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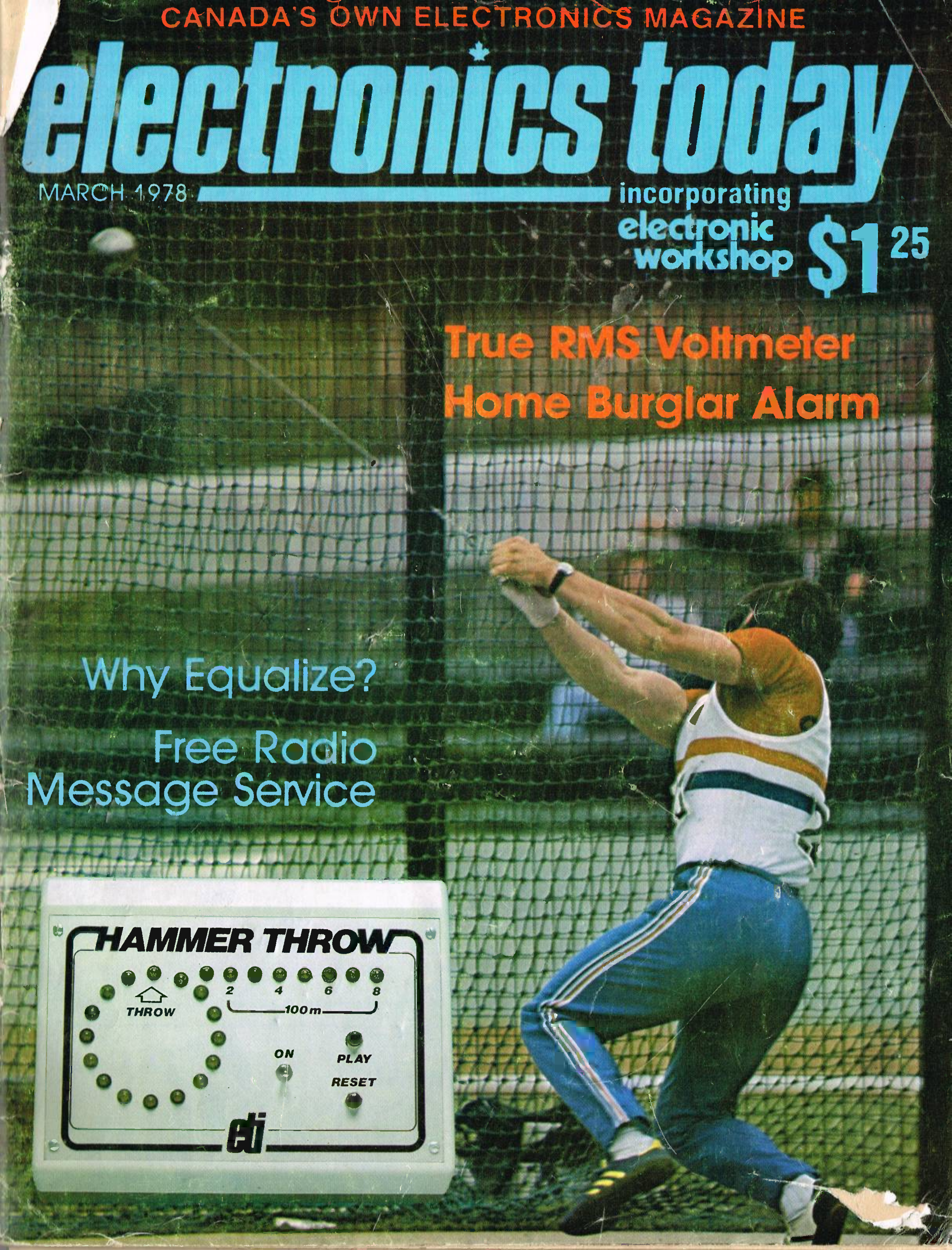

VOL. 2 NO. 3
MARCH 1978

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EDITORIAL AND ADVERTISING OFFICES
Unit 6, 25 Overlea Boulevard, Toronto, Ontario, M4H 1B1
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Printed by Livingstone Printing Ltd.
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
Electronics Today International
25-27 Oxford St., London W1R 1RF, UK.
Editor Halvor Moorshead.

Electronics Today International,
Ryrie House, 15 Boundary St., Rushcutters Bay, Sydney, Australia
Editor Collyn Rivers.

Electronica Top Internationaal,
Postbus 260, Emmen, Holland.
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electronics today international

CANADA'S OWN ELECTRONICS MAGAZINE

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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.

What Is Equalization?

A dictionary would tell you that to equalize is to make equal or uniform. Wally Parsons, ETI's Contributing Audio Editor, discusses.

ALL MEN (and women) are said to be created equal, but they don't all stay that way. Some become fat, others thin, some tall, some short, some rich, others poor, in other words, some are more equal than others. And the same is true with audio electrical and acoustical signals. Sure, they start off okay, but as soon as a musician pushes a sound out of his horn and sends it hurtling to a microphone little gremlins start chewing at it as it makes its tortuous way through the air, into the microphone, the bewildering maze of wires and transistors, cutters, pickups, loudspeakers, listening rooms. Indeed, it often seems miraculous that it emerges from all this as something even vaguely resembling the original. And then, of course, there is man, ever ready to show mother nature the errors of her ways, tinkering with this signal to make it conform to his own concept of perfection.

Then, too, sound was never meant to be recorded; therefore we have no choice but to make our equipment conform to the nature of sound, because the laws of physics are definitely not going to change to suit our convenience, except possibly to do us mischief, as outlined in Murphy's Law.

Okay then, what do we equalize, how do we equalize, and, for that matter, why do we equalize?

TYPES OF EQUALIZATION

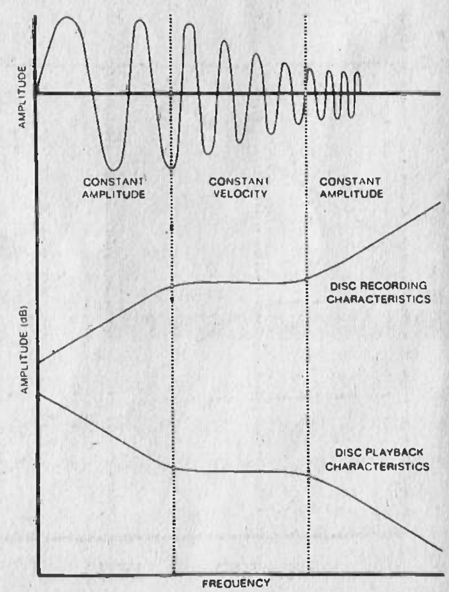
Equalization can be divided into two basic types as regards to function: **Correctional,** in which the purpose is to correct for faults in some part of the chain which produces deviations from flatness in frequency response, and **Adaptive** in which the response is deliberately caused to deviate from flat in order to improve the operating characteristic of some component in the chain, or to allow optimizing some other parameter, such as noise or distortion. In general, correctional equalization is introduced at an operator's discretion, while adaptive equalization is either imposed by physical conditions over which one has no control, or is established by agreed upon standards, or both.

ADAPTIVE EQUALIZATION

Several problems occur if we attempt to record or broadcast an audio signal and reproduce it with a flat characteristic from microphone to loudspeaker. As an example let us consider the case of a cutter engraving a disc recording. If a constant voltage is applied to the cutter, this will translate as a constant velocity of cutting stylus motion. Now, suppose we attempt to record a signal of 1000 Hz and at an amplitude such as to produce a stylus velocity of 10 cm/sec. One cycle will occur in 1 msec and will result in a stylus swing in each direction of 0.025 mm, that is, it will reach maximum displacement in one-quarter cycle which takes 0.25 msec. Now, if we record a signal of 100 Hz at the same velocity, the amount of displacement will be TEN TIMES the 1000 Hz value, or 0.25 mm. 20 Hz would cause a swing of 1.25 mm. Now try to visualize this on a microgroove disc. To allow adequate spacing between grooves, even with no safety factor, would require spacing the grooves at least 2.5 mm apart. If we recorded over a total of 7 cm of the record surface at 33.3 rpm our maximum possible recording time would be just over 8 minutes. Remember, this assumes no guard space between grooves, which would easily cut this time in half, and assumes we can use the full 7 cm, which would be most unlikely for reasons beyond the scope of an article on equalization. It also assumes we can cut such an amplitude without running into formidable problems with the cutter, and that we can find a pickup which would trace such an amplitude. Remember, too, that 10 cm/sec is not that high a velocity. Clearly, some compromise must be made.

This adaptation consists of "equalizing" the recording system so that all frequencies below an agreed upon standard, namely 500 Hz will be recorded at a constant **amplitude** rather than velocity. This results in an attenuation at the rate of 6 db/octave, and in actual practice this curve is modified at frequencies below about 80 Hz.

Fig. 1. Characteristics of phonograph recording and playback processes.



Now, it just so happens that magnetic pickups are velocity responsive devices, that is they give equal output voltage for equal stylus velocity. Since our recording was made with constant amplitude below 500 Hz, the velocity falls as frequency goes down, and if played back with a magnetic pickup the response will fall at the rate of 6 db/octave. Clearly, we must now equalize this response by introducing a response characteristic which increases at this rate as frequency is reduced.

The reader who has been following this closely and who has some knowledge of noise (not the kind sometimes called "music", but the other kind) will realize by now that if we continue to record at a constant velocity as frequency rises above our 500 Hz turnover, eventually the point will be reached at which noise generated by surface irregularities in the recording will be equal to or greater than our signal. In addition, noise generated in the pre-amplifier will assume a high level in comparison to the signal. The reader will also realize that this does not have to be, since if we continue to record at a constant **amplitude** we can overcome this noise in much the same way as we did at lower frequencies, i.e.: increase response at the rate of 6 db/octave as frequency rises (this is just another way of describing a 6 db/octave roll-off as frequency drops). It will be appreciated that this could result in stylus velocities beyond the capabilities of the pickup cartridge, and indeed this is one reason for the use of a modified constant amplitude characteristic; statistical distribution of energy also alleviates some of the potential problems, which makes a fair amount of boost possible.

Another example of this type of equalization is encountered with magnetic tape. A completely loss-free system would show a playback response characteristic which rises at the rate of 6 db/octave when the tape is recorded with constant flux in the gap, which, in turn, is the result of constant current through the coils of the record head. However because of the tape and head characteristics the response will begin to level off and ultimately drop as frequency rises. The 6 db/octave slope can be readily corrected (and is) by an appropriate low frequency boost circuit in the playback system, but the high end loss is primarily the result of tape self-demagnetization and playback head losses, with the tape loss characteristic playing a prominent part at lower speeds. This means that inherent tape noise will eventually swamp the signal so that we cannot restore flat response in the playback equalizer, but must do so during recording.

Unlike disc recording the amateur tape recordist is in a position to impose operating conditions during the recording process which conflict with the realities which we have been discussing. Increasing the signal level during the recording process brings the risk of overloading the tape or requires a reduction of overall level which causes deterioration in signal/noise ratio. Boosting response during playback also increases noise in the active region of the equalizer. As is so often the case in audio work, the end result is a compromise or, with luck, a fine balance between various conflicting requirements.

At the present time it is not my intention to get too involved with equalization circuits; volumes have been written on individual aspects of equalization and doubtless more will be written in the future. However, a brief examination of methods is useful in order to understand the proper use of equipment.

Fig. 2 shows a simple high-pass passive filter, having a first order, or 6 db/octave slope, along with a normalized frequency and phase response curve. Fig. 3 shows its low-pass counterpart. In figs. 4a and 4b we see these circuits modified to provide low boost and high boost respectively. Notice that the low-boost circuit and its characteristic are derived from the low-**pass** circuit while the high-boost circuit and its characteristic are derived from the high-**pass** circuit. It is also clear that a low-pass and a high-cut characteristics are, strictly speaking, the same thing, and a high-pass and low-cut are also equivalent. This has led to the mistaken belief which is often encountered even today that bass boost and treble cut are the same thing and vice-versa. An examination of the modified circuit used to provide actual boost immediately shows the fallacy of this notion. **ALL** equalizers **attenuate ALL** frequencies equally, except in the relatively narrow region in which boost or cut is required.

LOCATION OF CIRCUIT

The location of an equalizer in a circuit usually involves the reconciliation of several conflicting elements, and is generally determined by the type of circuit (bass boost, treble boost, etc.) nominal signal level, and the operating band.

Since a bass boost circuit involves considerable reduction in mid and high frequency level it is generally desirable to put most of the system gain before the equalizer in order that noise components may be attenuated along with signal. This is also true for high frequency attenuation. However, hum components would then be boosted along with signal; therefore, too much gain will require special attention to design aspects aimed at minimizing hum. Conversely, a high frequency boost circuit should be inserted early enough in the system as to raise signal above the noise of succeeding stages.

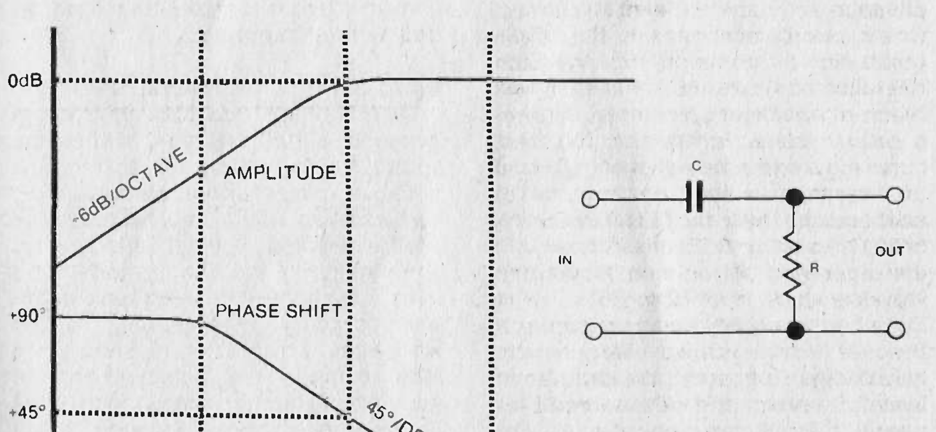


Fig. 2. Transfer characteristics of simple single pole RC high pass filter.

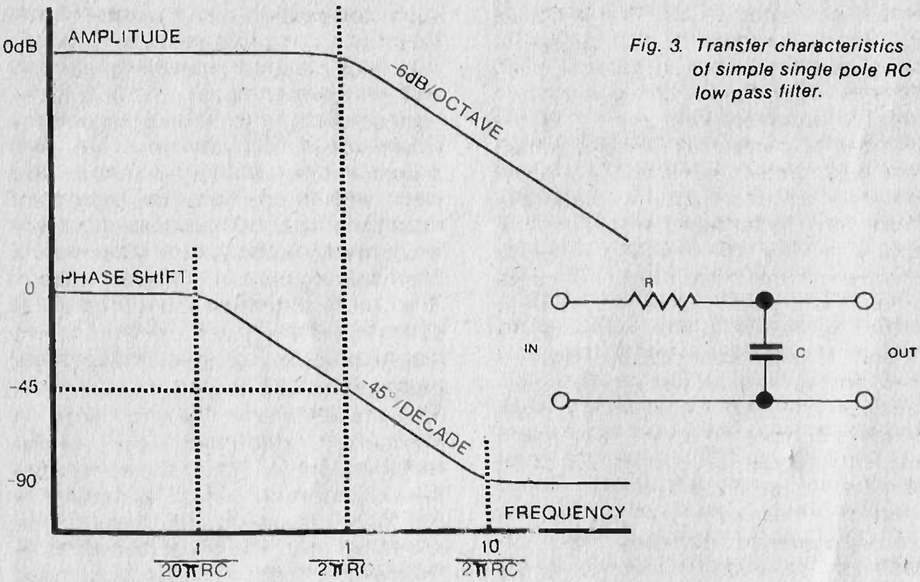


Fig. 3. Transfer characteristics of simple single pole RC low pass filter.

Where signal level is fairly high, as with some high output magnetic pickups such as Decca and Empire, our greater concern is amplifier overload particularly at high frequencies, in stages prior to the equalizer.

If you're starting to get the idea that perhaps equalization, when combined with pre-amplification, can best be accomplished when the equipment is designed for, and associated with, the components with which it is to be used, then you're right on target. Indeed, this is generally considered to be good practice in professional circles. Not only are tape equalizers incorporated into the tape machines with which they are to be used, but turntables may incorporate the required pre-amplifier/equalizer circuits. In over twenty years of audio work I have yet to comprehend the logic behind the common practice of building magnetic inputs into a control unit. But it helps to explain the differences often encountered between results published in equipment reviews and the users' own experience.

PHASE

Before moving on to the subject of corrective, or elective equalization some attention should be paid to the matter of **phase**. Hundreds of dollars are often spent on the construction or purchase of, for example, a phono preamp, and great attention paid to noise, channel balance, overload, accuracy of equalization (to within 2 db) and yet it is seldom realized that, in producing a stereo image, one of the three major factors in directional perception is relative phase. In addition, all matrixed 4-channel systems currently in use utilize a specified phase relationship between channels to encode and decode the additional channels. One common characteristic of all of the impressive demonstrations of quadraphony has been the use of very high quality components. Since equalizers are among the first functions to suffer in making economy cuts in domestic equipment, it's small wonder that quadraphony and even stereo reproduction in the home are often disappointing.

How does this happen? Take another look at the phase and amplitude characteristics in fig's 2 and 3. At the turnover frequency, that is the 3 db down (or up) point, the phase angle has shifted 45° and reaches its ultimate 90° shift a decade away, which also corresponds to the 20 db point. Now, if common 10% and 20% tolerance components are used in two different equalizers (example, each of a pair of stereo channels), the final curve may indeed be well within 1.5 to 2 db tolerance in each channel, but if each yields a difference in phase over a broad frequency band of as little as 30° the difference between channels may vary anywhere from 0° to 60°. This is quite considerable in comparison with the 90° shift called for in the parameters of **any** quadraphonic system, even the Dynaco-Haffler passive ambience network. As for stereo perception, although there is much disagreement among authorities as to the ear's sensitivity to phase shift, much of this disagreement involves steady tone conditions and single channel reproduction. When it comes to the perception of a synthesized stereo image as in present day 2-channel systems, as little as 15° has been observed to have a profound effect on imaging, particularly with regard to depth and elevation. In my own experiments I've been able to produce as much as 10 db channel level difference with no serious effect on the stereo image other than a shift in localization, and yet switching in a simple tone control circuit constructed of standard tolerance parts and with both channels nominally in the flat position will produce a subtle yet real change in image stability and solidity. The indication is quite clear: the precise matching of characteristics between channels is probably of even greater importance than the absolute accuracy of characteristics. This means precision parts and the associated costs.

CORRECTIVE EQUALIZATION

This might also be described as "discretionary equalization", since it refers to alteration of response in a manner and to a degree completely at the discretion of the operator. In a very real sense the concept of "correct" is irrelevent here. Adjustment is in accordance with the ear's own concept of right and wrong. Accordingly, no hard and fast rules can be laid down nor can definite "how to" instructions be given. However, most of the considerations already outlined apply here, so we may now proceed with an examination of the equipment and techniques available, secure in the knowledge that there is no such thing as magic, and that, rather than use technology to make things better, we can only use it to make things not as bad as they might be.

USES OF EQUALIZERS

Discretionary equalization is used because either we don't think the sound we're getting is correct, or because, correct or not, we don't like it and want to make improvements or produce some special effects. It's something like the photographer who uses a skylight filter to obtain a more realistic colour balance, and the one who uses a red filter to simulate a Martian landscape. Since recordings are made by human beings much of the time, who monitor through loudspeakers with their own characteristics in rooms with their own acoustics, and who apply their own judgement as to what the final sound should be like, it is not surprising that the music lover, or audiophile may be in disagreement with the producer from time to time. Perhaps you don't agree that the brasses needed a little extra bite by means of a 7 Khz boost, or you feel that the strings could have been brought more forward and made less disembodied by adding a little mid-range boost. Indeed, perhaps you don't mind a little structural noise of the concert hall, and feel that the producer sacrificed too much bass in order to suppress it. For this kind of correction you will want to use an "equalizer" or tone control of some kind to introduce the appropriate compensation. It may be sufficient to use a simple control circuit such as that outlined in figs. 5 and 6 to get some bass boost, or treble cut. But then there's no way for you to know the exact equalization used in the original recording, and even less chance that a simple circuit could duplicate its mirror image anyway, so at best you can still only adjust it until it sounds better.

If this is not satisfactory, you might try using what is called a "Graphic Equalizer", so called because such a device normally uses slider controls side by side and their settings provide a graphic representation of the response characteristic achieved. With this device you have several controls each of which controls the response over a narrow range of frequencies. It may divide the spectrum into as few as five broad bands or as many as thirty odd very narrow bands. These provide very precise response control indeed, but you still can only adjust response until it sounds right. Just like a simple tone control.

Another type of device is known as a "Parametric Equalizer" because it varies the parameters which define a response characteristic, that is centre frequency, bandwidth, and degree of boost or attenuation. In general, such devices offer fewer choices than a graphic equalizer with regard to the number of centre frequencies which may be operated upon simultaneously. However, in actual use it is generally more flexible largely because of the ability to vary the bandwidth and to choose centre frequencies. In some types it is possible to operate on the same frequency band twice or to operate on two closely spaced frequencies and to combine characteristics to obtain a final response which is completely unobtainable with any other type of component. The parametric equalizer has been widely used in professional work, and is the type of equalizer normally found on each channel of a recording or broadcast production console. There aren't too many in commercial production for consumer use yet, but there is every reason to expect that increasing numbers will be offered to the audiophile. For my money it is the preferred unit of choice for operating on the programme characteristics, provided it is not required to serve other functions.

The graphic equalizer is probably more familiar to the amateur, since it's been around much longer and several such units have been published as construction articles (See ETI Oct. 1977). The great virtue of this unit. especially the 1/3 octave type lies in its ability to notch out or boost one or more very narrow bands of frequencies, making it especially useful in compensating for the irregularities of such components as pickups (RIAA Equalized), loudspeakers, and room acoustics. Indeed one of its earliest and still common applications among professionals is in equalizing control room/speaker systems. For this purpose, the speakers are each fed with pink noise and the response measured via a calibrated microphone and a real time analyzer or other means of measuring response characteristics. The graphic equalizer is inserted in the speaker channel before the power amplifier and adjusted until the desired response (usually flat) is achieved. This procedure is then repeated for each speaker in the system. It should be noted that the response is valid only for the location of the measuring microphone and for the exact acoustical conditions which exist at the measurement.

Two problems arise here, especially for the amateur audiophile. To begin with the previously referred to phase problem can impair the system imaging characteristics. On the other hand it may improve a characteristic which was already deficient because of the phase shifts inherent in the previously uncorrected irregularities.

