLAR MOTOROLA

"It will be another 5 to 10 years before solar energy will be cost-effective for all general-purpose applications around the home, but it can be harnessed now, most economically, for supplying moderate amounts of energy to a lot of remote locations." — this according to Dr. Arnold Lesk, Manager of Motorola's Solar R&D effort.

The products being introduced by Motorola include two Solar Modules: A 48-cell array and a 36-cell array. Both are composed of inter-connected 3-inch silicon wafers and are available in a variety of series-parallel interconnect options, offering different voltage/current output combinations. Both are intended for energizing remote, unattended equipment such as microwave relays, navigational aids and forestry equipment, and a host of other systems. Energy storage, when required, is provided by batteries. For such applications, these solar arrays offer service-free operation through continuous recharging during sunlight hours.

Motorola's entry into the solar cell field is a result of three years of research during which ribbon fabrication technology as well as wafer technology were investigated. Present product is made with high-technology 3-inch wafers, packed into high-density arrays by means of staggered row-mounting.

The Motorola solar panels demonstrate advanced technologies in all facets of manufacture, including cell design, interconnection and encapsulation. Typical of these is the cell itself, which has a textured surface consisting of a dense population of tiny pyramids. These pyramids repeatedly intercept any reflected portions of impinging light rays, causing numerous secondary absorptions. As a result of this action, plus the silicon-nitride antireflection surface of the wafer itself, less than 1% of the impinging sunlight is lost to reflection.

Other innovations are technically interesting. For example:

The surface of the wafer is metalized with a series of very thin concentric metal rings, each of which is contacted by six connecting bars (bonding pads) running from the inner circle to the outer edge. This form of contact metalization masks only a very small portion of the wafer from the impinging light rays, yet it effectively covers the entire wafer surface for uniform collection of charge. Moreover, the six bonding pads enhance reliability by providing a considerable degree of redundancy; if one contact were to open, the change in output would be less than 3%.

Outputs of the new Motorola solar modules are proportional to the number of cells (wafers) employed. With each solar cell capable of producing in excess of a half-watt of peak power at 25°C, the 48-cell array (representing a 4ft ^{2} area) is rated at a nominal 26 watts of power. In a typical application, a solar powered energy source for a medium sized radio repeater station is estimated to cost around $3000. This would include the solar array, voltage regulator, storage batteries and mounting hardware.

Further information may be obtained from: Bob Hammond, Marketing Manager, Solar Operations, P.O. Box 2853, Phoenix, Arizona 85062.

CMOS-SOS

No, it's not a chip in trouble, in fact the 'silicon on sapphire' process of fabricating CMOS chips is breathing new life, and new speed into this established family. After overcoming various technical problems of the SOS process, Hewlett-Packard has introduced a 16 bit microprocessor set with some quite impressive features. The advantage of the SOS process is in allowing very low parasitic capacitances in the interconnections and junctions within the chip. The result is that much less charge has to be pushed around, leading to high speed at low power.

The MCC CPU chip itself operates at a maximum clock rate of 8 MHz, executing a full 16 bit register to register add in only 875 ns, and dissipates a meagre 350 mW, driven from a single 12 V supply. The 8 k ROM and 2 k RAM chips designed for the set in SOS, are also unusually fast, with access times of 50 ns and 80 ns respectively. Also available is an interface chip to hook up with the IEEE 488 standard interface bus.

Clearly, this family is going to give bipolar products a run for their money. Don't look for a one board SOS home microcomputer kit yet however, the MCC is aimed chiefly at control applications rather than for processing, in that it has an abundance of instructions oriented toward bit and bit cluster applications. But we will no doubt be seeing SOS processors in many shapes and sizes before too long.

EYE TRACKING

Novel information regarding the way people look at instrument panels, drive cars, operate machinery etc. is being gathered by researchers at Honeywell Inc. (Minneapolis, Minn.). Eye movements are monitored by an infrared device refered to as an 'oculometer'. The researchers believe that their efforts will one day lead to machinery which may be controlled simply by looking at it. This is not new, school teachers have used this technique in the classroom for years.

G.S.R. ERROR

It appears that an error crept into the G.S.R. Monitor article (June p.12). The background article referred to as being in the September 76 issue actually appeared in the April 77 issue.

SKEET

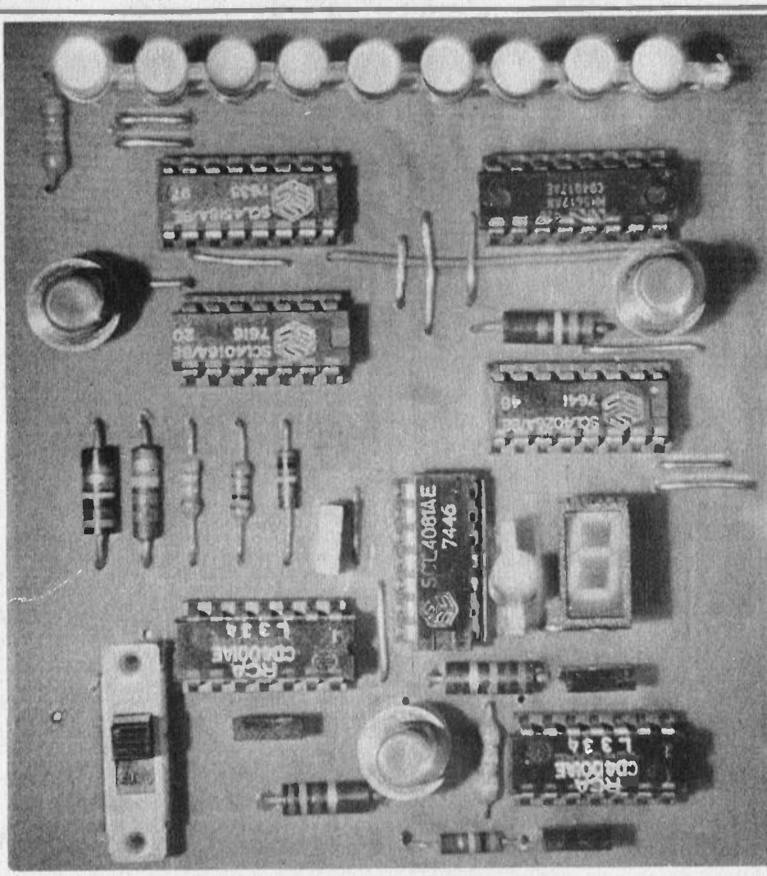
SHOOT IT OUT WITH OUR ELECTRONIC GAME!

by R.G. Cooper

TO BE ENTERTAINING an electronic game should require a fair amount of skill and a little bit of luck. Skeet satisfies both these requirements.

A line of LEDs represent the flight path of the skeet. The gun is "laid" on the last LED of the flight path and no aiming is involved, the point of the game being to correctly estimate the "lead" and fire the gun at the right moment to intercept the skeet at the last LED. Another LED indicates the travel time of the shot; at the instant this LED extinguishes another short duration monostable applies a pulse to one input of an AND gate. If there is coincidence with the lighting of the last LED of the skeet flight path this registers a hit. A monostable stretches the output pulse to light the score LED for about a second and at the same time the pulse advances the score counter by one. Immediately after, whether a hit has been made or not, the skeet flight path counter locks in the nine position.

Pressing the skeet release button starts another "bird" on its way, but at a speed which is impossible to predetermine; this is where the luck,



How it works

In Fig. 1 IC1 is one of ten decoded counter that simulates the passage of the skeet. Since the nine output (pin 11) is connected back to enable input the counter will remain locked on the count of nine. Pressing the skeet release (S1) button resets the counter thus removing the inhibit and another cycle takes place. The eight output (pin 9) is used to clock IC2/1 a BCD counter that tallies the number of skeet flights. This output also is applied to one input of an AND gate (IC6/3); the other input detects the arrival of the "shot" so that an output from the gate denotes a hit. This "hit" pulse is passed to the input of IC7 to count the score and is also used (stretched by a one-shot, IC3/3 and 3/4) to light the score LED via IC6/2. In this and other instances where we need a logic output in addition to lighting a LED we can't get away with hooking the LED directly to the output — the output is literally "flat out" and the voltage is well below an acceptable "1" level. However the unused gates and the AND package fill the bill nicely as buffers and we can still dispense with current-limiting resistors.

The "gun" consists of two simple standard CMOS monostables in series (IC4). The first one has a time constant representing the time of the shot travel to the target and necessitates the "lead". It drives a LED through a buffer (IC6/4) to allow timing judgements to be made during play. The second one-shot provides a short pulse after the first is completed. This is applied to the other input of the coincidence gate (IC6/3) for scoring purposes.

The score remains unlit until IC2/1 reaches a count of eight. When this happens its Q3 output inhibits further clocking and enables the display enable input of IC7, lighting the score display which signals the end of the round.

A standard CMOS simple oscillator is formed by IC3/1 and 2 with the resistor being split into five sections. Four of these are shunted by transmission gates (IC5) so that they may be shorted out as required to control the frequency of the oscillator. The remaining resistor ensures that the oscillator always has some resistance in circuit since there are times when all transmission gates are "on". The oscillator is running at all times when power is applied.

The enable of IC2/2 is tied to that of IC1. Since the enables of these counters require opposite polarities when one is running the other is halted, and vice versa. Tied into the same line is the control of transmission gate 4 which straddles the largest resistor in the oscillator string. This gate is open when IC2 is running so that it is clocked at high speed. This is the "shuffling" operation; Q1, Q2, and Q3 open the other transmission gates when they are "on". While the counter is enabled this leads to a very odd oscillator output since it varies in frequency for every cycle. This is incidental, however as all we require is a "fast stir". When IC2/2 is disabled it serves as a latch to hold the setting of the transmission gates as randomly determined at the time of latching.

Thus providing the "random" speed adjustment for the flight of the skeet

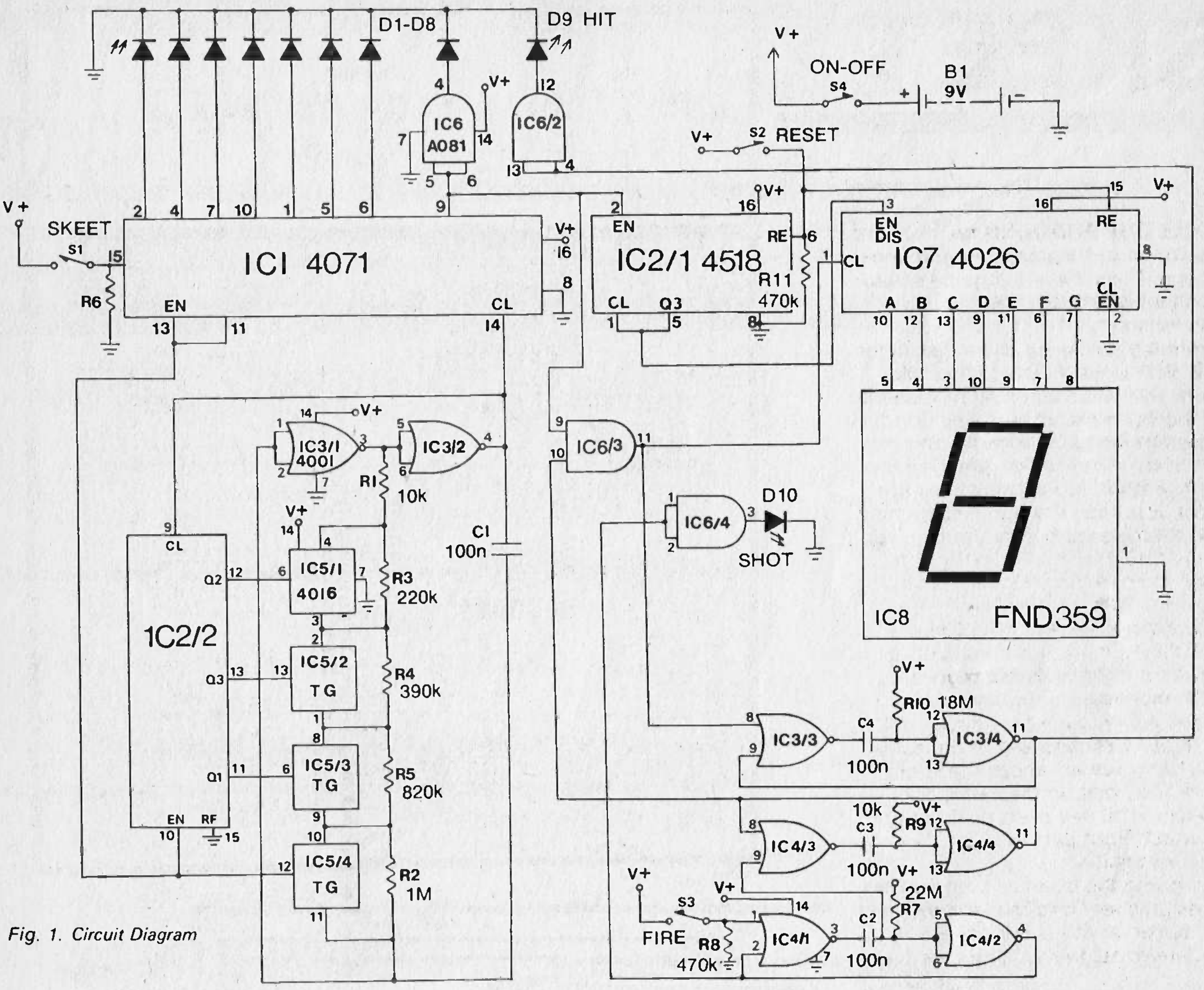


Fig. 1. Circuit Diagram

Parts List

Resistors all ¼W 10%

R1	10k
R2	1M
R3	220k
R4	390k
R5	820k
R6	470k
R7	22M
R8	470k
R9	10k
R10	18M
R11	470k

Capacitors

C1	100n
C2	100n
C3	100n
C4	100n

Semiconductors

IC1	4017 1 of 10 decoded counter
IC2	4518 dual BCD counter
IC3	4001 quad NOR gate
IC4	4001 quad NOR gate
IC5	4016 quad bilateral switch
IC6	4081 quad 2 input AND gate
IC7	4026 decode counter
IC8	FND359 7 segment LED display
DI-8,10	red LED (9 off)
D9	green LED

Miscellaneous

B1	9V battery
S1-3	S.P.S.T. momentary push to make switch (3 off)
S4	S.P.D.T. miniature slide switch

Battery connector, perforated board, insulated connecting wire, 10 Molex pins, case as required.

and the skill come in. You cannot become used to firing the gun at the same position in the flight path as the speed can be any one of eight different values determined randomly during the time between the locking of the flight path counter and its resetting.

After eight shots the readout lights up with your score. In competitive play the score would be reset and your opponent would then take eight shots. For practice the score need not be reset and the counter would continue to register.

When power is first applied any condition of the counters is possible, including invalid states. Press the skeet release button first and allow the skeet to complete one cycle. Press the score reset button and you're ready to begin a game.

Construction

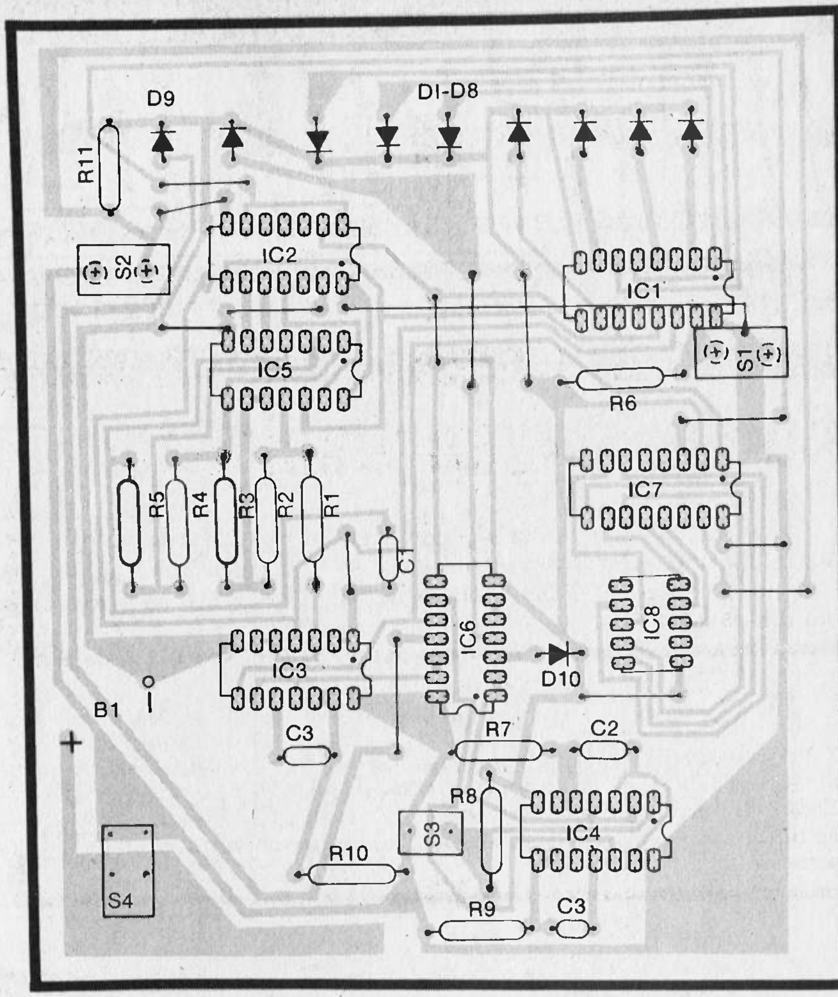
Everything but the battery mounts on the printed circuit board. The board is securely mounted to the front panel by means of the three push buttons. Cutouts are needed for the LEDs, on-off switch, and the readout. Molex pins should be used for the readout to raise it to the proper level (the width between pin rows is only 0.2 inches so it will not fit a standard socket). If sockets are used for the ICs they should be of low profile type in order to fit beneath the panel. To maintain the same height above the board, and to provide stiffening, 0.3 inch strips of perforated board were used between the leads of all the LEDs. A miniature switch will fit below the panel if a slot is cut to provide clearance for the handle.

Note that the LEDs are not all similarly oriented on the board.

There is nothing to prevent you from changing the values of resistors and capacitors to obtain different time constants. In general increasing

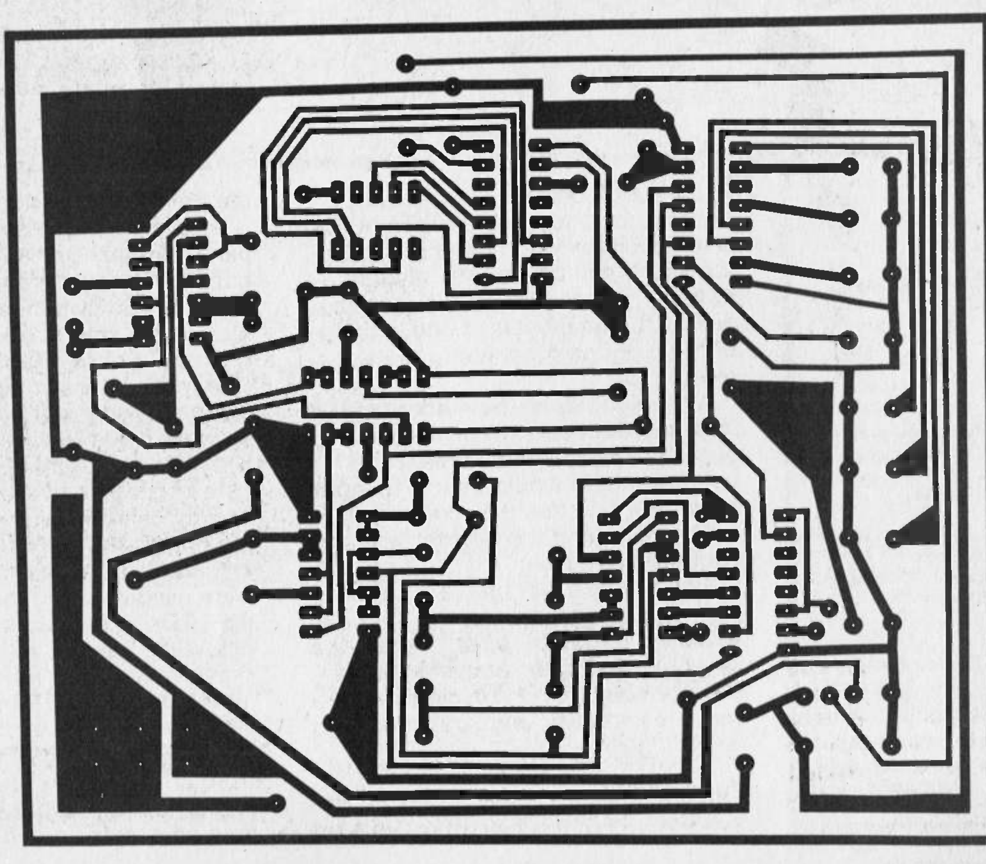
SKEET

the value of the resistor or capacitor (or both) in a circuit will increase the time constant. In the case of the oscillator and the coincidence one-shot this will make the game easier and in the case of the gun monostable more difficult. When changing the speed of the oscillator it is best to change the capacitor since this will change all the speeds in the same proportion. The gun time constant should not be so long in relation to the highest speed of the flight path that you don't have at least 3 or 4 LEDs clear to determine the skeet speed before firing. ●



Component Overlay

PCB Foil Pattern (full size)



How to make sound solder joints

THE ART OF SOLDERING

SUCCESSFUL SOLDERING

SOLDERING is an art — an art that is essential to the successful building of fault-free electronic circuits. Suppliers of kits tell us that about 90% of all problems with kits are due to faulty solder joints. Hence the beginner (if he wishes to avoid much frustration and, possibly expensive service charges) *must* learn to solder correctly.

In essence, soldering is a method of making joints having low electrical resistance. It is not primarily used to give mechanical strength. If mechanical strength is required the component must be separately supported.

SOLDER

A good solder should have a low melting point, low electrical resistance, should go very quickly from liquid to solid state (and vice-versa) and should be capable of rapidly fusing to the metal surfaces being soldered.

Solder, for electrical connections, is an alloy of tin and lead. Pure tin melts at 327°C and is plastic over the range 183° to 327°C, whilst pure lead melts at 232°C and is plastic over the range 183° to 232°. Either metal, used alone, is unsuitable as any movement whatsoever whilst the soldering metal is in its plastic state will result in a faulty joint. However when lead and tin are mixed it is found that the melting temperature of the mixture is lower than for either metal alone, and the plastic temperature range is decreased. When the composition is 63% tin and 37% lead the mixture has no plastic region and goes from solid to liquid at precisely 183°C. However, in practice it is found that a very small region of plasticity is desirable in a solder for electronics, and the usual proportions are 60 percent tin and 40 percent lead. Other alloys are made for special purposes, but only 60/40 composition should be used for electronic purposes as this provides optimum-strength with lowest-resistance of electrical joints combined with the most desirable plastic range of about 5°C.

THE NEED FOR FLUX

All metals (even when freshly cleaned) are covered with a non-metallic film of oxide which prevents solder fusing to the metal. The oxide has a surface tension which effectively *isolates* the solder from the metal. For a reliable, low-resistance joint this oxide *must* be removed during the soldering process, and this is performed by the use of a flux.

The flux used for electrical soldering is a high grade of wood or gum resin together with a small quantity of activator. The molten resin effectively wets both solder and metal, whilst the activator dissolves the oxides on the surface, allowing the solder to flow freely and form a molecular bond with the metal.

In order to dispense automatically the correct amount of flux, modern solders have the flux contained within cores in the solder itself. Five cores are generally used, throughout the entire length of the solder — so no additional flux is needed. Any excess flux hardens on the surface of the joint but it is completely non-corrosive.

Fluxes are also made for non-electronic uses. These are generally acidic and must *never* be used on electronic equipment as component leads and printed circuit board tracks will be corroded. Additionally the use of such a flux will completely void any warranty on a kit or electronic equipment on which it is used.

SAVBIT SOLDER

If the soldering iron has a copper bit the copper will gradually be dissolved in the molten solder. Thus the tip wears away quickly and requires constant filing and retinning. To overcome this problem some soldering iron tips have a copper core with an outer skin of iron. Another solution to the problem is to add a small amount of copper to the solder alloy. This prevents the absorption of further copper and greatly extends tip life. Such solder is known by the trade name of 'Savbit' The use of the Savbit solder can extend the life of ordinary copper tips up to ten times.

SOLDERING IRONS

There are many types of soldering iron on the market and those most commonly used in electronics may be grouped into three main categories.

(1) Quick heat irons.
(2) Continuous heat irons.
(3) Temperature controlled irons.

QUICK HEAT IRONS

Irons of this type generally work from a transformer which supplies a low voltage at very high current. The two main types are solder guns and low-voltage irons.

The solder gun passes a current of about 50 amps at 0.5 volts through a short length of copper wire, thus heating it quickly to very high temperatures. These irons usually include a reel of solder which is automatically fed to the tip each time a trigger is pressed.

The low-voltage irons have a copper tip against the rear of which a carbon contact is made. A current of around 30 amps at 3 volts is passed via this contact whenever the sleeve switch is actuated.

Both these types of iron are ideal for

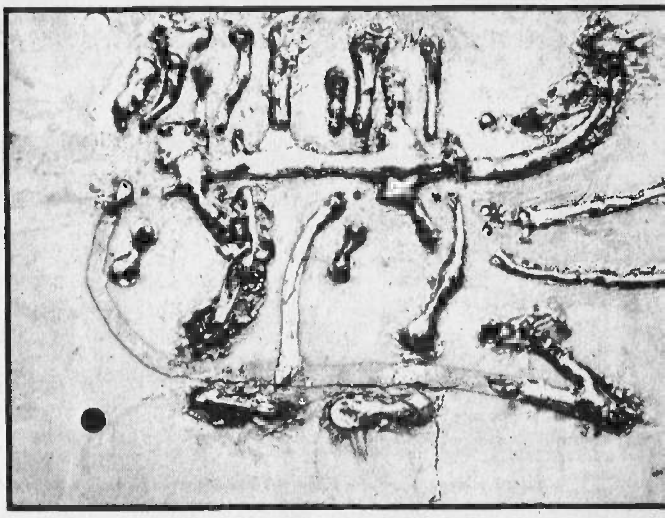
intermittent handyman use. However some irons of this type do not have an electrostatic screen on the transformer (which means delicate ICs and transistors could be damaged by leakage currents) and, if used improperly, are prone to overheat components and possibly damage them, and/or, the printed circuit board upon which the components are mounted. Such irons, (i.e. those without electrostatic shields) should therefore only be used for general electrical work for soldering to chassis and other tasks where large reserves of heating power are required. They are not recommended for printed circuit board assembly or general electronic service work. So before buying an iron of this type do make sure it has an electrostatic screen between the primary and secondary windings of the associated transformer.

CONTINUOUS HEAT IRONS

The most universally used irons are those of the continuous heat type. These irons are heated by an element of resistance wire wound around (but insulated from) the rear end of the copper tip. They are manufactured in wattages from about 6 to 250 watts but, for average electronic work, those most commonly used are rated between 15 and 30 watts.

These irons are slow to heat and hence are usually left running continuously. Such operation, although adequate, causes problems with oxidation of the tip. The tips therefore require constant attention and fairly frequent replacement.

Many workers find that these inexpensive irons are entirely adequate despite the drawbacks of continuous operation. They are light, cheap and well insulated.



How not to do it! So much heat has been applied to this board, that in places the tracks have been damaged. In other places insufficient heat, or improper fluxing, has caused the surfaces to be not wetted properly. So much solder has been applied that one does not know where the tracks really are, or whether the joints are good or not! A kit supplier would be quite justified in refusing to accept responsibility for a project, built this way and not working.

HEAT CONTROLLED IRONS

For continuous use on a production line or in an electronics laboratory a temperature controlled iron is often used. Those irons are relatively expensive but are unsurpassed, for accurate soldering and for minimizing damage to components and printed circuit boards due to overheating.

A typical temperature controlled iron, (manufactured by Weller), uses a switch operated by a magnet and spring assembly (within the barrel of the iron) to control temperature.

When a ferromagnetic material is heated it is found that at a certain temperature, which depends on the material, all magnetic properties are lost. This temperature is known as the Curie point and is typically 1000°K for iron, 633°K for nickel and 1393°K for cobalt. Thus by alloying these or other ferromagnetic materials the Curie point of a material may be set to any required temperature.

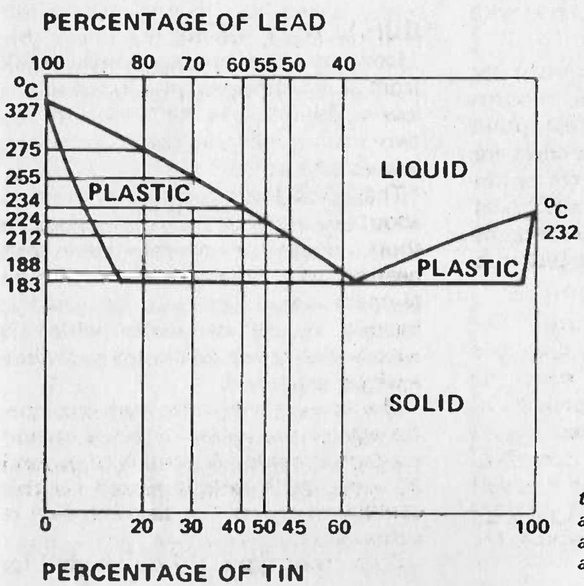
On the Weller iron the tip has a small piece of material at the rear, called the sensor. This is designed to have a specific Curie point. When the tip is cold the sensor attracts the magnet and hence the switch closes heating the iron. When the tip reaches the Curie temperature the sensor pad is unable to hold the magnet which is then forced back by the spring. The switch therefore opens, removing power to the iron. It can be seen therefore that the iron will switch on and off automatically to maintain the desired temperature.

These irons operate from a 24 V transformer within the stand and are supplied with a variety of tips of different shape and operating temperature. The temperature is stamped on the base of each tip. Thus, one can pick the operating temperature most suited to the class of work.

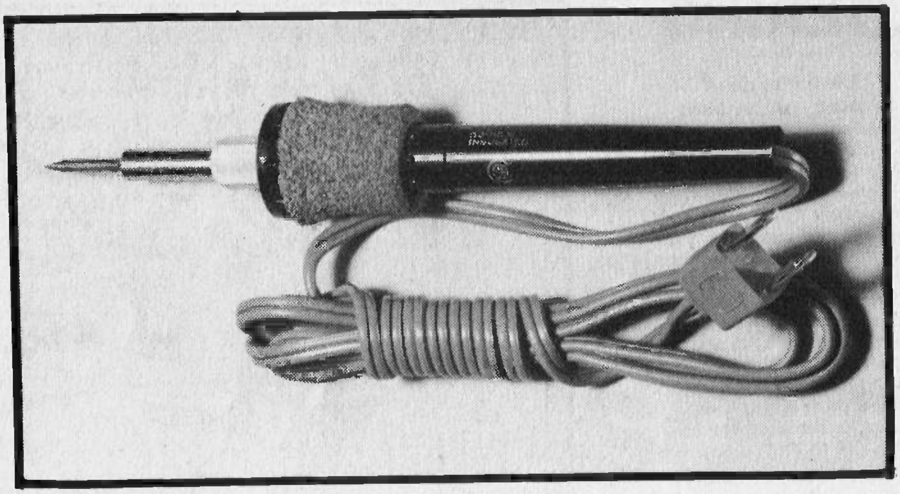
Where 60/40 solder is being used for new soldering, a bit temperature of 250°C (500°F) will be adequate. This allows adequate margin over the melting point of 215°C to allow for heat conducted away by the component or terminal etc. Savbit No 1 solder melts at a slightly higher temperature, and for this a temperature of at least 275°C (550°F) is required.