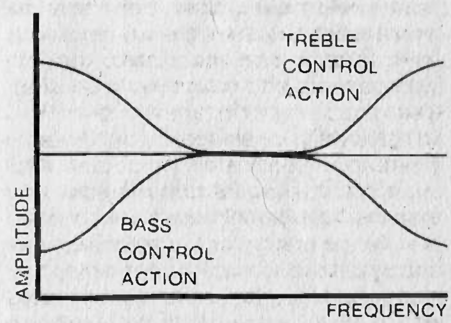


Fig. 6. Range of boost and cut action available from "treble" and "bass" controls.

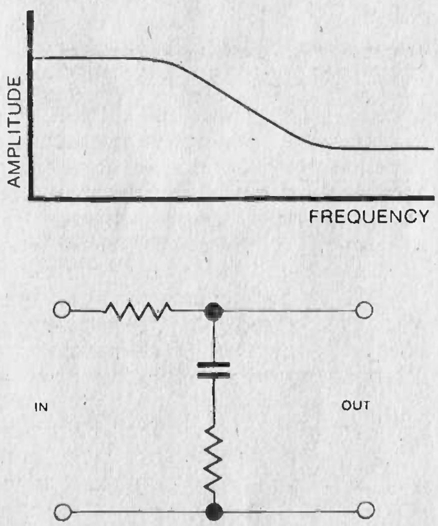


Fig. 4a. Low boost circuit and response curve.

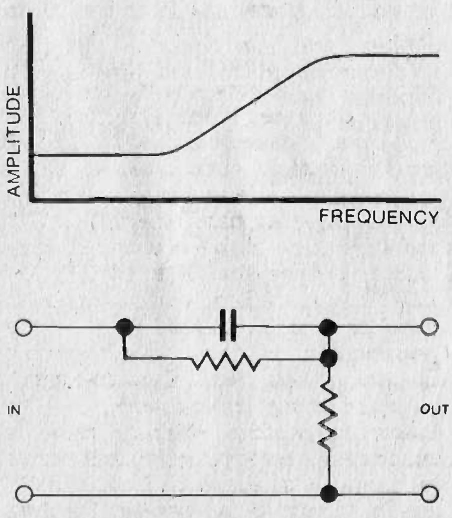


Fig. 4b. High boost circuit and response curve.

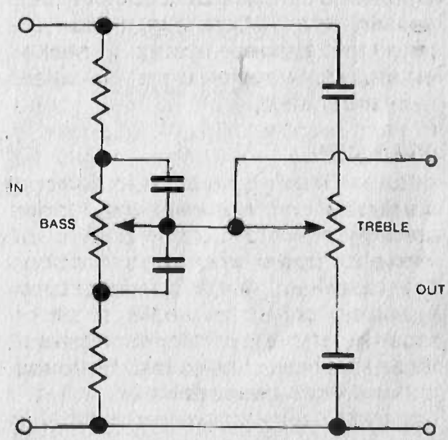


Fig. 5. Typical tone control circuit.

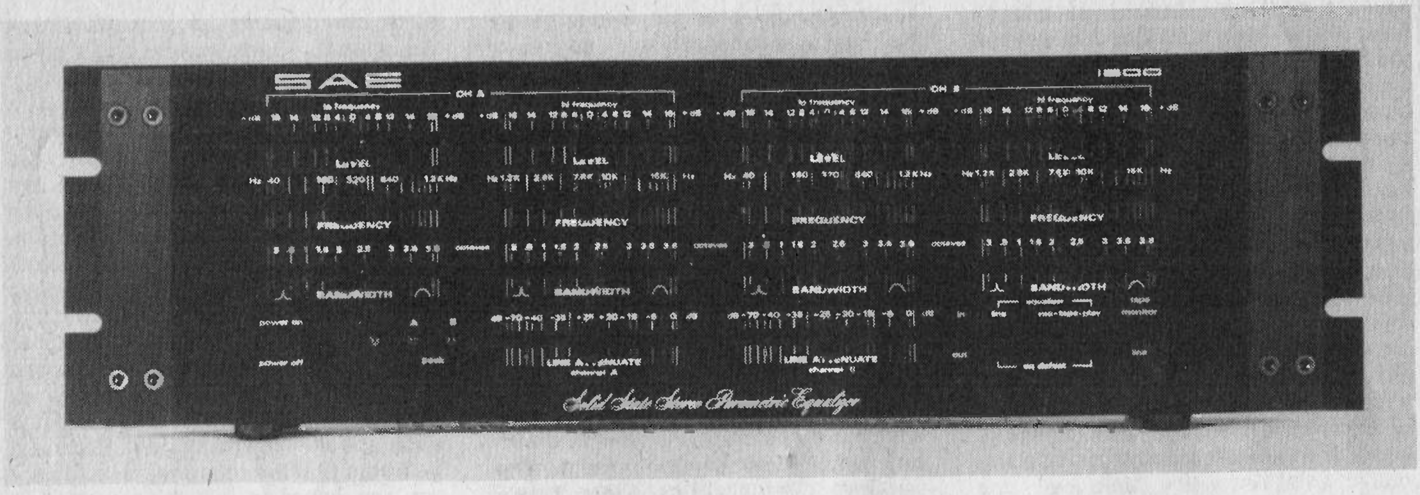


Fig. 7. Example of a parametric equalizer, in this case a two band, two channel model, the SAE 1800.

In other words, the assets may exceed the liabilities, both in qualitative and quantitative terms. That's what I mean by an inability to lay down hard and fast rules.

The second problem is more serious. After purchasing or building such an equalizer, especially the graphic type, there is the temptation to use it to correct faults in what is actually a poorly designed system. I can recall one enthusiast who built such a unit from a kit and installed it in a system which was little more than junk. Aside from sounding terrible, it also burned out speakers and destroyed an output transistor.

Why? Well, take a look at fig. 8 which is the representative response and impedance curve which might be expected of a small bookshelf speaker of the sort which promises to outperform speakers three times its size and selling for ten times the price. Although it may boast a response down to 30 Hz, its response at that frequency is down a good 20 db, and its impedance is equal to the voice coil resistance, around 6 ohms. Now, to flatten the response of such a speaker requires a power increase of 100 times. If, in order to operate at high sound levels, it requires 10 Watts of power in its mid-range, 1000 Watts would be required. That's quite a lot to demand of a 60 Watt amplifier whose ratings are already optimistic, to say nothing of what such power would do to the poor little speaker. Actually, most such equalizers only offer about 12 db of boost, but if this is combined with a so-called loudness control, it's easy to see the kind of abuse possible.

Another point worth considering is that if you use the control to correct for the equipment faults, you can't use it for programme correction at the same time. If you have 10 db boost available and you use it all for boosting speaker response, you have nothing left over for programme correction. And the amount of boost available is a function of the equalizer and power amplifier reserve plus speaker handling capacity.

Another consideration is energy distribution. A common assumption is that power levels tend to be about the same at all frequencies. This is not true, as is demonstrated in fig. 9 an energy distribution curve averaged from the results of a variety of studies. It shows that the largest amount of energy in orchestral music is concentrated in the range between about 100 Hz and 500 Hz dropping off rapidly above and more gradually below this band. It should also be remembered that 500 Hz is a common cross-over frequency in 3-way speakers and that with a current trend away from constant resistance networks, many such speakers exhibit high impedances and considerable reactive components in this region, which imposes severe limitations on the power capabilities of many amplifiers. This, in turn, limits the usefulness of many equalizers in this region, especially in the boost mode, where the result is often high distortion and damaged equipment. With a great deal of hard rock, electronic and synthesizer music high frequency energy tends to be considerably greater than with orchestral music. Now, the tweeter of a 60 Watt speaker system may be capable of handling typically from 5 to 10 Watts of actual power. This is reasonable enough in relationship to the orchestral distribution curve, but an excessive boost in the region handled by the tweeter carries with it the distinct possibility of requiring it to handle anything up to the full output of the amplifier, which may be 60 Watts or more. Further, if the tweeter level is padded down (with an L-Pad, I hope) to match the other drivers, this might save them, but much of this power is then dissipated in the pads, and may exceed their ratings.

Then there's the tape recorder. Remember the high frequency boost in the record mode? Well that uses up part of the headroom available. You now have the same problem as with speakers.

And one final word about high frequency boost. **Every** phonograph pickup has limitations to its trackability. And when it does mistrack it can generate very large high frequency components, which is one reason why it sounds so bad. Moreover groove damage also results, and even if the damaged groove is later tracked with a better pickup there are still quite a lot of extraneous high frequency "garbage" signals generated. If you boost these component signals along with the desired signal, the rest of your equipment doesn't know the difference and will react in the various ways already outlined.

CIRCUIT LOCATION

The same considerations apply as were outlined with regard to adaptive equalizers; locate at a high enough point in the system to avoid overload problems without boosting noise, and still be functional. Most commercial control units provide a recording output and a monitor return just before the volume control which effectively bypasses all controls including volume and tone, which are used only for monitoring. Installing a graphic or parametric equalizer at this point usually is the most satisfactory as it can then be used for recording and for listening. In general, the most useful

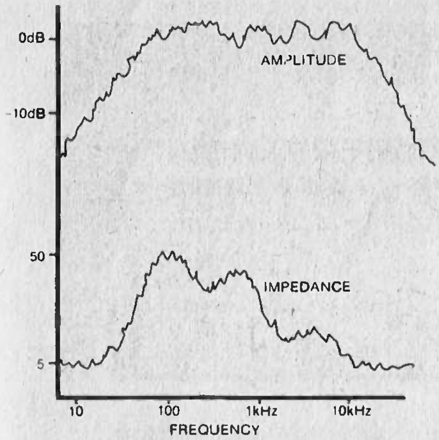


Fig. 8. Typical performance characteristics of low price small size (and high hype) loudspeaker example.

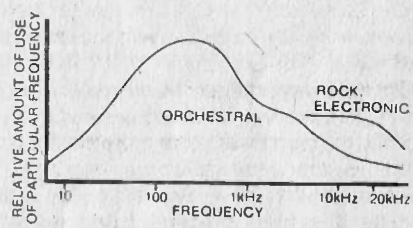
point is at the same level as is used for switching. However, if the device is used to correct for speaker/room acoustics the more logical location is immediately before the power amplifier. Unfortunately, this cannot be done with most receivers or integrated amps without opening and modifying the unit. And this is a good argument for the use of separates. But that's another story.

CONCLUSION

At this point the reader might well wonder what useful purpose is served by these equalizers. In many instances they do more harm than good, but this is largely the result of using them as a substitute for good design. If the performance level sought requires the use of a large Klipschhorn, get a Klipschorn, not a $99.95 super compact speaker special and a magic box. It won't do the job. DO pay attention to the acoustics of the room and the proper placement of a well designed speaker driven by a suitable amplifier. And DO use pickups and other equipment of appropriate quality. Then select the appropriate equalizer if you have use for one, and use it to deal with lesser acoustic problems or to make small alterations to programme quality.

In light of this the reader may be interested to know that my own system uses no equalizers of the discretionary type. A set of large transmission line speakers in a properly treated room provides high level performance through the full audible range with an imaging matched by only a very few professional systems. Phono preamps are matched to their own Stanton and Shure pickups each on its own turntable, and are not interchanged. The only need for additional equalization occurs occasionally when taping radio and TV broadcasts and 78 rpm discs. Under those conditions a fixed equalizer is designed and inserted in the line. So far the only real problem encountered is with a particular recording in which a phasor is used. In derived quadraphonics it sweeps the signal around the room and drives the cats crazy.

Fig. 9. Energy distribution graph for various types of music.



Hammer Throw

An exciting game of skill and luck that will help pass those long and lonely winter evenings.

IF, LIKE MOST of the ETI staff, you have more brains than brawn, and would not boast about the quality of either, it is likely that the mere thought of swinging a massive weight around your cranium is enough to strain your bodily systems. This probably means – and we are sorry if this comes as a disappointment – that your chances of selection for the Olympic hammer throwing team are, shall we say, nil.

Some may say that this is a pity as the sheer thrill of an event such as the hammer throw is probably very stimulating to those chunky brutes that are lucky enough to be able to take part. This is where we come to the rescue with our armchair version of the game. We think it has a number of distinct advantages over the real thing. One of these is that anyone, from an anemic sparrow upwards, can play the game. A second being that it is nowhere near as messy if, when playing in your living room, you get things wrong.

The game, as can be seen from our photographs, has a front panel with a circle of sixteen LEDs together with a line of eight LEDs at a tangent to the circle.

To play, after pressing reset, firmly press the play button. The LEDs in the circle will light one at a time simulating a spot of light moving in a circle. At the same time a distinctive, not to say loud, sound will be generated.

The spot will at first travel slowly round the circle, but will soon begin increasing in speed until it is travelling quite fast.

The object of the game is to release the play button at the instant that the 'top' LED of the circle is lit. If successful the line of LEDs will light to indicate your score, the faster the spot was moving when you scored the more will be your score. If you miss, the circle of LEDs will continue to rotate at the same speed as they were when you played.

BIG ONES AND LITTLE ONES

A game will consist of, say, eight rounds – the score from each being added to the last. At the end of a game the person who scored the most is the winner. The skill comes in deciding whether to go for a number of low scores that are relatively easy to get, or for a few big ones.

As befits the design of a project of this nature we were in convivial mood and pleasant surroundings when we first discussed the game. We produced the first design sketch (well a few lines on a napkin – yes in the tavern again) which used digital devices. Upon seeing this some likely person said that he thought most games featuring LEDs designed over the past few years should generically be called "spot the 4017".

Our initial reaction was to defend our design but a moment's thought showed that he had a point – the 4017 CMOS counter is over-used when it comes to games. At this stage we decided to rise to the occasion and produce the game using an all analogue approach.

The result can be seen in the circuit diagram. We are pleased with this circuit: It uses some unusual ICs and features a number of interesting circuit blocks – and of course there is not a 4017 in sight.

CONSTRUCTION

Construction of the game is greatly simplified if the PCBs are used. Three boards are required, one for the power supply, one for the display, and finally the main control board. Begin by building and testing the power supply. Take care to ensure that all components are mounted as shown in our overlay.

Next assemble the control and display boards. These carry a large number of components and mistakes made during assembly can be difficult to trace later – so take care at this stage. Do not insert the link between IC3/4 and IC9 at this stage.

It is best to test the boards before mounting them in the case, as it is difficult to get to some of the devices when the boards are in their final

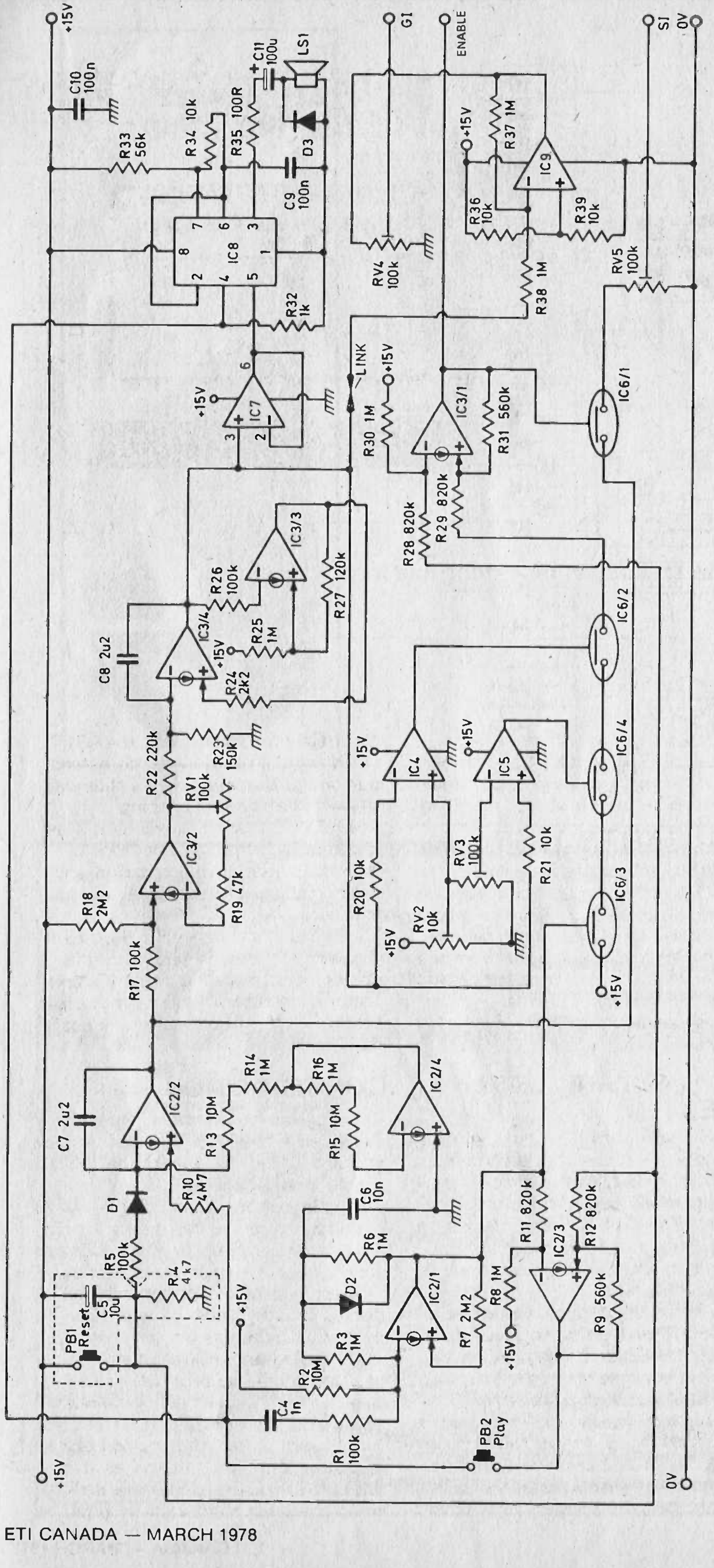


Fig. 1. Full circuit diagram of hammer throw control board.

positions. We used a sloping front Vero box to house our game and the general layout adopted can be seen from our photographs.

SETTING UP

There are five preset potentiometers on the board and all must be correctly set up before the game can be played.

The first adjustment to be made is to RV4. To calibrate this control first press the reset button and then the play button for a few seconds. At this stage a sound should be heard from the speaker and the game display LEDs should be seen flashing. Adjust RV4 until the LEDs produce a continuously rotating spot of light. The speed at which the circle of light rotates can be adjusted by RV1.

The next operation is to set up the score display. To accomplish this, press reset and then operate the play button until the spot of light is rotating at maximum speed. Release the play button and enable the score display by applying a positive pulse (from supply) to the junction of R29 and IC6. RV5 should now be adjusted so that the seventh score LED is just extinguished and the eighth lit.

The final adjustments concern the 'window' discriminator. To make this adjustment R38 (the end remote from IC9) should be connected to the slider of RV2. Adjustment of RV2 should illuminate successive LEDs of the game display. RV2 should be set to the point at which the top LED just extinguishes and the LED to the left just lights.

Now connect the input of IC9 to the slider of RV3. Adjust this pot so that the top LED just extinguishes and the LED to the right is just on.

This completes the adjustments and the link omitted during construction, should now be fitted.

Now is the time to get in training and, if you're good enough, you may yet make it to Moscow.

BUY LINES

The "line-o'-leds" ICs are made by Siemens. Their distributors are: Amphion in Halifax, and Moncton; Prelco in Montreal, Ottawa and Mississauga, Carsten in Toronto, Electrical Supplies Ltd and WES Ltd. in Winnipeg, Radio Supply and Service in Regina, Cardinal Industrial in Edmonton, Paar Industrial in Calgary, with RAE Industrial and Western Telecom in Vancouver.

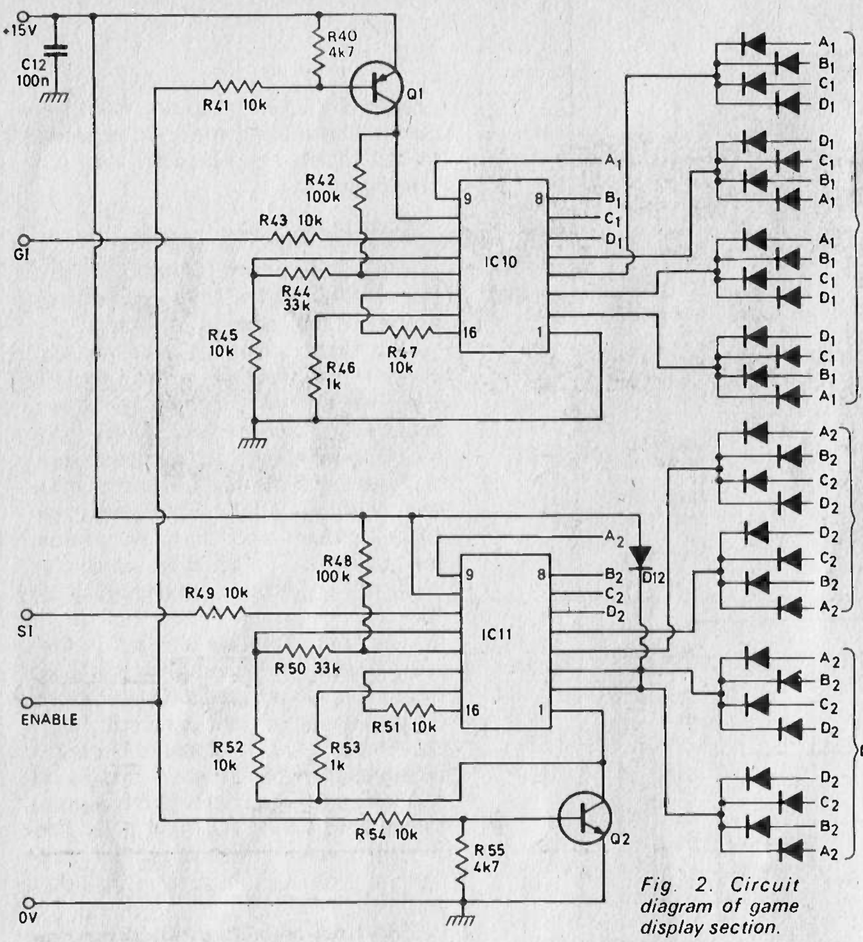


Fig. 2. Circuit diagram of game display section.

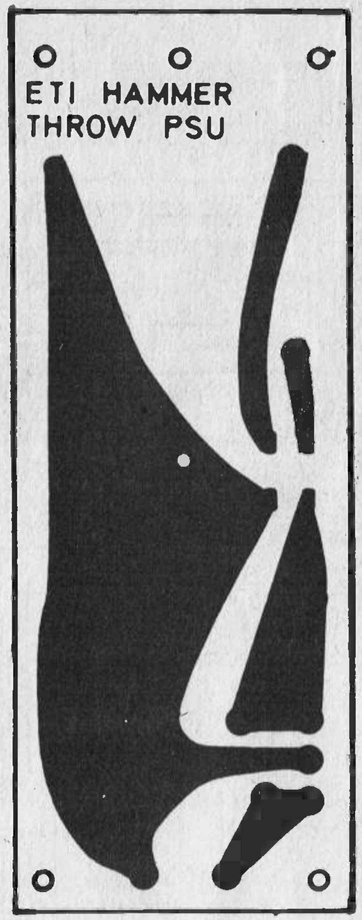


Fig. 3. Foil pattern of power supply board shown full size (120 x 45 mm).

PARTS LIST

RESISTORS all ¼W 5% unless stated

R1,5,17,26,42,48	100k
R2,13,15	10M
R3,6,8,14,16,25, 30,37,38	1M
R4,40,55	4k7
R7,18	2M2
R9,31	560k
R10	4M7
R11,12,28,29	820k
R20,21,34,36,39,41,43 45,47,49,51,52,54	10k
R22	220k
R23	150k
R24	2k2
R27	120k
R32,46,53	1k
R33	56k
R35	100R
R44,50	33k
R19	47k

POTENTIOMETERS

RV1,3,4,5	100k min hor trim
RV2	10k min hor trim

CAPACITORS

C1	1000u	25 V electrolytic
C2	220n	polyester
C3	470n	polyester
C4	1n	polystyrene
C5	10u	25 V electrolytic
C6	10n	polyester
C7,8	2u2	polyester
C9,10,12	100n	polyester
C11	100u	25 V electrolytic

SEMICONDUCTORS

IC1	78L15A
IC2,3	LM3900
IC4,5,7,9	741
IC6	CD 4016
IC8	555
IC10,11	UAA 170 (SEIMENS)
Q1	2N3905
Q2	MPS6515
D1,2,3,4–12	1N914
LEDs 1–24	.2" type
BR1	4 pin DIL TYPE: 0.9 A 400 V

TRANSFORMER

T1	120V — 15 V 6VA

LOUDSPEAKER

LS1	Telephone type insert or similar

SWITCHES

PB1,2	Push to make
SW1	SPST toggle

CASE Vero type 65–2523

MISCELLANEOUS
PCBs as patterns, LED mounting clips, fuse and holder to suit.

For clarification of component notation see "Reader Service Information".

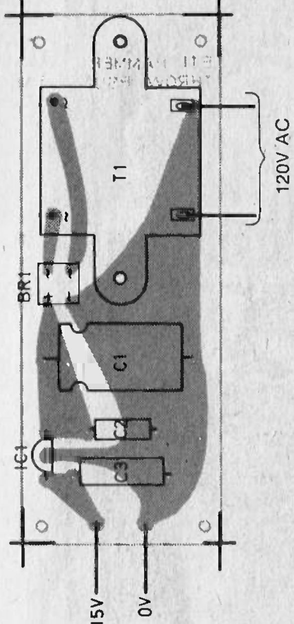
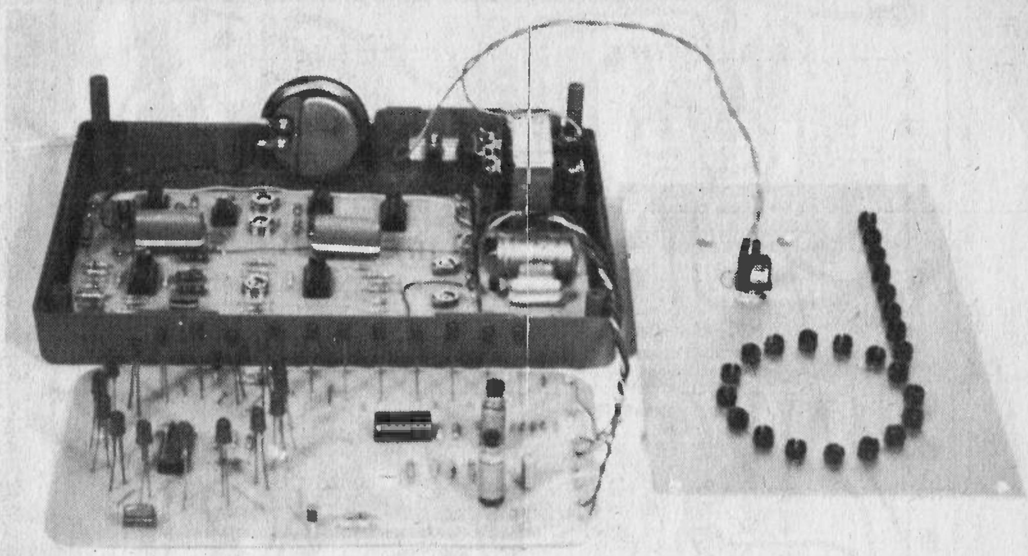


Fig. 4. Component overlay of PSU, line ground is connected to T1 by a solder tag under the mounting bolt.



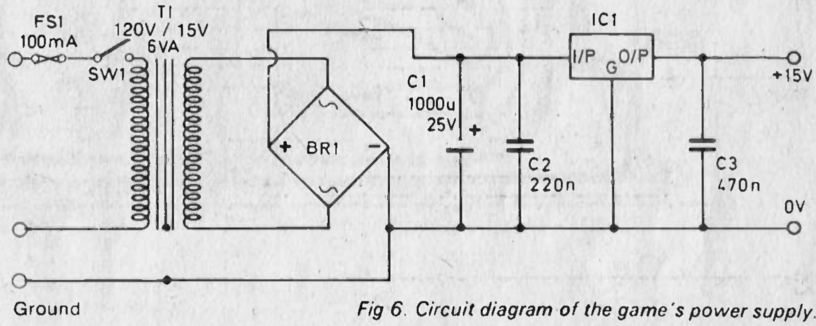
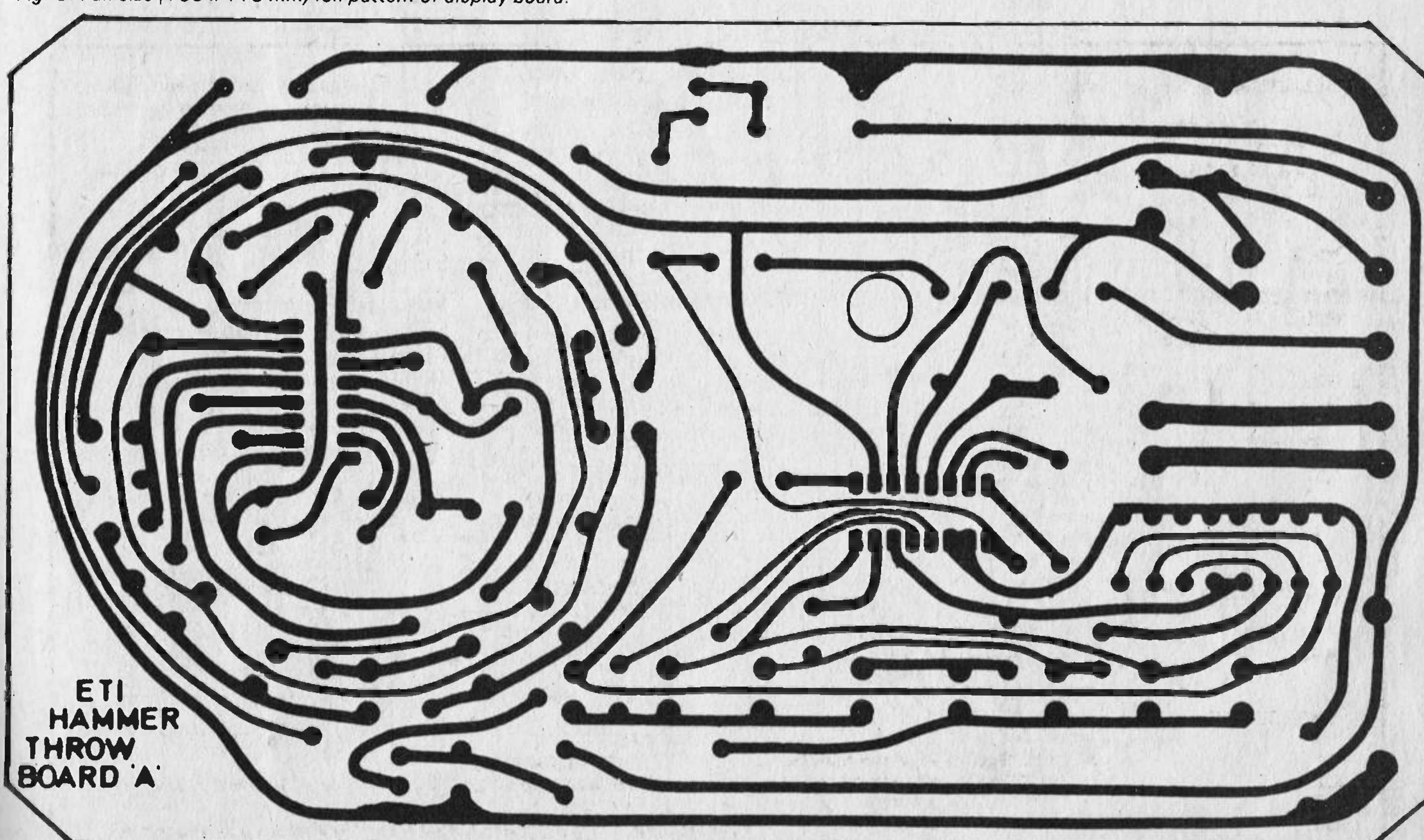


Fig 6. Circuit diagram of the game's power supply.

Fig. 5. Full size (160 x 110 mm) foil pattern of display board.



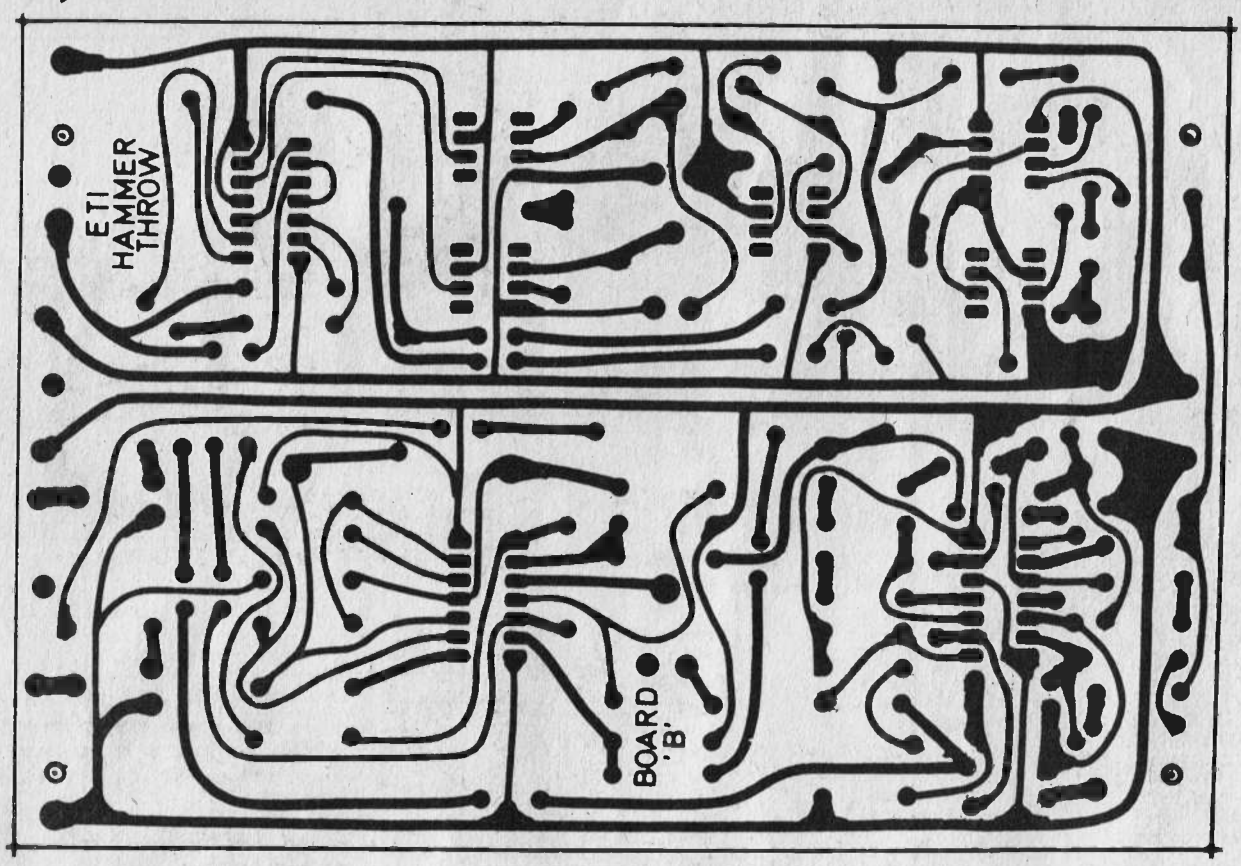


Fig. 7. Full size foil pattern of main control board (160 x 110 mm).

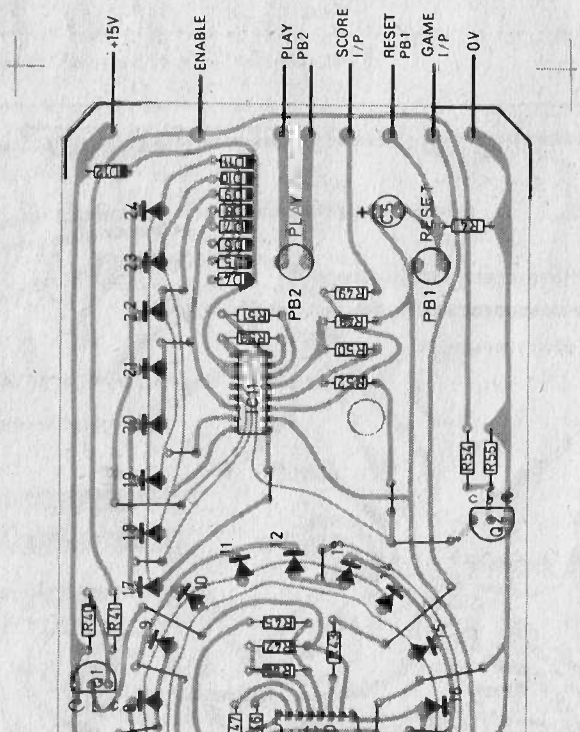


Fig. 8. The overlay for score board

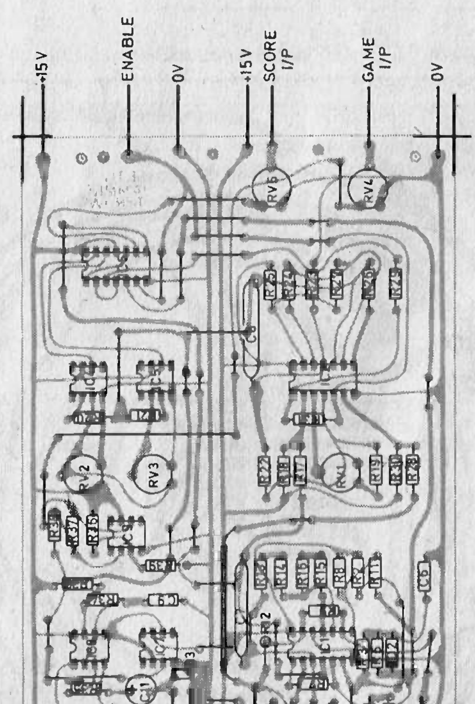


Fig. 9. Overlay for the control board. *Continued on page 20.*

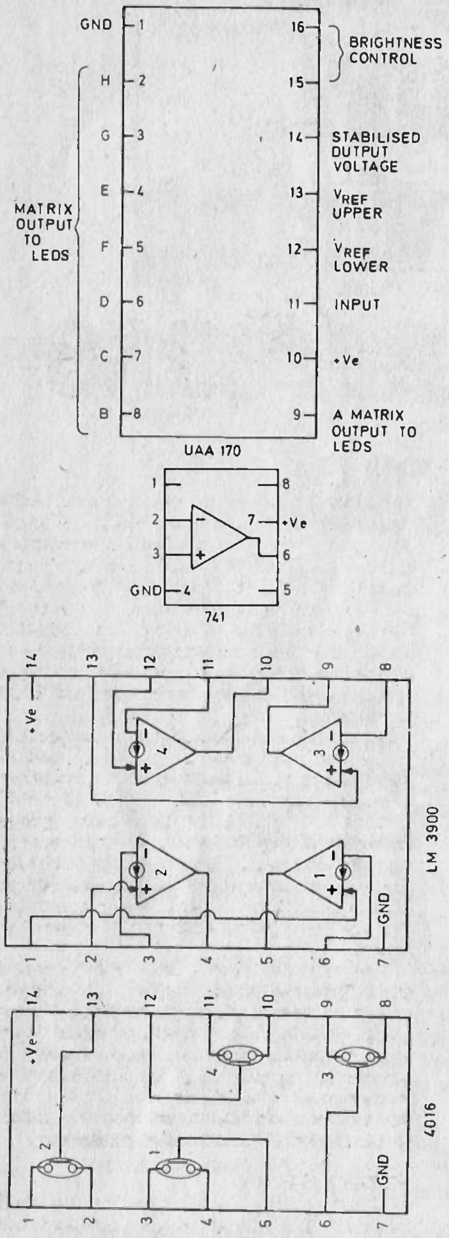
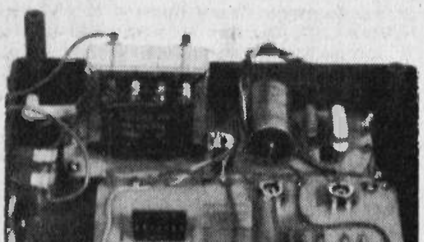


Fig. 10. Pinouts for the integrated circuits used in the hammer throw.



HOW IT WORKS

The circuit may be broken down into a number of major blocks — viz the display sections for both game and score, a voltage controlled oscillator, a ramp and hold circuit whose output controls the oscillator, a 'window' discriminator, a sound generating circuit and finally a power supply. As well as these major blocks there are also a number of latches, buffers and switches that are necessary for circuit operation.

The block diagram shown in Fig 11 shows most of the circuit blocks and, together with the circuit diagram, should be read in conjunction with this how it works.

SYSTEM OPERATION

The game display is based on a UAA 170 IC. This device is for driving LED displays and when connected to a line of sixteen LEDs will illuminate any one of these depending on the magnitude of the analogue voltage applied to its input. For the game display we need to produce the effect of a spot of light moving in a circle. To achieve this we arranged the sixteen LEDs in a circle and feed a sawtooth waveform into the UAA 170. A moment's thought will show that this will produce the desired effect.

In order to make the display rotate slowly at first, but speed up as play proceeds, we made the sawtooth generator voltage controlled. The control voltage is produced by a ramp and hold circuit which is reset to zero at the start of play, but begins to ramp up, thus increasing the sawtooth's frequency as play continues. When the play button is released, the voltage reached is held by the ramp and hold configuration until it is reset. This voltage is used for score purposes as described below.

The game requires that if, at the instant of releasing the play button, the 'Top' LED of the game display is lit, a score is indicated, the magnitude of the score being proportional to the speed at which the circle of LEDs was moving at the instant of release. From the description of the game display it will be seen that in order to light a specific LED the voltage input to the display driver must lie within a specific voltage range, thus in order to detect whether or not the 'top' LED is on we must look at the output of the sawtooth generator (this is input to UAA170) and decide whether it lies within the range that will light the specific LED at the instant the play button is released. The circuit that accomplishes this is the 'window' discriminator.

This is formed from two voltage comparators together with two analogue switches. Detailed action is described below, but briefly the circuit, when fed with the sawtooth output, will provide an indication whenever this waveform passes through an (adjustable) 'window' voltage range.

At the instant that the play button is released a short pulse is produced from a monostable. If this pulse is coincident with an indication from the window circuit that the top LED is on we must arrange to indicate a score.

The score must be proportional to the speed of the LED circle which is in turn proportional to the voltage level reached by the ramp and hold circuit. Thus, to produce a score, we feed the output from the ramp and hold, via an analogue switch, to a second UAA 170. This second display consists of eight LEDs in a line.

This completes a brief description of circuit action; we shall now deal with each block in more detail.

RESET CIRCUITRY

The game is initiated by operation of the reset button (PB1). This zeros the ramp and hold circuit described below, as well as setting latch 1 IC2/3 and resetting latch 2 IC3/1. Latch 1 enables the play button when its output is high (set) — latch 2 enables the score display when low (reset), the game display when high (set)

Each latch is based on two of the amplifiers of an LM 3900 Quad Norton amplifier package. This device is unusual in that instead of amplifying the difference in voltage applied to its input terminals, it amplifies the difference in input current.

The + and — inputs of these Norton amplifiers are both clamped to one diode-drop above ground and thus all input voltages must be converted to currents (by resistors) before being applied to the inputs. This is the basis for the current-mode (Norton) type of operation.

In operation the current flowing into the + input must equal that flowing into the — input, the difference between the current demanded and the current provided by an external source must flow in the feedback circuitry.

Operation of both latches is the same and we shall only describe the action of latch 1.

Assuming that the latch output is low (the latch is reset) the current injected into the — input of IC2/3 will ensure that the output remains low. If now sufficient current is injected into the + input the output voltage will rise as the device attempts to reduce the input current differential to zero. Positive feedback via R9 will enhance this action and cause the amplifier to latch high. This is because the current injected into the + input via R9 in this case is greater than that into the — input due to R8. A positive pulse via R11 to the — input will however once again bring the output low.

C5 and R4 ensure that when power is first applied the game is reset.

RAMP AND HOLD

The ramp and hold action is provided by IC2/2 and IC2/4. A positive voltage via R5 and D1 causes the output to ramp down while a similar voltage via R10 causes the output to ramp up. The reset button causes the downward ramp while play causes an upward ramp.

In any sample and hold application a very low input bias current is required if the hold period is to be stable. The existence of matched amplifiers within the LM3900 allows one amplifier to bias another.

In operation the LM 3900 requires a bias current to be applied to its — terminal. IC2/4 has its + terminal grounded and feedback applied via **R15** and R16. The output voltage of this device will attain a level such that the current fed back via these resistors is equal to the bias current demanded by the input. This same current will flow via R13 and R14 into the — input of IC2/2 reducing the effective bias current of this amplifier to almost zero. D1 isolates this bias current from the rest of the input circuitry.

If now a positive current is injected into the — terminal, the output voltage will fall as it attempts to feedback a current of this value in order to reduce the input current differential. This constant current across C7 results in a linear voltage ramp appearing across C7. Input to the + terminal causes a positive going ramp, to the — terminal a negative going ramp.

The rate at which the voltage across C7 changes is proportional to the value of the constant current supplied which is in turn proportional to R5 and R10. As R5 is some 40 times larger than R10, the ramp down (reset) is far quicker than the ramp up.

The output from the ramp and hold circuit is fed, via IC6/1 to the score display and via IC3/2, a non-inverting scaler, to the sawtooth VCO.