For unsoldering, a higher temperature is needed again. This is because the surface of the solder becomes oxidized and heat flow is impeded. Hence for general service work we recommend a tip temperature of at least 315°C (600°F) and perhaps even 370°C (700°F) for large connections etc.

The tips for temperature controlled irons are all iron plated and should never be filed. The tips are cleaned during use by wiping on the small sponge supplied. This should be kept



How melting temperature and plastic range are affected by the relative percentages of lead and tin in a solder alloy.



The Craftrite 7365 soldering iron. Ideal for general use, this one is rated at 27 watts and takes all standard 1/8 inch thread-in copper tips.

damp. If really dirty,steel wool or fine emery paper may be used.

If you can afford it this type of iron is by far the best available. The plated tips will last ten times as long as plain copper types, the irons heat in 30 to 40 seconds and have heating power equivalent to a 100 watt continuous type.

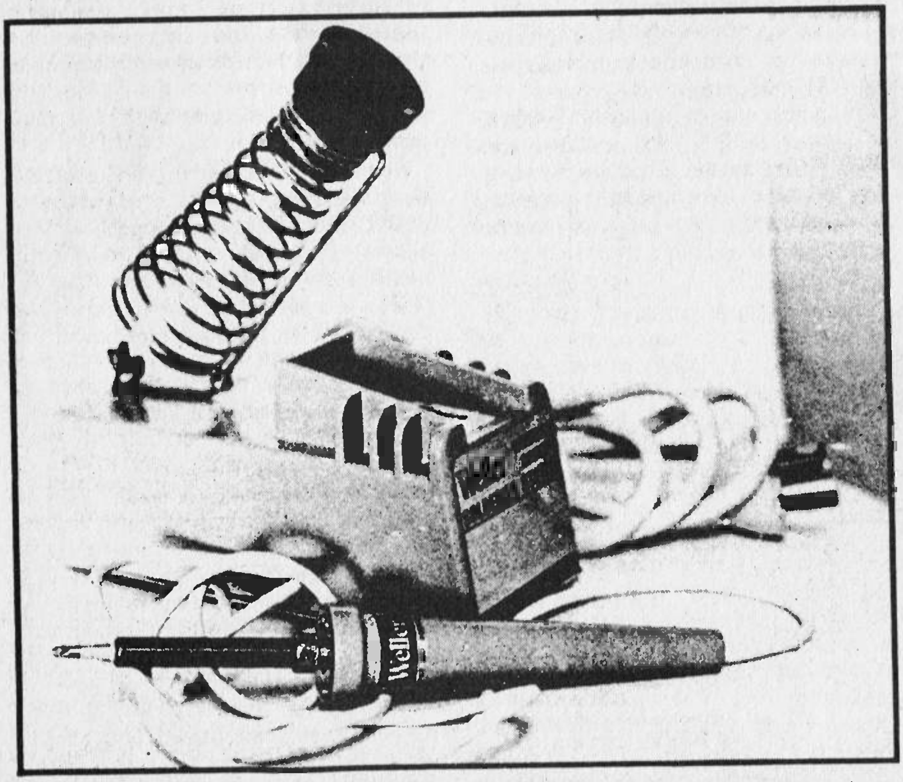
TINNING THE IRON

To make sound solder joints it is necessary to keep the tip of the iron clean and well tinned.

Iron-clad tips only need to be cleaned occasionally with fine emery cloth whereas copper tips will need to be dressed with a file and retinned at regular intervals.

Whenever a plain copper tip becomes pitted, and oxidation scale builds up between the heating element and the tip, the efficiency of the iron will drop considerably. To recondition the tip, clean off the oxide scale and, while the tip is cold, dress it with a file to remove the pitted surface on the end of the tip.

Heat the iron and apply solder at the lowest iron temperature which will melt it. Wipe the iron on a damp cloth or sponge until the whole tip is covered with a bright coating of solder. The iron is now ready for use.



The Weller temperature controlled iron. The stand incorporates a transformer in the base. The operating temperature is selected simply by changing the tip.

PREPARING COMPONENTS AND LEADS

Most components have plated or pre-tinned leads which will accept solder without any special preparation. However, if the components are old, or the leads tarnished or oxidized, the leads to be soldered should be cleaned and tinned before attempting to solder them in position. To do this apply the iron and the solder to the lead until a uniform coating of solder is obtained. If the lead is unusually dirty, and will not take solder, pull it through a piece of doubled over emery paper.

Stranded hook-up wire should be prepared by stripping away about 7 mm of insulation from each end. The strands should then be twisted together and tinned, and as detailed above, before the wire is attached.

MECHANICAL ASSEMBLY

When assembling components to printed circuit boards the component leads should be fitted through the correct holes and spread slightly so component is held firmly in position. Always mount the component such that its value, if printed on it, is visible. This facilitates later servicing.

Components are inserted from the non-copper side of the board, as shown in various pictures throughout this article, (this may seem totally obvious to experienced enthusiasts but it's surprising how often we and kit suppliers come across boards on which the components have been mounted on the copper side!).

When fitting components to terminals or tag strips turn the lead half way around the lug or tag so that the component is mechanically secure. This prevents component movement (and resulting faulty joints) whilst the solder is cooling. Do *not* make a full turn, or more, around the lug as this will make it very difficult to remove the component in later servicing.

SOLDERING

The iron must be fully up to working temperature and the tip clean and coated with 'wet' solder. It should actually look 'wet' and shiny. If it doesn't, touch it briefly with the solder and wipe off surplus onto a damp sponge.

Now press the tip against the terminal (or printed circuit board track) *and* the end of the component lead. Preheat like this for two to three seconds.

Still keeping the iron in position, apply solder to the joint — never to the iron. Continue to apply solder only long enough for the solder to flow evenly over the joint. After removing the iron you must let the joint harden before moving either the component or the PC board. Then snip off any excess lead.

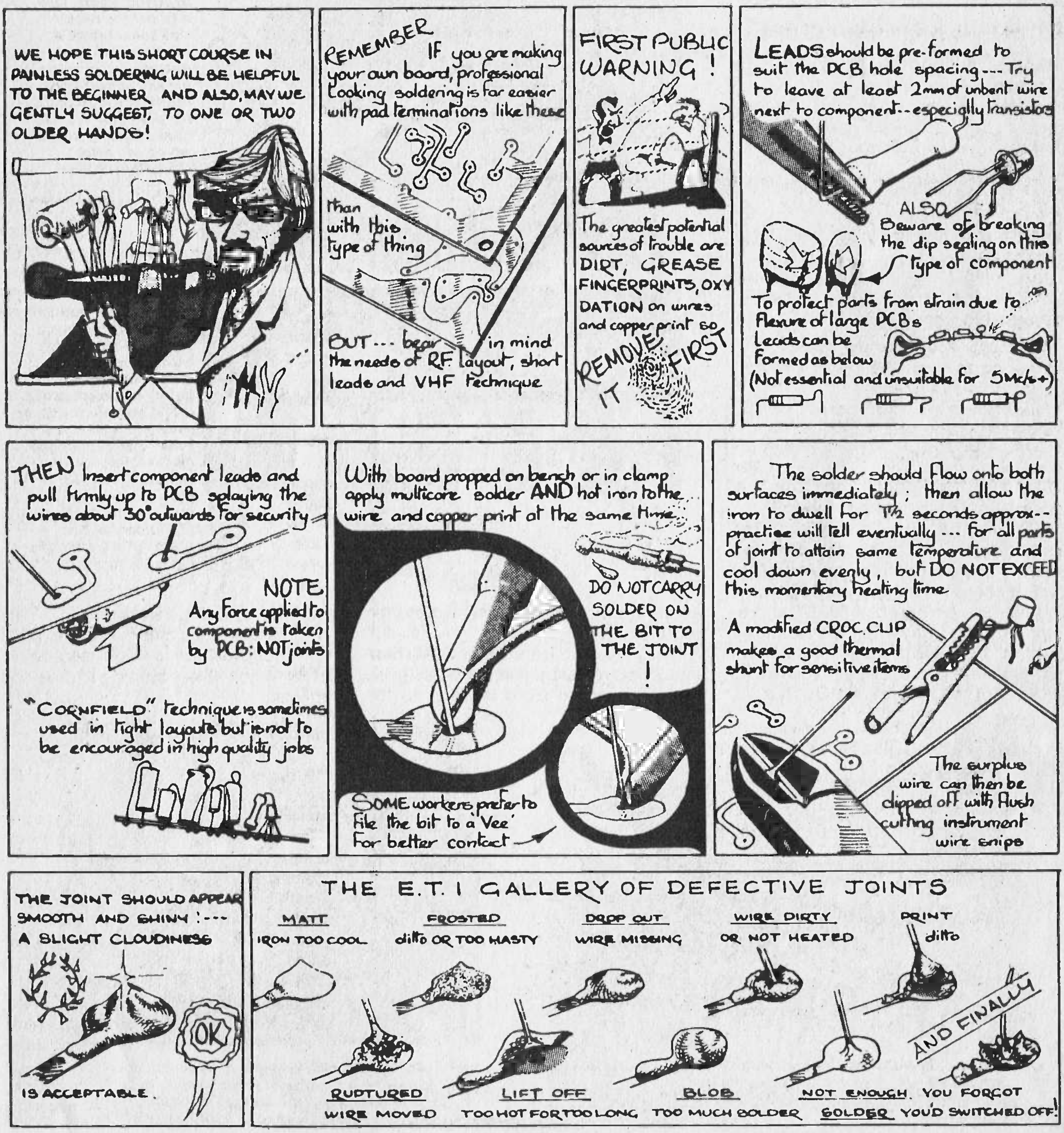
A correctly soldered joint should be bright and smooth. Poor joints look crystalline and grainy or, the solder tends to be in blobs (that is solder has not wetted the surface properly).

Take care not to apply too much solder as it is difficult, then, to see if the joint is a good one. Solder bridges may also be formed.

REMOVING COMPONENTS

If it is necessary to remove a component from a printed circuit board the solder should be removed from the joint by 'wicking'. To do this remove about half an inch of insulation from a piece of stranded hook up wire, dip the prepared end into liquid resin and lay it on top of the solder joint. Then apply the flat tip of the iron above the wire and joint until the solder melts and is sucked up by the wire. Repeat the procedure if necessary to remove all excess solder from the joint. Alternatively a proprietry product such as 'Dri-wick' (braid that is pre-fluxed) may be used.

THE ART OF SOLDERING



After all excess solder is removed, it is a simple matter to pry the component loose. Removing components by means of vacuum solder suckers (although effective) must be done carefully as there is a tendency for the devices to lift the copper tracks from PC boards.

Solder bridges, if they occur, should be removed by wicking and resoldering, or by heating the bridged area with the iron and wiping quickly with a soft cloth or with a brush to remove excess solder. Resolder the cleaned joints using less solder.

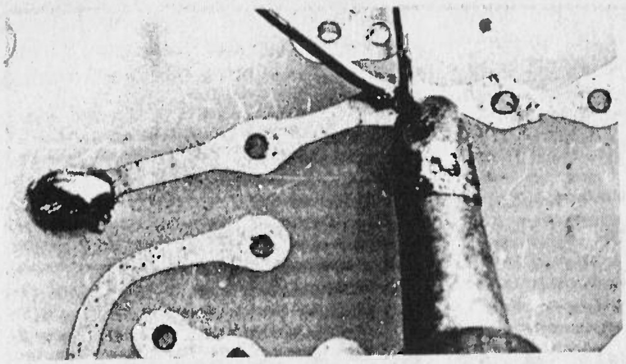
SOLDERING SEMICONDUCTORS

Some semiconductors will be damaged if subjected to too much heat. Hence transistors and ICs, etc, should be soldered quickly and cleanly. If you doubt your ability to do this — use a heat shunt (eg pair of long nose pliers), between the end of the lead being soldered and the transistor body, to divert heat from the device.

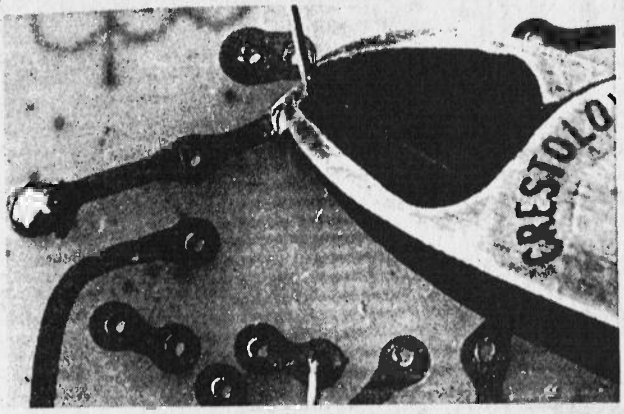
Integrated circuits based on MOS or CMOS technologies are particularly prone to damage during soldering not only due to heat, but also by electrostatic charges or leakage currents. These devices should be left with their pins inserted in the black conductive foam (in which they are usually supplied) until they are to be used. Avoid touching the pins of the IC as even static discharges from the body can possibly cause damage.

MOS and CMOS ICs should be the last components to be fitted to the board. They should be inserted quickly and cleanly, and the power supply pins should be soldered first. The remaining pins may then be soldered without fear of damage. Beginners may well find that it is safer to use an IC socket for MOS and CMOS ICs as soldering directly to the IC is not then required.

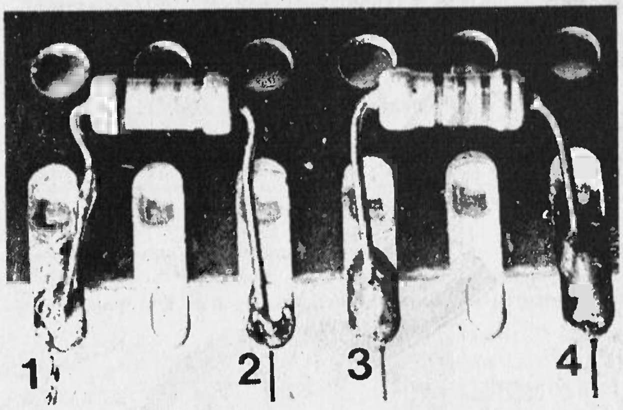
Many modern irons have a very high insulation resistance and are designed for low leakage from the line to the equipment being worked on. This reduces the possibility of damage to ICs and provides a high degree of safety when live equipment must be repaired.



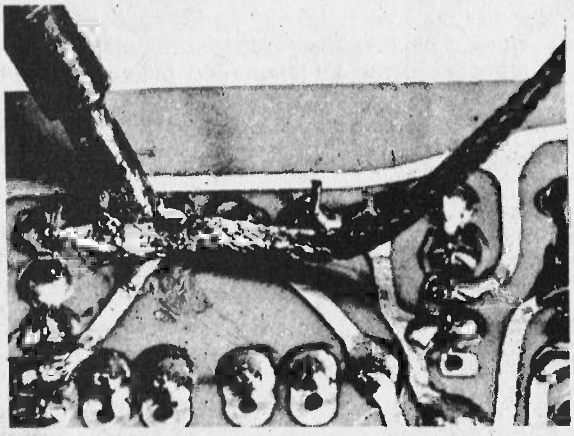
Apply the soldering iron to the joint so that the track and component wire are heated together for about two seconds, then apply the solder to the joint — NOT to the soldering iron tip.



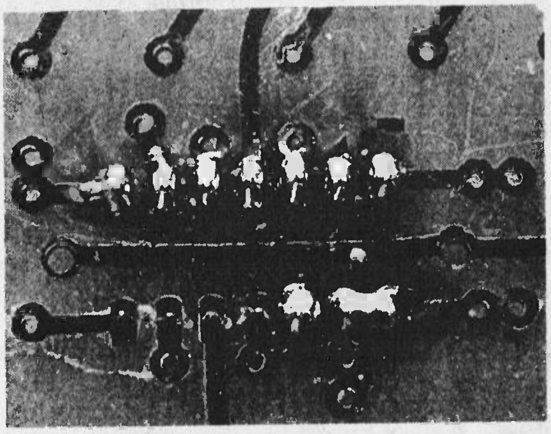
Allow the joint to cool completely undisturbed and then snip off the excess component lead. A joint that has been disturbed during cooling will appear crystalline and will probably be 'dry' (that is it will have a high resistance which will affect circuit operation).



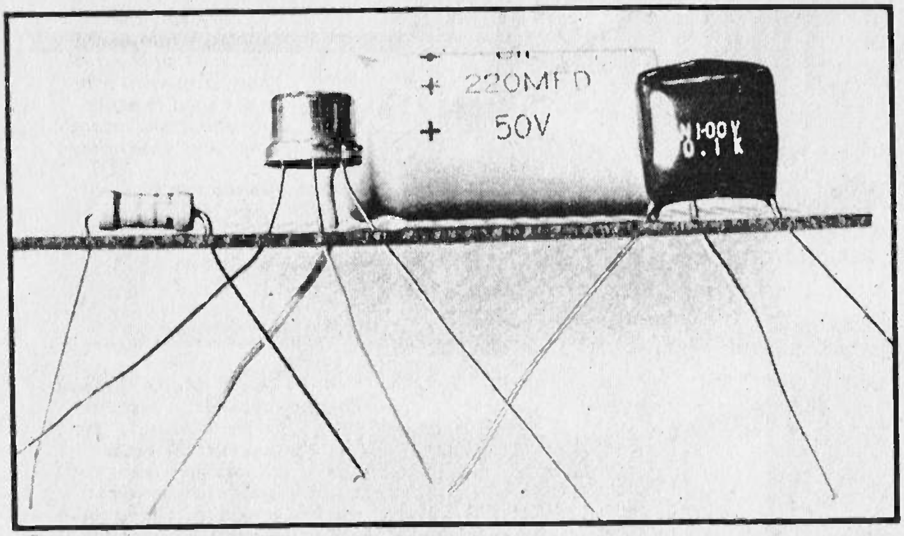
How the joint will appear — from left to right (1) too much heat results in solder leaving the joint. Movement of joint before it cooled results in the crystalline appearance. (2) Insufficient solder and not enough heat — solder has not wetted the tag. (3) A good joint should be smooth and shiny. (4) Solder has not thoroughly wetted the tag — the joint could be dry.



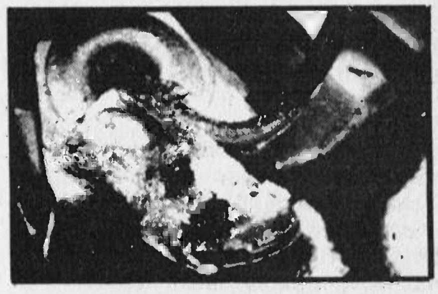
To de-solder using wick, apply the braid over the joint and place the iron on top of the braid until the solder is sucked up.



Too much solder may cause solder bridges. The top row of joints to this IC are fine but on the bottom row too much solder has resulted in a solder bridge.



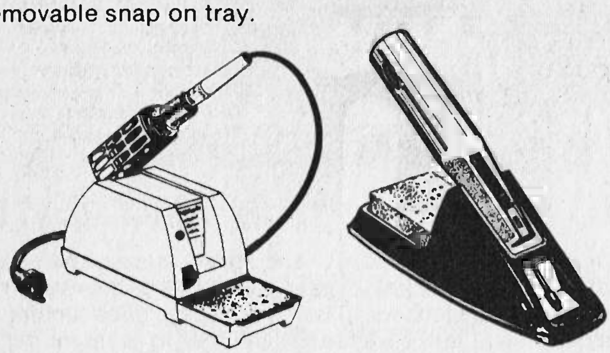
When components are fitted to a board the leads should be splayed, as shown, to keep them in position prior to soldering. Position components so that values and voltage ratings may be seen. This facilitates later servicing.



In a 'dry' joint the solder has not properly bonded to either or both metal surfaces, or the joint has been moved during the plastic region of the molten solder. Such joints have a high electrical resistance and low mechanical strength. Here this resistor lead 'dry-jointed' to a potentiometer may be pulled away quite easily.

SOLDERING-THE TOOLS

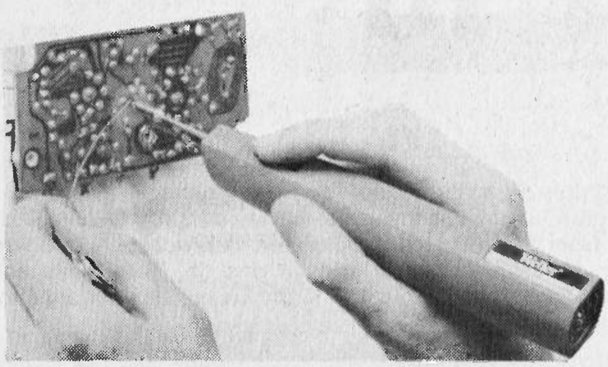
The Ungarmatic controlled soldering station. This low voltage system (24V) is available in 315°C, 370°C and 425°C preset temperatures — depending on the element used. The station carries the transformer, iron holder and a bit wiper in a removable snap on tray.

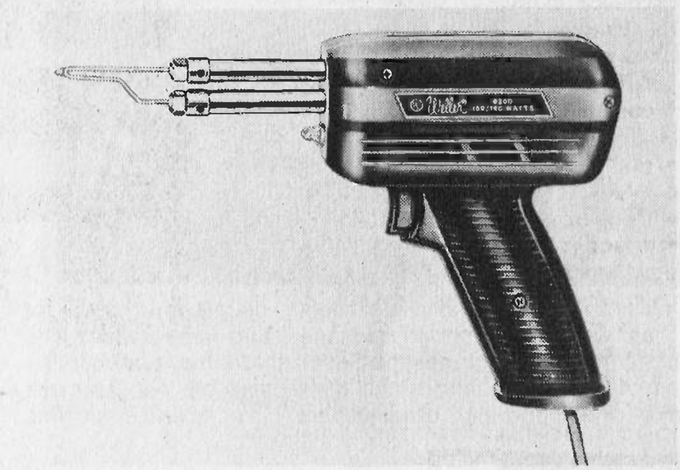


A rechargeable iron from Ungar, shown above is the #194 which is an "instant" heat iron that recharges from the soldering station. Has a power on indicator with trigger interlock when not in use, light for working area and a built in sponge tray for tip cleaning.

The unit will make from 50 to 100 normal joints per charge and will charge overnight from full discharge. Battery life is said to be a minimum of 1,000 recharge cycles.

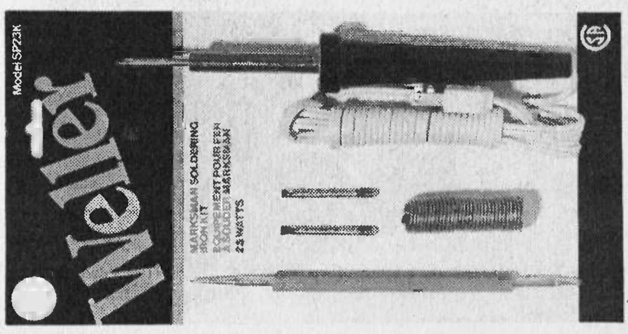
A very similar product is available from Weller; although the Weller iron has a separate charger, in almost every other respect the irons are similar. The Weller Cordless Soldering Iron is shown in use below.





Shown above is an instant heat soldering gun from Weller. This is the type mentioned in the text under "Quick Heat Irons". Unfortunately these are rather heavy for normal use but are ideal for handyman use. Various tips are available for cutting tiles, expanded polystyrene etc.

A general purpose hobby kit is also available from Weller (see below). The Marksman iron is sold on its own or as part of a larger 12 piece kit in a rigid plastic case which is designed for the hobbyist and includes model making tools etc. The kit shown contains the Marksman 25 watt iron, three different bits, some solder and a wire bending and "joint prodding" tool.

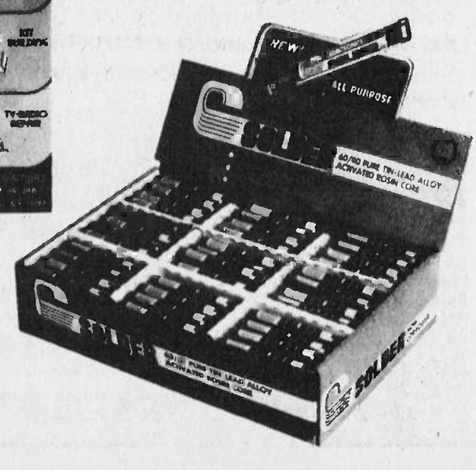


SOLDERING-THE TOOLS

Part of the large Kester Solder range is shown below; they produce a number of different solder products for electronics use in addition to their extensive range of industrial products and chemicals. A booklet entitled *Soldering Simplified* is available free for the asking from them or via distributors.

Most solders are available in reels — usually sold by weight — as well as in small packages and dispensers. The Multicore range includes the Savbit solder mentioned in the text of the article and small dispensers or "serviceman's paks" of this and their other general purpose and electronic type solders are available — see above. It is worth knowing that large savings can be made by buying a reel rather than the handy dispensers. Their range includes solder for stainless steel and silver as well as that for aluminum.

Another range of solder products is available from Chemtronics. Once again these come in handy dispensers and reels. These people also market a range of aerosol products for the electronics industry including freezers and flux removers.



A simple desoldering tool from Len Finkler. Working like a bike pump in reverse these simple devices provide one quick suck from a sprung loaded plunger. This normally removes all solder from a heated joint in one go. Both these and suction bulbs, available for many irons, are suitable for hobbyist use.

DIGITAL FREQUENCY METER

five ranges up to 10 MHz

THIS DESIGN USES A HANDFUL OF STANDARD COMPONENTS TO ACHIEVE A SUPERIOR PERFORMANCE TO MANY COMMERCIAL UNITS.

BASED UPON A SIMPLE TTL 'heart', this project grew and grew until it formed a 'Short Circuit' which occupied more development time than most major projects!

The basic range of the DFM extends to 9.9 MHz, and will reliably read down to about 1 Hz. By use of the boards the design will pose no problems to the constructor (we hope!) and should prove straightforward in use.

TAKING READINGS

Let's tell you how to use it first. A reading is taken upon operating the 'Test' button. The display will change - upwards - for a short period, halting once the value of the frequency under test is reached. Before making a second test, clear the display, else the reading will be cumulative.

Calibration is always the biggest hurdle with DFMs, and we must confess we've found no new ladders. One suitable method is to use a high-quality 'scope. The time base is usually very accurate on the better models, and each range can be set up using this signal. Failing this, a signal generator of good quality is needed. Each range is independent of the others, and so will need adjusting separately.

CONSTRUCTION

Assemble the boards as per the overlays. Space within the case is restricted, so it might be a good idea to complete the interwiring before positioning them within the case. Drill all the mounting holes etc, and get an idea of lengths of wire beforehand - it'll look neater if you don't have six mile of cable where two inches would suffice.

Don't forget either to drill the holes into the back panel for calibrating the range pots; it's much easier to carry out this function once the meter is built and working.

FREQUENTLY USED

Just a few points about operation. The input squaring circuit is reasonably sensitive, about 100 mV is all you need to ensure correct operation. If you intend to pump appreciably more than this into the terminals, may we suggest an attenuator?

The units may well read with inputs lower than 100 mV, but is not intended to do so. Accuracy will depend on how it is set up, and on the higher ranges may be affected by length of leads from input circuitry etc, so keep these to a minimum.

How it Works

The input signal is amplified and squared to TTL level by Q1 Q2 and the Schmitt trigger IC1, which means that a train of squarewaves at the input frequency will be presented to the NAND gate IC2c. This will be held open for a preset time depending on the frequency of monostable IC3.

This period is set by the timing (range) components RV1-5, R7-11, C2-6 respectively. The switch bank selects which network is in circuit, and hence for how long IC2c allows the input pulse train to pass. The train is inverted by IC2d and fed to the counter chain IC4-6.

The number of pulses reaching IC4 will represent the input frequency, provided the period of IC3 has been correctly set (calibrated). RV1-5 is adjusted to achieve this.

IC2a and IC2b 'debounce' SW6, the test button, and IC7-IC9 and DISPLAY 1-3 display the input frequency.

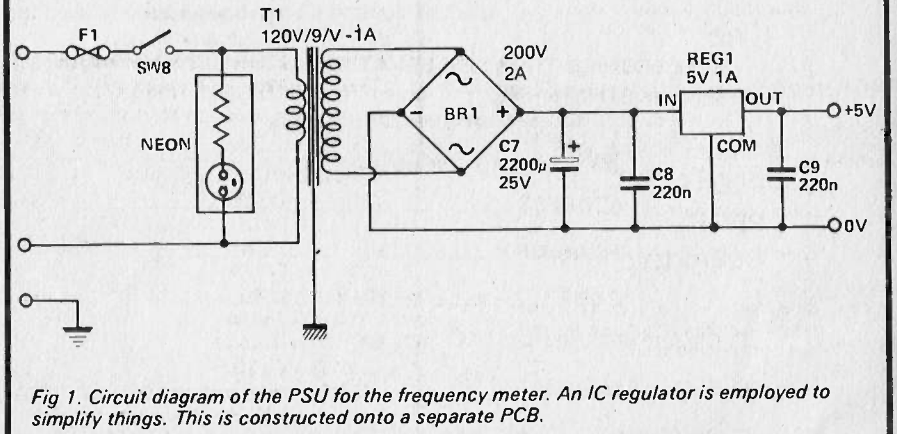


Fig 1. Circuit diagram of the PSU for the frequency meter. An IC regulator is employed to simplify things. This is constructed onto a separate PCB.

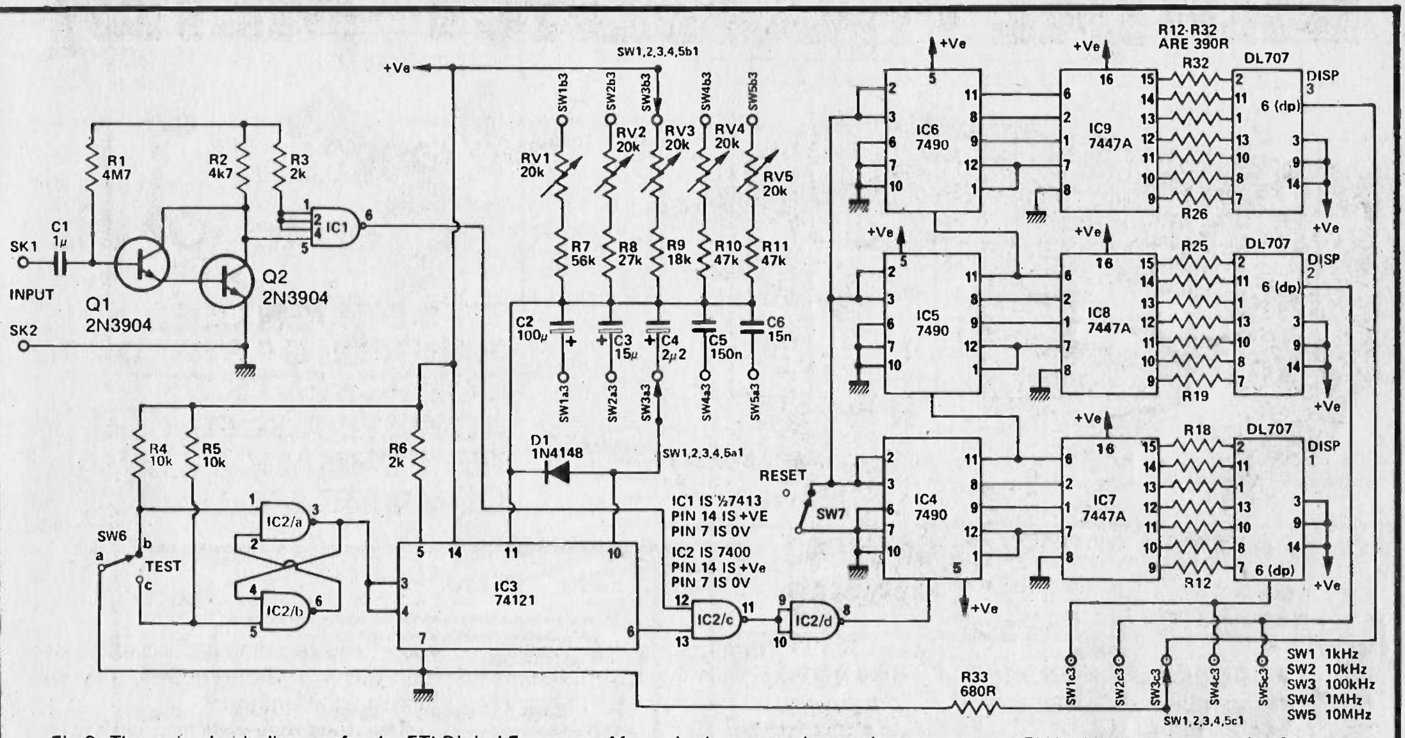
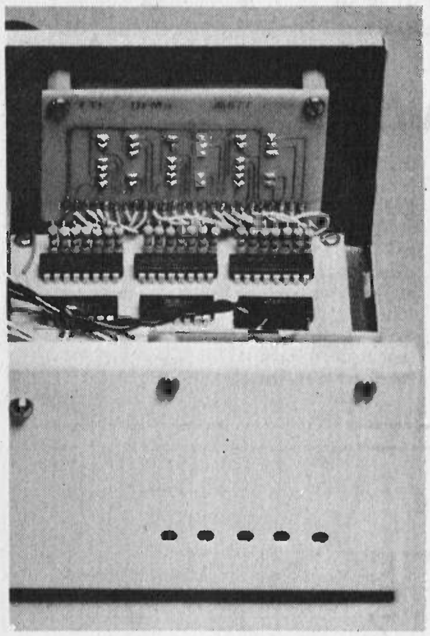


Fig 2. The main circuit diagram for the ETI Digital Frequency Meter. In the centre the ranging components RV1-RV5, R7-R11 and C2-C6 can be seen. The resistor chain R12-R32 is to control the current (here brightness) of the three displays.



An internal view of the meter, showing the mounting of the display board onto the back of the front panel. Note that the display resistors, R12-R32 are mounted vertically onto the PCB.

Down at the right hand lower corner the calibrating holes for the range pots can be seen.

Parts List

RESISTORS All ½ W 5%

R1	4M7
R2	4k7
R3,6	2 k
R4,5	10 k
R7	56 k
R8	27 k
R9	18 k
R10,11	47 k
R12-32	390
R33	680

CAPACITORS

C1	1 u Polyester
C2	100 u 10 V tantalum electrolytic
C3	15 u 16 V tantalum electrolytic
C4	2.2 u 25 V tantalum electrolytic
C5	150 n polyester
C6	15 n polyester
C7	2,200 u 25 V electrolytic
C8,9	220 n polyester

SEMICONDUCTORS

Q1,2	2N3904
IC1	7413
IC2	7400
IC3	74121
IC4-6	7490
IC7-9	7447A
D1	IN4148
DIS.1-3	DL707
REG.1	7805 with T0220 case.
BR1	200 V 1.6 A bridge rectifier

POTENTIOMETERS

RV1-5 20 k Multiturn (.75" type or sim.)

SWITCHES

SW1-5 The switches used in the prototype are available through W.A. Components Ltd. 65 Granby St., Toronto, Ontario M5B 1H8. The ordering codes are: 5 x 338 636 (switches), 2 x 338 563 (packs of 4 buttons), 1 x 338 254 (latching assembly).
A 6 pushbutton assembly which is electrically suitable but differs mechanically is available from Dominion Radio at $2.75, cheaper than the imported version. The switch PCB will need slight modifications to incorporate this switch set.

SW6	S.P.D.T. momentary action
SW7	S.P.S.T momentary action
SW8	On/off 1A 120V type

TRANSFORMER

120V/8.5V 1A (Hammond 166 J8)

MISCELLANEOUS

Case, fuse holder, fuse, line cord, neon, 2 mm red and black sockets, PCB posts, wire, nuts, bolts etc.

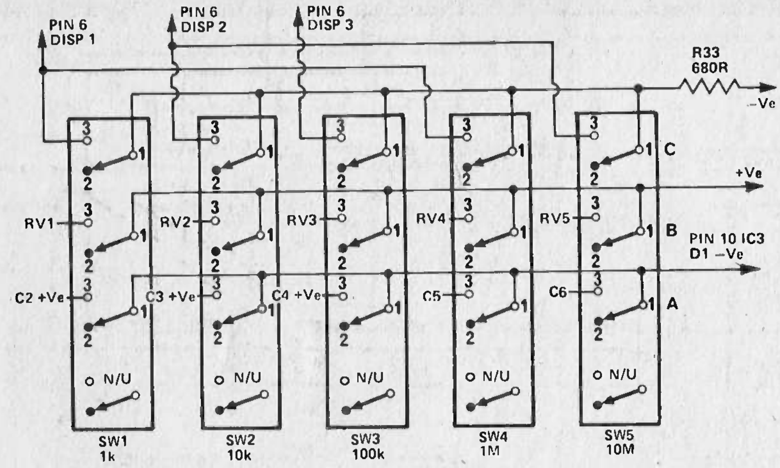
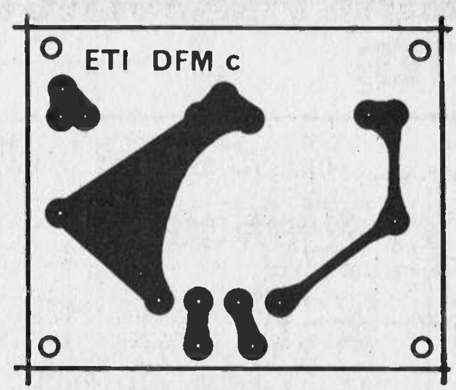
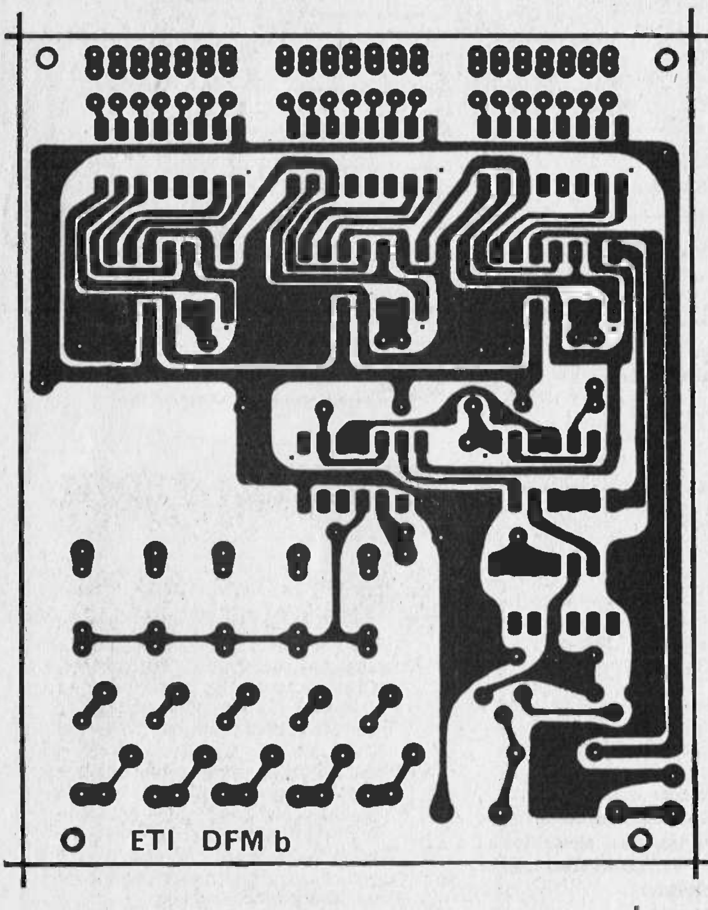
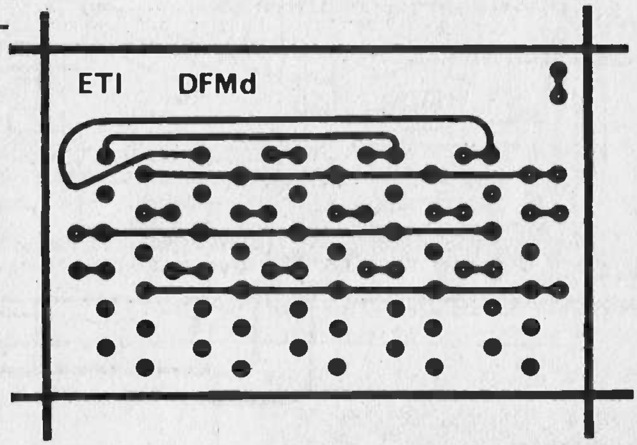


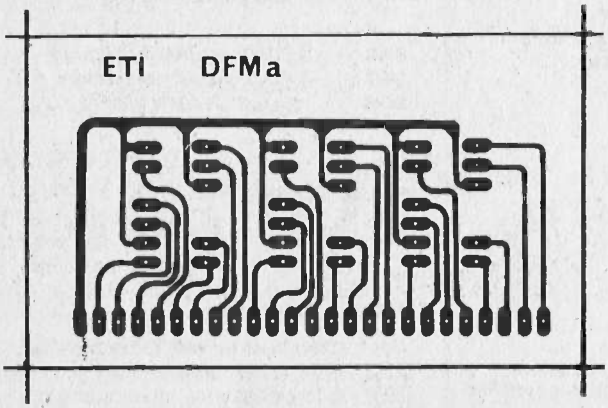
Fig 3. The switching circuit for the DFM. Connections are achieved via the PCB which solders beneath the actual bank

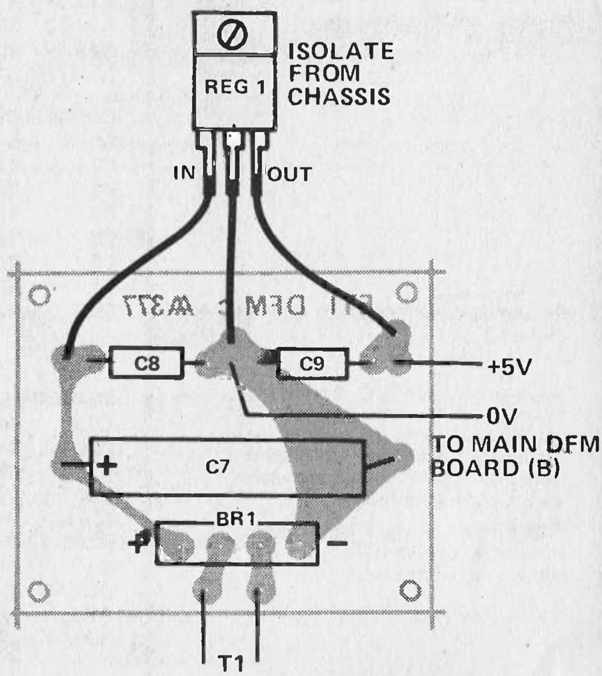




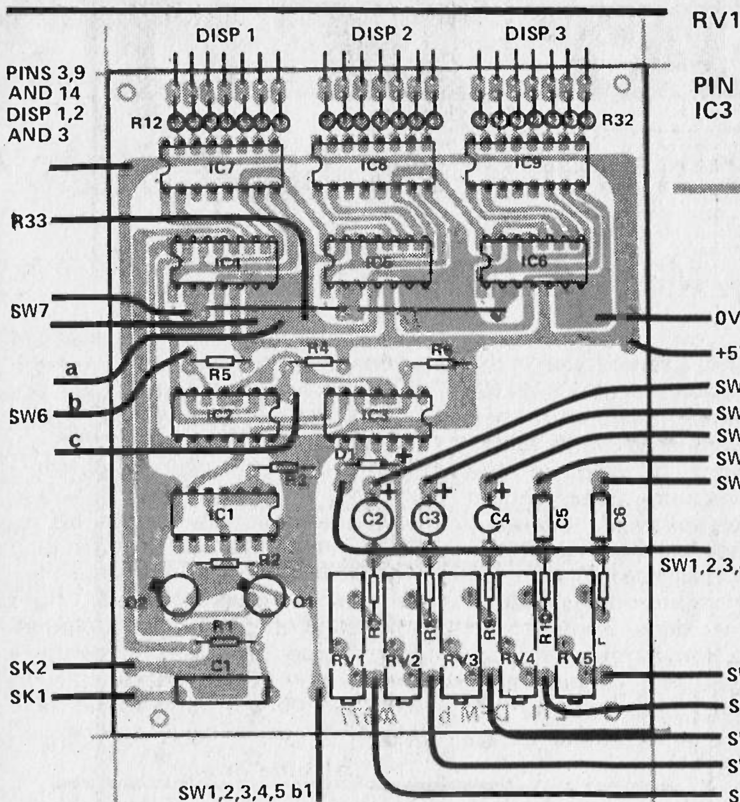


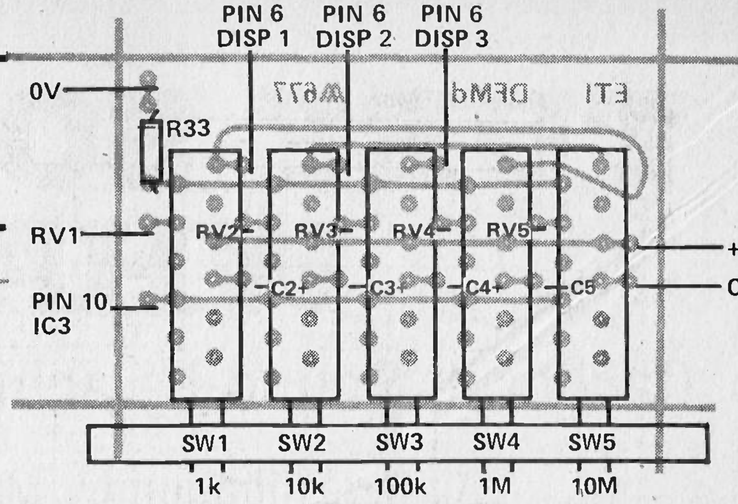
Above and below left: The foil side PCB patterns for the four DFM boards. These are all shown full size and from the 'foil side upwards' direction.
Below right: Component Overlay for the PSU board, DFMs. Reg. 1. is mounted off board.

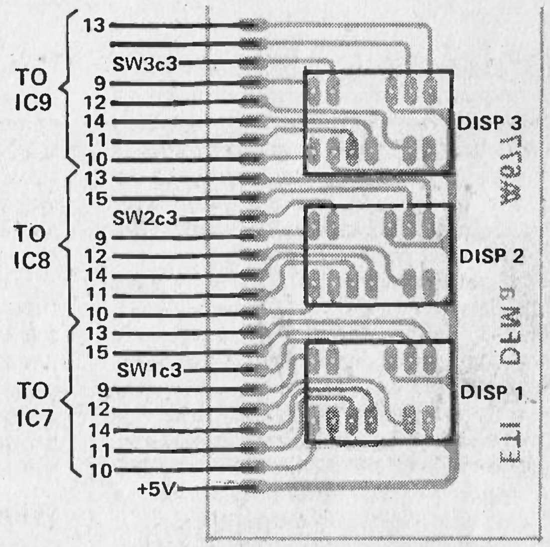




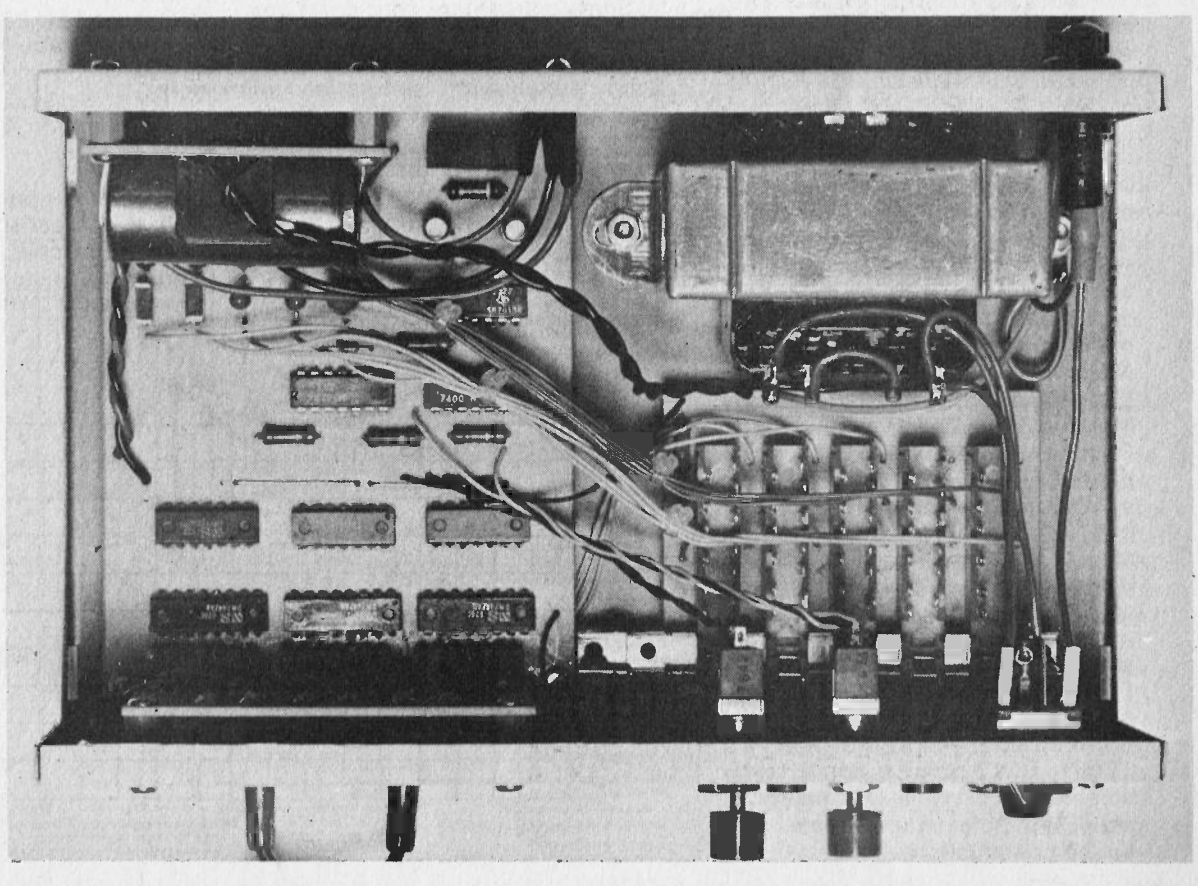
DIGITAL FREQUENCY METER







Above: Component Overlays for the main three printed circuit boards. Assembling these is the major part of the electronics work involved in constructing the DFM.



An aerial view of our meter. This shot shows the layout of the boards inside our case. To the top left is the PSU board mounted vertically over the main board.

TUBE SOUND — WHAT IS IT?

Can Tube Sound be achieved from Solid State?

An investigation by Mark A. Sawacki, M.Sc.

"Almost every musician has a favourite amplifier which may be so old it's held together with chewing gum and solder, but it produces exactly the sound he wants, so he has to carry it around with him, practically packed up in cotton wool to keep it from falling apart."
(Melody Maker, April 10 1976, Karl Dallas)

A reasonable man might be expected to ask the following: "If you want it to sound like tubes, why not use Tubes? and the two answers to this would be — reliability and weight — "When he plugged into a transistor amp, it didn't distort, it reproduced a more accurate sound, but he didn't like it, neither did the rest of the band . . ."
(International Musician and Recording World, July 1976, Ray Hammond)

IN ORDER TO UNDERSTAND the above quotes more fully an early history and problem presentation will clarify the situation. Recently there was a great hulabaloo about "tube sound" in the instrument market and way back in March/April 1976 Tony Reeves (Curved Air basist) told Melody Maker's Karl Dallas that — Dick Parmee, a Cambridge University graduate, currently with PA:CE as an Electronics Research Engineer, had analysed this dilemma and solved the problem of making a transistor amp sound like a tube amp.

Around the same time I visited another Cambridge factory — H.H. Electronics, where boss Mike Harrison showed me (with great pride) his new baby — V.S. Musician — a 100W RMS power combination amplifier, as well as his V.S. Musician Solo model, both *tube sound* constructions.

During that same year, the Japanese musical giant YAMAHA came onto the market with their G100 series of solid state amplifiers, claiming to capture the warmth of vacuum tubes without sacrificing the reliability, economy or portability of transistor amps.

This phenomenal "vacuum tube" market explosion has been in answer to a deep seated and widespread conviction amongst guitar sounds and certain other instruments, a tube amplifier will produce a far better *sound* than a transistor amplifier.

Filling a vacuum

Some elementary considerations should be taken into account before delving into this issue any further. Solid state with simulation of "vacuum tube sound" of the 60's is now a fact, although for people not *au courant* with the problem, it will appear as a regression, which is of course completely untrue. Since I intend to concentrate on the technical side of the *"tube* phenomena", I will leave all associated aspects, ie psycho-social, for others to deal with (only recently I heard it said that it was a mass hypnosis phenomena — upon which I prefer not to comment!)

In my analysis of the problem, I set up a small experiment. I tested eight internationally known instrument amplifiers, all from highly reputable manufacturers, and chosen as representative of both solid-state and tube technologies. The choice itself was made by a person who is not associated with the investigation, but who did have 15 years musical stage experience!

The first 'tube' group:

1. Marshall MK2 — Master Model 100W Lead Amp (GB)
2. Fender 160 PS Vocal Amp (USA)
3. Fender Guitar/Bass 300 PS Amp (USA)
4. Roost Session Master SM 100R Combination Amp (GB)

and these are presented in Table 1 with a comparison of the T.H.D. (Total Harmonic Distortion) in % plotted against specific amplifier power output in W.RMS as shown in Fig. 1.

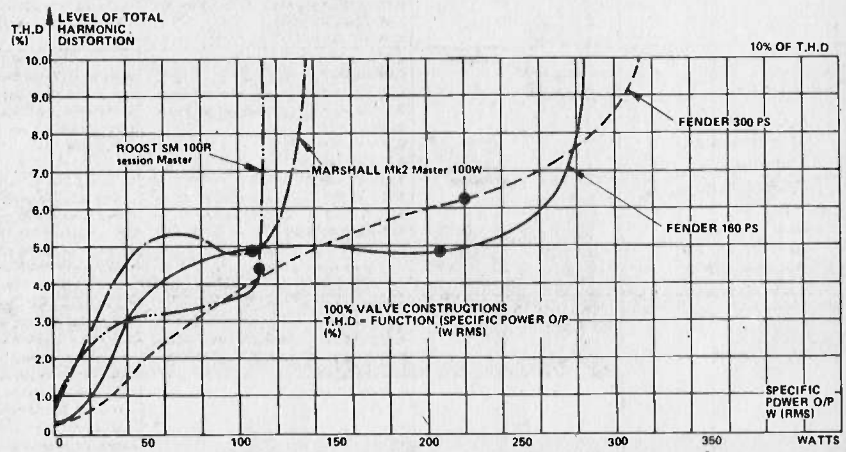


Figure 1.

● POWER AT ONSET OF CLIPPING
REF. STATED O/P IMPEDENCE
MEASURED WITH 1 kHz I/P SIGNAL

TYPE	COUNTRY of origin	POWER OUTPUT in Watts R.M.S., ref. 1 kHz	TOTAL HARMONIC DISTORTION measured at specified power level	COMMENTS
MARSHALL Mk.2 Master Model 100 W Lead Amp.	G.B.	136.1 W R.M.S. at 10% THD into 8 R 107.2 W R.M.S. at clipping into 8 R	4.75% at 100 W 5.25% at 80 W 5.05% at 50 W 3.81% at 30 W 1.45% at 10 W 0.57% at 1 W Measured into 8 R dummy load ref 1 kHz	Classical European design. Only slight changes since introduction 10 years ago. Still very successful. 'Marshall sound' - predominance of 2nd. harmonic in overload.
FENDER 160 P.S. Vocal Amp.	U.S.A.	285.0 W R.M.S. at 10% THD into 2.6 R 206.0 W at clipping into 2.65 R	4.95% at 160 W 4.82% at 100 W 0.42% at 10 W 0.23% at 1 W Measured into 2.6 R dummy load ref 1 kHz	Very special piece of equipment - original design for musician P.A. system. Incl. built in graphic equaliser & four channel mixer.
ROOST Session Master Combination Amp.	G.B.	110.5 W R.M.S. at clipping into 8 R	3.80% at 110 W 3.85% at 80 W 3.22% at 10 W 1.85% at 10 W 0.55% at 5 W Measured into 8 R dummy load ref 1 kHz	Relatively 'new' when compared to other three. Very orthodox in design. Predominance of 2nd. harmonic in overload.
FENDER 300 P.S. Bass Amp.	U.S.A.	312.5 W at 10% THD into 8 R 220 W R.M.S. at clipping into 8 R	5.94% at 200 W 5.14% at 150 W 3.74% at 100 W 2.14% at 50 W 0.38% at 10 W 0.26% at 5 W Measured into 8 R dummy load ref 1 kHz	Popular on both sides of Atlantic. Very high power delivery, quite high THD levels, but still respectable for tube amps. Predominance of 2nd. harmonic in overload.

Table 1.

Omission

Someone is bound to ask at this point why I did not include makes such as VOX, Orange, Carlsboro, or Leslie, and the reason is that this selection was purely a matter of individual preference (as well as limited space!)

The second "solid-state" group consists of:—

1. H.H. V.S. Musician Reverb 100W Amplifier (GB)
2. Pearl Vorg 102 Guitar Combination Amp (Japan)
3. Dynacord Emminent 200 vocal Amp (West Germany)
4. "MM" Electronics — AP 360 Dual Power Slave (GB)

and the results of experiments on this group are presented in Table 2 and Fig. 2.

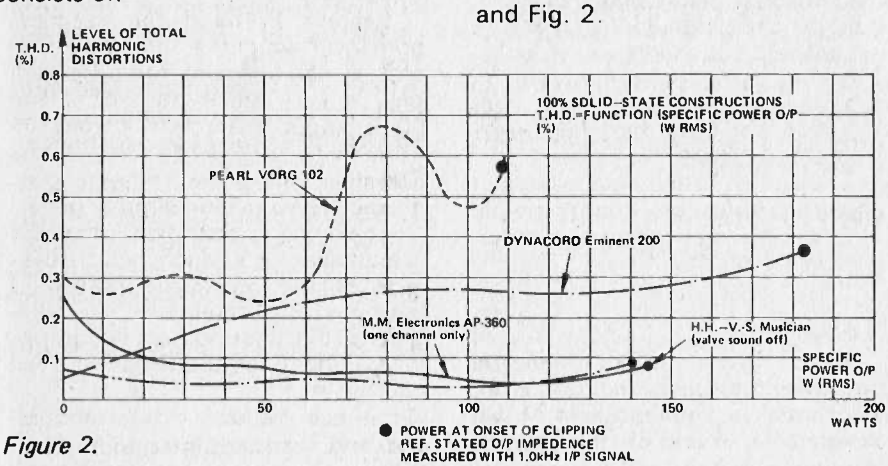


Figure 2.

TYPE	COUNTRY of origin	POWER OUTPUT in Watts R.M.S., ref. 1 kHz	TOTAL HARMONIC DISTORTION measured at specified power level	COMMENTS
H.H.V. S. Musician Reverb 100 W Amp.	G.B.	144.03 W R.M.S. at clipping into 4 Ohms	0.04% (20.5%) at 100 W 0.06% (13.9%) at 80 W 0.06% (13.8%) at 80 W 0.08% (14.5%) at 40 W 0.1% (13.2%) at 20 W 0.14% (9.8%) at 10 W 0.24% (3.2%) at 1 W Measured into 4 Ohms dummy load, ref 1 kHz	T.H.D. values in brackets correspond to 'valve sound' in 'ON' position. AS000 - 15/A - is the tone correction module. (Anatomical heart of 'valve sound').
PEARL VORG 102 Guitar Combination Amp.	JAPAN	108.16 W R.M.S. at clipping into 4 Ohms	0.47% at 100 W 0.67% at 80 W 0.28% at 60 W 0.27% at 40 W 0.28% at 20 W 0.26% at 10 W 0.30% at 1 W Measured into 4 Ohm dummy load ref 1 kHz	Quite a high level of THD as far as solid-state technology is concerned. Recently extremely popular due to proven reliability and good sound produced.
DYNACORD Eminent 200 System Amp.	WEST GERMANY	182.5 W R.M.S. at clipping into 2 Ohms	0.27% at 140 W 0.27% at 100 W 0.085% at 10 W 0.05% at 1 W Measured into 2 Ohms dummy load ref 1 kHz	Very popular on Continent because of good quality sound & reasonable price. Built-in EQ-graphic, reverb, as well as 6 channel mixing system and 100 W R.M.S. approx. monitoring amp.
'MM' Electronics AP - 360 Dual Power Slave Amp.	G.B.	140.28 W R.M.S. at clipping into 8 Ohms	0.080% at 140 W 0.04% at 100 W 0.04% at 70 W 0.035% at 40 W 0.04% at 20 W 0.075% at 5 W Measured into 8 Ohms dummy load ref 1 kHz	Very low level of THD by any standard. AP-36 is a Dual Power Slave Amp. of very universal character. Input sensitivity approx. 0.5 V R.M.S. Both channels driven simultaneously produce quite a high power level suitable for P.A. systems, stereo systems, discos, etc, etc.

Table 2.

Drawing in the threads

From Fig. 1 and 2 it is possible to draw the following conclusions; the transistor amp output is relatively distortion free and a typical THD figure across the whole power range would be about 0.15%. The figure of 0.05% is also realistic, even just prior to clipping.

Similar figures were also confirmed sometime ago by D. T. N. Williamson in a series of articles proposing standards of THD for audio amplifiers at full rated power output and they have been generally accepted as the target figures for high quality audio.

We found the average level of THD for our four tube amplifiers was somewhere in the region between 4.5% and 6.5% (for power levels measured at onset of clipping). When the power output was slightly increased, then the THD shot up rapidly reaching 10% with no problem!

This fairly high THD level however, is nothing new as far as tube technology is concerned, and even well-known European standards and specifications often rate power output at 10% THD for tube amps. To conclude then, the most important difference noticed in the performance between tube and transistor amps was the overload distortion characteristics of the output stages. Comparing the value (0.04 — 0.4% THD for solid state systems to 2.0 — 10% in tube constructions), it is quite easy to see the differences and consequently understand the changes in the specific character of both types of sound. All these THD tests were carried out with a single 1KHz sine input which is acceptable as a very useful standard. A practical set-up arrangement for this test is shown in Fig. 3.

Further into tubes

Before discussing the problem any further, a little mathematics will help in clarifying it. In order to examine the construction of sound (even single frequencies), it is necessary to use analysis of the tonal spectrum characteristics. The oldest historical model known was developed by a German physicist and physiologist, Herman Helmholtz in 1863, at present the most popular method used is the "Fourrier Array".