NON-INVERTING SCALER

The scaler is required because the output from the ramp and hold configuration can vary over nearly the whole supply voltage whereas the VCO requires only a small voltage swing to provide the required frequency change.

The scaler is based on another Norton amplifier arranged as a non-inverting amplifier. Feedback is applied via RV1 and R19 and output is fed to a potential divider formed by R22 and R23 and thence to the VCO.

VOLTAGE CONTROLLED SAWTOOTH OSCILLATOR

The VCO is formed by IC3/3 and IC3/4. Action of IC3/4 is much the same as that of IC2/2 described above. The special input bias circuitry is not required as there is no hold requirement.

IC3/3 acts as a comparator and circuit action is as follows: while the output of IC3/4 is high and ramping down (input to — terminal) the current into the — input of IC3/3 due to R26 is greater than that to its + terminal due to R25 – its output is thus low.

As the output of IC3/4 ramps low however, there comes a point where this situation is reversed. The output of IC3/3 goes high (this state being maintained by positive feedback via R7 which injects a large current into the +input of IC3/3 as R7 is much smaller than R25.)

The output if IC3/4 thus goes high, restoring current flow via R26 and starting the cycle again.

By varying the current injected via R22 the time taken for the output of IC3/4 to ramp down to the point at which the comparator triggers is lessened. This results in an increase in the frequency of the sawtooth.

The output from the VCO is fed to the game display section RV4, to the 'window' discriminator formed by ICs 4 and 5 and via IC7 to the sound generator *IC8*.

WINDOW DISCRIMINATOR

The window discriminator is formed by two comparators IC4 and IC5 and two of the analogue switches in IC6.

Operation is as follows: If we assume that the output of the sawtooth VCO is high and ramping down the voltage on the – input of IC4 will be higher than that on the + input (a reference level established by RV2) and its output will be low. The output of IC3 will be high as the input to its + terminal is higher than that to its – input.

As the voltage ramps down, a point will be reached where the output of IC4 goes high as the voltage at its – input falls below that set by RV2 at its + terminal. At this stage the outputs of both IC4 and IC5 are high, as IC5 has not switched. As the voltage continues to ramp down, however, the voltage on IC5's + input falls to a point below that on its – input and the output of this IC goes low.

Thus the outputs of both ICs will be high for a small range of input voltages (the window) defined by the difference in voltage between the sliders of RV2 and RV3.

The outputs of these ICs are fed to the inputs of two analogue switches. A positive voltage applied to these switches turns them "on".

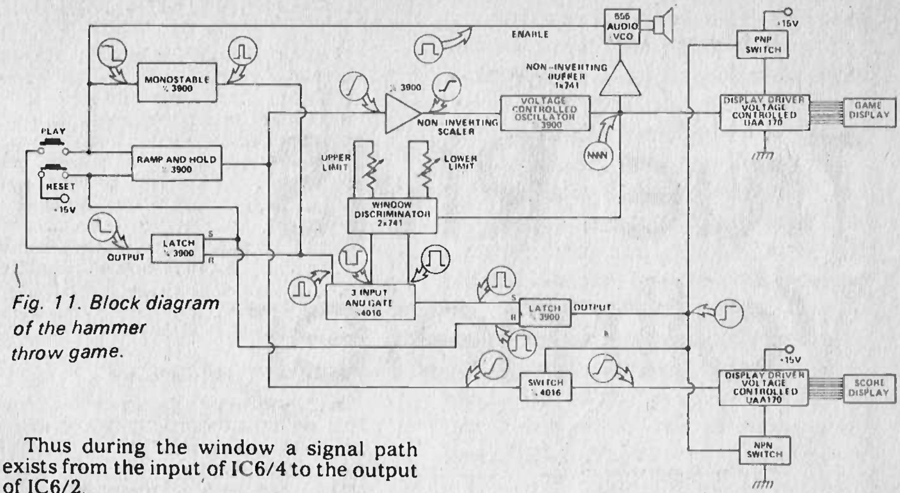


Fig. 11. Block diagram of the hammer throw game.

Thus during the window a signal path exists from the input of IC6/4 to the output of IC6/2.

MONOSTABLE

The monostable is formed by IC2/1 this produces a short positive going pulse upon receipt of a negative spike produced by the release of the play button.

Current injected into the — terminal via R3 will normally hold the output low, however a negative pulse applied via C4 and R1 will "rob" this current from the input and causes the output to go high.

R7 latches the gate in this state after the negative pulse is removed. At this stage C6 begins charging, feeding back an increasing amount of current to the — input as the voltage at the junction of R6 and R3 rises.

There comes a point when this current is greater than that fed back via R7 and the output returns low. Diode D2 rapidly discharges C6 to provide reliable re-triggering.

The leading edge of the output pulse is coincident with the release of the play button. This pulse is used to turn on analogue switch IC6/.3 It will be remembered that if the voltage of the VCO is within the 'window' at this point — switches IC6/4 and IC6/2 will also be on. This allows the supply voltage input to IC6/3 to set latch 2 and thus initiate the required actions, ie, blank game display, enable score display, etc.

The monostable also resets latch I IC2/3 to remove supply from the play button, this prevents cheating.

GAME DISPLAY

The output of the sawtooth VCO is fed via an inverting buffer, IC9, and a potential divider, RV4, to the input of IC10 a UAA170. The input circuitry of this device consists of a series of differential amplifiers with one input of each connected to the input terminal (pin 11) via an emitter follower. The other input of each is connected to a point in a potential divider chain consisting of equal value resistors. The differential amplifiers thus operate as analogue voltage comparators and as the input exceeds the reference voltage of a particular comparator, the output of that comparator will change state.

To reduce the package pin-out the LEDs of the display are not driven individually but are arranged in a four by four matrix pattern controlled by the row and column outputs of the UAA170 (A-D and E-F respectively). By enabling the appropriate row and column output any one of sixteen LEDs may be selected. The matrix outputs are controlled by the internal logic of the UAA170.

The resistor chain R42, R44 and R45 sets up the reference voltage inputs of the device. The voltage on pin 12 establishes the lowest voltage to which the UAA170 will respond. If the input voltage is below this point the first LED of the display remains lit. As the voltage rises above this level the first LED is turned off, the second on — as the input rises the spot moves up the chain, until the voltage reaches that set on pin 13. This is the maximum voltage to which the display responds and if the input is taken above this level the last LED remains lit.

In addition to defining the indication range the voltage between pins 12 and 13 determines the abruptness of transition between any two LEDs. With this difference set to 1V4 the light point glides smoothly along the scale, with increasing voltage difference the passage becomes more abrupt until at 4V the light spot jumps from one LED to the next. We have set this voltage to a point between the two extremes.

The resistors R46, and R47 control the brightness of the display. Q1 supplies power to the display and is driven from latch 1 IC2/3. This, you will recall, is reset, ie, its output is low, at the start of a game. A low voltage applied to Q1 via R41 turns this transistor on and enables the display. The latch is returned high at the end of a game, this turns Q1 off and blanks the display.

SCORE DISPLAY

The score display is formed by a second UAA170 (IC10). Much of the circuitry is the same as that of the game display except that we only wish to display eight LEDs. The diodes from unused outputs to the +VE supply act as 'dummy' LEDs, restricting the display to eight LEDs, you could use LEDs for extended scoring — but a larger box is needed. This display is powered by Q2 which is again fed from the output of latch 1 (IC2/3. This time, however, the display is blanked, Q2 off, when the latch is low and enabled, Q2 on, when the latch output is high.

SOUND GENERATOR

The sound is generated by IC8 an NE555 operated in its astable mode.

The reset pin(4) is normally held low by R32 and hence circuit action is inhibited. A positive voltage applied from latch 1 via the play button enables the sound during the game.

The output is frequency modulated by applying the output of the sawtooth VCO, via buffer IC7 to provide the necessary low impedance drive, to the voltage control input (pin 5) of IC8.

True RMS Voltmeter

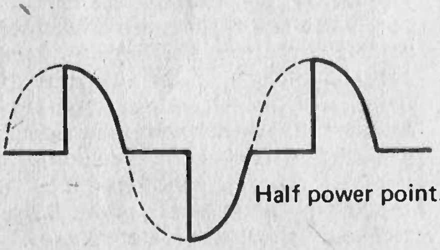
The use of a special IC results in performance greatly improved over conventional designs.

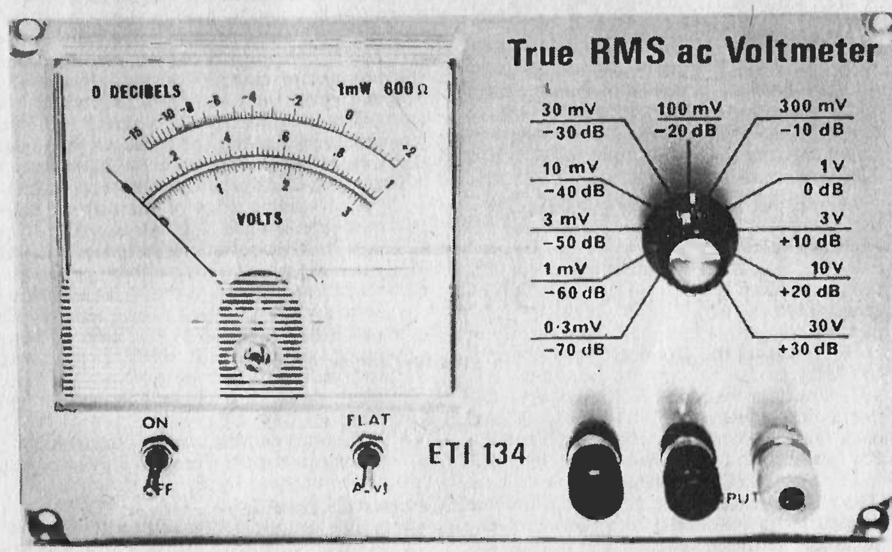
MOST METERS which can measure ac signals do so by rectifying the signal and then measuring the average voltage. With a sinewave the average voltage is 0.637 of the peak voltage while the rms value is 0.707 of the peak. Therefore a correction factor of 1.11 is built into the meter to give the rms value of the signal.

Provided you stick with sinewave signals these meters are adequate. With any other waveform, however, they are not accurate. With a square wave the error is 11% and with pulse wave forms the error increases.

Before continuing we should explain what rms means and its significance. Without getting mathematical, the rms value of any wave form is the same as a dc value which would produce the same heating effect in a resistor. For example:

Power in a load can be varied by using phase control (i.e., light dimmer) where the time the load is connected to the line is variable. The rms value is difficult to calculate except at the point where it is half on—half off. The power then is obviously half power.





If the input voltage is 120 V and the load is 120 ohms the power (maximum) is given by

$$P = \frac{E^2}{R} \text{ or } \frac{120 \times 120}{120} = 120W$$

Half power therefore is 60W. The voltage corresponding to this is given by

$$E = \sqrt{P \times R} \text{ or } 85 \text{ V (rms)}$$

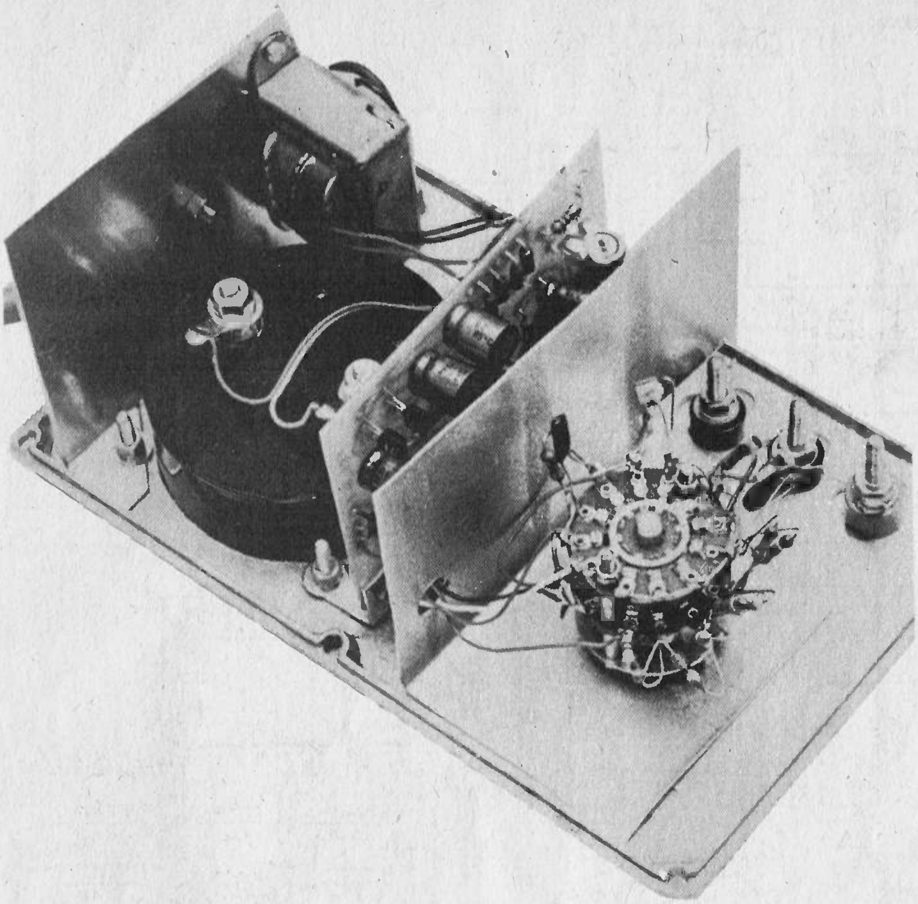
On a "normal" meter this will read 60 V or an error of 30%.

This design uses an rms detector IC, which is basically a small, special-purpose analogue computer to mathematically calculate the true rms value for any waveform.

DESIGN FEATURES

The design of the voltmeter is basically simple, starting with an attenuator in the front end, then an amplifier with a high input impedance and switchable gain which, with the attenuator, gives the range selection. A filter is then added to give the "A" weighting and the rms detector IC (LH0091) does the rest.

The output of the input amplifier is 60 mV, independent of range selected, for an input corresponding to the full scale reading. This gives a maximum gain of 46 dB on the 0.3 mV range. There is a loss of about 2.3 dB in the filter (at 1 kHz) and the spare amplifier in IC2 is used to provide a gain of 20 dB giving 500 mV (for full scale reading) before the rms detection is done. The



SPECIFICATIONS

Meter Type	rms reading ac only
Ranges	0.3, 1, 3, 10, 30, 100, 300 mV 1, 3, 10, 30 V
Accuracy	+3% nominal (crest factors up to 3) −8% at crest factory of 10
Input Impedance	1 megohm in parallel with 25 pF
Weighting Networks	Flat or 'A' weight
Frequency Response	10 Hz – 20 kHz

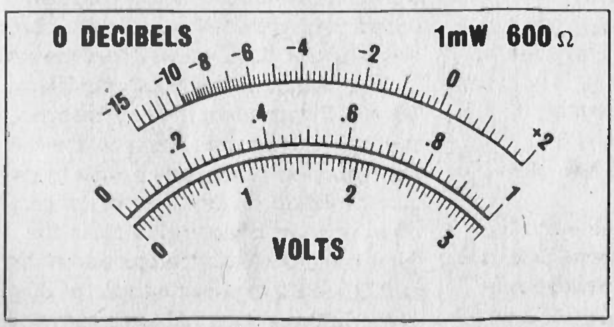


Fig. 1. Meter scale shown full size.

rms detector section has unity gain with 500 mV rms in giving 500 mV dc out.

However things are never that simple. With a total of 60-odd dB gain, along with the requirement for a 1 M input impedence, we have an excellent formula for an oscillator. With the third try (yes, we have failures too) with adequate shielding and layout, stability was obtained and this final design is presented here.

The spare IC in the LH0091 is normally used to buffer, filter or amplify the output of the rms converter (see data sheets in this issue) but we used it before so as to buffer the filter network and save an additional op amp (the input of the rms converter is only 5 k ohms). The output voltage from the converter is only 500 mV but this is adequate to drive a meter. We could have provided more gain in the buffer stage so giving a higher output but this would lead to greater errors with high crest factor waveforms.

We have limited this instrument to ac signals as this eliminates the need for balance controls to correct for drift when measuring low level signals. This normally is of no consequence as most signals, i.e., output of a tape recorder, sound level meter, etc., have no dc component. If dc capability is needed, capacitors C1, 8, 9, 14, 15 and 16 have to be shorted out, a zero adjustment potentiometer added to IC1 along with the potentiometers needed to offset adjust IC2 (see data sheet).

CONSTRUCTION

If the printed circuit board is used along with the layout and shields as described there should be no problems with construction. The wires associated with the rotary switch should be no longer than necessary to minimise any pickup. The box should be grounded to the line ground and the front panel ground terminal (left hand one) should also be connected to ground.

USE

When measuring low level signals there may be 60 Hz pickup unless the common side of the input signal is connected to ground. This may be done either in the unit under test or on the meter (hence the ground terminal). Also with the meter terminals open circuited the meter will give some reading. However, as the output impedance of low level signals (0.3 mV and less) is normally relatively low this is usually no problem.

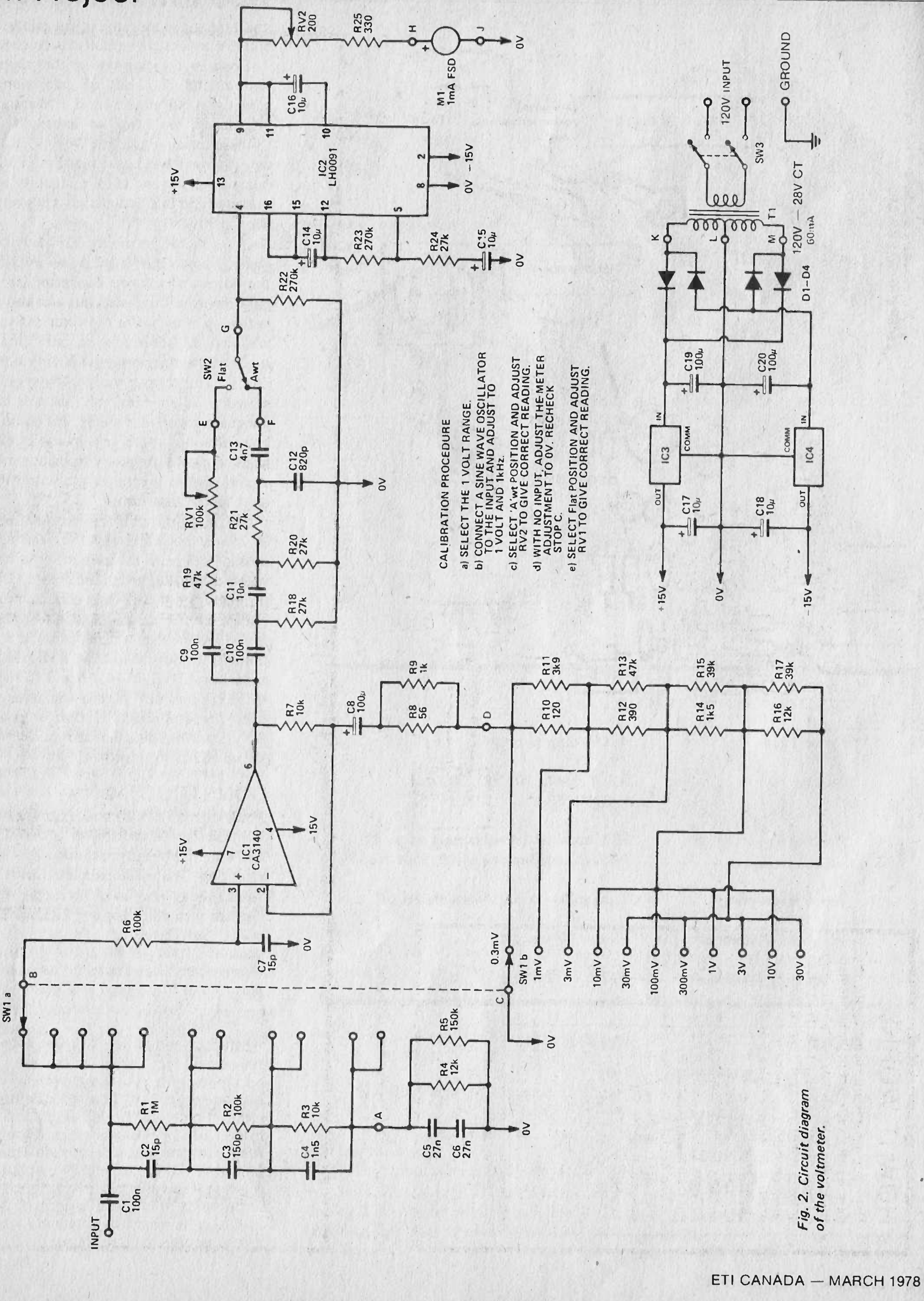


Fig. 2. Circuit diagram of the voltmeter.

HOW IT WORKS

The input signal is attenuated by the network R1–R5 and C2–C6; the appropriate attenuation is selected by SW1a. This gives 0 dB, 20 dB, 40 dB and 60 dB. The output of SW1a is buffered by IC1 which is a FET input op-amp. This amplifier has a gain which is switchable giving 5.56 dB, 15.56 dB, 25.56 dB, 35.56 dB and 45.56 dB. By selecting a combination of these two variables the eleven ranges from 0.3 mV to 30 V are obtained. The output of IC1 for full scale reading is 60 mV.

The output of IC1 goes to the 'A' wt filter network and also directly (via R19) and RV1) to SW2. This selects either 'A' weighting or flat response. As the filter has 2.3 dB loss at 1 kHz the "flat" position is also attenuated (hence R19, RV1) to maintain calibration.

The rms detector IC provides a gain of 20 dB before the detector; the output of the detector is about 500 mV for full scale reading.

The power supply is simply a full wave rectified supply giving both plus and minus voltages of about 20 V, which are then regulated to ±15 V by IC3 and IC4.

A Kit of parts for this project is available from Dominion Radio, see page 60.