According to Fourrier's hypothesis, every periodical wave can be divided into a family of sinusiodal harmonic components (Fourrier

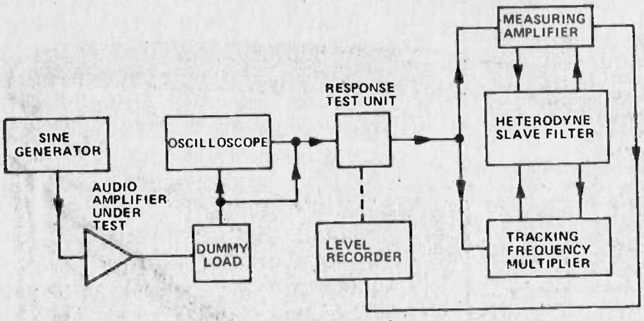


Figure 3.

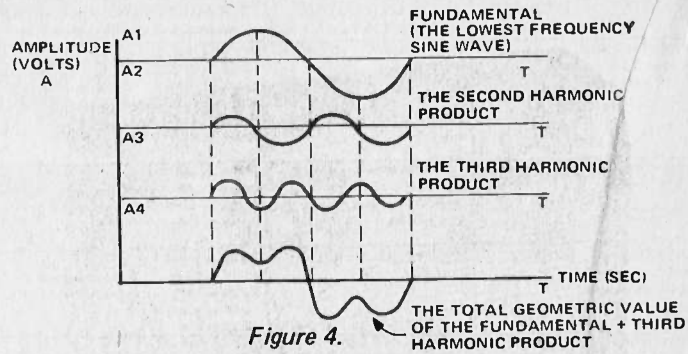


Figure 4.

components) where the lowest Fourrier component is equal to the fundamental frequency and the others are: 2f, 3f, 4f, and so on.

A 1Khz frequency sinusiodal signal will produce the subsignals, 2Khz, 3Khz, 4Khz etc. The THD waveform of an amplifier's output is quite a complicated construction, and its final shape depends not only on the operating frequency but also on its amplitude and phasing. See. Fig. 4.

The above analysis is quite complicated and requires a lot of calculation to obtain the results of the number of harmonic amplitudes. In the simplest one dimensional case there may be expressed as:

$$f(x) = \tfrac{1}{2} A_0 + \Sigma A_h \cos 2\pi (h\,x/a) + B_h \sin 2\pi (h\,x/a) \ldots$$

where $h = 0, 1 \ldots$

where $f(x)$ is known as the repetition interval (o, a) and it is required to determine the values of Fourrier components Ah and Bh. Naturally, we also have from the orthogonality of the sine as well as the cosine, the functions:

$$Ah = 2/a \int_0^a f(x).\cos 2\pi (h\,x/a)\,dx$$

where: $h = 0, 1 \ldots\ldots$

and respectively

$$Bh = 2/a \int_0^a f(x).\sin 2\pi (h\,x/a)\,dx$$

where: $h = 1, 2 \ldots\ldots$

To solve this problem computing numerical methods were developed and for more technical areas, the most common method is — Heterodyne/Resonance Analysers and Spectroscope filters.

In the circumstances it seemed worth while to obtain more detailed information, with special emphasis on the systems analysis. This is available by measuring the amplitude of each of the harmonic separately. For this experiment a band stop filter rejects the fundamental for measurements of THD, but the band pass filter can measure distortion components individually. The next two tests carried out on tube as well as solid state construction show the harmonic spectrograms in Fig. 5.

The amps tested had a similar power output of approximately 100W. RMS and again the difference is easy to see.

Tube:-

Apart from the fundamental frequency the dominant line is at 2f, with 'train' of amplitudes at 3f, 5f, 7f where 4f, 6f and 8f attain quite a low level.

The relatively distortion free solid-state amps have harmonics which are much lower but still with a 3f predominance. 4f, 6f, and 8f are difficult to measure because of very small magnitudes and their influence in our analysis can be easily ignored.

We can now say that tube amps have a tendency to clip on the *second* harmonic which gives that characteristic 'punch', whilst a transistor amp tends to clip on the third harmonic, which *may* create a quite unpleasant sound. Of course 'unpleasant' or 'pleasant' is really a question of personal taste.

Solutions

Bearing in mind the practical non-linear amplifier (and any real amp is non-linear to some degree) only a very small portion of the Input/output characteristic may be considered substantially linear. The curvature of the input/output char-characteristic (nonlinearities) generally give rise to distortion known as "non-linear" distortion, which consists of the previously presented harmonic distortion tests, in addition to the intermodulation group, where the intermodulation groups consists of:

Difference frequency Intermodulation and Transient Intermodulation

Even a brief look at other methods of testing shows the effects of nonlinear distortion in both types of amplifiers, but the

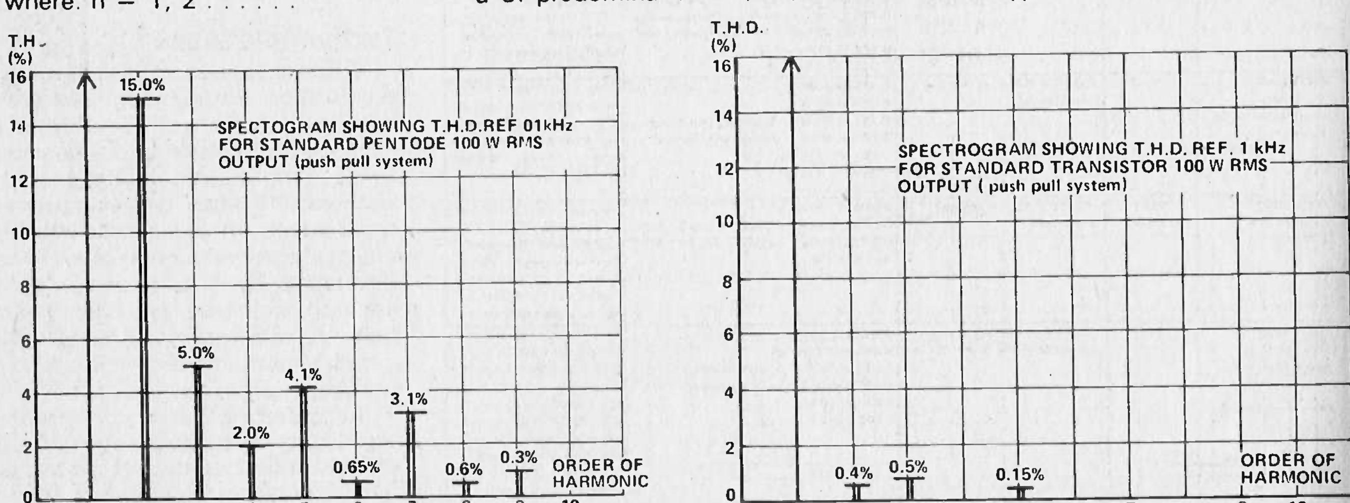


Figure 5.

The Yamaha G-100-B-212.,

The H.H. V.-S. Musician.

complexity of the work should I imagine be undertaken by scientists as part of a (really fascinating) research program.

Buttons for tubes

Let's look at what the Japanese YAMAHA engineers call the 'vacuum tube' sound, and what Cambridge H.H. Electronics claim as the 'valve sound' — As can be expected both are 'combotype' designs in principle but designed as a real effort to simulate the tube 'character'. The Yamaha G 100B 212 amplifier 's THD figures plotted against specific power output, ref. 1Kz is shown in fig. 6.

And now the H.H. V.S. Musician Reverb Model 100W — Fig. 7.

Both the Yamaha G100B 212 and the H.H. V.S. Musician were tested in conjunction with a good quality electric guitar and the test was carried out purely from the subjective sound quality point of view.

Subjected subjectively

Both amps performed satisfactorily, however, the sound which was obtained was entirely different in both cases. H.H.'s amp and gave a more dynamic sound, very harsh but with good sustain when the 'valve sound' footswitch was on. At the same time, the sound produced was very similar to an overworked valve amp, but the overall volume was controlled by the 'Master Volume Control'. This was found to be a useful feature when very loud volume is a problem, but when the wound-up 'tube amp' sound is still necessary. This particular feature has a wide range of applications as far as studio recording work is concerned.

After testing the Yamaha G 100B 212 amplifier, one important difference emerged, namely the total absence of an ON/OFF pedal footswitch. However, the manufacturer provided a special rotary 'Distortion Control' specially for this function. By experimenting with the different distortions and volume settings, an unusually wide variety of effects was obtained, but to get really clean chord playing, rhythm guitar work or certain keyboard instruments, this control was set on the 'minimum' of its rotation.

The Yamaha G100B 212 compares with the H.H. V.S. Musician producing a much more mellow tonal character, whether it was played softly or at a crashing volume, but on the other hand the H.H. amp would be much more effective in pure rock practice or whenever a lot of high frequency was desired.

To summarise, one can say that as long as the manufacturers analyse those designs on the market with the best tube performance, and then try to obtain this by duplicating it will be possible for him to construct a solid state amp with realistic tube sound. But, of course, whether musicians will go for one or the other, remains a question of personal preference. ●

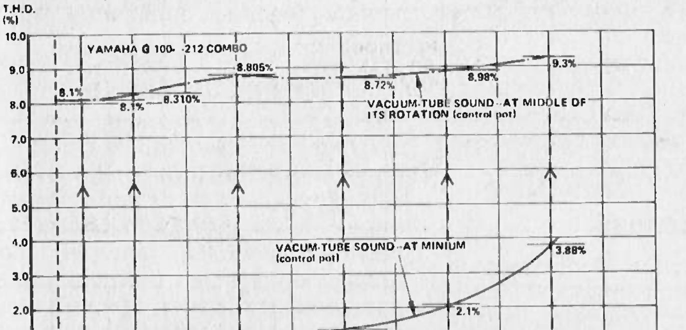


Figure 6.

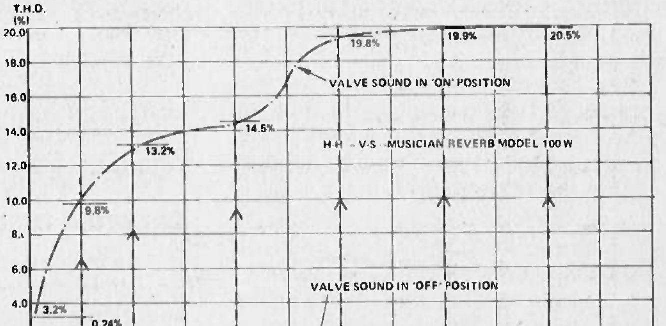


Figure 7.

TUBE SOUND – on the rebound?

GEORGE CHKIANTZ
AND
RICHARD ELEN

TWO PROFESSIONAL SOUND ENGINEERS HAVE A NEW AND HIGHLY CONTROVERSIAL THEORY

IMAGINE YOU ARE SITTING in front of a pair of monitor speakers, listening to some master tapes. The first track you hear is a very good, high quality mix, such as one could find on a fair proportion of modern albums. But the second one is something else.

It too is of very high quality, but in addition, it seems to possess an undefinable 'something'. Could one call it 'clarity'? 'Rightness'? It's hard to put into words, but there is something about that mix which makes it stand out from the rest — an unknown factor which is nonetheless sufficiently noticeable for you to be able to tell the difference every time. Yet you know that the two recordings were made in the same studio, on the same desk, the same tape machines — every link in the chain from microphone to master tape was duplicated. But with one subtle change. The first mix was monitored on a pair of loudspeakers driven by a modern, high-power transistor amplifer with a rated output of 300 W.

The second, on the other hand, was reproduced using a pair of old, glowing tube amps running a mere 50 W — and unaccountably sounding just as loud.

Why the difference?

Tubes versus Transistors

It is important to state at the outset that this is intended to be a *qualitative* analysis of points that we have found to be significant over the past few years during our work as sound recording engineers. Some of the points we are raising are nowadays being discussed, but not all. One reason for this is that several of them are not easily measured — rather, they are experienced.

Until a couple of years ago, the standard answer you *would* get from any audio engineer worth his salt when asked the question, *"Why do tube amps sound better than transistors?"* would have been, *"Well, tubes produce more even-harmonic distortion, and you like the distortion, which is more 'musical' than the distortion produced by a transistor amp. And after all, transistors are more distortion-free than tubes ever were."* He would not give the same answer today, unless he spent all his time listening to test instruments instead of real signals.

A big question that needs to be discussed is, *"Is distortion just a variable you can measure with test equipment or is high fidelity, in the words of a well-respected manufacturer, 'the closest approach to the original sound'"?* What does it mean to have 0.0001% distortion when it *still* doesn't sound like the original (and when the speakers are adding a further 2%)?

Distorted assumption

The tacit assumption in the design of so-called 'tube sound' musical instrument amplifiers is: *"Musicians like the sound of a tube amp. Tube amps may have more distortion. Therefore musicians like the distortion."*

Manufacturers thus have produced special boxes which may introduce up to 25% second harmonic distortion and come up with a 'tube-sound' which some musicians like, but many describe as 'sounding more like a fuzz-box than a tube amp.

This seems to suggest that maybe we aren't just talking about distortion. And when we come on to studio monitoring amplifiers, we find that there is little significant difference in distortion figures between solid state and thermionic equipment. It must be remembered, however, that we are *not* suggesting that *all* tube amps are 'nice', or that *all* transistor amps are 'nasty'. The major problems inherent in transistor amplifier design are the direct result of developments made on tube circuits. Also, these problems, as will be seen, do not necessarily apply to low-power transistor amplifiers, as usually encountered in Hi-Fi systems, for example. The problems seem to arise in the method and amount of negative feedback application.

Feedback and its Misuse

There are two major ways of applying negative feedback to an amplifier: in the form of an overalll feedback loop from one end of the amp to the other, or by the application of an individual feedback path per stage. Tube amplifiers like the Leak 'triple-loop feedback' series primarily used the latter method with little additional overall feedback (typically about 10dB). Certain other tube amps relied mainly on a high degree of overall feedback, producing a sound much closer to that of transistor amps. The main reason for the choice of overall feedback was probably economic: a glance at the handful of components underneath a certain well-respected Hi-Fi amplifier will indicate how cheap they must have been to produce. But it appears that it is the amps which used a feedback loop on each individual stage that produce the oft-mentioned 'tube sound'.

OK, you may well ask, what's so bad about high levels of overall feedback? Well, there are three main reasons, one well-known, the others totally unheard of.

T.I.D.

Almost everyone has heard of manufacturers recent efforts to eliminate the newly-discovered menace of Transient Intermodulation Distortion, but in case you missed the blurb, it may be broadly described as the type of distortion produced when the rising edge of a waveform applied to the amplifer input is rising faster than the total transit time (i.e. the time taken for a signal to pass through the amplifier). This may result in a spike of pure, 100% distortion, whose output power may well be limited only by the peak handling capacity of the PSU.

There are two ways of dealing with this. Either the initial stages of the amplifier must be capable of taking the maximum level of applied signal with no feedback applied, or the rising edges must be 'slowed down' to avoid this effect. This latter solution is the one generally used today, and is equivalent to placing a low-pass filter on the input to limit the frequency response to something below the level of 'flat from DC to Light'.

TID was not so prevalent in certain tube amplifiers because of a) locally applied feedback on the stage itself, thus reducing the transit time (and perhaps electrons crawl more slowly across semiconductors than they travel *in vacuo*?), and b) because the linear portion of the tube transfer curve is longer than the corresponding area of transistor characteristics.

The Ricochet Effect

A point often not considered in amplifier design is that as negative feedback across the amp is increased, the 'back-to-front-impedance' (ie the impedance from the output looking back up the feedback loop towards the input) is reduced. Some high-power transistor amps have loops with an impedance of only a few ohms. This is because transistors are generally low impedance devices.

But how does this affect the sound produced? Let us remember that amplifiers are most often used to drive loudspeakers which are placed in fortunately-not-normally-anechoic rooms. A loudspeaker can also be considered as a microphone, and as such it is evident that whilst sound is being projected into the room, it is also being picked up (after multiple bouncing around) by reflection and reconverted into voltages, which will appear at the amplifier input, if the back-to-front impedance is sufficiently low. Another way of looking at this would be to say that the loudspeaker is loaded by standing — and other pressure-waves in a non-linear manner with respect to frequency. The total effect is that the same source-/amplifier/speaker system will sound different in various locations, whilst a system without this effect will vary only slightly according to room acoustics. We noticed this whilst working in a studio fitted with Tannoy Monitor Reds in Lockwood cabinets, (over)driven by well-known 300 W transistor power amps. The act of panning a signal from left to right changed its sound dramatically, far more than one would expect in an acoustically treated studio environment. Replacing the studio amps with our venerable bottles, however, removed the effect almost entirely, most other factors (including subjective SPL) remaining equal. The transistor amps, of course, had a high level of overall feedback, unlike the valve system.

Frequency Response

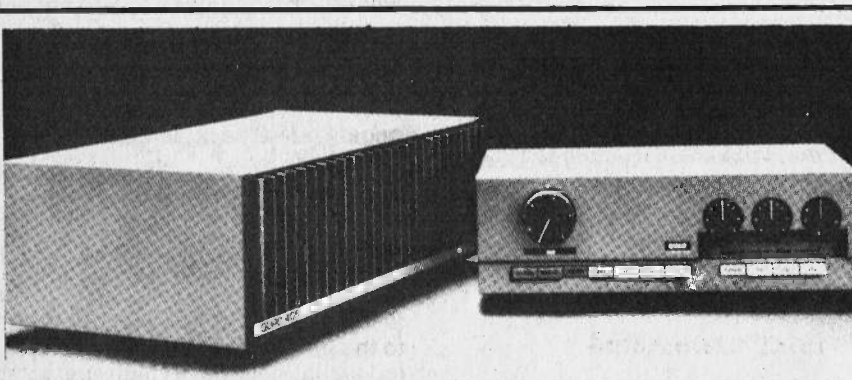
One problem with evaluating the Ricochet Effect and, to some extent, TID, is that they are very hard to measure objectively. You should be able to pick them out by listening critically, but using normal test procedures it will often be found that there is little descernible difference between 'nice' and 'nasty' sounding amps on the test bench. However, one point of comparison has emerged. It was found that a number of 'nasty' sounding amplifiers had a very limited frequency response with the feedback loop removed (one specimen managed 3 dB points at 250 Hz and 8 kHz approx., with 40 Hz and 15 kHz over 10 dB down!), whereas our 'nicest' amplifier was found to be nearly flat under similar conditions (3 dB points at 40Hz and 15 Khz approx.). However, this point required further investigation, and removing a feedback loop is not always as easy as one would wish!

As far as the ricochet effect is concerned, we have so far found no other satisfactory explanation for the phenomenon (try it and see!).

Conclusions

The result of a few years of subjective listening tests has indicated very firmly to us that the 'second harmonic distortion' explanation for the 'tube sound' *IS LARGELY A FALLACY.* And after all, we were always taught at college that push-pull output stages *cancelled out* even harmonic distortion. Where is it all coming from?

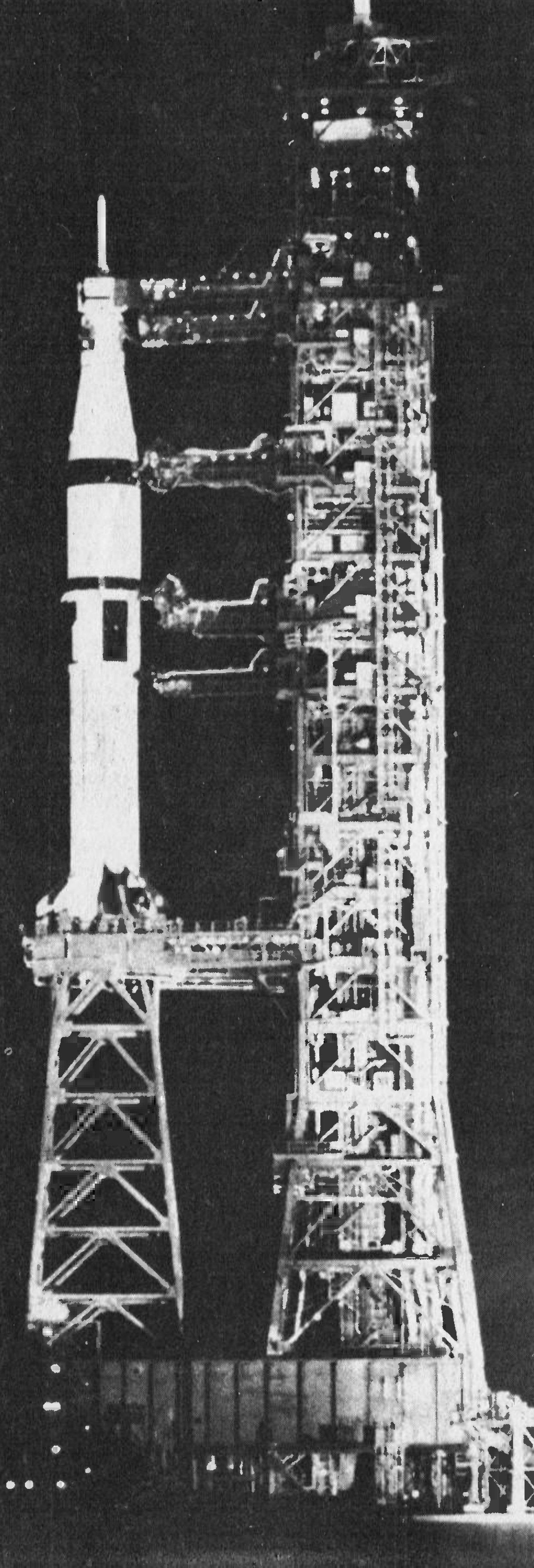
We feel that the major reason for *certain* tube amps sounding better than *certain* transistor amps is largely the result of the use of smaller amounts of overall feedback in the former, resulting in less TID and a reduction of what we have termed the hypothetical 'Ricochet Effect'. Some transistor



The Quad 405 'current dumper'. This hi-fi amplifier is finding support from the advocates of thermionic emission amplification for its 'non-transistor' sound. A possible explanation of this is the absence of the usual feedback arrangements in the circuit. Operation of the 'current dumping' output stages is basically that of a high current switch, controlled by a low power class A amplifier.

continued on page 35

THE KENNEDY SPACE CENTRE



By Mike Howard

PERHAPS THE BEST known of the NASA installations lies on the east coast of Florida midway between Jacksonville and Miami in the area known as Cape Canaveral. Here stands the major U.S. spaceport and launch facility now named the John F. Kennedy Space Center. Within the Center are two separate federal facilities, one operated by NASA, the other by the U.S. Department of Defense.

The NASA Kennedy Space Center (KSC) occupies some 88,000 acres of land and water on and around Merritt Island extending 35 miles north-south with the Atlantic on its eastward side and the Indian River to the west. Adjacent to NASA KSC is the Cape Kennedy Air Force Station managed for the Department of Defense by the U.S. Air Force. This is the prime station of the Eastern Test Range which stretches southeast 10,000 miles to the Indian Ocean.

KSC's History

On 8 July 1947 Cape Canaveral was formally approved as a test area for long-range guided missiles under the management of the USAF. This site was chosen largely because of the chain of islands, spread out in a south-easterly chain as far as Ascension Island, which were suitable for the tracking stations required to monitor the flight of research missiles. Shortly after the Cape was established, Congress authorised the acquisition and construction of the Atlantic Missile Range, later renamed the Eastern Test Range and extended into the Indian Ocean.

The Cape became operational as a missile test centre in May 1949 with the first launch on 24 July 1950. That first vehicle was a 56ft missile called WAC-Bumper. Its first stage was a modified V-2 with a WAC Corporal as the upper stage. The launch pad was a concrete slab, the service structure a painter's scaffold, and the launch team utilised a tar-paper shack (formerly used as a bath house) as the launch control centre. From these meagre beginnings would develop the most advanced launch facility in the world.

In October 1958 the National Aeronautics and Space Administration was established. Just 3 days short of a year earlier the Soviet Union had orbited Sputnik 1, the Earth's first artificial satellite, followed a month later by Sputnik 2 with a canine passenger. It was small consolation that a U.S. Army team headed by Dr Kurt H. Debus, who later became Director of the Kennedy Space Center, had succeeded in placing Explorer 1 into orbit on 31 January 1958 since it was only a fraction of the weight of the Sputniks.

With this state of affairs NASA came into being with the peaceful exploration of outer space as its goal. To this end the Launch Operations Center was established at the Cape. On 5 October 1958 Project Mercury got underway and on 5 May 1961 Alan Shepard made a brief sub-orbital flight to become America's first astronaut; however, once again, the Russians had beaten the U.S. because on 12 April of the same year Yuri Gagarin had become the first man to orbit the Earth. In 1960 John F. Kennedy had been inaugurated as President of the United States and on 25 May 1961, just 20 days after Shepard's flight, he committed the U.S. "to achieving the goal, before this decade is out, of landing a man on the Moon and returning him safely to the Earth . . ."

The results of that commitment are now well known in the steady progression up through Mercury, Gemini, and finally to Apollo. Kennedy was not to see his dream fulfilled. On 22 November 1963 the President who pushed America to the Moon was assassinated while riding in a motorcade through Dallas, Texas. Seven days later the new President, Lyndon B. Johnson, officially directed that the NASA Launch Operations Center and Station No. 1 of the Atlantic Missile Range be renamed the John F. Kennedy Space Center.

KSC Facilities

Few people realise that the Kennedy Space Center is also a wildlife refuge. On 2 June 1972 the entire spaceport was incorporated into the Merritt Island National Wildlife Refuge which comprises some 145,000 acres of land and water. In spite of the occasional rocket blastoffs, racoons, bobcats, alligators and wild pigs roam the scrubland; rare species of birds, such as the Bald Eagle and Peregrine Falcon can also be found within the bounds of the space centre. When the KSC site was acquired NASA took over some 3306 acres of citrus trees which it now leases to their former owners. Also on site are 2 fishing camps and three private burial grounds.

Functions of the Space Center's life that are not required at the launch complexes are mostly grouped together in the Industrial Area. Here in the Headquarters building are located the Director of the Kennedy Space Center, procurement, programme management, legal, and other support functions.

The largest structure within the Industrial Area is the Manned Spacecraft Operations Building (MSOB) in which Apollo-type spacecraft underwent modification, assembly, and preliminary checkout. Within the MSOB are two 50ft altitude chambers capable of simulating altitudes up to 250,000 ft for the testing of spacecraft and systems. Here, also, are the astronaut quarters and medical facilities. Instrumentation to receive, monitor, process, display and record information from the space vehicle during pre-launch, launch and immediate post-launch activities, is located in the Central Instrumentation Facility. The Flight Crew Training Building is the KSC equivalent of the Mission Simulation and Training Facility at the Johnson Space Center, Houston. Here astronauts and flight controllers practise for manned flights utilising the computerised mission simulators.

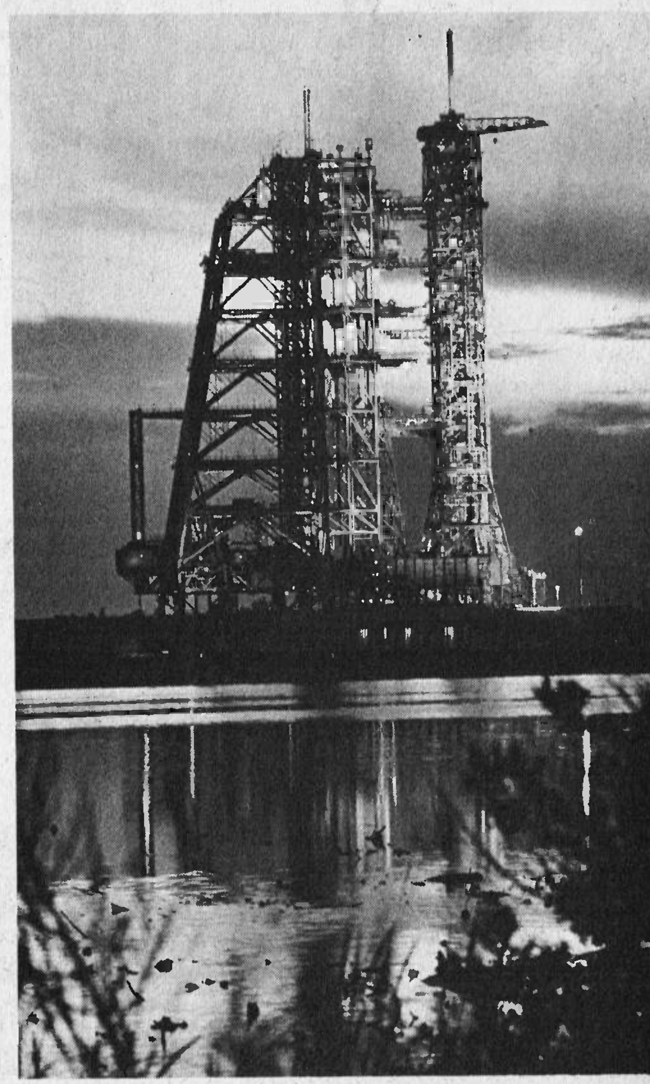
Other facilities within the Industrial Area include a cafeteria, fire station, occupational health building, security offices, warehouses, and specialised laboratories for spacecraft pyrotechnics checkout.

The Kennedy Space Center is also responsible for the launch of unmanned space vehicles from the Western Test Range at Vandenberg Air Force Base in California.

Launch Complexes

Within the Kennedy Space Center are fourteen main launch complexes plus launch and test facilities for Polaris, Poseidon, and Minuteman missiles. Some sites are no longer operational but all have played their part in the history of U.S. space exploration.