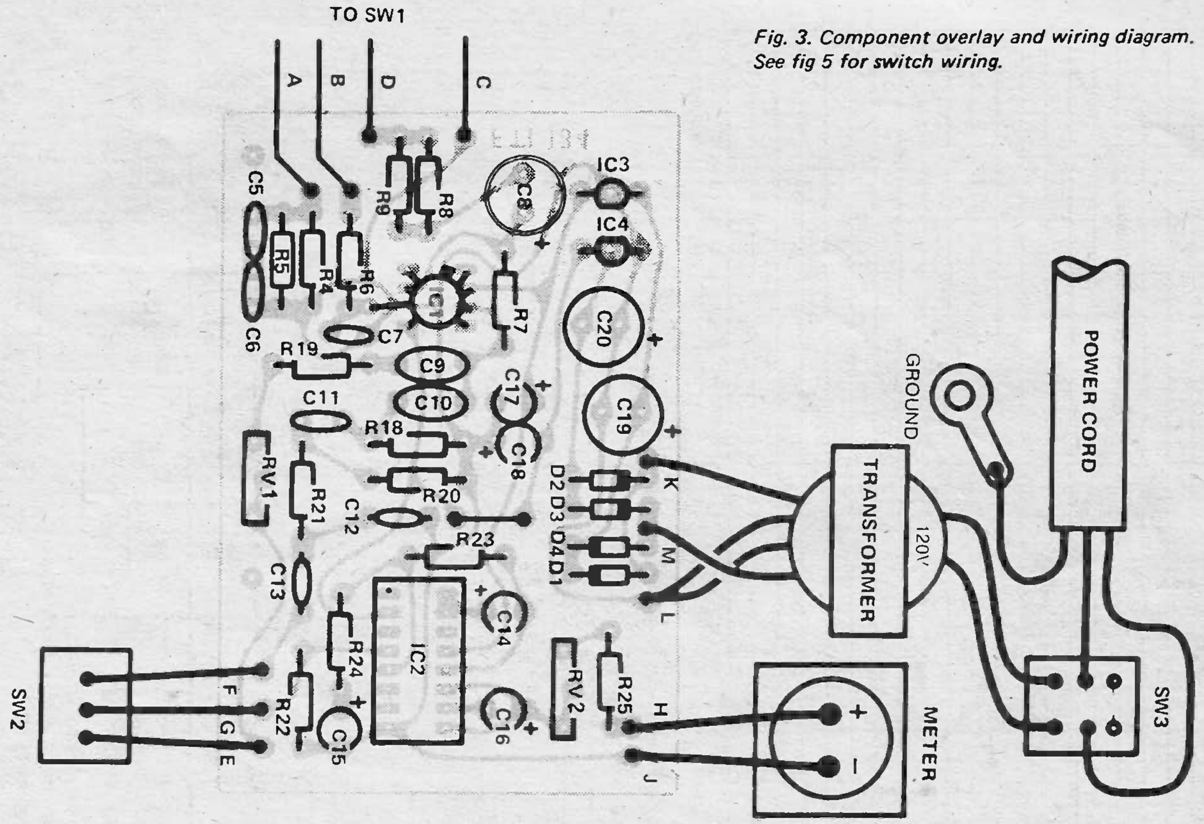


Fig. 3. Component overlay and wiring diagram. See fig 5 for switch wiring.

PARTS LIST

RESISTORS

All ½ W 5%, except where marked.

R1	1M	1%
R2	100k	1%
R3	10k	1%
R4	12k	1%
R5	150k	
R6	100k	
R7	10k	1%
R8	56	1%
R9	1k	
R10	120	1%
R11	3k9	
R12	390	1%
R13	47k	
R14	1k5	1%
R15	39k	
R16	12k	1%
R17	39k	2%
R18	27k	
R19	47k	
R20,21	27k	
R22,23	270k	
R24	27k	
R25	330	

POTENTIOMETERS

RV1	100k	trim
RV2	200 ohm	trim

CAPACITORS

C1	100n	polyester
C2*	15p	ceramic
C3*	150p	"
C4*	1n5	polyester
C5, 6*	27n	"
C7	15p	ceramic
C8	100μ	25V electro
C9, 10	100n	polyester
C11	10n	"
C12	820n	ceramic
C13	4n7	polyester
C14-C18	10μ	25V electro
C19, 20	100μ	25V electro

* These capacitors should be as accurate as possible as they affect accuracy above 10kHz.

SEMICONDUCTORS

IC1	CA3140	op amp
IC2	LH0091	RMS converter
IC3	78L15	regulator
IC4	79L15	regulator
D1-D4	1N4001 or similar	

MISCELLANEOUS

PC board	ETI 134
SW1	2 pole 11 position OAK switch
SW2	SPDT minature toggle switch
SW3	DPDT minature toggle switch
T1	Transformer 120V/28Vct (28V ct)
M1	Meter 1mA scaled as shown

3 terminals (red, black green)
Box
Metal brackets and shields (see Fig 7)
3 core flex and plug
Aluminum front panel
16 pin socket for IC2
Knob

For clarification of component notation see "Reader Service Information".

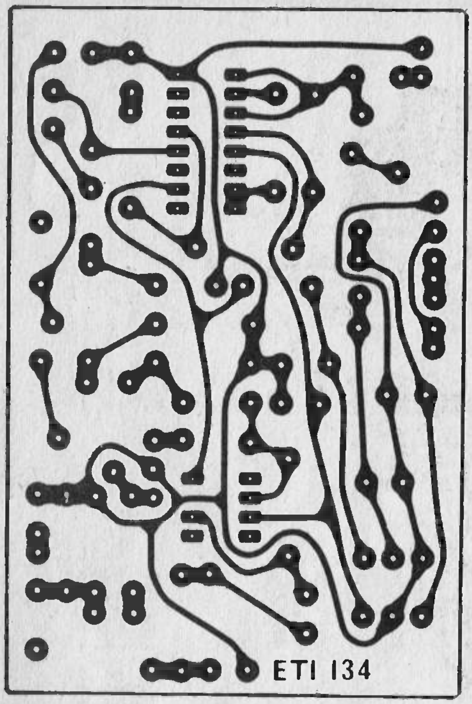


Fig. 4. Printed circuit layout. Full size 90 x 60 mm.

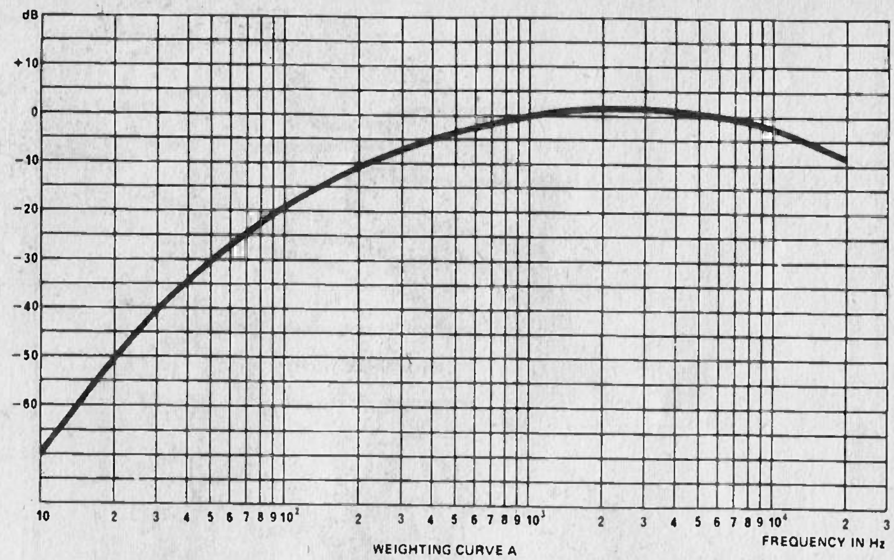
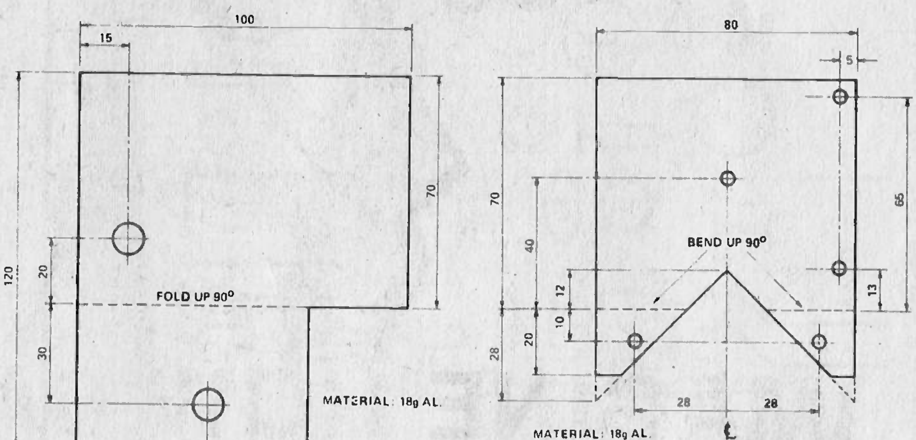


Fig. 6. The response in the " A " weight position.



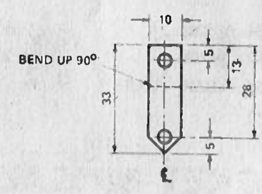


Fig. 7. The details of the shield, transformer bracket and PC board support bracket (two required).

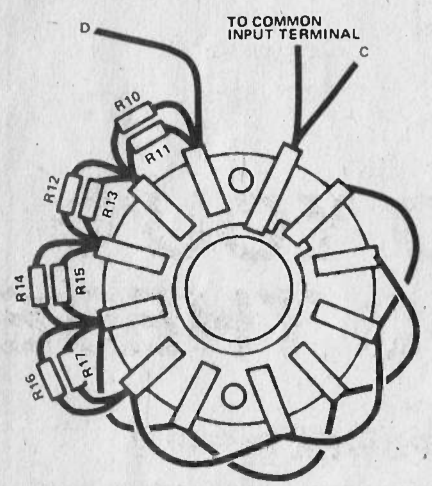
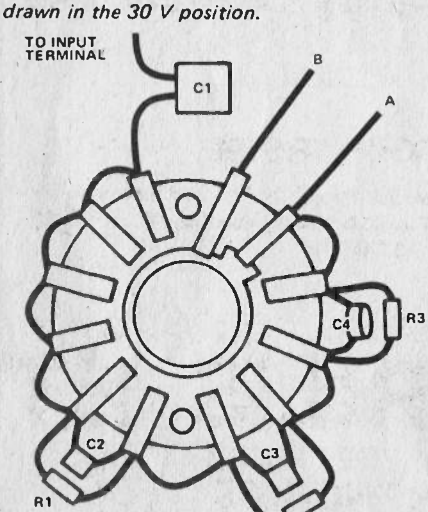
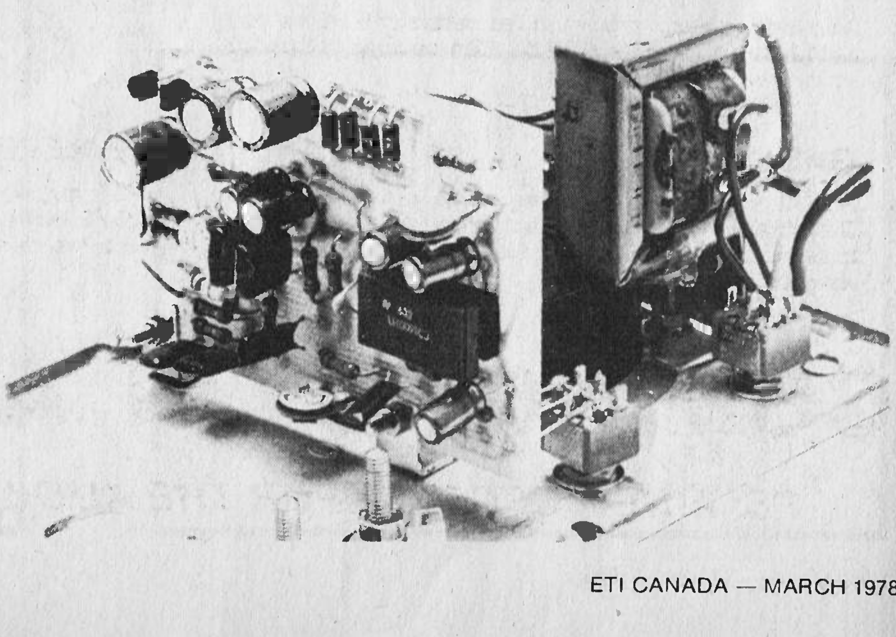


Fig. 5. Connection of the range switch drawn in the 30 V position.





House Alarm

In these days of increasing crime and vandalism an alarm system for the home can add greatly to ones peace of mind. To be effective however, not only must the alarm circuitry be well designed, it must also be correctly installed. This article describes a sophisticated alarm system and how best to commission it.

WE HAVE EXAMINED the most common forms of break-in and came up with an overview of the burglary problem, and how to most effectively counter the would-be thief. First we'll look at general techniques, and then we present a quite sophisticated alarm unit with a variety of useful features.

We cannot emphasize enough though, that any alarm system — no matter how sophisticated — can only be of use if it is installed correctly. Further the installation of an alarm should only be considered as part of a general awareness of the need for greater attention to be paid to security. For this reason, before going on to describe the alarm in detail, we shall deal with domestic security in general, the installation of alarms and how the specification of our alarm evolved.

HOW THEY GET IN

Nearly 30% of all burglaries are committed by thieves entering via unlocked doors or windows. A further 24.4% are committed via forced door locks, and about the same percentage via forced windows.

Thus nearly four out of five potential breakins can be avoided by installing adequate door and window locking mechanisms.

Use 'deadbolt' locks on all external doors. These can only be opened with a key — even from the inside — so that even if a thief enters via a window he cannot remove any large items as the doors remain locked and few thieves will risk passing out items through a window.

Do have locks fitted by an experienced locksmith unless you have experience in this field — and do not fall for the door-to-door lock salesman — it is not unknown for such people to retain a duplicate key.

Consult a security expert about window locking devices. Innumerable types are available for metal, wood framed and sash windows. A burglar might break glass but few will risk climbing through a window frame with broken glass in it.

The precautions outlined above will reduce your chances of being burgled by about 80% — the remaining 20% can be reduced to almost zero by installing a good burglar alarm. The emphasis must be on the word good, a poor alarm may go off erratically, or worse, not at all.

SENSORS

For most premises, it is necessary to install sensors to protect front and rear doors, windows and garage entrances.

A few forced entries are made through the walls or roof or very occasionally via the floor. Although rare, such forced entries may be guarded against by placing sensors in a strategic passage or area through which any intruder is likely to pass.

The simplest and most reliable switching device for alarm installations is the magnetic reed switch. This consists of a pair of ferromagnetic contacts in a small hermetically sealed glass enclosure. The switch contacts are cantilevered from the ends of the glass tube and overlap slightly at the centre, with a small air gap between them.

When a magnet is brought near the reed switch, the attracting forces increase and overcome the stiffness of the reeds, bringing them into contact. When the

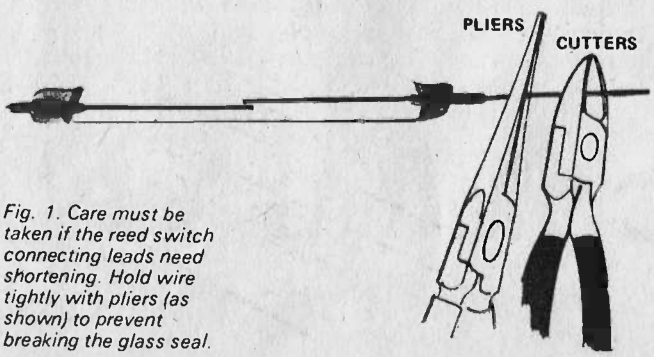


Fig. 1. Care must be taken if the reed switch connecting leads need shortening. Hold wire tightly with pliers (as shown) to prevent breaking the glass seal.

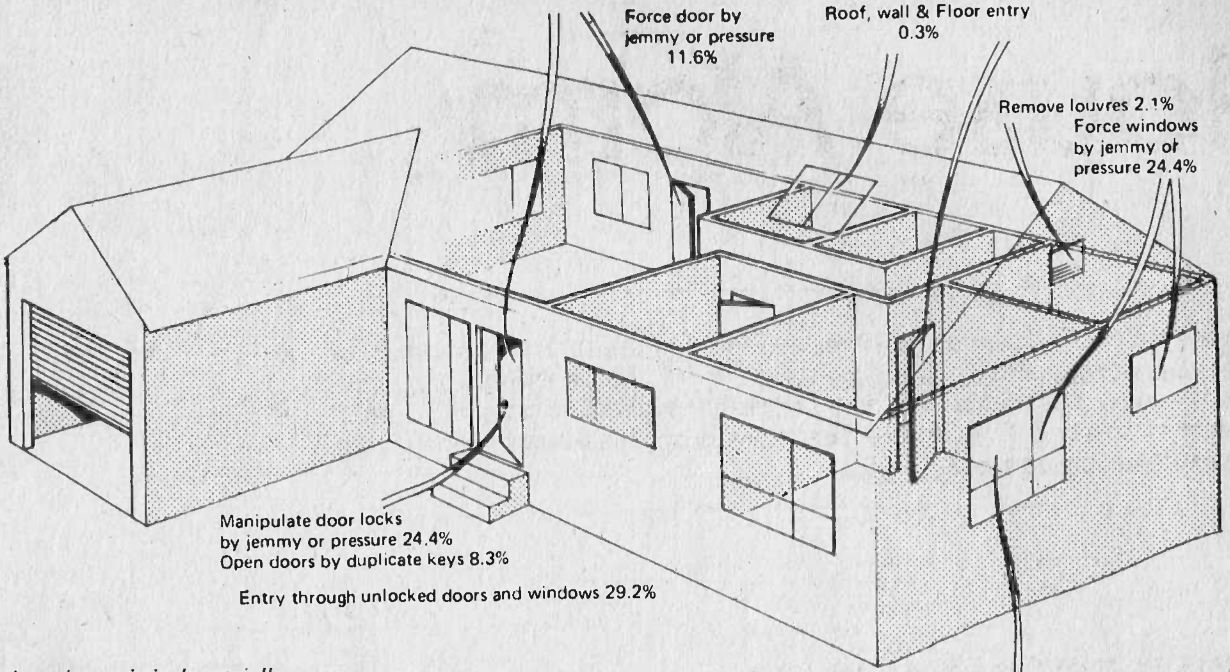


Fig. 2.
This sketch shows how criminals generally enter a house. Note the very high percentage of entries made via doors and windows found unlocked.

magnet is removed, the contacts open. The relative distance for pull-in is less than for drop-out, a valuable feature as small movements of doors and windows will not cause false triggering.

Reed switches purchased for alarm installations must be of a type specifically intended for the purpose — standard reed switches will not do.

Many professional security companies install reed switches and magnets encased in plastic mouldings. Whilst these are neat and simple to fit, it is better to conceal both reed and magnet within the framework of the door or window to be protected.

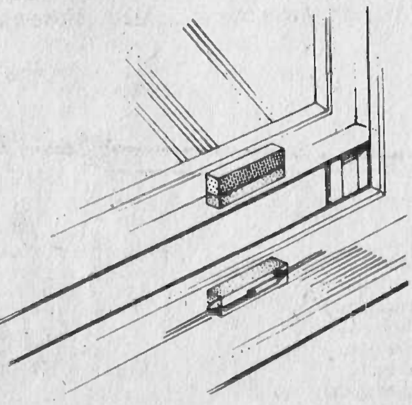
In Figs. 3 and 4 we show just two of the various methods of fixing the reeds and magnets (note that the magnet is to be fitted to the moving part of any door or window).

Window glass may be protected by glueing on a loop of aluminium foil tape (or using a self-adhesive type of foil). The foil is quite thin and breaks if the glass is fractured. Foil will deter all but the most determined of burglar. After all, why risk being caught when next door does not seem to be protected by an alarm?

Vibration sensors may be used to protect large areas of glass but these are prone to false triggering during thunderstorms etc.

Many other types of intruder sensing devices may also be included in the system. Pressure mats for example can be placed under carpets in strategic passageways — or even under the door mat. The mats contain a large number of normally open contacts some of which will be closed when the mat is trodden on. The system can also include more sophisticated intruder detectors

Fig. 3. Set the reed switch into the window frame and the magnet into the moving part.



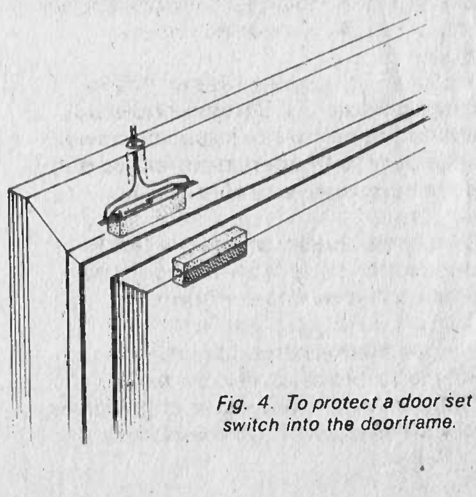


Fig. 4. To protect a door set the reed switch into the doorframe.

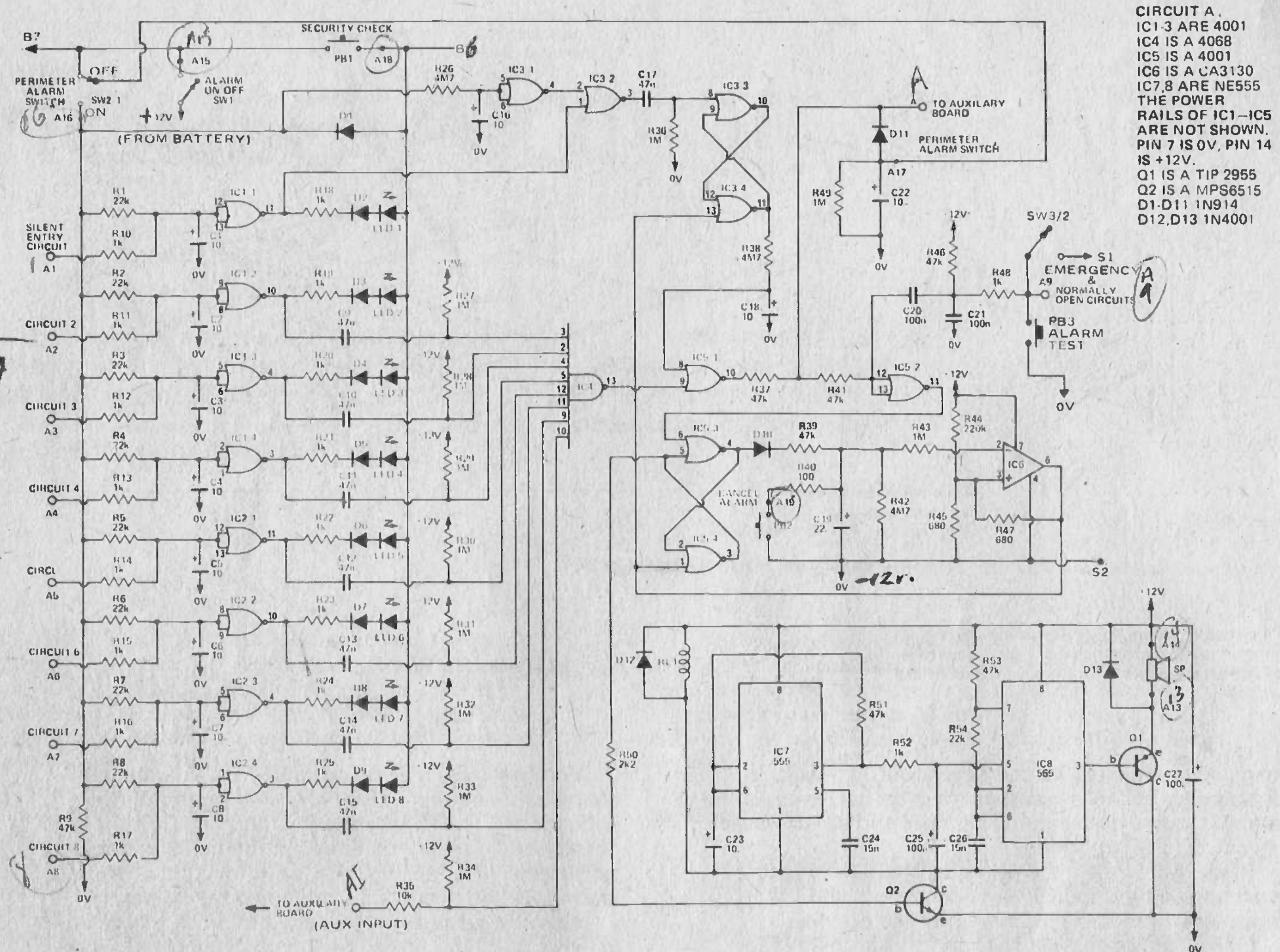


Fig. 5. Circuit diagram of the 'A' board.

such as infra-red type sensors.