Heading northward from Port Canaveral the first call is at the pads of Complexes 5 and 6. It was from here that the early Mercury-Redstone flights began. Today the pads are part of the Air Force Space Museum and are marked by a plaque commemorating Shepard's 1961 flight and a full-scale Mercury-Redstone space vehicle on the site of that historic launch. The Museum also incorporates Complex 26 from which Explorer 1 became the first U.S. satellite. Moving on, the next stop is the Delta launch complex 17. From this area the versatile Delta vehicle has launched many well known unmanned spacecraft including Echo, TIROS, Relay, Telstar, Early Bird and Intelsat. Further north are the twin pads of complex 36 from which the Atlas-Centaur launches lunar and planetary spacecraft including Surveyor, Mariner, and Pioneer. Launch Complexes 12 and 13 were once the launch sites from which Atlas-Agena vehicles boosted Ranger and Lunar Orbiter spacecraft on their moonbound journeys. More Atlas-Agenas ascended from Complex 14 during the Gemini programme to place target docking vehicles into Earth orbit. This site was also used to launch the later Mercury missions using the Atlas booster. Moving

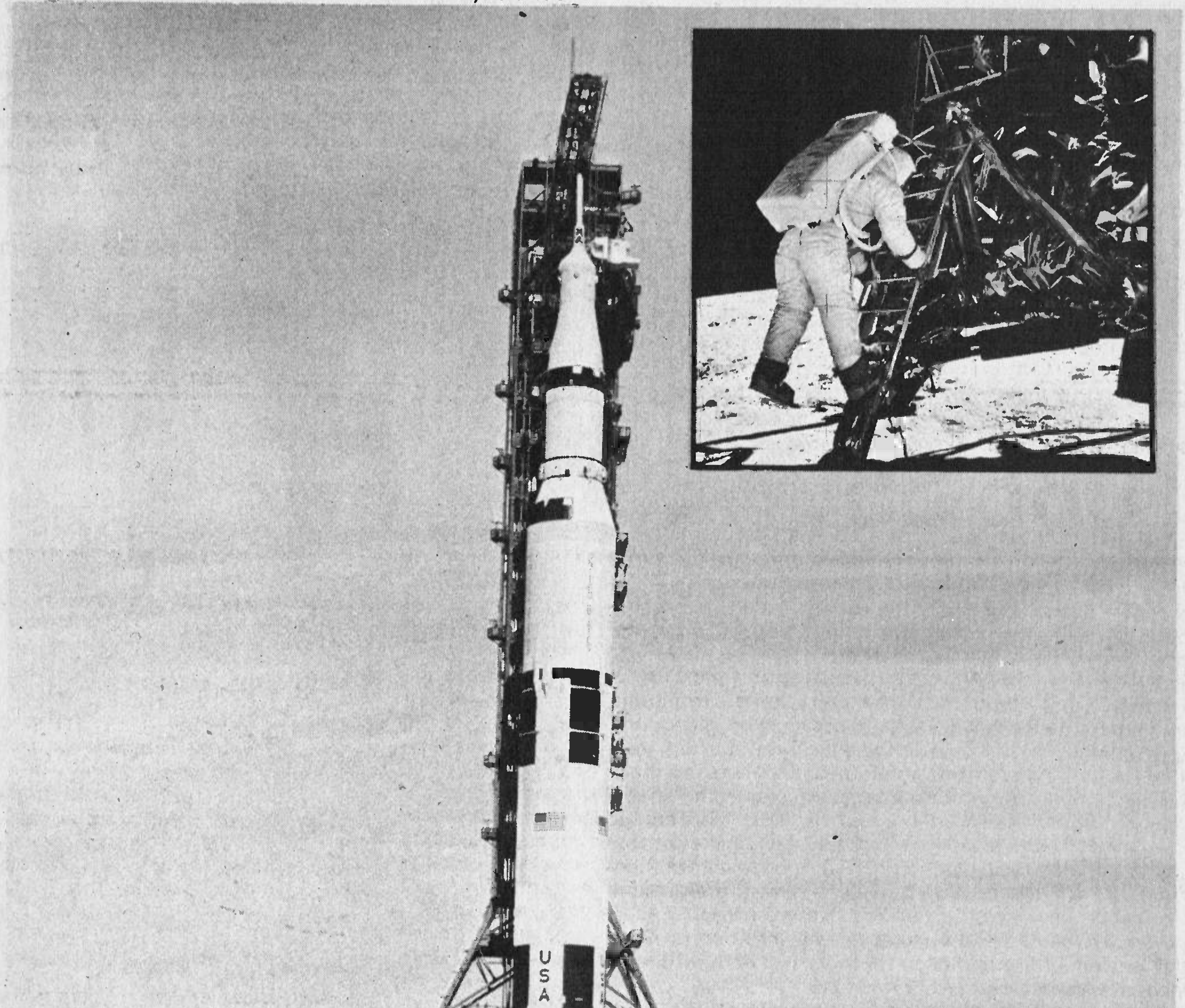


on again we encounter Complex 16 which was utilised for Apollo Service Module static tests. Complex 19 was the launch site for the Titan II vehicles that placed 10 Gemini crews into Earth orbit during a 20-month period commencing in March 1965. Complex 34 and the adjacent Complex 37 were used for Saturn I and IB flights preceding and during the Apollo programme. The first manned Apollo launch took place from Complex 34 on 11 October 1968 when a Saturn IB lifted the Apollo 7 crew to orbit. Both complexes have since been deactivated and dismantled.

The next area reached is the Air Force's Integrate-Transfer-Launch (ITL) Facility. This site includes Launch Complexes 40 and 41. From here the Air Force conducts its own launch programme using Titan III-C vehicles. Tital III-E/Centaur vehicles which NASA uses to launch Helios solar probes and Viking Mars landers are also launched from LC-41 when the facility is under the operational control of NASA. The last stop in the tour of launch complexes is at the twin pads of Kennedy Space Center's Launch Complex 39.

Launch Complex 39

Inspecting Complex 39 is like moving to another world where everything is larger than life. The main components are the Vehicle Assembly Building (VAB), the Launch Control Center, three Mobile Launchers, two Crawler-Transporters, the Crawlerway, a Mobile Service Structure, and two launch pads. Standing 526 ft high, 716 ft in length and 518 ft wide the Vehicle Assembly Building covers 8 acres of ground. Due to its immense size the VAB can sway up to 12 inches in strong winds and is equipped with a gravity ventilation system which forces a complete change of 130 million cubic feet of air every hour to prevent condensation and fogging within the structure.



Once the space vehicle has been assembled and checked out in the VAB aboard its Mobile Launcher (ML) it is ready for transfer to the pad. To accomplish this one of the Crawler-transporters moves under the ML and raises it ready for transfer. This vehicle itself weighs some 6 million pounds, is 131 ft long, and 114 ft wide. At a maximum speed of 1 mph the transporter moves out of the VAB carrying its precious cargo along the specially constructed crawlerway, the overall width of which is equal to an 8-lane highway. On arrival at one of the 3000 ft wide pads the transporter gently lowers its load and backs off. Pad A is some 3.5 miles from the VAB while Pad B is about a mile further on. In the centre of each of the octagonal pads is a 390-by-325 ft reinforced concrete hardsite, the top elevation of which is some 50 ft above sea level to allow the rocket's thrust chambers to rest above a 650-ton mobile flame deflector.

The Mobile Service Structure weighs over 10 million pounds and contains 5 service platforms from which the space vehicle can be serviced at the launch pad. This 410 ft high structure is also moved by the crawler-transporter.

Alongside the VAB, and connected to it by an enclosed bridge, is located the Launch Control Center (LCC) — a far cry from the blockhouses at other launch pads. The first floor of the LCC contains offices, a dispensary, and a cafeteria. The second floor houses telemetry, measuring, and check-out systems used during assembly in the VAB and later at the launch pad. On the third floor are the four firing rooms (only 3 are fully equipped) and their respective computer support rooms. Viewing of the firing rooms and the launch area is possible through specially laminated and tinted glass windows on the LCC's mezzanine level.

To date Launch Complex 39 has been used to launch Saturn V's for the Apollo and Skylab programmes and Saturn IB's for Skylab and the Apollo-Soyuz Test Project. In the future the Space Shuttle is planned to use complex 39's facilities. To accommodate the different vehicle, modifications to the Launch Pads, Mobile Launchers, and VAB will be necessary and these are already the subject of design contracts. In addition, since the Shuttle will return to a runway landing, a 15,000 ft long by 300 ft wide landing strip with associated overruns, apron, taxiway and access roads has been constructed to the north-west of the VAB on a northwest-southeast alignment.

Epilogue

The John F. Kennedy Space Center is also a site of great historical significance. During construction of the Center archeologists unearthed traces of pre-Christian human activity, Indian burial mounds and refuse piles, and signs of French and Spanish occupation. The petrified bones of prehistoric mammals were dredged up from the Banana River at the same time. It was with these discoveries in mind that Professor Charles Fairbanks of the University of Florida remarked that "this was one of the areas where Western civilisation came to the New World, and now it is the area from which our civilisation will go forth to other worlds".

Acknowledgements

We thank Spaceflight for their kind permission to reproduce this article.

TUBE SOUND

continued from page 31

amps — mainly low power types — sound particularly good because they have better linearity, lower transit times, etc. and hence less Ricochet and 'less objectionable' TID (the 100% momentary distortion of TID is far more painful if the amplifier is capable of delivering 500 W than if it can only manage 2!)

And more conclusions

We have found that a well-designed, high quality tube amplifier can provide adequate power reserve (remember that a Tannoy Red, for instance, only takes 35 W, not 300!) and distortion figures and is capable of reproducing the textural subtlety of the original performance with ease, whatever the apparent performance on non-typical test tones may indicate. After all, it is usually music we are listening to, with all its transients and non-cyclic waveforms, rather than what to us are rather boring sine waves. This aspect of 'high-fidelity' — the ability of the system to reproduce the subtle nuancesoriginal instruments — is something that few transistor amplifier (with the possible exception of amps like the Quad 405, with its totally different feedback 'technique,' the H/H monitoring amplifier with its special attention to TID, and the Vertical FET amplifier (Yamaha) which seems to exhibit all the 'niceness' of a valve amplifier), have shown themselves able to do.

Of course, a few people will contest our results, which are admittedly only preliminary: we say to them, listen for yourself and see. it is likely that more distortion will be caused by a closed mind than will be exhibited by the (tube) amplifier under examination!

ABOUT THE AUTHORS:

George Chkiantz is a freelance sound recording engineer of long experience with many of the world's top recording groups, including Family, King Crimson, Led Zeppelin (including 'Whole Lotta Love'), Hawkwind and many others. He also received a Gold Disc for the Rolling Stones album 'It's Only Rock 'N' Roll.

Richard Elen is Studio Manager of KPM Sound Studios, Denmark Street, a part of EMI Music Publishing. He is quite clever as well.

Both are moderately sensible. ●

introduction to THE OSCILLOSCOPE

OF THE MANY INSTRUMENTS required to service, test and maintain electronic systems, the cathode-ray oscilloscope must be the most versatile and useful. Other names are derivatives from the full name – the C.R.O., CRO (pronounced crow), oscilloscope and scope. Early works also refer to it as an oscillograph.

THE CATHODE RAY TUBE

The first cathode-ray tubes were experimental, designed to investigate the nature of beams of particles produced in thermionic-diode arrangements operating at extremely-high voltages.

Figure 1 shows the three stages in developing the basic cathode ray tube. Fig. 1(a) is a thermionic diode — a tube diode. The cathode, heated by the current passing through it, emits electrons into the space around it. These, being negatively charged, are attracted to the positive anode.

The greater the voltage between the cathode and anode the greater the velocity of the electrons. If a hole is made in the anode, as in Fig. 1b, many of the electrons will pass through, forming a diverging beam on the other side of the hole. When a phosphor powder is placed on the inside of the tube the electrons reaching it cause it to glow as they give up their kinetic energy. The powder re-emits this energy as photons of visible light. Early researchers' tubes did little more than this. The nature of cathode rays was studied in the early 1900s by such famous names as Goldstein, Braun, Crookes, J. J. Thompson, Rontgen, Coolidge and Dumont. Experiments showed that the beam could be deflected by a permanent magnet and by electro-magnetic and electrostatic fields. Prior to 1897 interest had been in physical-science investigation – not in the measurement of electronic signals. Then in 1897 K.F. Braun produced the first basic measuring device from the CR tube.

FURTHER DEVELOPMENTS

However for the CRO to become a useful, practical instrument more development was needed. From Fig. 1b it can be seen that the beam of an elementary device is badly defined and floods over the entire area of the phosphor. A tube or grid arrangement placed between the cathode and anode causes the beam to pass through the anode more cleanly, because of the negative repulsive effect of this tube assembly. The whole assembly – cathode, anode, grids and tube – is

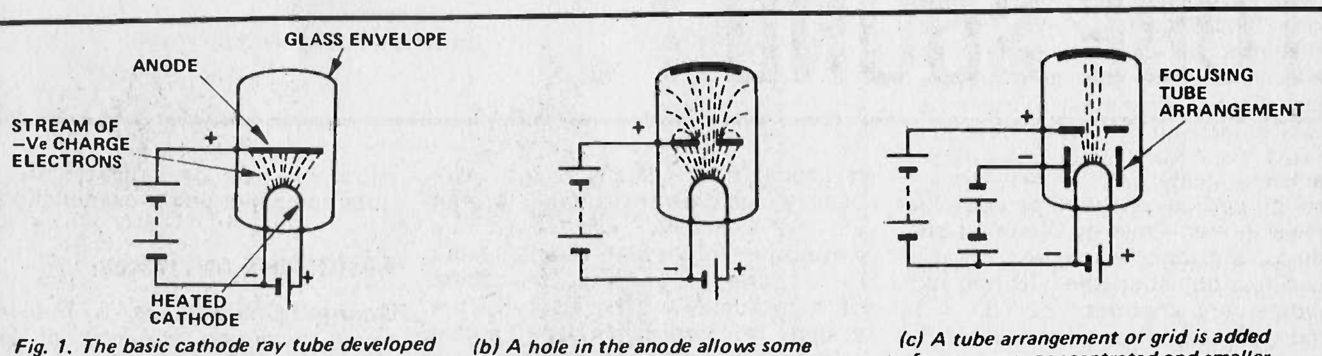


Fig. 1. The basic cathode ray tube developed through stages to provide a gun aimed at a screen.
(a) Thermionic diode in which the cathode is self-heated to cause liberation of electrons which move to the positive anode.
(b) A hole in the anode allows some electrons to pass through to the phosphorescent screen.
(c) A tube arrangement or grid is added to form a more concentrated and smaller size beam.

Phosphor European/ U.S. code	Fluorescence	Phosphorescence	Persistence	Burn resistance	Relative luminance	Comments
GP/P2	Bluish-green	Green	10 µs–1 ms	Medium high	55%	Medium speed oscillography
GM/P7	Purplish-blue	Yellowish-green	100 ms–1 s	Medium	35%	Low speed oscillography.
BE/P11	Blue	Blue	10 µs–1 ms	Medium	15%	Best photographic writing speed.
GH/P31	Green	Green	10 µs–1 ms	High	100%	General purpose oscillography. Brightest available phosphor.
GR/P39	Green	Green	5–100 ms	High	50%	Sampling oscillography.

Fig. 2. Chart showing characteristics of oscilloscope screen phosphors.

called the electron gun. Its full design is quite complex: Other elements are used to make electron-lenses (akin to optical lenses and light) to provide focus control and intensity control, the former adjusts the spot shape and size on the screen, the latter the current flowing in the electron beam.

The choice of phosphor on the screen determines the persistence (the length of time the spot glows after removal of the beam) of the display. The storage effect of various phosphors enables CROs to be made so that beam energy can be dispersed as light over time durations varying from microseconds to milliseconds. Fig. 2 is a guide to the selection of a phosphor. Manufacturers often offer a choice of screen persistence values to suit various applications. Fast moving spots, where the spot is likely to reappear on the same point in a short time, require short persistence. Long-persistence screens are suitable for slowly changing signals. (See the discussion of storage methods in the next part.)

ELECTROSTATIC DEFLECTION

The next refinement provides a method by which the beam can be made to deflect under the control of electrical signals. Fig. 3 shows how this is done for one axis, using the electrostatic method. A voltage difference of zero between the deflection plates allows the beam to pass along the tube axis undeflected. Any voltage differential will cause the beam to be deflected towards the more positive plate. Thus we have a way to cause the beam to move in the vertical direction (called Y-axis or Y plates). A further two plates set at right angles to these (the X plates) will cause the beam to deflect in the horizontal plane when a voltage is similarly applied to them. Beam-intensity control by electrical means is defined as the Z control.

Electrostatic deflection is the easiest to deploy for voltage measurements because deflection is proportional to applied voltage. Small cathode ray tubes usually use electrostatic deflection. Large tubes, such as those used in television systems or large-screen teaching oscilloscopes, usually use magnetic deflection because electrostatic deflection would require very high deflection voltages. These do not have deflection plates set inside the tube, but make use of magnetic fields created by electromagnet coils placed around the neck of the tube. The deflection in this case is approximately proportional to the current in the coils.

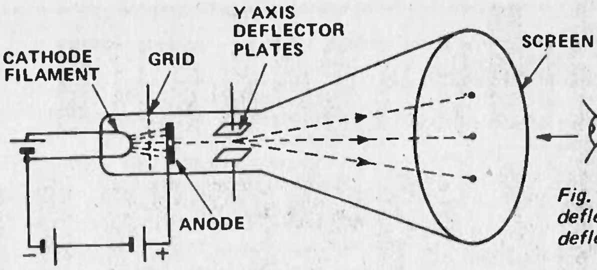


Fig. 3. The electron beam can be deflected by voltages applied to deflection plates.

Cathode ray tube design (for CROs and TV) has remained reasonably static since the late 1930s, the only obvious differences being in the linearity of beam sweeps and the shortness of tubes for a given screen size in television applications. Figure 4 is a modern oscilloscope with the cover removed to show the tube. From the instrument viewpoint the differences have been improvements in frequency response, spot control, linearity of sweep and a wider choice of phosphors. In addition the development of tubes with more than one gun and deflection system (some dual-beam oscilloscopes, but not all, use separate beams for each channel) and storage tubes which enable the effective persistence to be varied at will have greatly improved the versatility of today's instruments.

TURNING THE TUBE INTO A MEASURING INSTRUMENT

In the majority of cases the CR tube is used to produce a graphical display with the amplitude of a signal being expressed in the vertical (Y) direction and its variation with time being along the horizontal (X) direction.

Time-base: If the X plates are driven by a signal voltage that increases proportionally with time the electron beam will be deflected across the tube at a steady speed. When the signal returns to its original value the spot returns to begin the next sweep. The waveform required to produce such linear deflections is a sawtooth. (During return the beam is normally blanked out.) This provides a sweep function. The period of the sawtooth determines the time taken to cross the

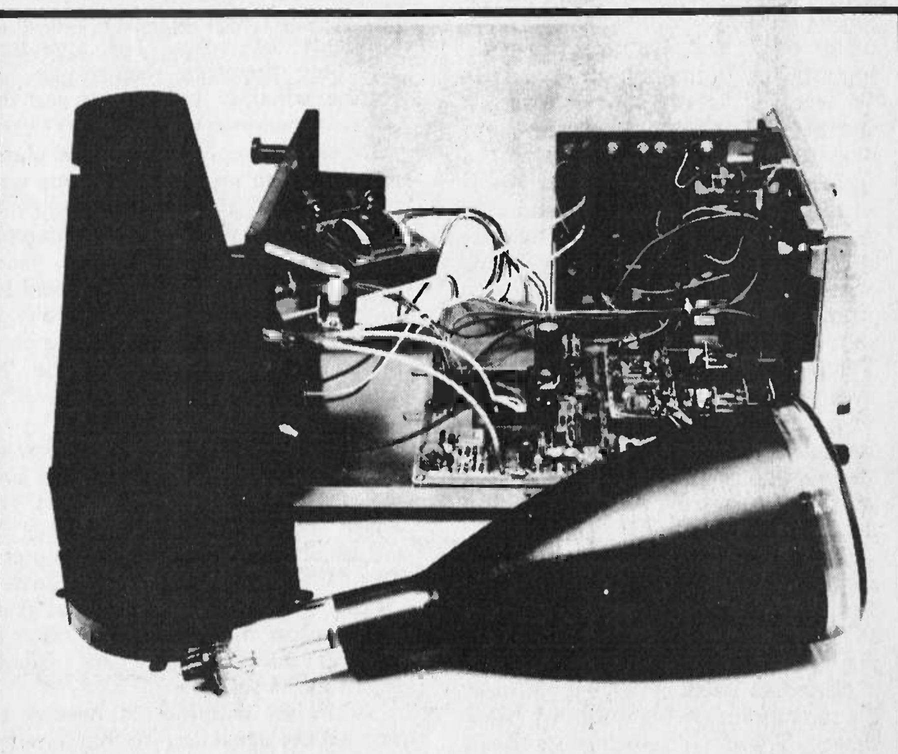


Figure 4. The insides of a modern oscilloscope — the Trio CS-1562. The cathode ray tube and its cover have been removed. Note the tube is much longer than the tubes used in TV sets (when you consider the small screen area). The cover screens the electron beam from the influences of stray magnetic and electric fields.

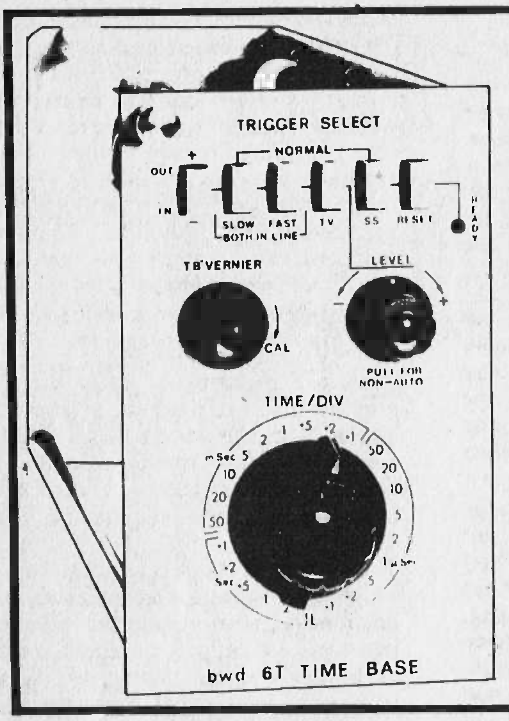


Fig.5. Controls of a basic time-base unit include those shown on this plug-in. Terminology is generally the same for all makers but layout and controls will vary.

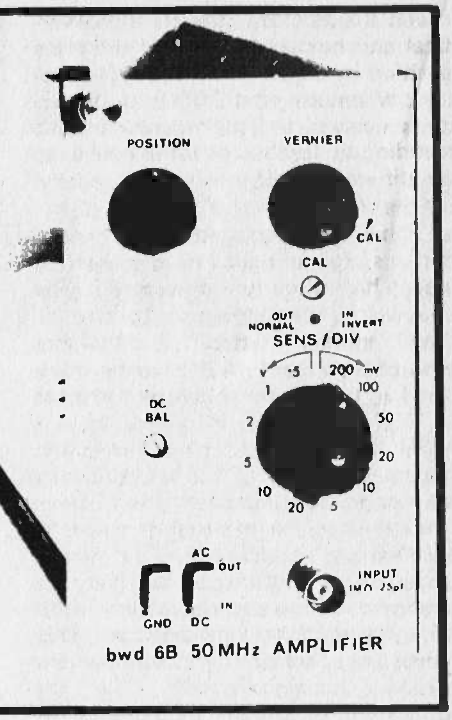


Fig.6. Single-trace amplifier unit. These can be used for both vertical and horizontal amplification.

screen; this is expressed in the units of time per division (screens are divided into a grid of centimetre squares by means of plastic graticule or by engraving the inner face of the tube). A selector switch in the time-base section of the panel enables the sweep rate to be chosen to suit the period of the signal being examined. Basic units will have time bases which range from 0.5 μs to 0.1 seconds per centimetre; sophisticated units can go as slow as 10 seconds per division to as fast as 1ns per division. (Special "sampling" plug-ins can provide 10 ps/division.)

The time-base sawtooth generator is an integral part of all CRO measuring instruments. The accuracy of the rates are determined by circuit components — more expensive units can provide more-accurate information. A further control in the time-base section (See Fig.5) allows the switch-selected sweep rate to be varied continuously. This is usually referred to as a vernier control. When making time measurements, such as waveform period, it is important to set the vernier control to the calibrated position.

To obtain a static display (where each cycle of a periodic signal overlays the previous one) the time-base must be synchronized with some point of the input signal. That is, the time-base is caused to begin its sweep across at the same point on the waveform being viewed. The circuit which does this is called the triggering circuit. Triggering can be taken from either an internal or an external source. When switched to 'internal' it is possible to vary the voltage level of the signal operating the trigger. Thus the sweep may be adjusted to commence at a chosen point on the waveshape. An 'auto' control position provides automatic selection of the voltage level for most reliable triggering.

Time Base Amplifier: The voltage required to deflect the beam over the full X (or Y) traverse is of the order of hundreds of volts. The time-base generator therefore requires an 'X' voltage amplifier between it and the plates.

In certain applications the X plates are used with signals in the same way as Y plates — that is without a time-base signal. In such cases considerable amplification may be needed. More versatile CROs offer plug-in facilities for the X input to give the user a wide choice of functions from the one unit. Simple units however, have the 'X' amplifier wired in permanently.

Vertical Inputs: At the same time as the time-base circuits sweep the line across the screen the 'Y'-plates are driven with a voltage proportional to the amplitude of the signal of interest. This causes the beam to be deflected in the vertical direction whilst it is swept across the screen. The result is the graphical display of signal amplitude versus time.

Again an amplifier is needed to increase the signal level so that a useful vertical deflection results. Such an amplifier must be able to amplify the incoming signal without distortion to provide vertical sensitivities up to 10 mV/centimetre (typically the most sensitive range of educational units), or maybe as high as 10 μV per division (in sensitive oscilloscopes). The insensitive end of the range will usually be around 50 V/cm but special units (for electrical supply authority use) provide for much higher levels. (Attenuator probes also enable high voltage signals to be investigated.)

The application needed from Y-amplifiers can, therefore, rise to 100 000 on the most sensitive range. In addition it is important that the gain be constant over the bandwidth of the signals being monitored.

Basic units provide amplifier response flat from dc to a megahertz or more. (Bandwidths are defined between points 3 dB down from maximum.) Magnetic-deflection display monitors will only reach 20 kHz whereas sophisticated high-quality instruments have bandwidths rising to 350 MHz. Sampling plug-ins provide bandwidths equivalent to dc to 1 GHz.

Vertical amplifier controls are usually grouped together on the front panel, as are time-base controls. Figure 6 shows the panel layout of a 50 MHz bandwidth amplifier. From this it may be seen that the vertical sensitivity is selected by a switch and that the y amplifier has a 'vernier' sensitivity control which must be at the 'calibrate' position when measurements of signal amplitude are being made.

The position of the trace on the screen depends upon the standing voltage applied to the plates. On both

Y and X axes extra controls enable vertical and horizontal shift of the trace position by adjustment of the bias applied. When using the CRO to probe circuits involving ac signals combined with standing dc levels – as is the case in ac amplifiers for example – the dc level on the Y signal causes the trace of the ac signal to be displaced vertically and, perhaps, to go right off the viewing area. This difficulty is overcome if you couple the circuit signal to the Y-amplifier via a capacitor. The ac signal then centres itself on the screen at the position chosen by setting the vertical shift control. This method is acceptable provided frequencies below the cut-off of the RC filter produced are not wanted. Measurement of very-low frequency to dc signals must be dc coupled on the ac/dc selector switch provided. A further switch position enables the input to the plates to be brought to its dc zero position. This helps the operator to establish where this level is on the screen

Signal Input Connections: Oscilloscopes for use with frequencies below about 1 MHz can make use of separate plug-in/screw-down banana-plug terminals. More usually, however, the input to the Y amplifier, and perhaps to the external trigger, will use standard BNC connections. These are designed for use with coaxial cable and coax should be used for all except the shortest end connections to the circuit. The input impedance characteristics are usually quoted – 1 megohm with 20-100 pF shunting capacitance being typical values. In some applications the CRO must be matched to reduce reflections – in such cases the input might be 50 Ω or 600 Ω. For fast rise-time studies it is necessary to ensure that the capacitive value presented does not reduce the overall bandwidth by shunting the device to which the CRO is connected. In exacting cases, needing high input impedance and small capacitance, special probes are used. These are described later.

Calibration of the Time Base and Y-Amplifier: The value of electronic components may drift with time, altering the sweep rate and vertical amplifier values from those indicated by the selector switch. To enable the operator to check these, more advanced oscilloscopes incorporate a special circuit that provides a fixed-frequency, fixed-amplitude square wave signal for calibration purposes. A typical signal would be 1 volt peak-to-peak. As it is derived from the line frequency its time duration is also quite accurate.

MULTIPLE TRACE OSCILLOSCOPES

Measurement situations involving oscilloscopes more often than not require display of comparative information between two points in a system – the relative input and output signals in an amplifier response test, or the phase shift between two signals across a filter stage. Single-beam oscilloscopes are very limited because they cannot provide as much information to the user as a unit that can compare the waveforms at two points simultaneously. Three distinct alternatives are available to provide dual beam operation:

Separate gun: These use two, physically-separate, electron beams and deflection systems that are mounted inside the tube envelope. The beams may be generated by splitting the beam from a single gun. These are generally referred to as dual-beam units (dual-trace is a term reserved for the next method described).

Each beam has its own Y-input panel with a complete set of controls as described earlier. Dual-beam units drive both X-scans with a common set set of deflection plates (as in Fig.7) but some (rather rare) oscilloscopes enable each time-base to scan at a different rate.

In general, dual-beam units are less common because of the higher expense compared with the next method.

Electronic switching – chopped mode: The deflection response of an electron beam is rapid enough to allow it to be directed from one position to another at a speed exceeding the scan rates used with the signal being viewed. Fast electronic switches are used to switch the common single beam between two (or three or four) Y-inputs. Figure 8 illustrates this. Appropriate blanking (that is reduced Z intensity) is applied when needed, when the beam is chopping from one trace to the other. If the chopping rate is chosen to be at least 100 times faster than the highest frequency to be viewed the two traces appear as separate traces. Hence the name "dual-trace" for this method. In reality the traces are not continuous but are made up of dash-spaces. A hundred dashes across a screen produces a virtually continuous trace to the eye. The limit of usefulness is reached when the inbuilt chopping rate comes close to the upper frequency being viewed thus producing a dashed-line trace in which the dashes are of length equal to wanted signal features. At this point information is lost.

As far as the user is concerned there are still two groups of Y controls – just the same as for a dual-beam arrangement. The difference arises as the position chosen on the selector switch where a 'chop' mode must be selected.

Chopped operation ensures that the time relationship between the two signals is faithfully presented: phase

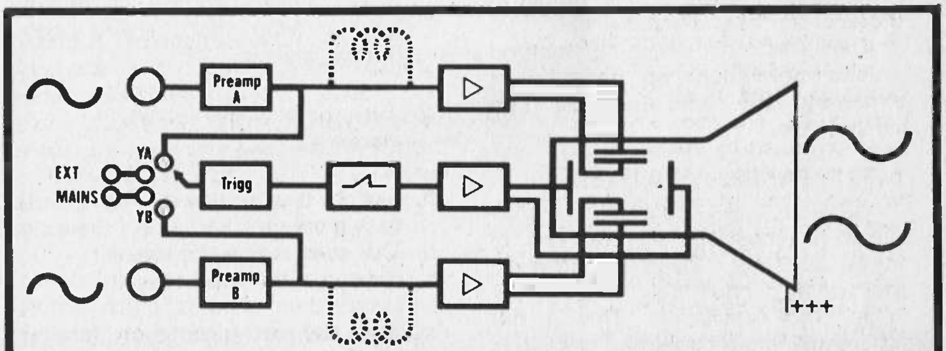


Fig.7. Schematic of Philips 3232 dual beam oscilloscope. Common x plates provide scan for both beams, separate y plates deflect the two distinctly separate electron beams that are derived from a common gun.

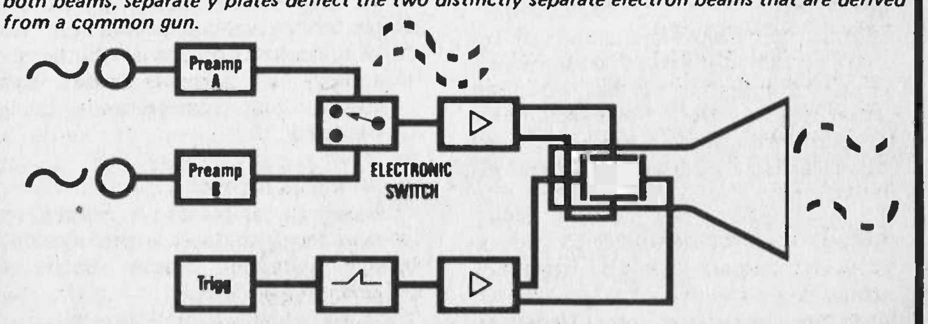


Fig.8. Electronic-switching enables a single-beam and deflection system to provide dual-trace operation.

measurements are also accurate (that is, providing the input amplifiers to each are identical).

Chopped operation will also display two simultaneous, non-recurrent signals, such as transients induced at various points when a complex resonant system is excited by an impulse. It is quite suitable for low-frequency signals but less convenient as the frequency rises.

Electronic switching alternate mode: Switching can also be employed on a full alternate trace-by-trace basis. The first trace is a scan of channel 1, the next of channel 2 and so on. This does not suffer from the dotted defect with high-frequency viewing but it suffers from another deficiency in that the phase relationship between the two signals may not necessarily be as indicated on the screen.

The method is unusable for observation of "once-only" dual events because the second transient signal may have gone to zero by the end of the trace of the first simultaneous transient signal. The panel shown in Fig. 9 is typical of dual trace units. The selector switch enables choice of alternate, chop, channel 1, channel 2, and channel 1 plus channel 2 modes.

With two channel operation it is necessary to decide which input will synchronize the time-base scan. A switch provides the choice of appropriate internal triggering. Although only channel 2, for example, may be being viewed there are circumstances where it is desirable to trigger from the channel 1 signal.

The electronic-switching method enables more than two traces to be displayed– three and four-trace units are available.

DIFFERENTIAL AMPLIFIERS

Generally the dual-trace oscilloscope is recognised by two sets of input terminals. There is, however, another two-input unit that is for single trace operation. This is the differential input amplifier unit; it is normally provided as an optional plug-in.

Two two inputs are amplified by the high-gain differential arrangement of a dc amplifier. These are used when common-mode noise rejection is needed and when the difference between two fully floating inputs must be studied.

FINDING THE TRACE

Even experts can experience temporary difficulty when confronted with an unfamiliar oscilloscope – especially when it is complicated. Naturally it takes training to get the best from a unit. A basic difficulty is often finding the trace! These steps provide an efficient procedure that should be learned. Begin with the input to the Y-amplifier unconnected.

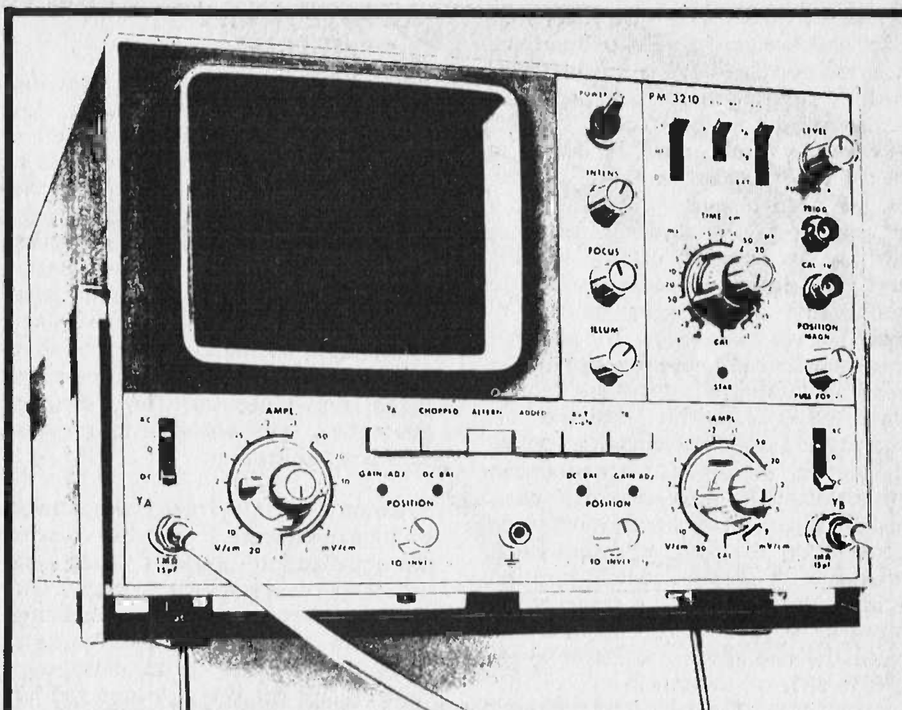


Fig.9 is the front view of a modern unit. It is given as an exercise. 1. Locate the controls that are associated with the quality of the dot produced by the beam. 2. Does it provide two trace display, and if so by what method? 3. Which is the time-base control area? 4. Where is a Y-amplifier vernier control to be found? 5. What are the input terminating conditions?

1. Ensure that the power is on. The on-off switch control is usually built in with the intensity knob, but not always.

2. Turn the intensity to 75% clockwise.

3. Switch the time-base (horizontal) to a medium speed – say 1ms/cm. This ensures that the screen displays a full line across the screen rather than a point which occurs when the scan speed is on the slow ranges.

4. Switch to auto triggering. This may be a marked position of the trigger control or a separate switch. This ensures that the trace is being triggered.

5. Switch to internal triggering. This is necessary for (4) above. Relying on an external signal to trigger the scan is unreliable – it may not be of adequate magnitude.

6. With this done slowly vary the vertical position control about its mid range point widening out to get the trace on screen.

7. The above may still not produce the trace. If not put the vertical position in its middle point and the gain at an insensitive value and begin a scan of the x-position control. This should be somewhere mid range. Too much x-shift can cause the trace to slide off screen.

Complicated oscilloscopes will invariably incorporate a variety of controls that may also need adjustment to find the trace. Space prevents a full guide to spot finding. Fortunately the more expensive units often provide a spot-finder button. Press it and the spot appears on screen enabling the controls to be adjusted accordingly to bring it back from the direction it flies too when the button is released.

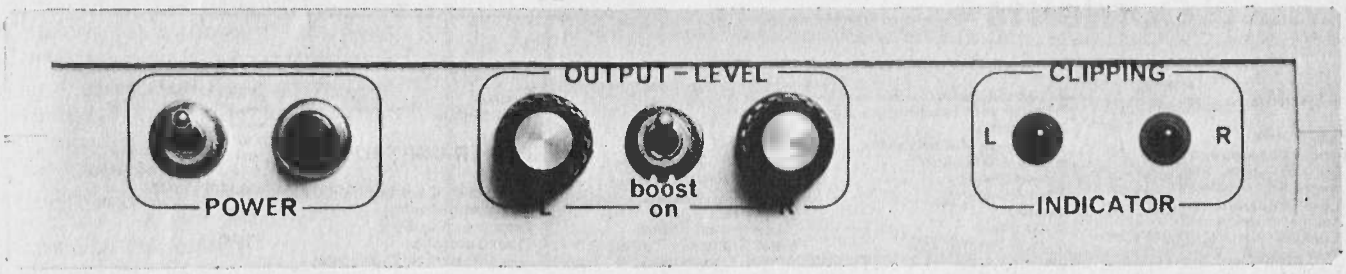
When the trace is located in mid screen the intensity and focus are then adjusted by switching the scan to the slowest rate to produce a spot. These should be adjusted to produce a small round spot without halo: stationary spots on screens should be avoided as this shortens the life of the phosphor at that point.

In part 2 we will continue the discussion on oscilloscopes providing understanding of storage kinds, the delay sweep mode, probes and special plug-ins.

SHORT CIRCUITS

ETI's series of straightforward projects, not necessarily simple in their operation but presenting few problems in building.

BASS ENHANCER



ONE UNAVOIDABLE DRAWBACK of small speakers, as compared to their larger brethren, is a lack of extreme bass frequencies. They may not 'boom' and they may sound less 'coloured', but they will NEVER play the lowest of the low as well!

There are many high quality 'mini-monitors' around these days — which possess an excellent sound overall, but are victims of their size with regard to bass response. Let it not go unsaid, however, that many of the aforementioned designs produce a bass signal which is incredible for that size!

As you may have gathered (or hoped!) from this little lecture, this circuit has something very much to do with small speakers. It is designed to compensate in some degree for this lack in the last octaves. Turning up the bass control on the amp doesn't help, indeed it will generally make things worse by introducing too much lift too high in the spectrum. Most tone controls have a turnover at about 100-200 Hz, and will thus still have some appreciable effect at 500 Hz or more.

What is needed is a boost below about 70 Hz, but above 15 Hz or so (to avoid amplifying warp signals) and this is impossible to apply with conventional tone controls.

ETI TO THE RESCUE!

Having now told you what's wrong with your 'Mighty Mouth XXV Mini Super Monitors' we'd better explain how we can help put it right. Our Bass Enhancer is designed to insert a 'hump' at precisely the aforementioned frequency, into that part of the overall frequency response where the small speakers are rolling off.

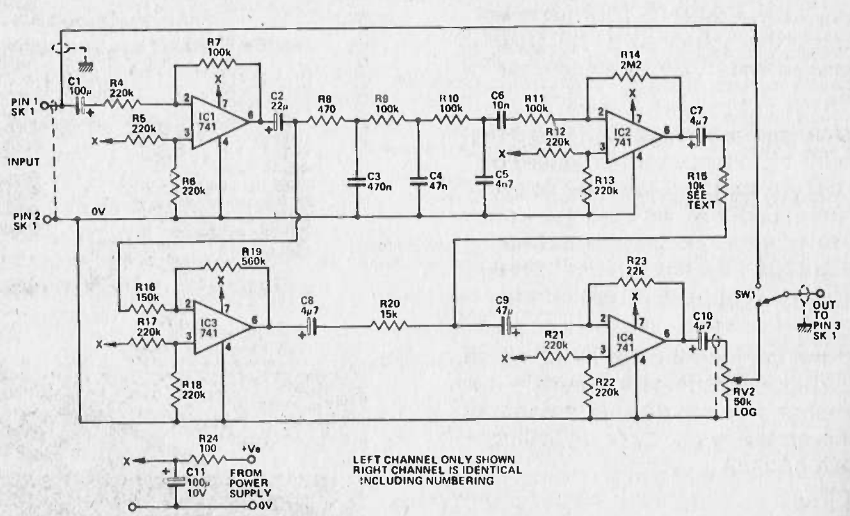
The circuit is of no use to you unless your speakers, and your amp, are capable of taking the extra punishment at these frequencies. Most hi-fi components are. Most of the cardboard boxes which are sold with music centres masquerading as loudspeakers are most definitely not. If in doubt, check with your supplier.

As a precaution against smoking ruins replacing your amplifier, we have included the ETI 'Overload' project to keep watch for clipping in the amplifier. With the bass enhancer in circuit, more power is drawn at the lowest frequencies, and amplifiers driven to distraction by these demands will exact revenge in horribly audible form.

How it works

IC1 and associated components form a buffer amplifier stage which isolates the lowpass filter R8, C3, R9, C4, R10, C5 from the source. C6 forms a single-pole roll-off filter to get rid of warp signals etc below the band with which we are concerned.

IC2 amplifies the low frequency signal fed to it from the filter, and so provides the 'enhancement' signal. This is mixed back with the amplified and buffered version of the input signal coming from IC3 in the mixer amplifier IC4. RV2 provides output level control, and SW1 allows the unit to be bypassed completely.



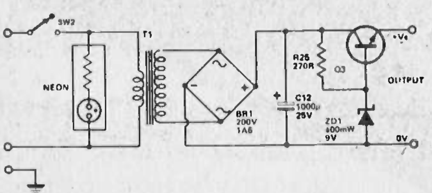


Fig. 1. Circuit diagram of the Bass Enhancer and power supply (left).

CONSTRUCTION

A single board layout has been chosen, to keep down the cost, and to simplify things as much as possible. The clipping indicator section is in fact the 'Overled' circuit previously published, so if you've already built this, you don't really need to build up this section of the unit. An AC supply was added simply because it is cheaper in the long run.

Standard — good quality — 741s are employed in the prototype, and very well they work too. However, there is a low noise 741, and 741N, which does have much less inherent noise if you find the noise is too high for your liking.

When building up the Over-Led, refer to the table given to set the values for your amp and speakers.

All that is entailed is to fit all the components to the board as shown in the overlay drawing, and wire up to the LEDs and controls.

GET SET...

Setting up the clipping indicator is best done with the enhancer out of circuit. Advance the volume until the sound from the speakers becomes distorted - it doesn't matter whether it's the amp or the speakers cracking up, something is! - and then advance the pre-set until the LED just lights. This is the point of 'no-fidelity'!

The enhancer goes into the chain between pre- and main-amps, or into the tape monitor circuits, and can be switched out of circuit by SW1. Set the output levels so that there is no 'jump' in level when you switch it in from another source path - it's much kinder on both your ears and your hi-fi.

Varying the value of R15 varies the amount of enhancement produced by the circuit, so if you think we've over (or under) done it - let's see you do it better! Our value gave consistent results on quite a few different speakers, from Wharfdale to Celestion.

If the clipping indicators come on you're driving things too hard — turn *something* down! (If your neighbours bang on the walls, floor or ceiling — move house!)

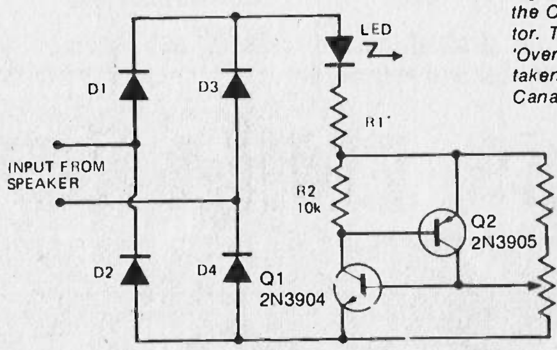
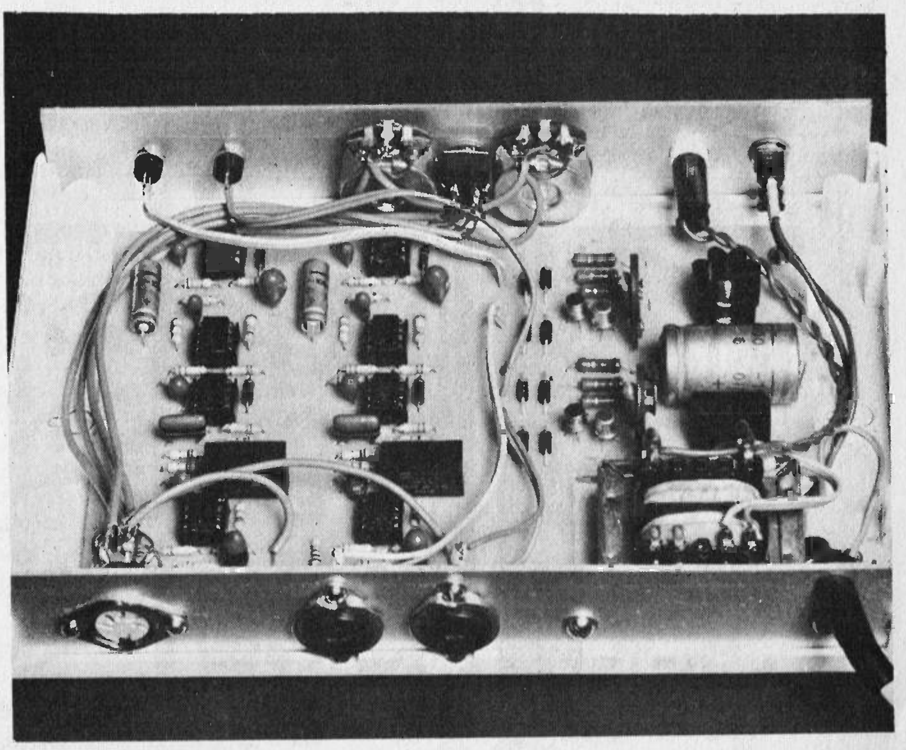


Fig. 2. The circuit of the Overload Indicator. This is the 'Over-Led' project taken from ETI-Canada, July 77.

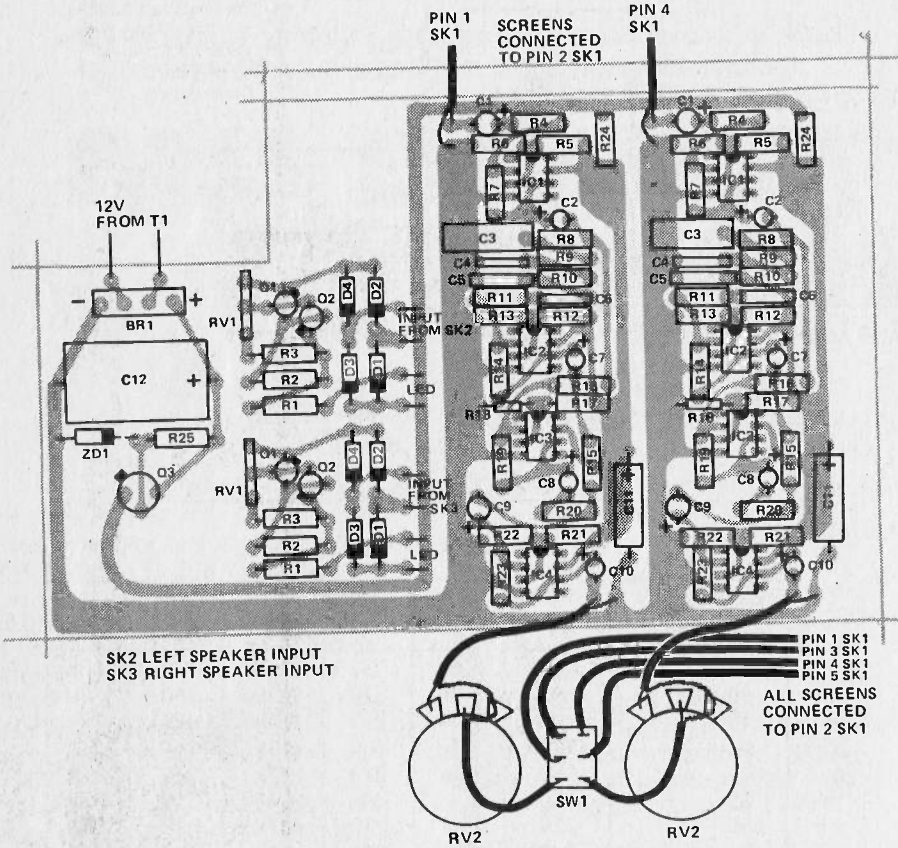
TABLE 1

RMS watts per channel	4Ω R1	4Ω R3	8Ω R1	8Ω R3	16Ω R1	16Ω R3
5	68	5.6k	82	8.2k	120	12k
10	82	8.2k	120	10k	180	18k
15	100	10k	150	15k	220	22k
20	120	12k	180	18k	240	24k
25	150	15k	220	22k	270	27k
35	180	18k	240	24k	330	33k
50	220	22k	270	27k	390	39k
75	240	24k	330	33k	470	47k
100	270	27k	390	39k	560	56k

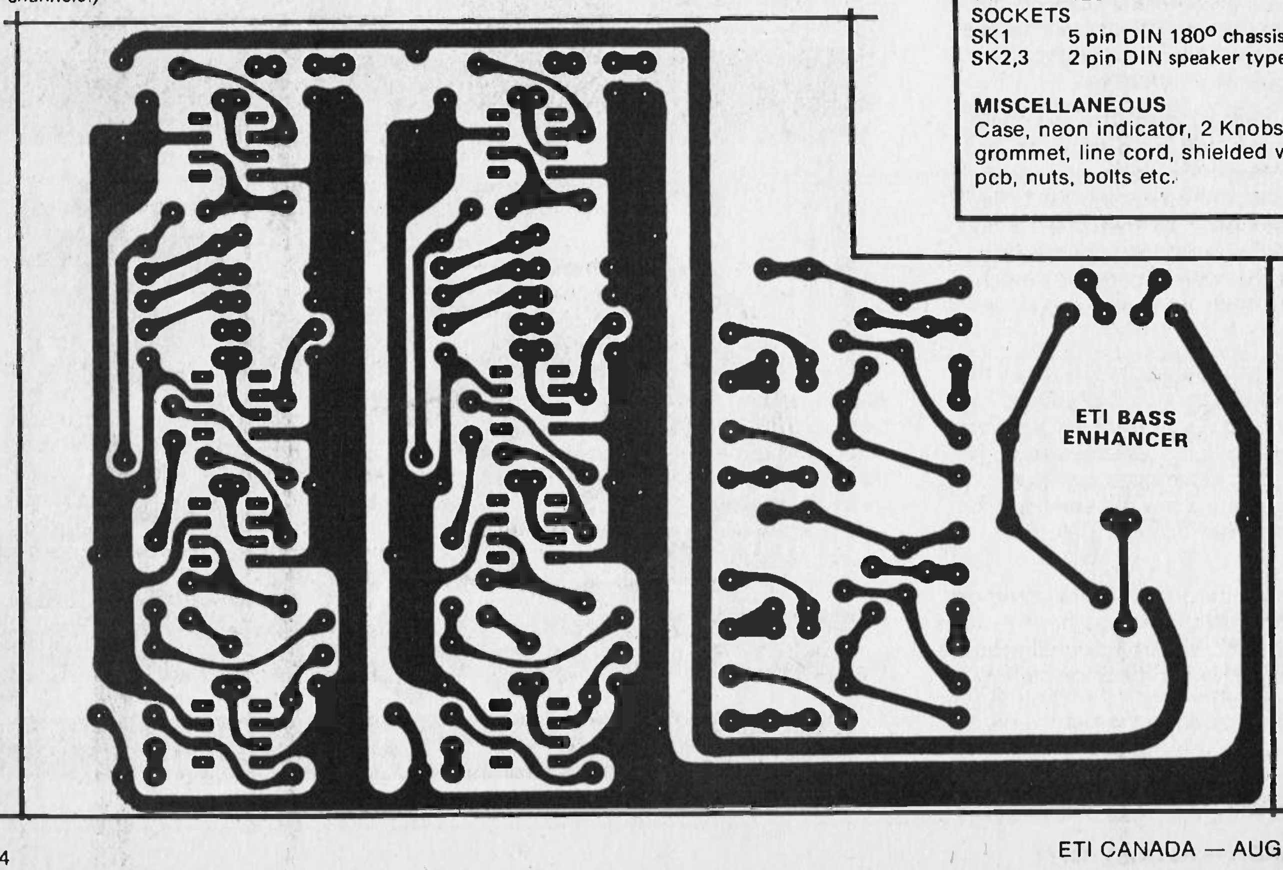
(SPEAKER IMPEDANCE)



BASS ENHANCER



Overlay and Foil Pattern for the Enhancer. Note that a single board construction is used, so that whilst the overlay may look complex it is in fact only half as bad as that! (Two identical channels!)



Parts List

ONE SET REQUIRED PER CHANNEL

RESISTORS	
(All ¼ W except where stated)	
R1	as table 1
R2	10 k ½ W 5%
R3	as table 1
R4-6, 12, 13, 17, 18, 21, 22	220 k
R7, 9-11	100 k
R8	470 R
R14	2M2
R15	10 k (see text)
R16	150 k
R19	560 k
R20	15 k
R23	22 k
R24	100 R
R25	270 R ½ W 5% (only 1 req.)
CAPACITORS	
C1	100 u 16 V tantalum
C2	22 u 16 V tantalum
C3	470 n polyester
C4	47 n polyester
C5	4n7 polyester
C6	10 n polyester
C7,8,10	4u7 16 V tantalum
C9	47 u 16 V tantalum
C11	100 u 10 V
C12	1000 u 25 V (only 1 req.)
SEMICONDUCTORS	
Q1	2N3904
Q2	2N3905
Q3	2N1893, 2N4001
D1-4	IN4001
ZD1	9 V 400 mW zener (only 1 req.)
BR1	200 V 1.6 A bridge rectifier (only one req.)
LED	.2" type red
IC1-4	741
POTENTIOMETERS	
RV1	1 k vertical preset
RV2	50 k log rotary
SWITCHES	
SW1	D.P.D.T. mini toggle
SW2	On-off toggle 120 V 1 A
TRANSFORMER	
T1	120V - 12 V 500 mA
SOCKETS	
SK1	5 pin DIN 180° chassis type
SK2,3	2 pin DIN speaker type

MISCELLANEOUS

Case, neon indicator, 2 Knobs, grommet, line cord, shielded wire, pcb, nuts, bolts etc.

SHORT CIRCUITS

CONVERT ANY 1mA METER MOVEMENT INTO AN ACCURATE TACHO FOR YOUR CAR WITH OUR

TACHOMETER

This design uses a single integrated circuit to provide an easily calibrated unit that will provide rpm indications of a wide range of engine speeds. It is suitable for automotive engines with standard ignition.

UNTIL TEN OR SO YEARS AGO, car tacho's were cumbersome mechanical devices usually driven via a flexible cable from gearing attached to the shaft of the vehicle's dynamo — or sometimes via the distributor shaft.

The advent of transistor technology changed all this and since then almost all car tacho's are electronically operated.

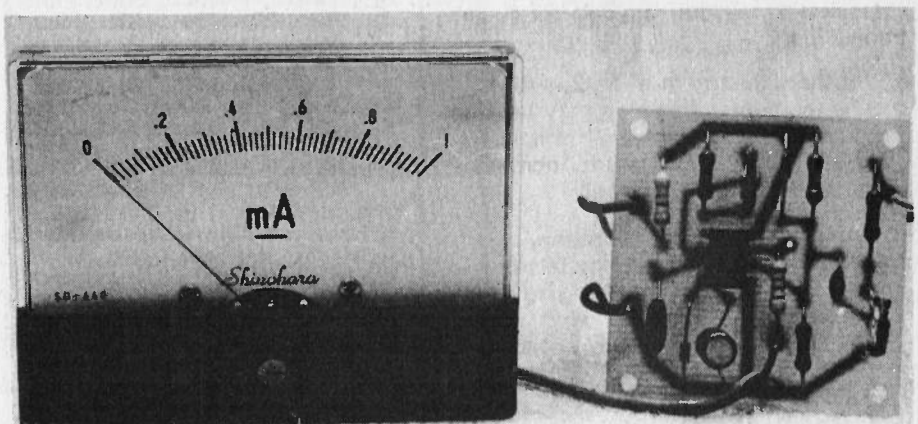
The basic principle is much the same for all electronic tacho's. An electrical signal taken from the low tension side of the distributor is converted into a voltage proportional to engine rpm and this voltage is displayed on a meter calibrated accordingly.

Most car tacho's are complex and expensive devices — but here's one with a difference! It is simple yet extremely effective. Its simplicity is due to our using one single integrated circuit rather than the more conventional multiplicity of individual transistors.

The unit will operate on both positive and negative earth vehicles and will also operate successfully and without modification with most types of electronic ignition systems as well as the more common electro-mechanical systems.

Construction

As there are so few components, construction is very simple and straightforward. Do make sure though that the 555 IC is soldered in the right way round — ditto the two diodes. Compare your work against our layout drawing as a final check.



Photograph showing a view of the completed pcb connected to a 1mA meter with 120° movement. The compact layout of our single integrated circuit design is apparent from this picture.

Any type of meter that has one milliamp full scale deflection can be used. This is a very common type of instrument and you should be able to obtain one new or secondhand with no difficulty. Ideally you should choose one that has 180° or 280° movement but these tend to be rather expensive. The meter size should be chosen to suit your proposed housing.

When the meter has been assembled connect it to the vehicle's battery and connect the input to the contact breaker side of the coil.

Calibration

We can think of three ways to calibrate this unit. The easiest method is to borrow an already calibrated tacho which can be temporarily connected to your car. RV2 may then be adjusted until the two readings agree over a range of engine speeds.

The unit may also be calibrated by the use of a signal generator to "simulate" the ignition system. Taking the number of cylinders in your engine, and dividing by two, gives the number of sparks per revolution. Hence, to simulate 1000 r.p.m. with a six cylinder engine, use a frequency of 3000 cycles per *minute*, or 50 Hz.

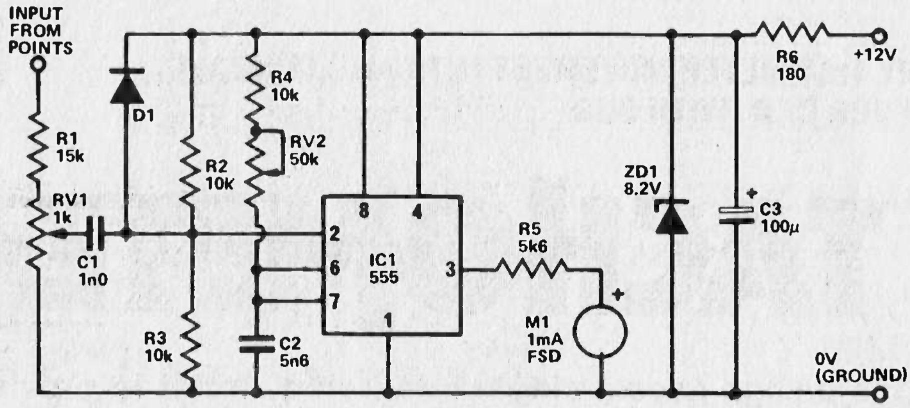
To put this more mathematically, adjust your taco so that:

$$f = \frac{N}{2} \cdot M \cdot \frac{1}{60}$$

Where N is the number of cylinders, M is the meter reading in r.p.m. and f is the generator frequency.

TACHOMETER

Circuit diagram of the tacho.



Our final method is to calculate the vehicle speed per 1000 r.p.m. in top gear and adjust RV2 accordingly. Needless to say, this is a *two* person operation.

If the adjustment of RV2 is found to be too coarse its value may be reduced to 25k or lower. If this is done it will be necessary to increase the value of R4 accordingly

Before making final calibration, adjust RV1 to eliminate any false triggering — check at all engine speeds. This unit may be used with either +Ve or -Ve ground vehicles — simply connect the battery leads as shown.

How it works

The 555 timer IC is used as a monostable which, in effect, converts the signal pulse from the breaker points to a single positive pulse the width of which is determined by the value of R4 + RV2 and C2. The mathematical formula is $T = 1.1 \times R \times C$ where R= R4 + RV2 (the section of RV2 in use) and $C = 5.6 \times 10^{-9}$ (Farads), and T= pulse length in seconds.

Resistors R2 and R3 set a voltage of about 4 volts at pin 2 of IC1. The IC is triggered if this voltage is reduced to less than approx. 2.7 volts (1/3 of supply voltage) and this occurs due to the voltage swing when the breaker points open.

An adjustment potentiometer RV1 enables the input level to be set to avoid false triggering.

Zener diode ZD1 and the 180 ohm resistor stabilise the unit against voltage variations.

Parts List

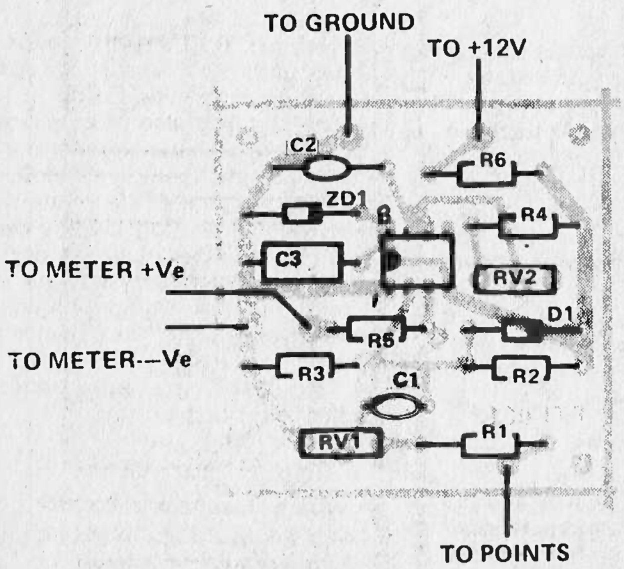
RESISTORS (all ½ W 5%)
R1 15 k
R2-4 10 k
R5 5k6
R6 180 R

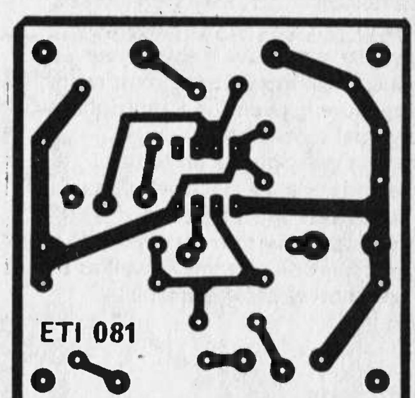
CAPACITORS
C1 1n0 polyester
C2 5n6 polyester
C3 100 u 10 V electrolytic

SEMICONDUCTORS
D1 1N914
ZD1 8V2 300 mW
IC1 NE555

POTENTIOMETERS
RV1 1 k
RV2 50 k

MISCELLANEOUS
PC board
Meter 1 mA FSD





The foil pattern of the printed circuit is shown full size above and to the right is the component overlay for the PCB.