The intruder alarm itself should be reasonably accessible to people entering and leaving the premises via a 'silent entry' door, but will be hidden from the sight of an intruder. The alarm's output stage should be a relay which latches when an alarm signal is received.

WARNING DEVICES

For household use a good quality 12 Volt bell should prove an adequate warning device. Being mechanically resonant, bells have a very high conversion efficiency; in fact, the average bell draws less than 500 mA at 12 V yet can be heard several hundred metres away.

Good sirens can be heard well over a few kilometres away, but they draw a lot of current and cost more than a good bell. Small cheap sirens cannot be recommended.

If at all possible, householders should make mutual arrangements with neighbours to contact the police if the alarm is heard. Similar arrangements should also be made so that neighbours can switch off the alarm when the police arrive.

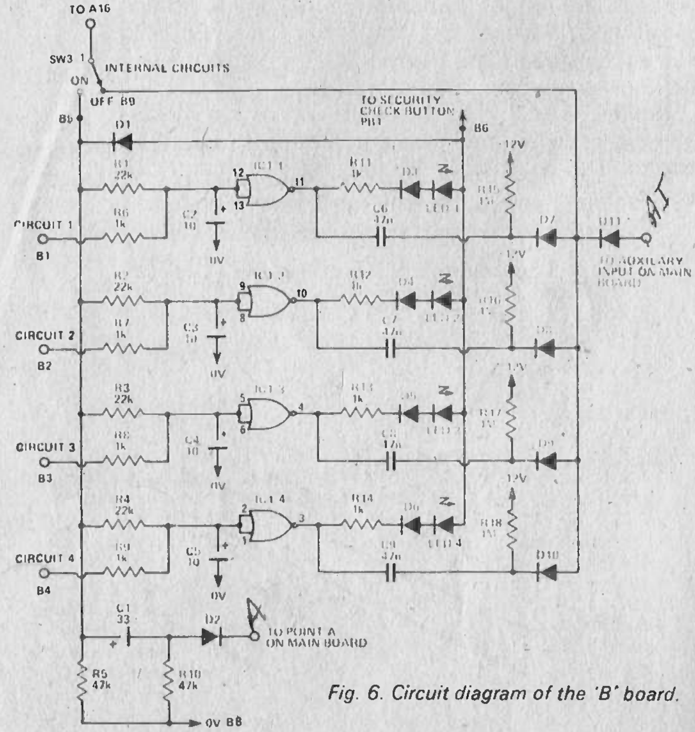


Fig. 6. Circuit diagram of the 'B' board.

An alarm which resets after a period of time, silencing the bell or siren, is a useful device that will be much appreciated by the neighbours. Care must be taken to ensure however, that the alarm when triggered and reset, still provides some measure of protection to the property.

Whatever the warning device chosen, it should be mounted unobtrusively high up in an inaccessible place. The leads to the device should be of an adequate gauge to avoid any voltage drop associated with a long run. The wires should be concealed from view.

We strongly recommend that a seperate 12 Vbattery be used in any burglar alarm. This should be checked at regular intervals to ensure it is still in good condition and should be replaced as a matter of course when it has been in service for a period of one year.

ALARM UNIT

The specification of our alarm unit is shown in Table 1. From this one can see that the alarm has seven 'normally closed' circuits (A2-A8) plus a silent entry

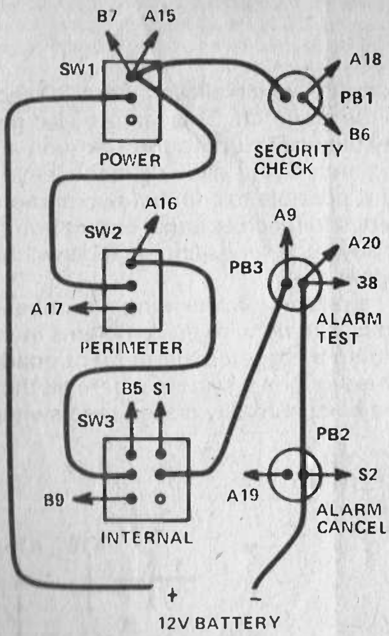


Fig 7 Wiring of the switches and pushbuttons

HOW IT WORKS

UNLIKE SOME ALARMS that use a single sensing loop with all the switches wired in series, this design features a number of different alarm groups. These are broken down into two groups designed for normally closed (N/C) switches – Perimeter Group (inputs A_1-B_8) and Internal Group (inputs B_1-B_4m) — together with one group for normally open (N/O) switches (inputs to A_0).

The inputs to each of the circuits described above have their own input circuitry.

PERIMETER CIRCUIT

The normally closed sensors associated with the perimeter circuit (inputs to A_1-A_8) are connected to the circuitry around IC_1 and IC_2.

These ICs are Quad NOR gates which, in this application are configured as inverters. The sensors are connected to the inputs of these gates via the resistors R_{10}-R_{17}. With the sensor switch closed the output of the associated IC will be high. If the switch is opened the output will go low as the inputs to the gates are then tied high via resistors R_1-R_8. R_9 is included to ensure that the inputs to the CMOS ICs are terminated under all conditions. The capacitors C_1-C_8, together with the resistors R_{10}-R_{17}, provide a filter to ensure that transients on the input lines do not trigger the alarm.

In each output of IC_1 and IC_2 there is a LED which is connected to the Security Check Button (PB_1). Upon operation of this button power is supplied to the LEDs which will light if the IC they are connected to has a low output, ie the input is triggered. The diodes in series with the LEDs are necessary because of the low reverse voltage breakdown of the LEDs. Diode D_1 supplies power to the input circuitry during the security check. The input A_1 provides the silent entry feature and is described below.

The other sections A_2-A_8 have their outputs fed via an RC network, which generates a negative pulse upon triggering, to one of the inputs of IC_4. Thus if any of the inputs are triggered a positive pulse at the output of IC_4 will result.

SILENT ENTRY CIRCUIT

With the silent entry circuit a 30 second delay due to R_{26}, C_{16} and $IC_{3/1}$ overrides the output of $IC_{1/}$ immediately after the alarm has been energised. After this time if the input is triggered the output of $IC_{3/}$ will go high, having been inhibited from doing so until now by the high output of $IC_{3/1}$, and will toggle the RS flip flop formed by $IC_{3/3}$ and $IC_{3/4}$, taking the output of $IC_{3/4}$ high. After another 30 second delay due to R_{38} C_{18} the input to $IC_{5/1}$ will be high and its output low.

TRIGGERING CIRCUIT

The same output results if one of the other inputs is triggered and the output of IC_4 goes high momentarily.

This output is used to toggle, via $IC_{5/2}$, the RS flip flop formed by $IC_{5/4}$ which is used to control the alarm and resetting circuitry described below.

$IC_{5/2}$ also has two other inputs. The first, consisting of the network R_{19}, C_{22} and D_{11}. This circuitry disables the alarm function when the Perimeter Switch is in the off position and for a short period of time after the switch is moved to the on position by holding the input of $IC_{5/2}$ high. This prevents spurious triggering.

The second input to $IC_{5/2}$ is from the normally open input (A_9), as well as the emergency and alarm test switches. If any of these switches are taken low a negative going pulse is coupled to $IC_{5/2}$ to trigger the alarm. These functions operate even if the perimeter sensors are off. This input can be used for emergency inputs such as fire alarms.

OUTPUT

The positive going pulse at the output of $IC_{5/2}$ sets the RS flip flop $IC_{5/3}$, $IC_{5/4}$ and in this triggered state $IC_{5/3}$ output is low and $IC_{5/4}$ is high.

The delay circuitry uses a CA3130 (IC_6) configured as a comparator. C_{19} is normally charged to +10V until the flip flop is triggered allowing it to discharge via R_{42}. When the voltage on C_{19} has fallen to about 20mV (the level set by R_{44} and R_{45} on the non-inverting input of IC_6). The output of the IC will go high resetting the flip flops formed by $IC_{5/3}$, $IC_{5/4}$ and $IC_{3/3}$, $IC_{3/4}$. R_{47} is included in the feedback loop to provide some hysteresis.

The output device can either be a relay or siren circuit. We have provided for both options. The siren output is formed by two 555s, one operating at a high frequency and driving the speaker via driver transistor Q_1 and the other at about 2Hz which is used to modulate the frequency of the first.

The relay and 555s are energised when Q_2 is turned on by the high output of $IC_{5/4}$ as the flip flop is set.

Addition circuits can be added in blocks of four at a time (as Board B) and connected to the Aux. input.

AUXILIARY BOARD

The circuitry of board B is almost identical to that of Board A. The main difference is that the negative going outputs of each IC are ORed using diodes D_7-D_{10} as opposed to a logic gate.

This board can only be energised if the perimeter board is powered up. The capacitor C_1, together with R_{10} and D_2, provide a short positive going pulse upon switch on to disable the main alarm for a brief period of time.

under these conditions.

We have provided a test button so that a check on the security of the house can be made before the alarm is set, indicating immediately which window or door is open.

As well as the external circuits the system has provision for connecting a number of internal circuits. These may be actuated by normally closed switches — in which case they should be connected to B1-B4 — or by normally open sensors connected to A9.

It may well be worth considering installing a series of emergency push buttons. Such switches should be mounted on the doorframes of the front and rear doors or in a readily accessible position near the doors. They enable the occupant to set off the alarm if a caller forces his way into the house when the door is opened. Although this is not a common event, emergency switches provide elderly or timid people with a feeling of security.

Use good quality bell pushes for these circuits and connect them to the A9 inputs on the circuit board.

FIRE ALARMS

Fire sensors may be wired across the A9 input. The actual fire sensors should be mounted in the ceilings of

PARTS LIST

BOARD A
RESISTORS all ½ W 5%

R1-8,54	22k
R9,37,39,41,46,51,53	47k
R10-25,48,52	1k
R26,38,42	4M7
R27-34,36,43,49	1M
R35	10k
R40	100R
R44	220k
R45,47	680R
R50	2k2

CAPACITORS

C1-8,16,18,22,23	10u 16 V tantalum
C9-15,17	47n polyester
C19	22u 16 V tantalum
C20,21	100n polyester
C24,26	15n polyester
C25,27	100u 16 V

SEMICONDUCTORS

IC1-3,5	CD 4001
IC4	CD 4068
IC6	CA 3130
IC7,8	555
LED1-8	.2" type LED
Q1	TIP 2955
Q2	MPS6515
D1-11	1N914
D12,13	1N4001

MISCELLANEOUS
PCB as pattern, 12 V 185R relay.

BOARD B
RESISTORS all ½ W 5%

R1-4	22k
R5,10	47k
R6-9,11-14	1k
R10	47k
R15-18	1M

CAPACITORS

C1	33u 16 V electrolytic
C2-5	10u 16 V tantalum
C6-9	47n polyester

SEMICONDUCTORS

IC1	CD 4001
D1-11	1N914
LED1-4	.2" type LED

MISCELLANEOUS
PCB as pattern.

GENERAL FOR BOARDS A & B.
SWITCHES

SW1	SPST toggle switch
SW2	SPDT toggle switch
SW3	DPDT toggle switch
PB1-3	single pole press to make push type.

MISCELLANEOUS
Case to suit, 12 V battery terminal strip, bell or speaker

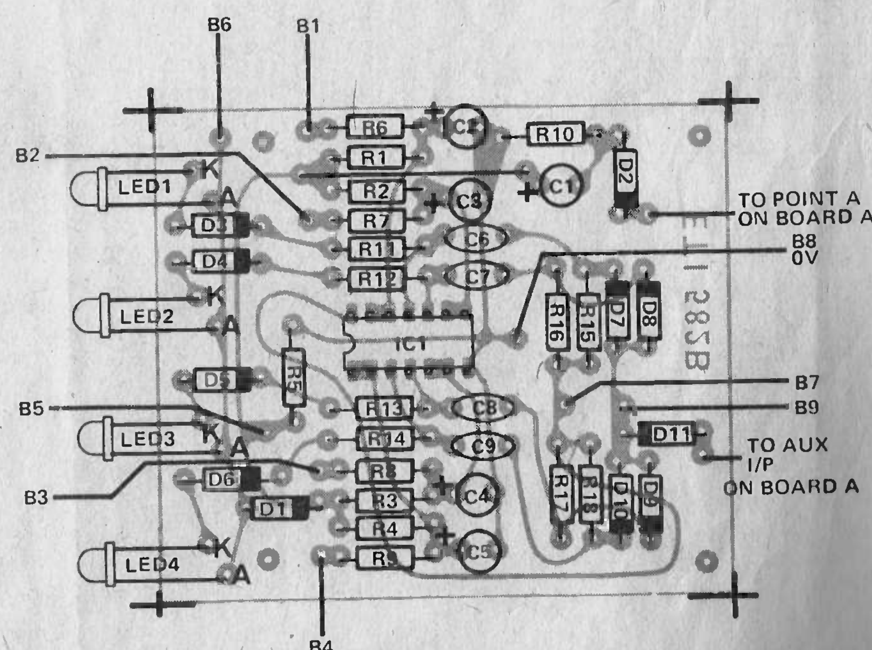


Fig. 9. Component overlay of the 'B' board.

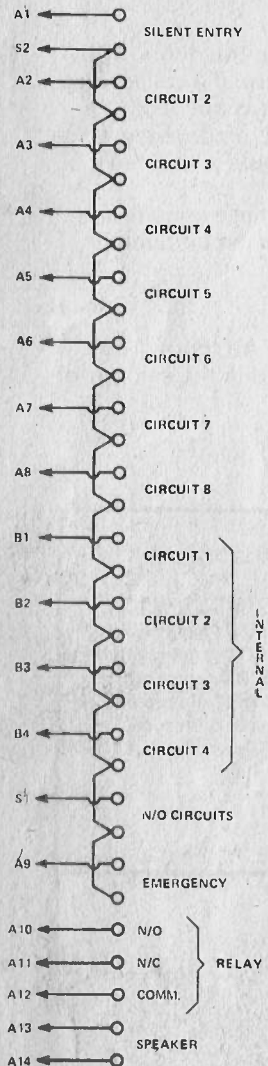


Fig. 10. Connection of the rear terminal block.

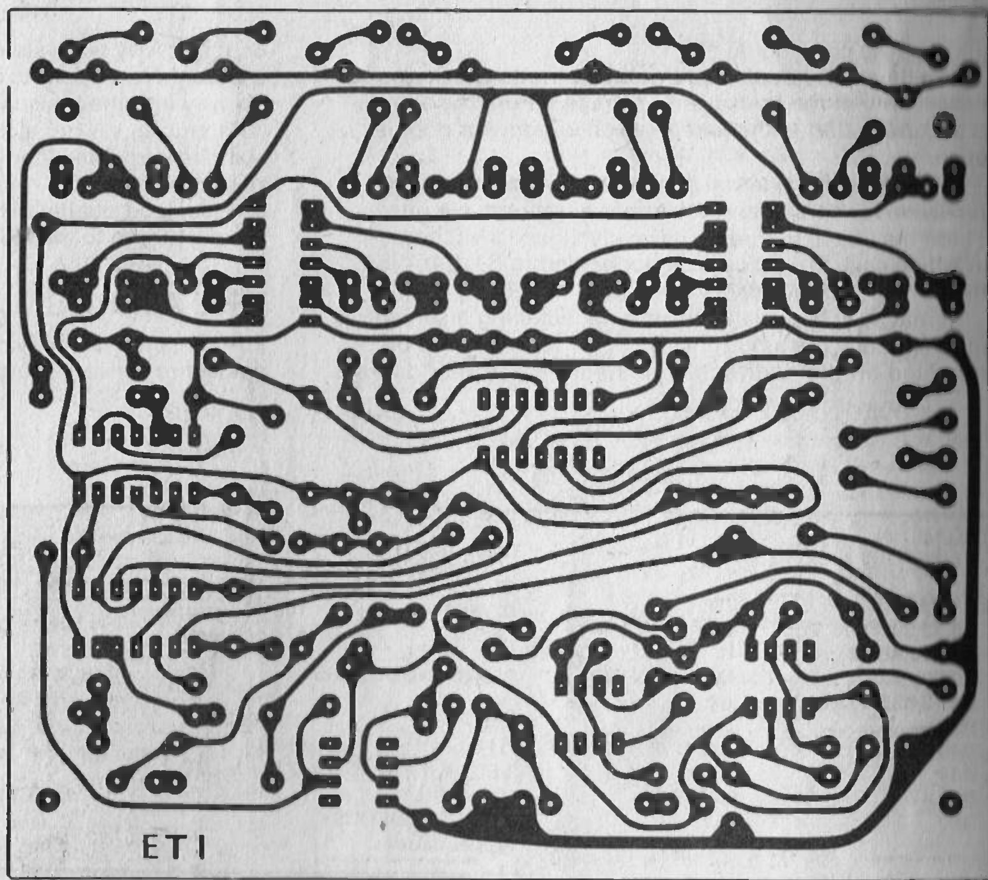
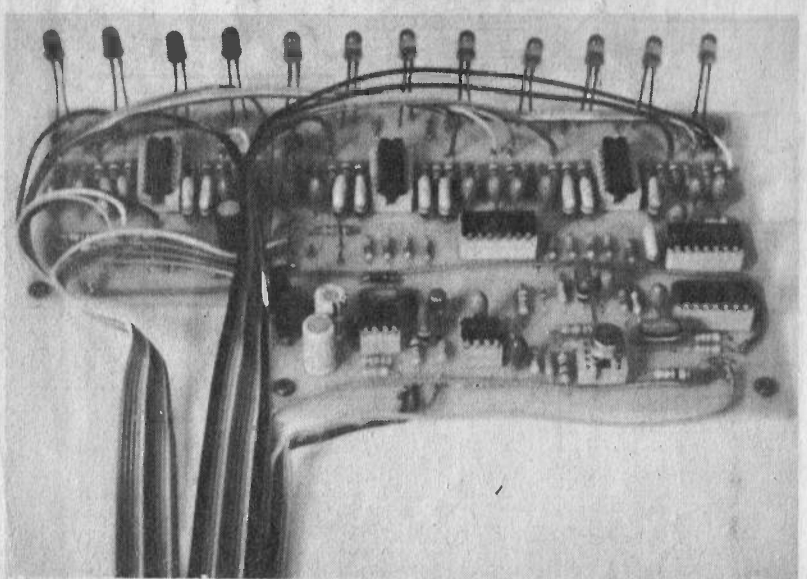


Fig. 11. PCB foil pattern of 'A' board shown full size (130 x 115mm).



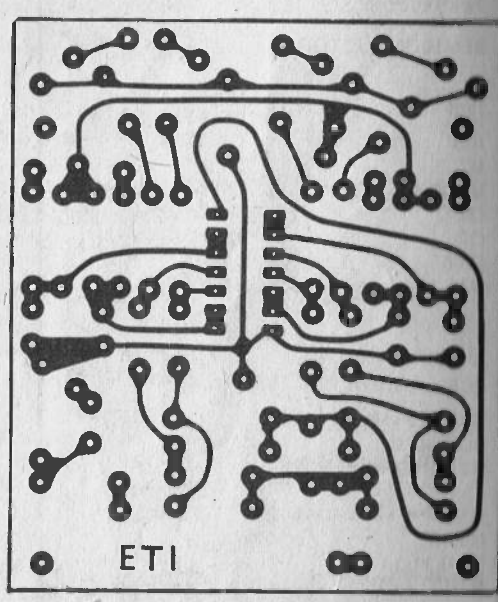


Fig. 12. PCB foil pattern of 'B' board shown full size (75 x 65mm).

rooms in which there is a fire hazard — kitchen, living room, rooms with electrical or heating appliances or where people smoke (don't forget the bedroom if you've a habit of smoking in bed!). Sensors should also be installed in the roof of the garage especially if this is attached to the house — the laundry, workshop etc.

CONSTRUCTION

Due to the number of components, it is recommended that the unit should be built using the PCBs shown here.

Assemble the components, watching the connection of all the polarised components. Also solder connection of all the polarised components. Also solder the CMOS ICs last and then solder pins 7 and 14 first. This allows the protection diodes inside the IC to be effective. The LEDs should be mounted parallel to the PCB as shown in the overlay as these have to protrude through holes in the chassis.

Boxing of the alarm unit is largely a matter of choice. Our layout can be seen in the photographs. Note that we did not fit a key switch to our alarm, but installed it in a locked cupboard which could also be used for the storage of valuables.

SPECIFICATIONS

Types Of Inputs	Silent entry Perimeter circuits Internal circuits Emergency circuits
Silent Entry	Single circuit, 30 s exit delay, 30 s entry delay.
Perimeter Circuits	7 circuits, N/C contacts, can be expanded in units of 4.
Internal Circuits	4 circuits, N/C contacts, can be expanded in units of 4. Any number of N/O circuits.
Emergency Circuits	Any number of N/O circuits. These circuits are active even if perimeter and internal circuits are switched off.
Current Drain	
Emergency only	2.5 mA
Alarm active	9 mA
Alarm sounding	500 mA
Alarm Time	12 minutes.

Texas Instruments 7647

This flexible device should prove useful in a variety of applications, including video games, alarms and more.

THE SN76477 is a bipolar/I^2L device that provides a noise source, VCO, low frequency oscillator, envelope generator, plus various mixing and control logic on a single 28 pin DIL package. By the connection of appropriate external components and application of logic level control signals a wide variety of complex sounds can be synthesized.

The block diagram in Fig. 1 shows the main circuit blocks, each of which is described in detail below.