Software Techniques for Hardware Men!

A CONSIDERATION OF THE METHODS USED TO APPLY DIGITAL CIRCUIT DESIGN TECHNIQUES TO SOFTWARE ROUTINES BY R. WILSON MSc

THE WEALTH OF EXPERIENCE possessed by engineers and technicians involved with digital hardware can be enormously valuable to them in their transition to microprocessor based designs. The thought of programming may distress the hardware man; howevever, to write efficient assembly language programs demands an intimate familiarity with the hardware structure of the particular microcomputer being used.

The flowchart forms the basis of program writing by ensuring that the logical sequence of events has been crystallised. Consider as an example a process control situation depicted in Fig. 1. After a controlled start, system initialisation can commence by processing the input data to check given interlock requirements. Satisfactory results allow the process to begin, otherwise the interlock failure is annunciated and a system stop ordered.

As the process continues towards a designated goal, periodic status checks of the system are required so that control action can be implemented. To ensure that actuators operate correctly the response to an action command is fed back for the system to monitor. This outline scheme typifies the use of flowcharts. Of course each block could be examined further resulting in a more detailed diagram.

Program Power

The real power of programs is their ability to make decisions. Examples of assembly language conditional instructions are:

JUMP and
INCREMENT SKIP IF ZERO

Jump instructions can either be mandatory, thus directing the program to an address which is accessed in all cases, or jumps can be conditional as illustrated by the following examples:

JZ could mean Jump if accumulator = 0
JC could mean Jump if accumulator carry = 1
JCN could mean Jump if a given CPU pin = 1

The number and type of jump instructions provided depends upon the particular microprocessor in use. The Increment Skip if Zero (ISZ) is useful for decision making because it can distinguish between zero and non-zero contents of index registers. A designated path can be followed in each case.

Interesting Routine

A program sector which is frequently used during a process forms a prime candidate for a SUBROUTINE. In Fig. 1, the block performing process status checks would clearly be used repeatedly. Two of the three

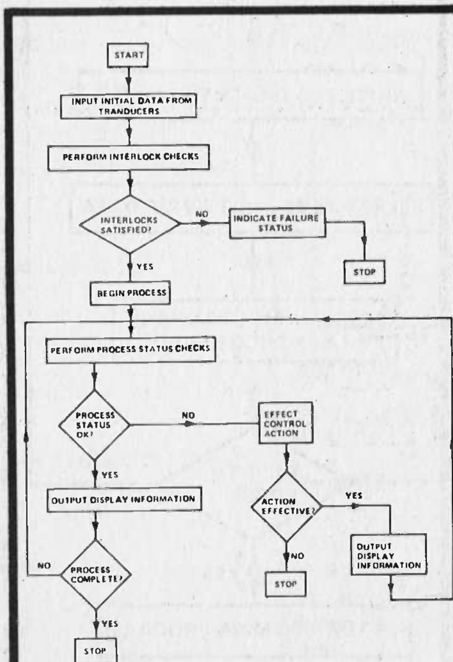


Fig. 1. A flow chart depicting a process control situation, requiring several interlock checks.

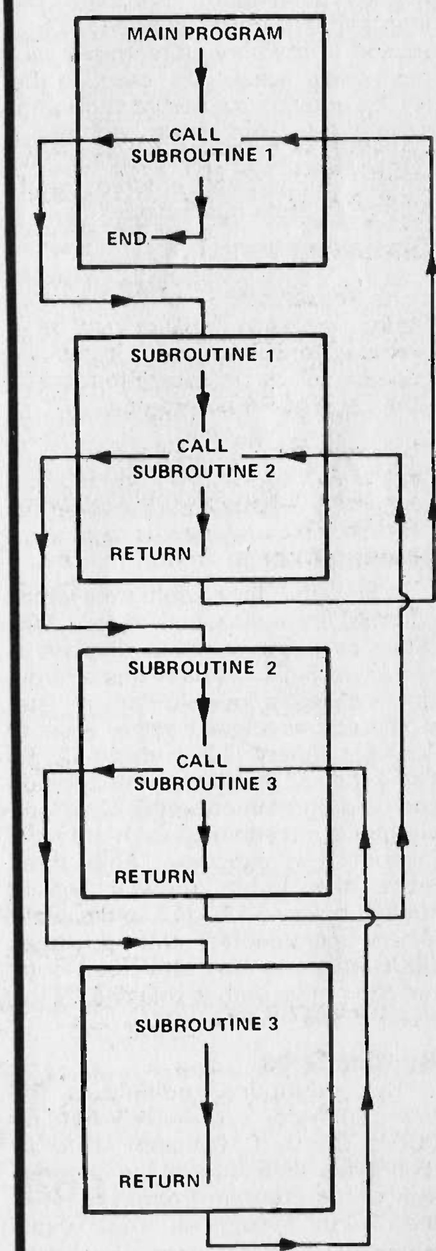


Fig. 2. An example of nested subroutines. In this case to a depth of three.

paths leading to this block are the result of decisions made during the on-going process. The subroutine would therefore be called into action by an instruction such as JMS 70 (Jump to subroutine at address 70).

To perform process status checks, data has to be input from transducers, stored, and then a number of successive decisions made based upon the data. These requirements can be met by using NESTED SUBROUTINES. A number of subroutines are written each one called by a preceeding member of the set. Fig. 2 shows subroutines nested to a depth of three.

After servicing a subroutine the micro processor needs to know the re-entry point in the program. This information is normally provided by a section of memory store known as a push-down stack. As each further level is entered in a nested subroutine system the latest return address is placed at the top of the stack thus causing previously entered return addresses to be pushed-down.

Displaying Versatility

A main application area of microprocessor systems is in replacing hard-wired logic by a stored program. A multiplexed seven-segment LED system for numeric display is usually implemented by interconnecting integrated circuits. The function of some of the circuits can be duplicated by a sequence of assembly language instructions.

Flowing Charts

A flowchart for a subroutine which controls the indications of four BCD digits by seven-segment displays is shown in Fig. 3. When this subroutine is called, a four-bit index register which acts as a loop counter is set to 1100 in binary (12 in decimal). By using the ISZ instruction the decision can be programmed simply. The loop counter is incremented each time this instruction is executed. After three loops the index register would contain binary 1111 (15 in decimal). When incremented this becomes 0000 ensuring that after four loops the alternative path is selected by the *"skip if zero"* part of ISZ.

Routine Subs

The subroutine multiplexes the four displays by successively sending 0001, 0010, 0100 and 1000 as digit drive data during the progression of the program from the first to the fourth loop. The BCD digits represented as segment drive data are thus automatically routed to the correct display from a common highway.

Each BCD digit was stored by four-bits of random access memory (RAM). The read/write facility was essential as variable data was being processed. When programming for dynamically changing data the locations of the various data sets in RAM must be constantly reviewed. A RAM map, Fig. 4, is a straightforward visual-aid which makes this task easier. A diagram is drawn showing the empty memory locations then as the instructions are written, the space in RAM can be thoughtfully allocated, modified and updated as necessary.

Finding Bugs

Engineers know that the phrase *"nothing ever works first time"* usually applies to hardwired designs. It also applies to software designs. When a comprehensive program has been written it will contain errors

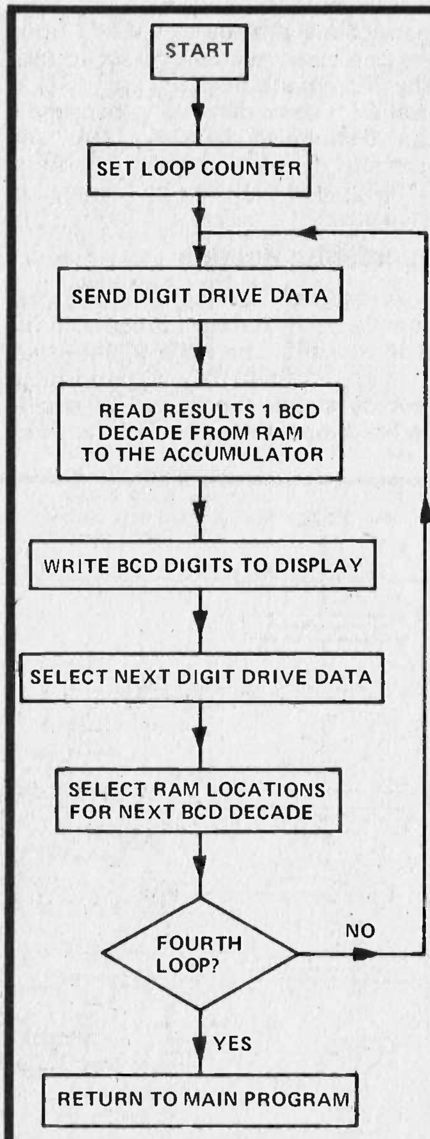


Fig. 3. This subroutine is to run four seven-segment displays, multiplexed and with each digit stored in RAM.

therefore a means of examining the program operation is required. The main sections of a microcomputer hardware structure, the CPU, memory, I/O, and clock can be simulated either by software in the form of a computer package or by

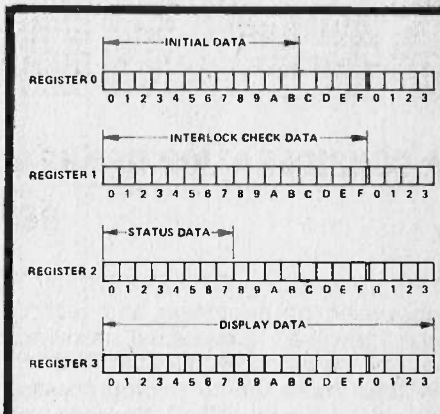


Fig. 4. A RAM map. This shows the memory locations available and what is stored in each.

special purpose hardware. Software simulators are available through various commercial time-share networks whereas hardware simulators can be obtained direct from microprocessor distributors.

The source program written in memonics needs to be translated into numeric patterns. The resulting data, called object code, is generated from the source code by using an assembler. Again hardware and software assemblers are available. The object code version of the program can then be presented to the simulator for testing.

Program testing is not a simple task but the following suggestions might be helpful. If possible, assemble the whole program from the outset, or at least assemble substantial segments such as subroutines. This ensures that the actual program is examined rather than a simplified version. By mentally working through the selected program segment using chosen test data the expected outcome can be predicted before beginning a test-run. The RAM map is often useful during this procedure. If a teletype is used to communicate with the simulator during a program test-run a printed record of the test process can be preserved for analysis later. As each section is tested, modified and eventually verified as correct, further segments can be processed. The aim is to commit the whole of the validated program to PROM.

The hardware man can be comforted in his venture into software by remembering that a microcomputer is, after all, an engineer's computer.

DESIGNING & USING ACTIVE FILTERS PART 1

A SHORT SERIES BY TIM ORR WHICH WILL ENABLE THE HOME CONSTRUCTOR TO UTILISE CIRCUITS OF HIGH COMPLEXITY AS EASILY AS PLUGGING IN A RESISTOR!

THERE IS NO DOUBT that active filters are very useful devices. Also, there is no shortage of literature on the subject. This would seem to suggest that designing active filters is a fairly straightforward business. Well, it is and it isn't. It is if you read *this* article. It isn't if you read the aforementioned literature. Most of the books on this subject have filled our heads with terms such as poles and zeros, Laplace transforms, transfer functions, etc, which haven't actually helped *us* to design anything!

Some basic theory

It is advisable quickly to run through some basic terms and expressions. Firstly, consider a simple low pass filter, Fig. 1a. The frequency response (Fig. 1b) is nearly flat until the break point, denoted by fb. After this point the response rolls off at 6dB/octave, that is signals above this frequency are increasingly attenuated. The break point is defined as being the frequency where the resistance equals the capacitive reactance. When this occurs, the output is attenuated to 0.707(−3dB) of the input. Although the resistance equals the capacitive reactance, the output is *not* half of the input. (This is because it is the vector sum of the two and hence equals 0.707 of the input!)

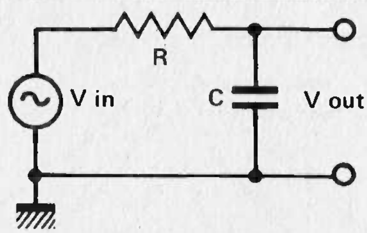


Fig. 1a. Simple low pass filter.

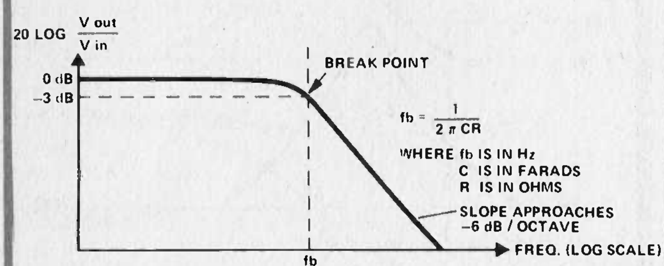


Fig. 1b. Frequency response of above.

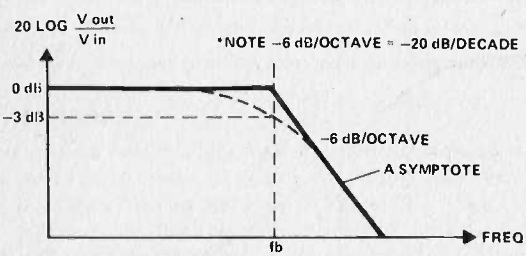


Fig. 1c. Approximation to response.

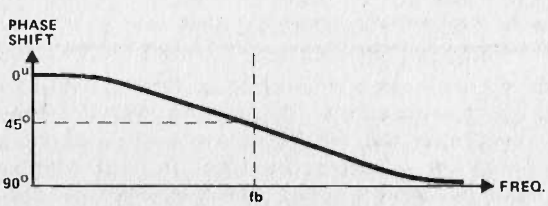


Fig. 1d. Phase shift v Frequency plot.

As the frequency response is a rather complex curve it is very useful to use a straight line approximation to it. These lines are called asymtotes (Fig. 1c). Note that the frequency response graph uses the convention of logarithmic scales, octaves or decades along the frequency axis, and dBs along the vertical axis representing output voltage divided by input voltage.

Phase shift with respect to frequency is also often plotted as in Fig. 1d. These two (the phase and frequency response plots) are known as Bode diagrams and are generally considered the most useful way of representing a filter's performance.

You will note that for the low pass filter of Fig. 1a, the phase shift starts at 0 degrees, is 45 degrees at fb and then approaches 90 degrees as the frequency approaches infinity. This is not an *active* filter, it is composed entirely of *passive* components which means that its output cannot be effectively loaded without changing its performance.

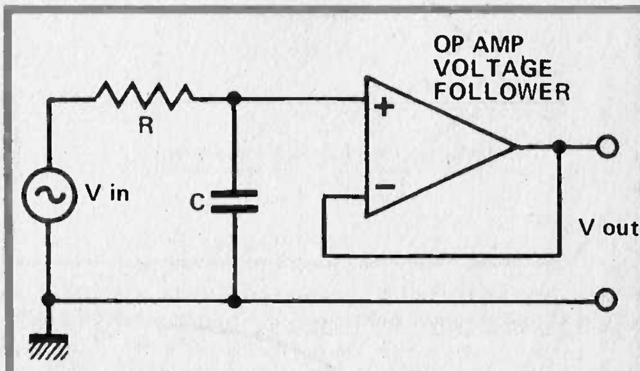


Fig. 1e. Active filter to perform the same task as the passive circuit of Fig. 1a.

Fig. 1e shows the same filter but in its active form, the op amp being used as a voltage follower serving only to isolate the filter's output. This type of filter is known as a First Order filter — a measure of the roll off slope.

When a more rapid slope is required, a higher order filter structure (one with more reactive elements) must be used. This is dealt with later.

ACTIVE FILTERS

Summary of low pass filter of Fig. 1.

Filter type	Low pass
Filter order	First order
Roll off slope	−6dB/octave or −20dB/decade (the same)
Breakpoint fb	fb = 1/2πCR Hz
Phase shift at fb	45°

TABLE 1.

Passing highs

Next, let us consider the simple high pass filter of Fig. 2a. It is the complement of the low pass filter, the elements having been interchanged. Therefore it is not difficult to accept the complementary phase and frequency response curves of Fig. 2b. Note that the break point is the same and so is the roll off slope.

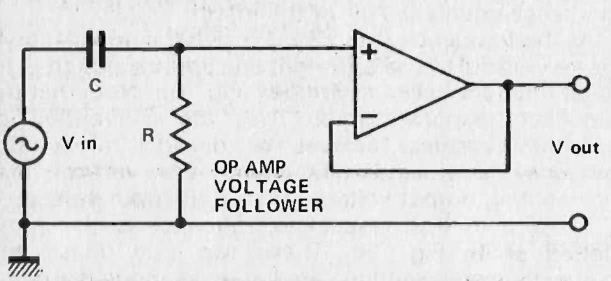


Fig. 2a. Simple high-pass active filter.

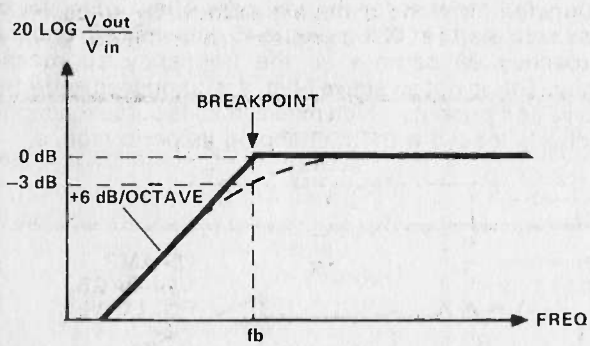


Fig. 2b. Frequency response of the high-pass filter.

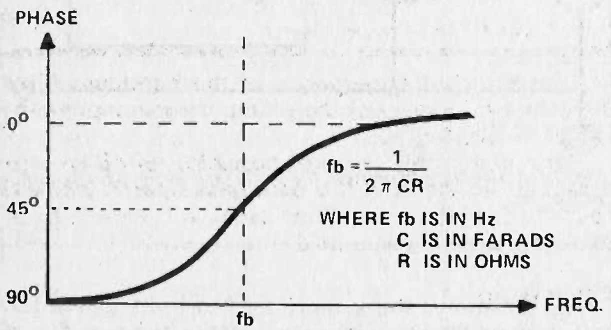


Fig. 2c. Phase v frequency plot of the same filter.

Summary of the high pass filter of Fig. 2.

Filter type	High pass
Filter order	First order
Roll off slope	+6dB/octave or +20dB/decade
Break point fb	fb = 1/2πCR Hz
Phase shift at fb	45°

TABLE 2

Passing bands

The next type to be considered is a simple band pass filter shown in Fig. 3a. Although it uses an inductor it is only to illustrate the bandpass theory. Later on in this series, inductors will be replaced by their active equivalents.

The frequency response (Fig. 3b) shows that this circuit is symmetrical, having roll off slopes of 6dB/octave on either side of its RESONANT peak. This

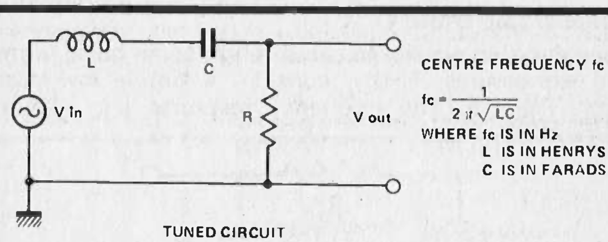


Fig. 3a. Simple band-pass filter.

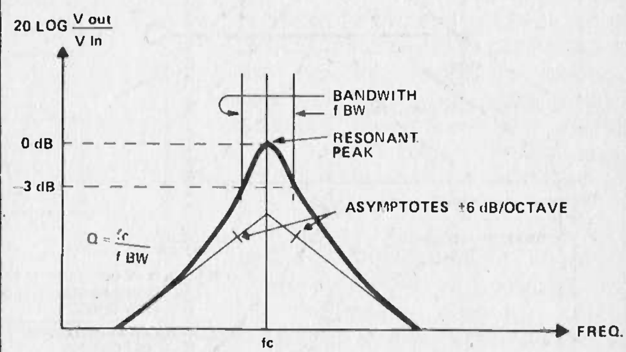


Fig. 3b. Band-pass frequency response.

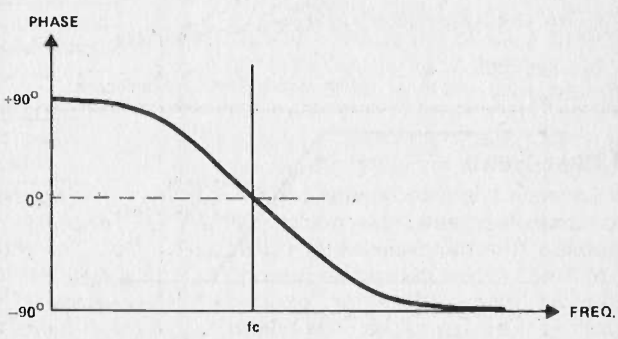


Fig. 3c. Band-pass phase response.

filter is known as a second order filter, because it has two reactive sections, the L and the C. The C is responsible for the +6dB/octave portion of the slope, the L for the −6dB/octave portion. But where these two slopes should meet, the response of the filter peaks and the slopes become much larger (Resonance). The sharpness of this peak is described as the Quality of the filter, the Q factor. Resonance occurs at a frequency known as the Centre frequency denoted by fc.

The bandpass filter is so called because it only passes signals within a certain bandwidth, which is defined as being the frequency range contained between the two points that are 3dB below the resonant peak. There is a fixed relationship between Centre frequency (fc), bandwidth (fbw) and Q factor, given by $Q = fc/fbw$.

The centre frequency is given by $fc \simeq 1/2\pi\sqrt{LC}$ Hz. This is only approximate, as it assumes that the value of R is relatively low. As R decreases, the Q factor increases. Thus R has the effect of damping the resonances, and so as it approaches zero ohms, Q approaches infinity.

The phase shift is shown in Fig. 3c. As this filter is a second order structure, then the total phase movement will be twice that for a first order structure, i.e. 180 degrees. Fig. 3d shows the phase and frequency responses for different values of Q. Note that a high Q has a very rapid rate of change of phase, a low Q has only a slow rate of change.

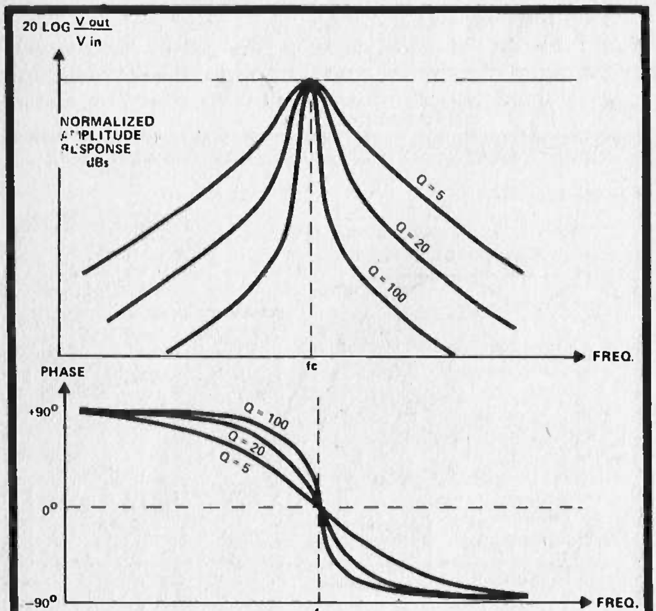


Fig. 3d. Effect of varying Q on the frequency and phase characteristics.

Time please

Bandpass filters also have a time response, as opposed to their frequency response. When an impulse is applied to a bandpass filter it rings (Fig. 3e). The filter oscillates at the centre frequency, fc, the amplitude of the oscillations decaying exponentially in time. The ringing time, T_r, is the time taken for the oscillations to decay to 37% of their initial value. Ringing time is related to the Q and fc by the following equation:

$$T_r = Q/2\pi fc$$

When a high Q filter has been constructed, it may prove difficult to measure its Q factor accurately due to the narrowness of its bandwidth. However, if the filter is made to ring, a reasonably accurate measurement of the Q can be obtained by measuring T_r and fc.

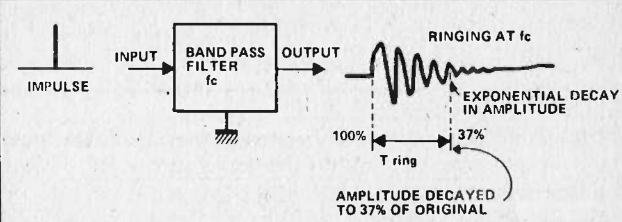


Fig. 3e. Ringing in a band-pass filter.

Filter type	Band pass
Filter order	Second order
Roll off slopes	+ and −6dB/octave greater near to resonance
Centre frequency fc	$fc \sim 1/2\pi\sqrt{LC}$
Phase shift at fc	0°
Q factor	fc/fbw where flow is the 3dB bandwidth
3dB bandwidth fbw	fc/Q
Ringing time, Tr	$Q/2\pi fc$

TABLE 3. Summary of band-pass filter.

Failed band

Another common filter structure is the band reject or notch filter. There are many ways of realising this filter, one of which is shown in Fig. 4. The input signal is subtracted from the bandpass output. By adjusting Ra with respect to R, complete cancellation can be obtained at fc.

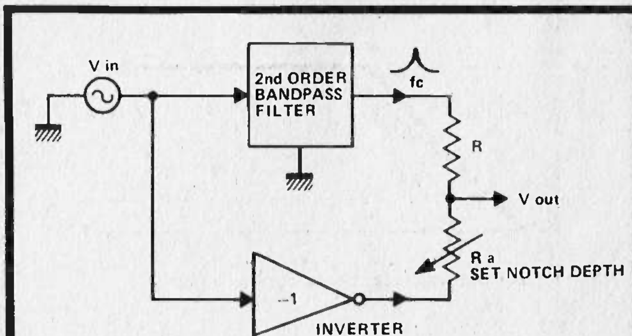


Fig. 4. Notch filter using Op-Amps.

Thus the centre frequency of the bandpass filter is the centre frequency of the notch, whose depth can be varied by altering Ra.

Very deep notches are possible, 50dB is easily obtained. As the Q of the bandpass filter is increased, so is the Q of the notch filter. However, Ra will have to be reset for each value of Q.

Filter Order

Consider the ideal low pass filter shown in Fig. 5a. Its response is flat right up until the break frequency fb. Frequencies above fb are attenuated to nothing! You won't be surprised to learn that filters like this don't exist. However, it is a common requirement to produce filters with very steep roll off slopes and this is achieved by designing filters with lots of sections, to increase the filter order. Each reactive element in the filter increases the filter order by one, therefore a low pass active filter with three capacitors is known as a third order filter and will have an ultimate roll off of three times 6dB/octave, which is 18dB/octave.

However, designing a third order lowpass filter is not just a simple case of sticking three first order RC circuits in a line. What you get when you do this is a very soggy curve indeed! The filter should be flat in the pass band, then it should turn over and rapidly assume its ultimate roll off slope. Examples of this type of Maximally flat filter are shown in Figs 5b and c. The effect of order number upon a bandpass filter is shown in Fig. 5d.

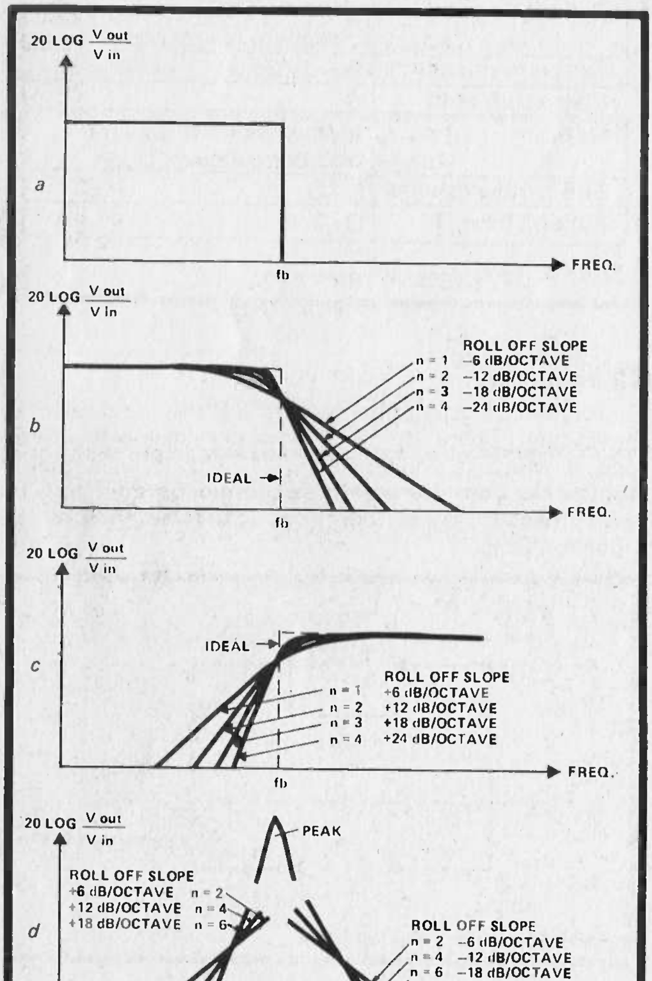


Fig. 5a. Ideal low-pass response;
b, Examples of maximally flat;
c, Filter responses;
d, Effect upon band-pass filter of increasing order number.

Later on in this series the circuit diagrams and design charts are given for various filter types and order numbers. It would seem that to get a filter to approach its ideal response, all that is needed is to increase the order number. This is in fact true, but there are certain tolerance problems. (When 8th order filters are designed, component tolerances of about 1% are required!)

Which filter shape?

The type of filter that is chosen to do a particular job will depend on what parameters are thought to be important. There are three basic characteristics to be considered (lowpass and highpass filters only).

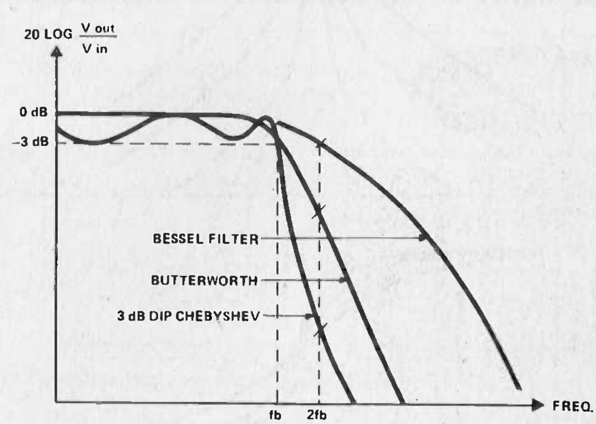
1. Good transient response.
2. Maximum flatness of the filter within its passband.
3. Maximum rolloff slope outside the passband.

The type of filter used should be chosen to fit the job that they are being designed for. The filters have been categorised into three basic types for the purpose of simplicity.

Filter number 1 is known as a Bessel filter. Its phase changes almost linearly with frequency. It is useful for systems where a good transient response is required, such as joining the dots up on the output of a digital to analogue converter. It has a very poor initial roll off slope.