SLF (SUPER LOW FREQUENCY OSCILLATOR)

The SLF can be operated in the range 0.1-30 Hz, the specific frequency is determined by a control resistor connected to pin 20, and a capacitor connected to pin 21. The frequency being given by the following equation:

$$F_{SLF} = \frac{0.64}{R_{SLF}\,C_{SLF}}\,\text{Hz}$$

VCO (VOLTAGE CONTROLLED OSCILLATOR)

The VCO provides an output whose frequency is dependent upon a voltage fed to its input, the higher the voltage the lower the frequency. The control voltage may be either the SLF output, or an external voltage applied to pin 16, the SLF output being selected when the voltage applied to pin 22 is a logic '1', and the external source when pin 22 is at logic '0'.

The "range" of the VCO is internally set at a ratio of 10:1. The minimum VCO frequency is determined by a control resistor connected to pin 18 and a capacitor to pin 17. This minimum frequency is given by the equation:

$$F_{MIN\,VCO} = \frac{0.64}{R_{VCO}\,C_{VCO}}\,\text{Hz}$$

The "pitch" of the VCO's output is changed by varying the duty cycle of the output. This is achieved by adjusting the ratio of the voltages at pins 16 and 19. The duty cycle is given by the following equation:

$$\text{VCO Duty Cycle} = 0.5\left[\frac{\text{V pin 16}}{\text{V pin 19}}\right]\%$$

leaving pin 19 high produces an output with 50% duty cycle.

MIXER SELECT C	MIXER SELECT B	MIXER SELECT A	MIXER OUTPUT
PIN 27	PIN 25	PIN 26	
0	0	0	VCO
0	0	1	SLF
0	1	0	NOISE
0	1	1	VCO/NOISE
1	0	0	SLF/NOISE
1	0	1	SLF/VCO/NOISE
1	1	0	SLF/VCO
1	1	1	INHIBIT

Table 1

NOISE OSCILLATOR

The "noise oscillator" supplies random frequencies for the "noise generator". The noise oscillator requires a 43 k resistor to ground at pin 4. The "noise oscillator" controls the rate of the "noise generator". An external noise oscillator may be used to provide this control. The external source is applied to pin 3 and provides an automatic override of pin 4.

NOISE GENERATOR/FILTER

The output of the "noise generator" feeds an internal noise filter. This "rounds off" the generator's output, reducing the HF content of the noise. The upper 3 dB point is given by

$$F_{UPPER} = \frac{1.28}{R_{NF}\,C_{NF}}$$

where R_{NF} and C_{NF} are external components connected to pins 5 and 6 respectively.

MIXER

The "mixer" logic selects one, or a combination, of the inputs from the SLF, VCO, and noise generator. Selection is according to Table X.

SYSTEM ENABLE LOGIC

The "system enable" input provides an enable/inhibit for the system output. The output is inhibited when the voltage at pin 9 is a logic '1', and enabled when logic '0'.

ABSOLUTE MAXIMUM RATINGS AT T_A = 25°C (Unless otherwise specified)

SUPPLY VOLTAGE, Vcc (1), PIN 15 .... 6.0V
SUPPLY VOLTAGE, Vcc (2), PIN 14 .... 12.0V
INPUT VOLTAGE APPLIED TO ANY DEVICE TERMINAL 6.0V
STORAGE TEMPERATURE .... −65°C to +150°C
OPERATING TEMPERATURE RANGE . −55°C to +120°C
LEAD TEMPERATURE 1/16 INCH FROM CASE FOR 10 SECONDS . +260°C

RECOMMENDED OPERATING CONDITIONS

	MIN	TYP	MAX	UNITS
SUPPLY VOLTAGE, Vcc1, PIN 15	4.5	5.0	5.5	V
SUPPLY VOLTAGE, Vcc2, PIN 14	5.7		9.0	V
OPERATING FREE-AIR TEMPERATURE	0	25	70	°C

OPERATING CHARACTERISTICS AT T_A=25°C AND V_{CC1}=5.0V

ONE SHOT LOGIC

The "one shot" logic can be used to provide sounds of a short duration. The duration of the "one-shot" is given by the following equation:

$T_{OS} = 0.8\ R_{OS}\ C_{OS}$

where R_{OS} and C_{OS} are external components connected to pins 24 and 23 respectively. The maximum duration of the "one-shot" is about two seconds.

The "one-shot" logic is triggered by the trailing edge of the system enable logic control signal.

ADL (ATTACK/DECAY LOGIC)

The ADL determines the envelope for the mixer's output. The envelope selected is determined by the ADL control inputs to pins 1 and 28, the output selected being shown in Table 2.

ENVELOPE GENERATOR AND MODULATOR

The attack decay characteristics of the output are determined by the components connected to pins 7, 8 and 10.

The attack and decay times are given by the following:

$T_{ATTACK} = R_A\ C_{A/D}$ secs

$T_{DECAY} = R_D\ C_{A/D}$ secs

where $C_{A/D}$ is the attack decay capacitor connected to pin 8, and R_A and R_D are resistors connected to pins 7 and 10.

OUTPUT AMPLIFIER

The output amplifier provides a low impedance output. The peak output voltage is determined by the following equation:

$$V_{OUT} = \frac{3.4\ R_S}{R_G}$$

where R_S is a summing resistor connected to pins 12 and 13 (set equal to 10 k) and R_G is a gain resistor connected to pin 11.

NOTES:

1. Supplies greater than 5VO may be used, in which case they should be connected to pin 14 to allow the internal regulator to supply the internal circuit requirements.
2. For dedicated sound logic inputs (pins 1, 9, 22, 25, 26, 27 and 28) may be hard-wired to high or low logic levels.

Fig. 2. Showing the various envelopes that the SN76477 circuitry can produce

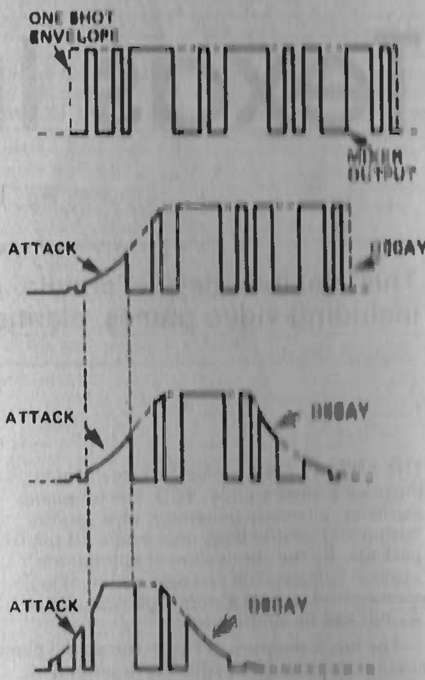


Fig. 1. A voltage fed to the input of the VCO will change the output frequency of this oscillator.

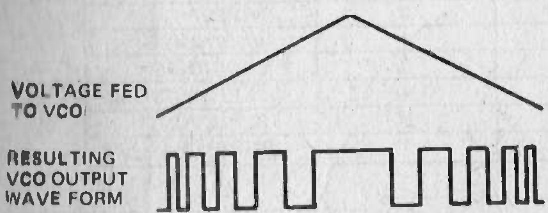
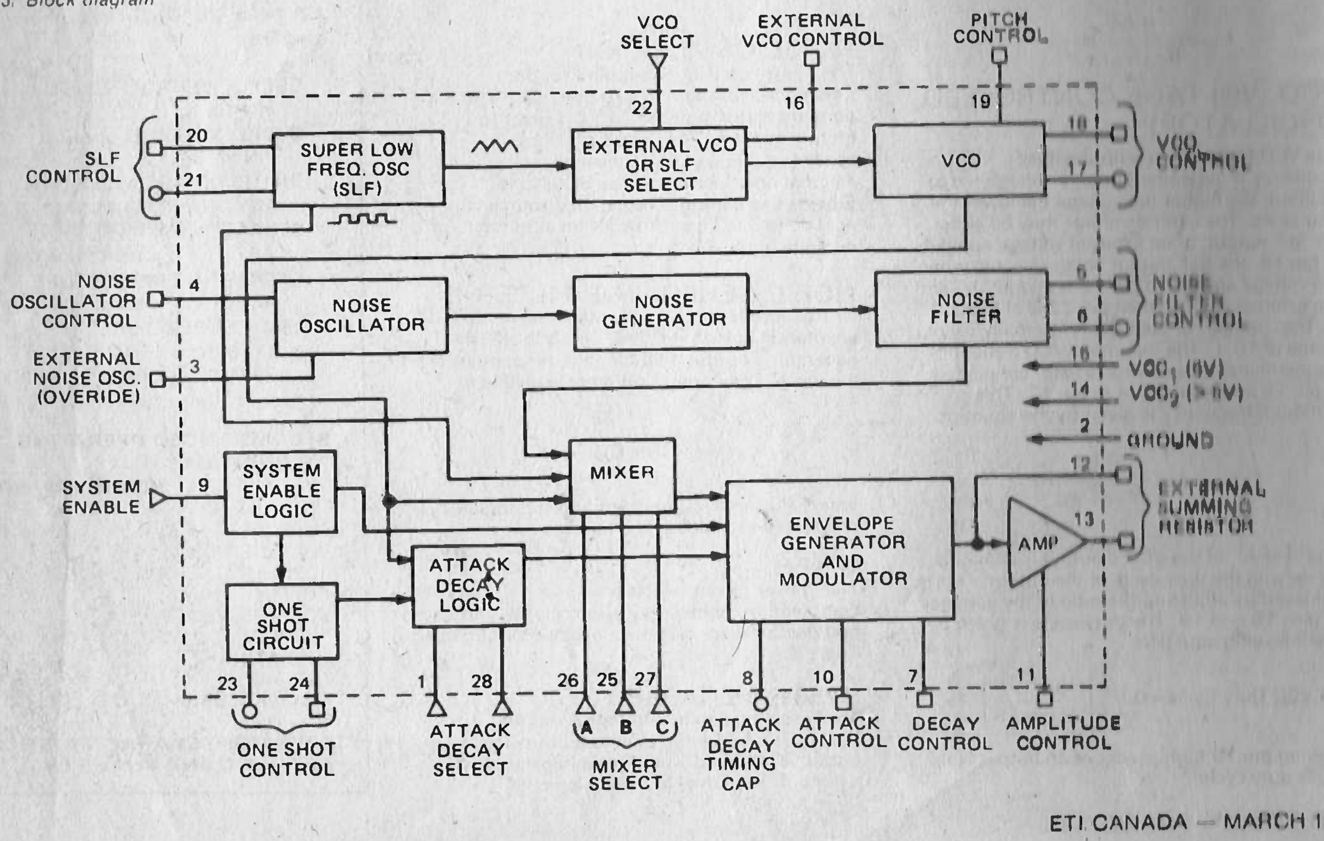


Table 2

ADL SELECT 1 PIN 1	ADL SELECT 2 PIN 28	OUTPUT
0	0	VCO
0	1	MIXER ONLY
1	0	ONE SHOT
1	1	VCO WITH FLIP FLOP

Fig 3. Block diagram



The Jungle Telegraph of the Twentieth Century

Trafficing: messages across the continent. Mike Goldstein VE3GFN, discusses this fascinating facet of amateur radio. It's also a service YOU can use.

WE'VE ALL SEEN THOSE African adventure films where, after several "I thought I saw a man in the bushes" sequences, the castaway hero and heroine clutch each other as the drums begin to beat! First, the drums are loud, as the locals "call up the troops". Then, a drummer further away picks up the message, and repeats it. We know that, within the hour, the message will have been relayed to the far reaches of the Dark Continent, and every jungle resident not otherwise engaged will be hastening to the slaughter. We leave our hero at this point, counting the bullets in his Webley revolver . . .

The modern version of this jungle telegraph is a message-handling service, operated by the radio amateurs of North America.

Briefly, this system consists of a series of message networks, spanning the continent, and which operates on a daily basis throughout the year. Operated by the radio "hams" as part of their hobby, the system enables messages (radiograms) to be relayed from network to network, until the last station receiving the message delivers it by mail or telephone.

This service is absolutely free, and YOU can use it.

This system of organized, message-relaying networks, known as the National Traffic System (NTS) is sponsored and administered by the American Radio Relay League, (yes, that's where the name comes from!) the international amateur radio organization in North America. The system was established for emergency communications purposes, and that is its prime function. However, emergencies being of a rather sporadic nature, the NTS practices daily on ordinary message traffic, and any message of a non-commercial nature is cheerfully accepted. Messages are the "lifeblood" of a message-handling network, so they are always "looking for business".

WHO CAN USE IT?

Any person who knows about the service can use it. The user need not be a radio amateur, but need only know a person who is; not all radio hams are involved in the system, but most can start a message on its way, one way or another. The service is absolutely free to the public, and radio amateurs are not allowed to accept any reward for the use of their services — beyond a grateful "thanks!".

There is no limit to the number of messages one can send, and the length of each message can be twenty or thirty words, if necessary. If proper addresses and telephone numbers are provided, to aid quick delivery, the originator often finds himself receiving a reply the same day! Not bad, for a free, volunteer service.

HOW DOES THE SYSTEM WORK?

The NTS is a sort of pyramid structure, with each level of the pyramid having liason with the next higher level, and the next lower level.

Basically, the NTS divides the continent into three Areas — the Eastern, Central, and Pacific Area — and each Area has its own major net, the EAN, CAN, and PAN. Representatives, or liasons, carry traffic back and forth between these three Areas. These liason operators form the Transcontinental Corps (TCC), the elite of the NTS.

Each Area has within it a number of Regions, each Region having its own Region Net. The 11th Region, for example, is made up of Eastern Canada — the Atlantic Provinces,

AMATEUR RADIO SIMULATED EMERGENCY TEST

A massive snowstorm struck the Toronto area on the afternoon of January 28, paralysing traffic, creating a breakdown in law-enforcement leading to looting situations, and stranding hundreds of motorists on Highway 401, in danger of freezing to death as the storm roared unabated through the afternoon.

As it became obvious that civic authorities were going to require assistance in handling a wide variety of communications problems, the local Amateur Radio Emergency Service swung into action.

Being a Saturday, a large number of ham mobiles were on the roads, stranded along with many others. These operators collected names and addresses of motorists wishing to advise families of their predicament, and kept the local police advised on traffic conditions.

Amateur radio stations were set up at Red Cross headquarters, the local office of the Ontario Hospital Association, local Borough utility offices, and at least one Metro radio station. Liason was established with the police radio system.

Soon, traffic reports from on-the-spot mobiles began to flow in to the radio station, for broadcast to the public. The police were receiving a steady stream of reports of accidents, and some looting areas. Borough snowplows and trucks were directed by radio to the worst-hit areas. Dozens of motorists were assisted by Red Cross, with food and accommodations, after being rescued by snowmobile teams coordinated by the radio hams.

Ham stations set up at various hospitals coordinated the arrival of frostbite and heart attack victims, coordinated by the O.H.A. office, using the radio system.

The amateurs, using vhf repeaters for local Metro-wide coverage, and high-frequency radio for the long-haul messages around the Golden Horseshoe, demonstrated the ability for assistance in disaster situations, for which they have long been famous.

The preceding situations represent the format for the annual **Simulated Emergency Test**, held on the afternoon of Saturday, January 28, by local-area radio amateurs.

Sponsored by the American Radio Relay League, this Test is an annual event designed to tackle the many problems that exist in any disaster situation, and examine methods of eliminating them, while training radio amateurs in the art of disaster communications. Amateur radio has a long tradition of rendering assistance in time of emergency, and it is only through constant training that the amateurs are able to field a large, experienced group of communicators when the need arises. This Test serves that purpose.

The Test is coordinated in the Toronto area by Michael Goldstein, helped by his assistants Paul Edgley, Lyle Stanway, Saltus Jones, Keith Ballinger, and over a hundred radio amateurs.

Quebec, and Ontario — and this 11th Region Net is called the Eastern Canada Net (ECN).

Each Region net meets once, before the Area net convenes, and once after the Area net has terminated — so messages are collected for transmission out of the Region, on the Area net, and incoming messages to the Region are picked up on the Area net, and distributed on the Region net. Therefore, ECN will send both a "Transmit" and a "Receive" liason to the Area net, EAN.

Each Region net is made up from liasons from the Section nets which (hopefully) cover all of the geographical area within the Region. The ECN collects liason stations from the Atlantic Provinces Net APN, the Quebec Section Net QSN, the Ontario Southern Net OSN, the Grey Bruce Net GBN, and the Ontario Phone Net OPN. Messages designated for addresses within the Region are passed between Sections on the ECN, while messages coming into the Region from the Area net are distributed on ECN to the proper Section net, for local delivery.

Section nets may collect representatives from local nets which are not affiliated with the NTS, or they may act as the only message-handling system within a Section, or a portion of the Section.

INSIDE

The National Traffic System represents the only "professional" side of amateur radio: the traffic schedules are kept religiously, the procedures are adhered to, and the normal "socializing" that is a tradition in amateur radio is highly discouraged on the formal nets . . . when they open for business, the discipline of the operators is immediately apparent.

The foundations of the NTS were laid down prior to the Second World War, and many of its present operators have been involved in handling traffic for more than thirty years — their proficiency can only be imagined!

To make life easy, the techniques and procedures used on all NTS nets across the country are identical; if a ham can operate in one net, he can operate in any of them, if his code speed is up to scratch. The formats of the radiograms are also identical.

While a certain percentage of NTS business is conducted on radiotelephone, the dyed-in-the-wool traffic operator is generally a "CW man" — a Morse code operator. The use of Morse code, far from becoming an obsolete technique, has been brought to a high level of efficiency on the traffic nets. Operators on traffic nets employ "perfect break-in" — a technique which allows them to receive and transmit virtually INSTANTANEOUSLY, and the faster they transmit, the better it works. The transmitting station can immediately detect noise on the channel, interference, or his chum on the receiving end trying to interrupt him.

Message-handling in the traffic networks is only one of many interesting aspects of the hobby of amateur radio. This pastime is pursued by hundreds of thousands of people, from all walks of life, across the continent! Not only does this activity provide many with excitement of being "in the midst of things", but it prepares and keeps in tune what can be in times of emergency a vital message service.

"73, old man. I must check into the local traffic net. I can hear the drums beating . . . "

The keen traffic operator often teaches himself to type, as Morse can be "copied" much faster on a typewriter than with a pencil. For several reasons, formal, written messages can be taken much faster on Morse code than by radiotelephone. The traffic nets use a system of abbreviations and coded instructions, which allow them to "say a lot" in a hurry. Net operation is often at speeds in excess of twenty-five words-perminute, at the higher levels.

THE AMERICAN RADIO RELAY LEAGUE
RADIOGRAM
VIA AMATEUR RADIO

NUMBER	STATION OF ORIGIN	CHECK	PLACE OF ORIGIN	TIME FILED	DATE
5	VE3ETI	14	TORONTO ONT.	1433	MAR 14

THIS RADIO MESSAGE WAS RECEIVED AT
AMATEUR STATION ________ PHONE ________
OWNER ________
STREET ADDRESS ________
CITY AND STATE ________

To B. CORNWELL,
44 AFFERTON DR.,
NANAIMO, BC.

WILL BE IN VANCOUVER NEXT WEEK, HOPE TO MEET YOU MON.
OR TUES. PLEASE TRY TO GET IN TOUCH WITH TONY.
REGARDS,

FRANCES

SENDER'S ADDRESS AND PHONE NUMBER FOR REFERENCE

	FROM STATION / TO STATION	LOCATED AT	DATE	TIME	OPERATOR
REC'D	VE3ETI	Toronto, Ont.	3/14	1444	cs
SENT	phoned				

YOUR REPLY TO THIS MESSAGE WILL BE HANDLED WITHOUT CHARGE BY THE RECEIVING STATION WHOSE ADDRESS IS SHOWN ABOVE. AMATEUR RADIO OPERATORS AND MEMBERS OF THE A.R.R.L. LICENSED BY THE FEDERAL COMMUNICATIONS COMMISSION, OFFER TO THE PUBLIC A MESSAGE SERVICE WITHIN THE U.S.A. AND ITS POSSESSIONS WHERE POSSIBLE. AS MESSAGES ARE HANDLED BY RADIO AMATEURS SOLELY FOR THE PLEASURE OF OPERATING, NO COMPENSATION CAN BE ACCEPTED BY A STATION OWNER, SO DELIVERY IS NOT GUARANTEED BY THE LEAGUE OR ITS MEMBER OPERATORS.

A typical radiogram . . . someday soon a friend of yours will be receiving a message on a form like this.

BITS, BYTES and BAUDS

by Bill Johnson VE3APZ

Many parts make a whole; Bill Johnson puts it all together.

DURING THE LAST few articles in this series, I have attempted to impart enough wisdom to those who have closely followed my every word to allow me to go into an explanation of a complete system in this article.

We have so far learned about various means of communication, the basic architecture of the computer, binary numbering systems, the structure of the bus, peripheral and memory addressing, I/O transfers, and mass storage devices.

My own introduction to computers took a very similar approach to the pattern that I have followed in these articles. When I had finished learning about the hardware, I could take apart a computer, shuffle the circuit cards, and put it back together, having confidence that I would get it working again. (Eventually).

If you are reading these articles to get an insight into how a computer works, then my task, but for a few details here and there, is finished. However, I suspect that many of you are interested in building a system of your own, and once it is built, would like to be able to do something with it.

At this point, you will most likely have a picture in your mind of a million and one things that have to be done inside a computer to make it work properly and in a usable fashion. You probably have a feeling of utter helplessness when faced with the enormous task of making a computer do something. A married man who buys a computer with no programs will find that by the time that he has something worthwhile to show for his huge investment in time and money, he doesn't have a wife to show it to. (Or vice versa). Don't worry, this is normal.

DOING THE HOUSEWORK

One of the 'extras' that should be bought with any computer is a set of programs known as an OPERATING SYSTEM. An OPERATING SYSTEM, or O/S for short, will usually come complete with the following:—

a) A TEXT EDITOR — for creating and changing SOURCE (human-readable) programs in plain language,
b) An ASSEMBLER — for changing SOURCE programs into OBJECT (or machine-usable) programs,
c) A LINKER — for taking the output program or programs generated by the assembler and joining, or LINKING them in such a way that they can run together in memory.

Some other handy programs that may or may not be included in an operating system package fall under the heading of SYSTEM UTILITIES. UTILITY programs allow the user to interchange files between peripherals, make corrections or PATCHES to previously-assembled programs without the need to re-assemble them, destroy old programs, change program names, search a program or file for any particular bit pattern or instruction, print a listing of a program, or a host of other small, but important chores.

One type of program that is only used while a program is being written and debugged is called an ODT, short for ONLINE DEBUGGING TECHNIQUE. This program, when run at the same time and in conjunction with a program that has just been written, allows the programmer to run his new program under very tight control, and with the option of printing a log of everything that his program does, starting it under simulation conditions, or stop it if it accidentally gets out of control. Since this type of program is generally used to get rid of bugs in a program, it is sometimes called a Dynamic Debugging Technique, after that other well-known bug-killer, DDT.