Filter number 2 is known as a Butterworth filter. It has the flattest pass band possible. Its other two parameters are a compromise. That is it has a reasonable overshoot and a fairly fast initial roll off.

Filter number 3 is known as a Chebyshev filter. It has some ripple in its pass band, although this is small, and a very fast initial roll off, and a poor transient response.



ATTENUATION AT FIRST OCTAVE (2 fb)

*3 dB CHEBYSHEV	17	28	39	51	62	75
BUTTERWORTH	12	18	24	30	36	42
η/FILTER ORDER	2	3	4	5	6	7

*NOTE THE IMPROVED ATTENUATION

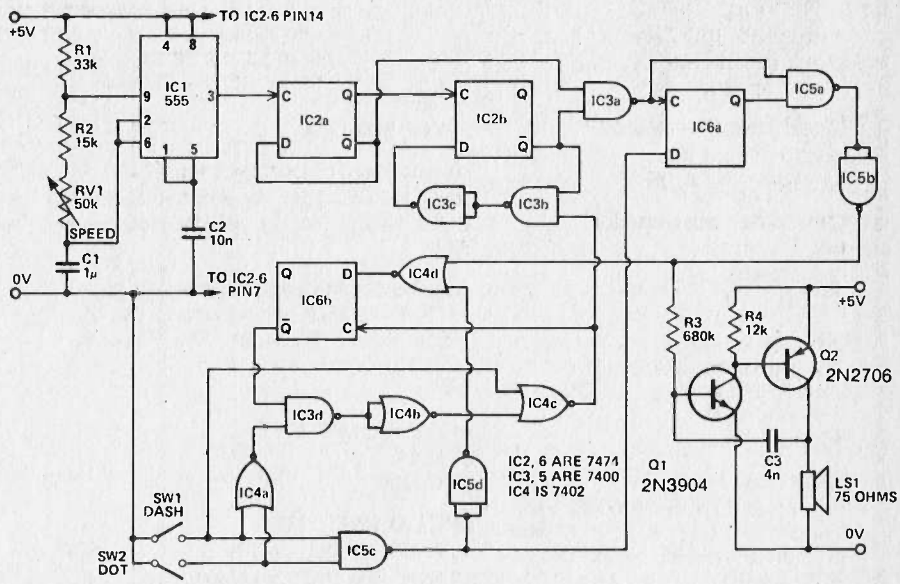
Fig. 6. Response of all three types of filter discussed, with table showing variation in attenuation between them.

Next Month: Full design charts and circuits for three types of Active Filter.

tech-tips

Tech-Tips is an ideas forum and is not aimed at the beginner. ETI is prepared to consider circuits or ideas submitted by readers for this page. All items used will be paid for. Drawings should be as clear as possible and the text should preferably be typed. Circuits must not be subject to copyright. Items for consideration should be sent to ETI TECH-TIPS, Electronics Today International, Unit 6, 25 Overlea Blvd., Toronto, Ontario, M4H 1B1.

TTL KEYER



This device can be used to send perfectly spaced Morse at very high speeds - up to twice as fast as with an ordinary Morse key. It uses six integrated circuits, and also requires two special switches, SW1 and SW2, which are described later.

To describe the operation of the circuit fully would take up over a page of ETI, and so a simplified explanation is given here. IC1 is a 555 timer connected as an astable multivibrator, whose frequency is varied by RV1. The output is fed to IC2a, a D flip-flop, which divides the input frequency by 2, producing a square-wave with a 1:1 mark-space ratio (dots).

If SW1 and SW2 are both open, the D inputs of IC2b and IC6a are both at logic 0, so that the dots from IC2a are inverted by IC3a, but blocked by IC5a. IC5b output is a 0, and so the audio oscillator made up of Q1 and Q2 and the associated components is disabled and no tone is fed to the speaker.

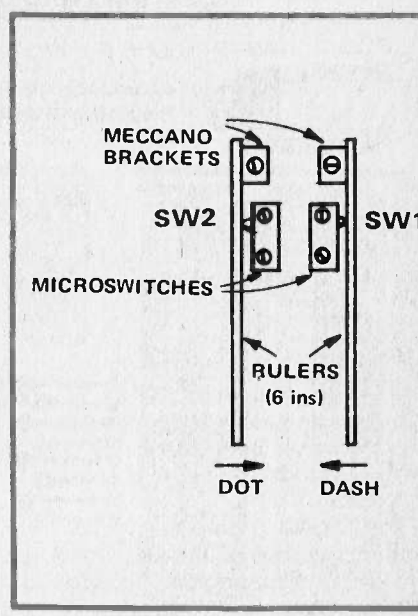
If SW2 is closed, IC6a's D input becomes logic 1. However IC6a's output can only change state on the rising edge of a clock pulse (i.e. the beginning of a dot). Hence if a dot has already started when SW2 is closed, it will not get through to the speaker, but the next dot will, because it will make IC6a's Q output to go to 1. Hence the dots now get through to the oscillator and successively enable and disable it, causing dots to be heard coming from the speaker. When SW2 is opened, if a dot is in progress it will continue until it has finished, and then at the beginning of the next dot, IC5a output will go low and no more dots will be heard. There is a short delay between the beginning of the dot and the Q output going low, which does cause a short 'blip' at IC5b output, but the blip is too fast to be heard.

If SW2 is closed, but SW1 open, IC4c output goes to 1 and IC2b's $\overline{Q}$ output is effectively shorted to its D input. This causes IC2b to divide the string of dots from IC2a by two. The outputs of IC2a and IC2b are combined by IC3a to produce a waveform with a 3:1 mark-space ratio (dashes). These are passed on to the audio oscillator just as before. The dashes, like the dots, are self-completing. Notice that IC4c output determines whether dots or dashes are produced.

While SW1 or SW2 is closed, IC6b's D input is fed from IC4c output and clock pulses come from IC5b. If SW1 and SW2 are both operated together, IC4a allows the output of IC6b to pass to IC4c, and IC4d inverts IC4c output again, so that IC6b $\overline{Q}$ output is shorted to its D input. Thus IC6b changes state every time a dot or dash begins at the output, and causes alternate dots and dashes to be produced. This is useful when sending a letter like C (dash-dot-dash-dot), as the switches SW1 and SW2 each need to be closed and opened just once.

It was found after the unit had been built, that it was difficult to send a letter like A (dot-dash) at high speed because SW1 had to be closed a fraction of a second after SW2, which was difficult to achieve at the first attempt. Hence IC5d and IC4d were added. When both switches are released, IC6b input becomes 0. A clock pulse is then applied to IC6b by the 'blip' described earlier. This makes the output go high, and if now SW1 and SW2 are closed simultaneously, the first thing to be heard in the speaker will be a dot.

SW1 and SW2 are push-button microswitches, and these are operated by means of a lever arrangement as shown in the diagram. Plastic rulers were used on the unit built because they are flexible.



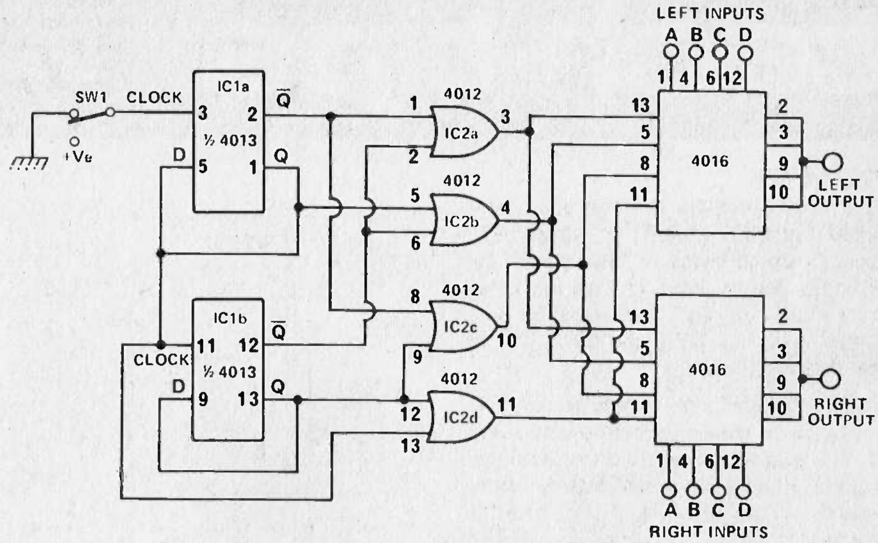
The component values shown around IC1 give a speed range of 11-30 words per minute. The upper limit can be raised by decreasing R2. I have so far reached a speed of 20 wpm on the unit, after only a week or so of using it. As it stands it is a Morse practise unit, but if IC5b output is taken to a transistor driving a relay, the relay contacts could be used in place of an ordinary Morse key in a C.W. transmitter.

Stereo Input Selector

T. E. Huffinley

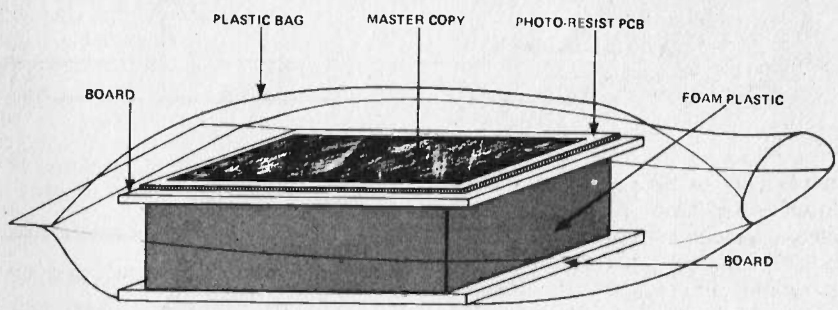
Four different inputs can be switched through by the continual pressing of SW1. IC1 is a dual 'D' type flip flop. The Q outputs are connected to the D inputs so that the clock inputs are divided by two. The two flip-flops are connected in series, giving a two stage binary counter.

IC2 is a quad OR gate. This is used to decode the four states of the counter. The outputs are used to control the quad switches of IC3 and IC4 (4016AE).



PCB Bag

L. Rink



A piece of foam plastic is placed between two boards of about the same size as the PCB being printed, on top of this is placed the Photo-resist PCB with the master copy transparency in position on top. The whole of this is put inside a plastic bag and then sqaushed flat.

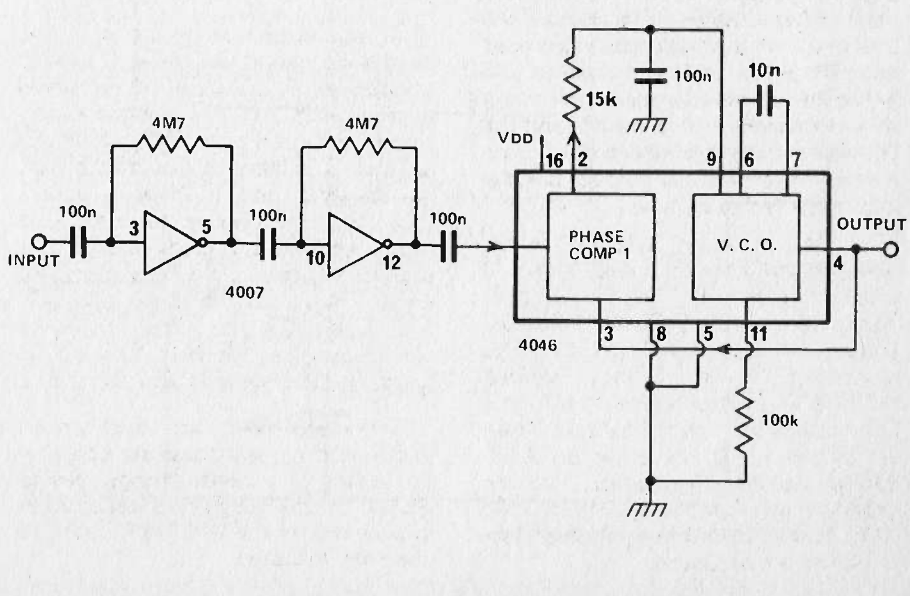
The end of the plastic bag is then sealed by folding over, and then when the pressure is released, and the plastic foam tries to expand, air pressure presses the transparency tight against the PCB and usually can hold it for several minutes.

Guitar Synthesiser

R. Barnett

This circuit uses a CMOS Phase Locked Loop, the 4046, to produce a very unusual sound from a guitar, which sounds something like a synthesiser.

The signal from the guitar is amplified by two of the amplifiers in the 4007. The amplified signal is used by the phase comparator to lock the VCO to the frequency of the note played. The VCO does not oscillate until a note is played, when using the low pass filter shown (i.e. the 15 k resistor and 100 n capacitor). If the value of the resistor is increased, the VCO oscillates continuously at about 1 kHz (with no input signal). This gives very smooth note changes. The basic frequency may be changed by varying the 100 k resistor.



tech-tips

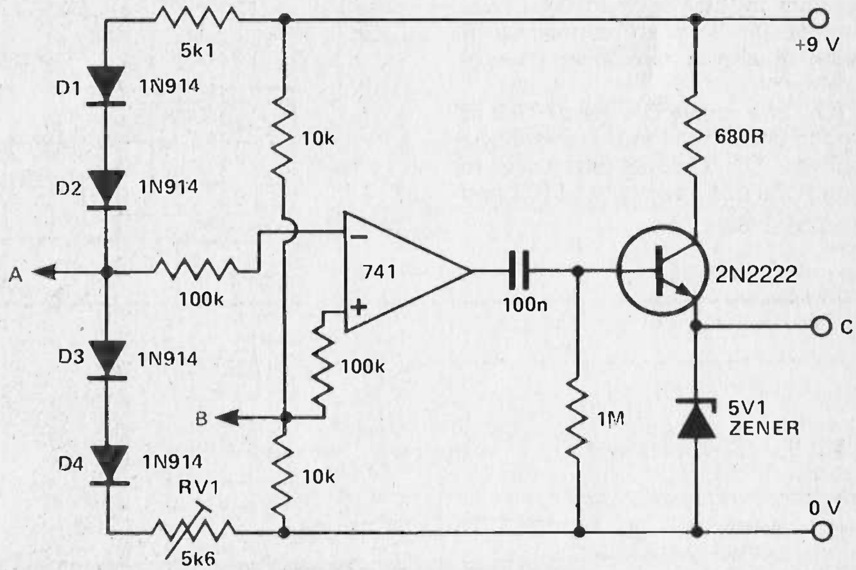
Thermo Touch Switch

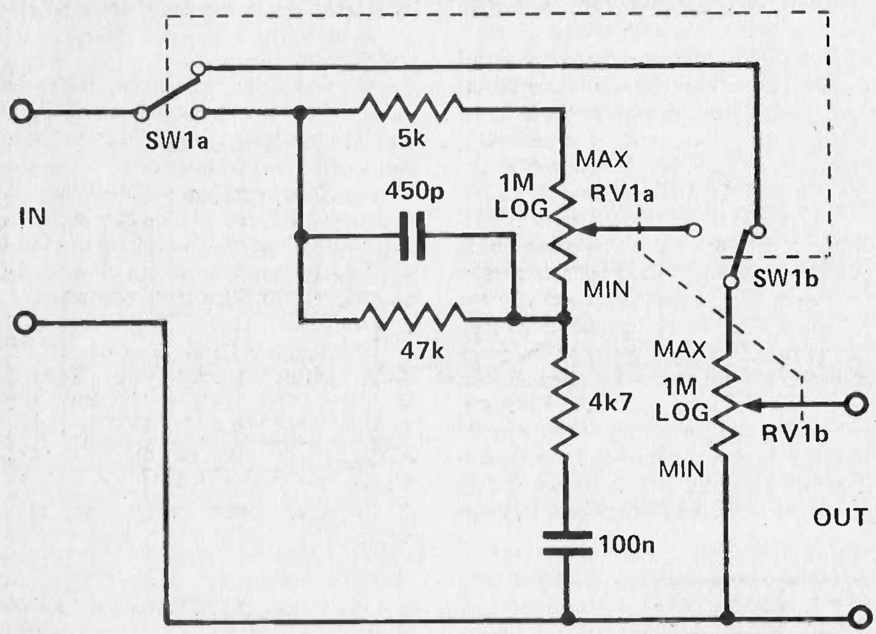
S. B. Dick

The following touch switch works on the temperature dependence of the forward voltage of silicon diodes. At 0 °C this is about 650mV, but drops by 2mV per °C increase in temperature.

When a finger is placed on D3 and D4 the voltage at A will drop below that at B and the O/P of the Op-Amp will go high, causing a TTL compatible pulse to appear at C. D1 and D2 provide compensation against ambient temperature changes. VR1 is initially set so that VA is greater than VB by about 10mV.

The system has the intrinsic advantage that it may be used in moisture-prone conditions in which ordinary touch switches would be most unsatisfactory due to their principle of operation.





Loudness Control

David Chivers

This loudness control works with the volume control to provide a more even listening contour. Since the human ear can hear sound in the middle of the audio spectrum better than at the extremities, it is desirable to attenuate high and low frequencies less than the middle frequencies as the volume is cut.

With SW1 on, bass and treble are boosted relative to middle frequencies. RV1a is ganged to the volume control, this varies the strength of loudness control so that at low volume the effect is more noticeable. This unit will replace the volume control in a present system, coming between the preamp and power amplifier. It a stereo unit is to be made, SW1 should be four pole two way, and it is best to have separate volume / loudness controls for each channel since four way potentiometers are hard to find.

ETI DATA SHEET

LM 2907, LM 2917 FREQUENCY TO VOLTAGE CONVERTORS — NATIONAL

The LM2907, LM2917 series are monolithic frequency to voltage converters with a high gain op amp/comparator designed to operate a relay, lamp, or other load when the input frequency reaches or exceeds a selected rate. The tachometer uses a charge pump technique and offers frequency doubling for low ripple, full input protection in two versions (LM2907-8, LM2917-8) and its output swings to ground for a zero frequency input.

Features

The op amp/comparator is fully compatible with the tachometer and has a floating transistor as its output. This feature allows either a ground or supply referred load of up to 50 mA. The collector may be taken above V_{CC} up to a maximum V_{CE} of 28V.

The two basic configurations offered include an 8-pin device with a *ground referenced tachometer* input and an internal connection between the tachometer output and the op and amp non-inverting input. This version is well suited for single speed or frequency switching or fully buffered frequency to voltage conversion applications.

The more versatile configurations provide differential tachometer input and uncommitted op amp inputs. With this version the tachometer input may be floated and the op amp becomes suitable for active filter conditioning of the tachometer output.

Both of these configurations are available with an active shunt regulator connected across the power leads. The regulator clamps the supply such that stable frequency to voltage and frequently to current operations are possible with any supply voltage and a suitable resistor.

Applications

The LM2907 series of tachometer circuits is designed for minimum external part count applications and maximum versatility. In order to fully exploit its features and advantages let's examine its theory of operation. The first stage of operation is a differential amplifier driving a positive feedback flip-flop circuit.

The input threshold voltage is the amount of differential input voltage at which the output of this stage changes state. Two options (LM2907-8, LM2917-8) have one input intermally grounded so that an input signal must swing above and below ground and exceed the input thresholds to produce an output. This is offered specifically for magnetic variable reluctance pickups which typically provide a single-ended ac output. This single output is also fully protected against voltage swings to ±28V, which are easily attained with these types of pickups.

Following the input stage is the charge pump where the input frequency is converted to a dc voltage. To do this requires one timing capacitor, one output resistor, and an integrating or filter capacitor. When the input stage changes state (due to a suitable zero crossing or differential voltage on the input) the timing capacitor is either charged or discharged linearily between two voltages whose difference is $V_{CC}/2$. Then in one half cycle of the input frequency or a time equal to $1/2\ f_{IN}$ the change in charge on the timing capacitor is equal to $V_{CC}/2 \times C1$. The average amount of current pumped into or out of the capacitor then is: $= V_{cc} \times f_{IN}$

The output circuit mirrors this current very accurately into the load resistor R1, connected to ground, such that if the pulses of current are integrated with a filter capacitor, then $V_o = i_c \times R1$, and the total conversion equation becomes:

$$V_O = V_{CC} \times f_{IN} \times C1 \times R1 \times K$$

Where K is the gain constant — typically 1.0.

Applications

- Over/under speed sensing
- Frequency to voltage conversion (tachometer)
- Speedometers
- Breaker point dwell meters
- Hand-held tachometer
- Speed governors
- Cruise control
- Automotive door lock control
- Clutch control
- Horn control
- Touch or sound switches

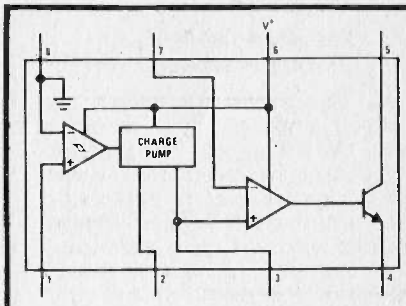
Absolute Maximum Ratings

Supply Voltage	28V
Supply Current (Zener Options)	25 mA
Collector Voltage	28V
Differential Input Voltage	
Tachometer	28V
Op Amp/Comparator	28V
Input Voltage Range	
Tachometer LM2907-8, LM2917-8	±28V
LM2907, LM2917	0.0V to +28V
Op Amp/Comparator	0.0V to +28V
Power Dissipation	500 mW

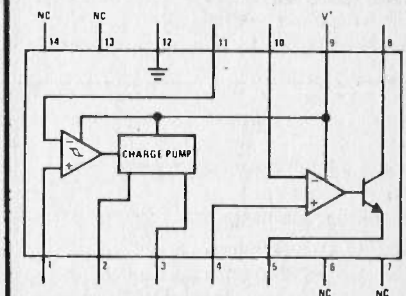
Choosing R1 and C1

There are some limitations on the choice of R1 and C1 which should be considered for optimum performance. The timing capacitor also provides internal compensation for the charge pump and should be kept larger than 100 pF for very accurate operation. Smaller values can cause an error current on R1, especially at low temperatures. Several considerations must be met when choosing R1. The output current at pin 3 is internally fixed and therefore $V_o/R1$ must be less than or equal to this value. If R1 is too large, it can become a significant fraction of the output impedance at pin 3 which degrades linearity.

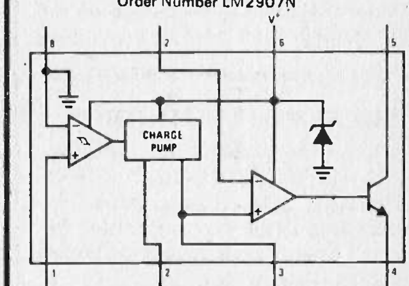
It appears R1 can be chosen independent of ripple, however response time, or the time it takes V_{OUT} to stabilize at a new voltage increases as the size of C2 increases so a compromise between ripple, response time, and linearity must be chosen carefully.



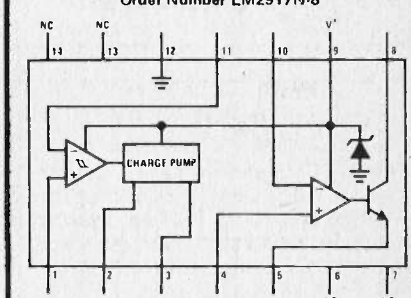
Order Number LM2907N-8



Order Number LM2907N



Order Number LM2917N-8



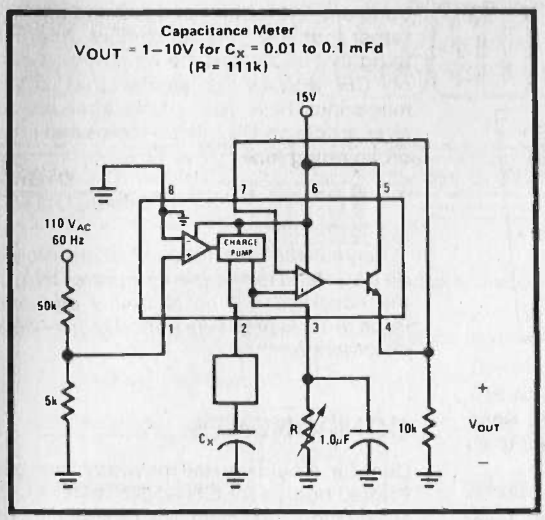
Order Number LM2917N

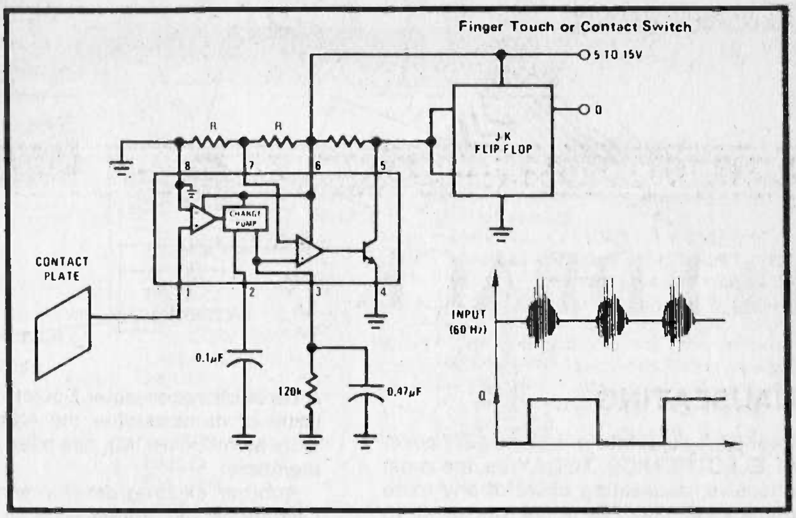
As a final consideration, the maximum attainable input frequency is determined by V_{CC}, C1 and I_2:

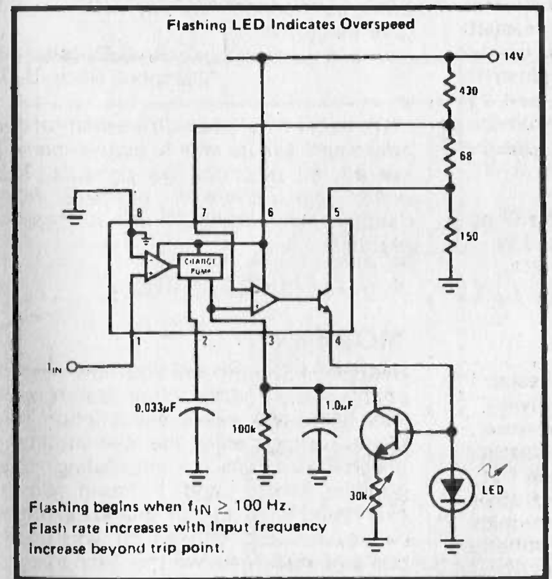
$$f_{MAX} = \frac{I_2}{C1 \times V_{CC}}$$

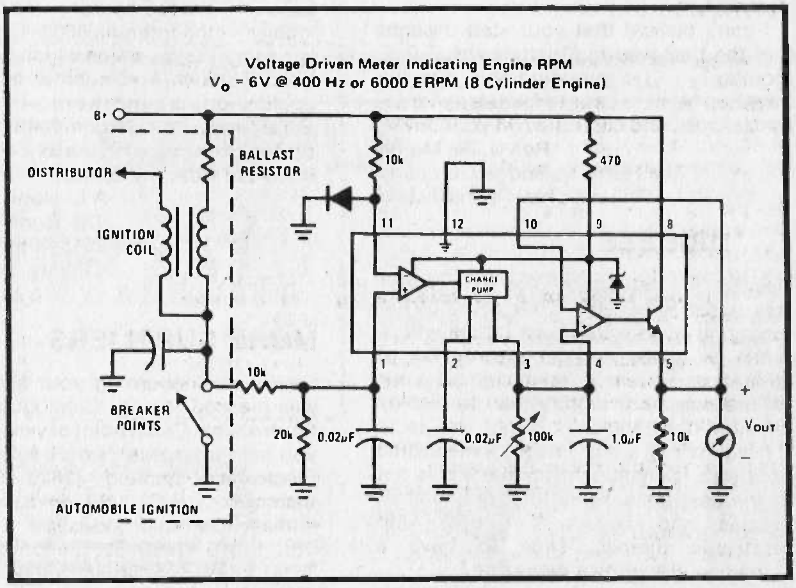
Using Zener Options

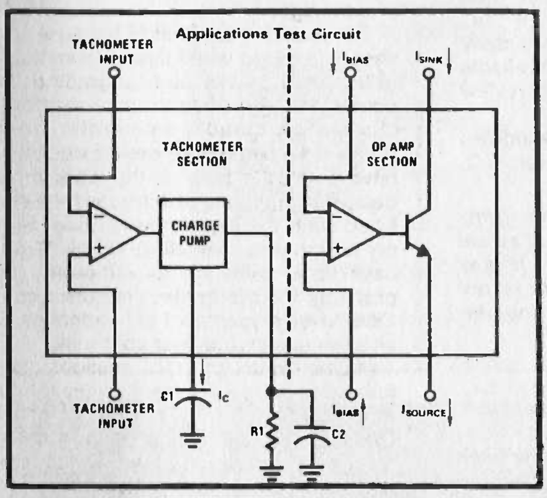
For those applications where an output voltage or current must be obtained independent of supply voltage variations, the

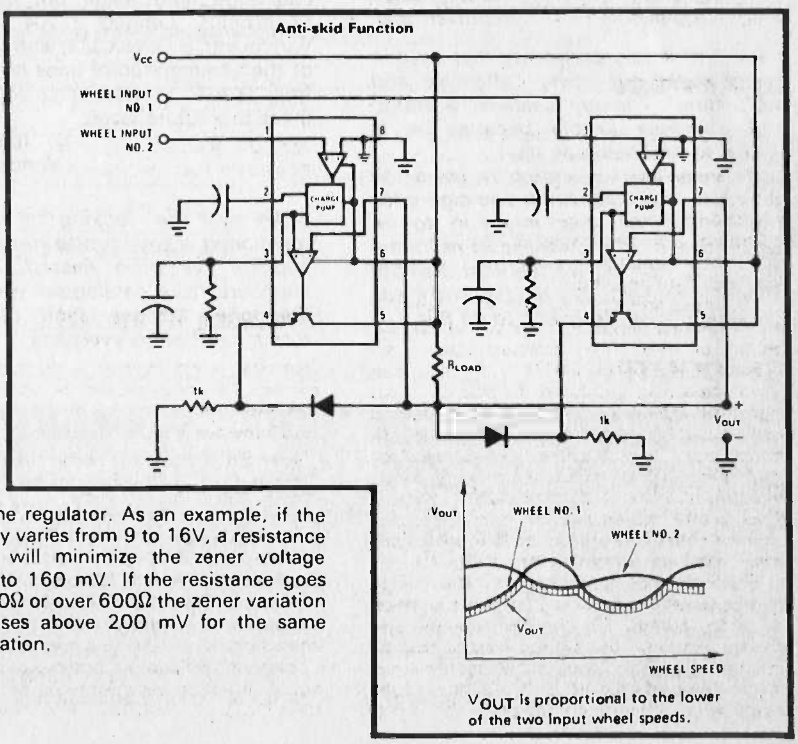












LM2917 is offered. The most important consideration in choosing a dropping resistor from the unregulated supply to the device is that the tachometer and op amp circuitry alone require about 3 mA at the voltage level provided by the zener. At low supply voltages there must be some current flowing in the resistor above the 3 mA circuit current to operate the regulator. As an example, if the raw supply varies from 9 to 16V, a resistance of 470Ω will minimize the zener voltage variation to 160 mV. If the resistance goes under 400Ω or over 600Ω the zener variation quickly rises above 200 mV for the same input variation.