An operating system consists of a main program, called a MONITOR, in addition to the above programs. The monitor, as its name implies, monitors the entire computer system. One of the most important functions of the monitor in this regard is to handle all input/output operations. The reason for this is simple: in a large system, if there are ten programs running concurrently in a computer, the chances are very high that each program will want to send or receive data to or from an external device at some time or another. If all these programs were allowed to handle their own I/O by sending instructions directly to discs, cassettes, terminals, etc., the result would be catastrophic — since each program would work without reference to all the other programs then one program could give a write instruction to a disc while another program is trying to read, giving unpredictable and usually catastrophic results.

For this reason, all I/O is handled by the use of I/O requests made by the user program to the monitor in the form of codes that can be left in a special place for the monitor to find them. The operating system will receive requests from all programs in the system, store the requests in strict order of priority, and execute the I/O as the speed of each requested peripheral permits. The monitor will take care of all HOUSEKEEPING, such as waiting for the device to be ready to accept commands, formatting the data into the right-sized blocks for transfer to the various peripherals, and each user program will simply have one command to execute to tell the monitor what to do and where it can find the data to be transferred.

SOMEBODY HAS TO LISTEN TO THE OPERATOR

Another of the monitor's functions is to handle all communications with the main console terminal, and its indispensible accessory, the computer operator. One may think that this is the same thing that was just mentioned under the heading of 'handling the I/O', but there is one major difference — that is that the console terminal can override all system activity. Through it, the operator can monitor, suspend, or change system activity simply by entering pre-defined commands. It should be understood at this point that these commands are not computer instructions themselves, but are sequences of letters that mean something to the monitor. For example, assume that a multi-task operating system for a large computer has just been started up. It does not know the time or date, since its memory has just been cleared. (From hereon, all computer output will be preceded by an asterisk (*), and all operator input will be prefixed with a dash (-)). Here is an example of typical computer-operator dialogue: chart A.

The above example shows how an operator can talk to the system, and monitor and control its operation by very simple means, using simple, English-like commands. However, before the programs listed above can be run like this, they must be specified, designed, flowcharted, encoded, and debugged.

ENTER THE PROGRAMMER

Since the human operator does not have to think like a machine in order to operate and understand a computer, it seems only fair that the human programmer should not have to remember the horrendous number of different possible combinations of bits that make up the machine's instruction repertoire.

Every instruction can be represented by a combination of bits, which form a binary number. When a program is loaded and executed, it must be in this form so that the computer can understand and make use of it. This does not mean that the programmer must code it in machine language though, and in practice this is very rarely done.

There are several 'levels' of computer programming languages today. At the machine level, every instruction occupies one, two, or three memory bytes in a microcomputer, depending on the instruction. The instruction itself is contained in the first byte, and if no memory reference is needed, then this represents a complete instruction. An example of this is the CMP A instruction in a 6800 which complements (changes all the zeroes to ones and vice-versa) the 'A' accumulator. Since no memory reference is needed, this is a single-byte instruction. If an instruction references data in another location that is very close, only two bytes may be required, since only the last eight bits of the address need be given, the first eight being the same as the instruction. If the referenced location does not have the same first eight bits of address, then the instruction must be a three-byte one to give the full sixteen-bit address.

EFFICIENCY VS. SIMPLICITY

Machine-level programming operates most efficiently at the time the program is run, because the code executed exactly fits the requirement of the program. It is, however, a very tedious language in which to program since a great number of instructions have to be written, coded, and debugged, to perform a relatively simple task. It is called, therefore, low-level programming.

In practice, when a programmer uses low-level programming (i.e. when he is forced to use it), he will write the coding in MNEMONIC form. This means that instead of writing down the actual numbers that the computer will use, he writes down simple combinations of letters that represent the instruction and are easy to remember. For example, the following code adds two numbers in the 6800: chart B

As you can see, it is much easier to remember the MNEMONIC forms of these instructions.

There is one handicap to the system described above, however. Let us assume that the above three lines form part of a large program. The program has been run and a bug has been detected, so the programmer wants to change the program by adding another instruction early on in the program. All instructions past this point where the change takes place will have to be moved down by one, two, or maybe

Chart A. Operator — computer dialogue.

*MASTER OPERATING SYSTEM VERSION 02B	Monitor announces that it has just been loaded.
*ENTER DATE AND TIME YY/MM/DD HH:MM:SS	Asks for date and time, and tells the form in which it is to be given.
-DATE 78/01/10 23:15:00	Operator enters information.
*DATE 78/01/10 23:15:00	
*READY	Computer indicates that it is ready.
-RUN PRINT	Operator tells monitor to run, or execute program called 'PRINT'
*PRINT 3 0100 05 01FF	Computer prints information about the program and starts executing it.
-RUN LISTING	Operator starts another program called LISTING
*LISTING 3 0200 05 05FF	Computer announces the loading of the second program, and runs it.
-RUN VERIFY	Operator tries to start another program.
*VERIFY *PROGRAM TOO BIG *LOAD ABORTED	Computer tries to load new program but finds that the two programs already in memory take up too much space for the new one to run.

At this point, the operator is uncertain how much memory is free. He thinks that he should have enough, but obviously, something has been put into the memory that he thinks is free. He thinks that there are 40 kBytes of memory free.

-FR	He asks the monitor to report on how much memory is free.
*20 kB FREE	Computer replies.
-SY	Operator asks for system status to find out why he only has 20 kB free.
*SYSTEM STATUS PROGRAMS IN MEMORY PRINT 3 0100 05 01FF LISTING 3 0200 05 05FF DATA 7 0600 00 2EFF	This tells him that one of the programs has loaded a data buffer that is taking up memory. He remembers that this is done by the PRINT job.
-KILL PRINT *EOJ PRINT 05	He gets rid of the PRINT JOB. Computer announces that the PRINT job has gone to end-of-job (i.e. has terminated) and gives its status code.
-RUN VERIFY	He tries again.
*VERIFY 3 0600 05 10FF	This time, since there is room in memory, the load is successful.
*EOJ LISTING 05 *EOJ VERIFY 05	If left alone, the jobs that are running will announce their end-of-job as soon as the tasks assigned to them are finished.

MACHINE CODE	MNEMONIC CODE	OPERATION
960024	LDA A 0024	Load accumulator A with the number in memory location 24
9B0025	ADD A 0025	Add accumulator A to the contents of memory location 25
970026	STA A 0026	Store the result in location 26

Chart B. Adding two numbers in 6800-ese.

more bytes. This may not seem to be important, until you realise that any memory location past that point that references another memory location as data will have to be changed, since the data address will now be different. See chart C.

It can now be seen that the LDA A, ADD A, and STA A instructions no longer reference the correct data, so they will have to be changed. The jump will also have to be changed, or the computer will start executing the data in location 28 as an instruction. As serious as this problem may seem, it can all be solved with the addition of LABELS into the source coding. A program with labels would look like this:

```
    START:  INSTR
            INSTR
            INSTR
            JUMP    CONTINUE
   DATA 1:  DATA
   DATA 2:  DATA
   DATA 3:  DATA
   DATA 4:  DATA
 CONTINUE:  LDA A   DATA 1
            ADD A   DATA 2
            STA A   DATA 3
            HALT
```

The above program, when converted to machine code by the assembler program, will generate the code as illustrated on the left side of the previous example, provided that you tell the assembler that the symbolic label START is equivalent to location 0020. All other labels will be assigned a value at assembly time referenced to this base address of 0020 and whenever an instruction refers to one of these labels, the assembler will substitute the calculated value of the label's address into the object program. This way, the program can be changed at will, leaving the tedious job of calculating all the memory location references to the number-crunching computer.

MACRO TO THE RESCUE

There are some limitations to the use of machine code. One is the number of instructions required to perform even simple tasks, and so the general rule when designing a system is to use a higher-level programming language if at all possible, such as BASIC, or FORTRAN. However, for those who are constrained for various reasons to use an assembler-level language (and there are many of them), there is a time- and exasperation-saving device known as the MACRO assembler. A macro assembler will perform all the functions of a straight assembler with a very popular addition: if you don't like the instructions that the computer is capable of decoding, then you can make up your own, using the instructions that are available, in groups. For instance, let us assume that you are writing a process-control system and you have a lot of long-winded multiplications to do. To multiply two numbers in a micro-computer can take twenty, thirty, or even more instructions at the machine level. Instead of writing down all these instructions every time you want to do a multiplication, the MACRO assembler allows you to specify your own MNEMONIC code and equate it with a long group of machine-level instructions. Every time the assembler recognises your own mnemonic in the SOURCE program, it generates all the instructions that you equated with it and puts them into the output OBJECT program.

HIGHER LEVELS YET

The higher-level programming languages such as FORTRAN, BASIC, COBOL, ALGOL, etc. have the advantage that they require the programmer to write only in simple, predefined, English statements. Their only disadvantage is that the code that they generate is sometimes much bigger, therefore taking more memory than the corresponding assembler-level equivalent program. They would therefore run somewhat slower, and be less attractive to the memory-bound small system user. However, during the last few years there have emerged some very good scaled-down, or limited-function versions of the high level languages quite usable by the small system user.

X=A+B

is an example of a BASIC program statement to add 2 numbers. BASIC and FORTRAN are very similar in their source language except that FORTRAN contains far more sophisticated statements. Also, BASIC and FORTRAN work in two different ways.

FORTRAN is a COMPILER-type language. That means that once the source program is written, it must be processed by the FORTRAN COMPILER program, which will generate output or OBJECT code that can be directly run by the computer without further need for the compiler program.

BASIC, however, is called an INTERPRETER-type language. There is no compilation process, and output object code cannot be generated, stored away, and used again on its own. When a program has been written in BASIC and the "RUN" button is pushed, the BASIC program is used as data for the machine language INTERPRETER program. The sequence of events that then take place is analogous to a line by line compile-then-execute process of the BASIC program.

The big advantage to a language such as BASIC is, of course, its interactiveness, that the programmer can interact directly with the program, a feature which makes personal computing so attractive.

Chart C. Problems if you don't use labels.

Old Address	Contents	New, after the addition of two instructions Address	Contents
0020	instruction	0020	instruction
0021	instruction	0021	instruction
0022	instruction	0022	instruction
0023	jump to 28	0023	instruction
0024	data	0024	instruction
0025	data	0025	jump to 28 (wrong, should be jump to 2A)
0026	data	0026	data
0027	data	0027	data
0028	LDA A 0024	0028	data
0029	ADD A 0025	0029	data
002A	STA A 0026	002A	LDA A 0024 (wrong, should be LDA A 26)
002B	instruction	002B	ADD A 0025 (wrong, should be ADD A 27)
002C	instruction	002C	STA A 0026 (wrong, should be STA A 28)
002D	instruction	002D	instruction

microbiography

Considering how little has been heard about Signetics in personal computing circles, the 2650 is a surprisingly powerful chip.

UP UNTIL 1975, Signetics was a fairly well respected IC manufacturer, especially in the area of bipolar circuitry. However, slipping behind in the field of MOS technology resulted in financial problems, and in mid '75 the company was acquired by Philips NV. The resultant sharing of marketing power and technology interchange has and will no doubt will result in benefits to both organizations.

2650

The 2650, then, may be considered as Philips' entry into the microprocessor field, and they're looking for some pretty hot action around this product. It is certainly no run of the mill mpu, and incorporates a number of very nice unusual features. Heavy emphasis has been placed on interface-ability. In fact, while other manufacturers almost universally released their XXX mpu PLUS support chips (without such support using the XXX is difficult), Signetics originally brought out the 2650 all by itself, for use with standard memories, latches, buffers, etc.

In addition, much attention appears to have been paid to "minimum" and small configurations. Indeed, the 2650 would look very good in a controller application, unlike some of the mpus which show up in big systems, but are not in their element as simple controllers.

HARDWARE

The 2650 fits into the set of specifications generally expected of a "third generation" mpu. On the hardware side this includes single +5V supply at 100mA, and single phase TTL clock of frequency DC to 1.25MHz, ie, the 2650 is all static logic. The instruction execution times range from 4.8 to 9.6 us

The block diagram and pinouts are shown in Fig. 1 and 2 respectively. The internal organization may be roughly divided into the three sections: Control, Addressing, and Data processing.

Control. In accordance with the emphasis on ease of interfacing, input and output to/from the mpu is asynchronous, which is to say external chips are not driven from the same clock that runs the mpu. Two control lines handle this function. OPREQ informs external devices that all information from the mpu is valid, and ready for action. When the memory, (or whatever it might be) has done its thing, it signals back via OPACK that it has finished.

Other control inputs include tristating controls for data and address buses, and a "pause" input. Control outputs are a memory/IO select line, read/write, write pulse, and run/wait indication. There are also interrupt request and acknowledge lines.

Addressing. The 2650 address bus is 15 lines wide allowing 32k of memory space. Referring to Fig. 1 again, one advantageous feature is the built in address adder which allows fast indexed addressing. In addition there is an 8 level subroutine return address stack.

Data processing. The main items required for this function are registers for storing data and an arithmetic logic unit for processing it. In the 2650 there are 7 data registers, but only four are in active use at one time. This is best visualized by numbering them R0, R1a, R2a, R3a, R1b, R2b, R3b. Those in use are then R0 plus the "a" series or the "b" series. Note that there is no accumulator as such, since in most instructions any register (of the four in use) can be the source or destination register. R0 would however, appear to be used in a more accumulator like fashion, with the other registers used as loop counters, index registers and so forth.

The "program" status word contains 14 bits (in 2 bytes) of information with the following functions. "Register bank select" determines which set of registers are in use, "Carry", "Logical/Arithmetic Compare", "Overflow", "With/Without Carry", "Interdigit Carry", are all used in processing data. A two bit condition code is provided which generally gives >/0/< or +/0/− based on instruction results.

The PSW also includes the stack pointer indicating the level of the subroutine stack in use, and the "sense" and "flag" I/O bits, to be dealt with in the next section.

The two PSW bytes may each be read from or written into.

INPUT/OUTPUT

Many forms of I/O are available, so that external circuit sophistication need only match requirements.

The very simplest I/O is the Flag/Sense output/input line pair. Flag may be set to 1 or 0 by writing into one of the PSW registers, while sense can be read similarly. With only two gates attached as buffers we have a Teletype interface!

The next level is the simplest parallel I/O scheme. Provision is made for a single 8 bit I/O port to be read from/written to in conjunction with any one of the four active mpu registers. This port would have a control and a data register. The mpu communicates with the port via the data bus, Memory/IO select line and upper two

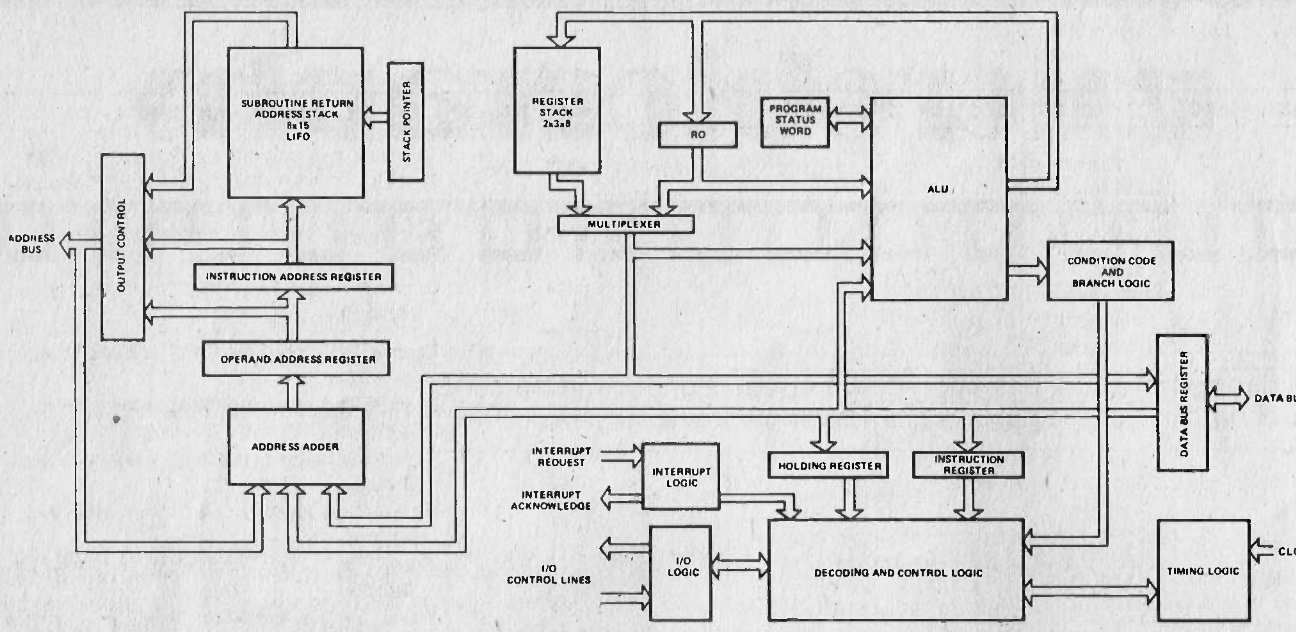


Fig. 1. Block diagram of 2650 internals.

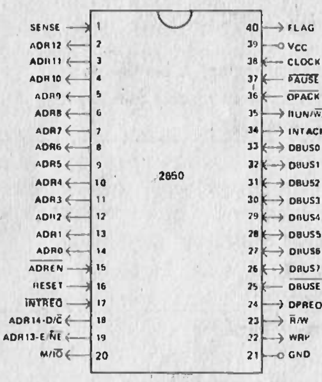


Fig. 2. Signetics 2650 pinouts.

address bits which select this mode of I/O and also tell whether the information on the data bus is data or control.

The next level of sophistication is the "Extended" I/O mode. In this case the same instruction is used, plus one byte to select one of 256 I/O port locations. Then the same lines plus 8 of the address lines to actually activate the appropriate I/O device.

Finally, one can always use I/O devices that masquerade as memory, which the mpu and programmer treat like any other memory location.

SOFTWARE

The instruction set, shown in Fig. 3, is a fairly standard collection. Some of the interesting features are the conditionals, and especially the addressing modes.

As previously mentioned, the two bit condition code determined by the result of some previous instruction will in one of three states. Each of the conditional instructions allows the programmer to include in the instruction a two bit code. Then, depending on the instruction the conditional action is, or is not taken if the programmer's code matches the condition code.

ADDRESSING MODES

Again, a respectable set of addressing modes are provided, register addressing, Immediate, Relative, Absolute, the usual favorites.

In addition there are Indirect and Indexed modes. Indirect addressing works as follows: the instruction op code is followed by two bytes which form the address at which to find the address of the operand. Indexed mode means that the two bytes following the op code, added to the contents of one of the mpu registers (selected by part of the op code) equals the address of the operand. But here's the real heart-winner, auto increment and auto decrement, which Signetics thoughtfully included. This mode is the same as indexed, but will automatically add (or subtract) 1 to/from the index register.

Suppose you've a table of data to be

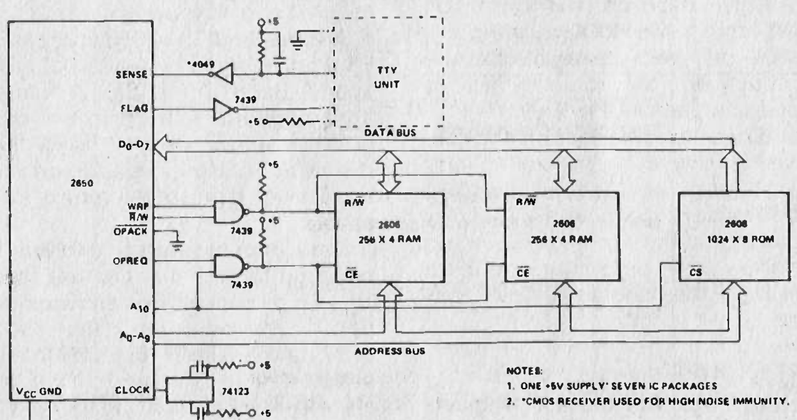


Fig. 4. A minimal system based on the 2650 and currently available parts.

Fig. 3. Instructions available on the 2650.

	MNEMONIC		DESCRIPTION OF OPERATION
LOAD/STORE	LOD	Z	Load Register Zero
		I	Load Immediate
		R	Load Relative
		A	Load Absolute
	STR	Z	Store Register Zero (r ≠ 0)
		R	Store Relative
		A	Store Absolute
ARITHMETIC	ADD	Z	Add to Register Zero w/wo Carry
		I	Add Immediate w/wo Carry
		R	Add Relative w/wo Carry
		A	Add Absolute w/wo Carry
	SUB	Z	Subtract from Register Zero w/wo Borrow
		I	Subtract Immediate w/wo Borrow
		R	Subtract Relative w/wo Borrow
		A	Subtract Absolute w/wo Borrow
	DAR		Decimal Adjust Register
LOGICAL	AND	Z	AND to Register Zero (r ≠ 0)
		I	AND Immediate
		R	AND Relative
		A	AND Absolute
	IOR	Z	Inclusive OR to Register Zero
		I	Inclusive OR Immediate
		R	Inclusive OR Relative
		A	Inclusive OR Absolute
	EOR	Z	Exclusive OR to Register Zero
		I	Exclusive OR Immediate
		R	Exclusive OR Relative
		A	Exclusive OR Absolute
COMPARE	COM	Z	Compare to Register Zero Arithmetic/Logical
		I	Compare Immediate Arithmetic/Logical
		R	Compare Relative Arithmetic/Logical
		A	Compare Absolute Artihmetic/Logical
ROTATE	RRR		Rotate Register Right w/wo Carry
	RRL		Rotate Register Left w/wo Carry
BRANCH	BCT	R	Branch On Condition True Relative
		A	Branch On Condition True Absolute
	BCF	R	Branch On Condition False Relative
		A	Branch On Condition False Absolute
	BRN	R	Branch On Register Non-Zero Relative
		A	Branch On Register Non-Zero Absolute
	BIR	R	Branch On Incrementing Register Relative
		A	Branch On Incrementing Register Absolute

	MNEMONIC		DESCRIPTION OF OPERATION
BRANCH	BDR	R	Branch On Decrementing Register Relative
		A	Branch On Decrementing Register Absolute
	ZBRR		Zero Branch Relative, Unconditional
	BXA		Branch Indexed Absolute, Unconditonal (Note 5)
SUBROUTINE BRANCH/RETURN	BST	R	Branch To Subroutine On Condition True, Relative
		A	Branch To Subroutine On Condition True, Absolute
	BSF	R	Branch To Subroutine On Condition False, Relative
		A	Branch To Subroutine On Condition False, Absolute
	BSN	R	Branch To Subroutine On Non-Zero Register, Relative
		A	Branch To Subroutine On Non-Zero Register, Absolute
	ZBSR		Zero Branch To Subroutine Relative, Unconditional
	BSXA		Branch To Subroutine, Indexed, Absolute Unconditional (Note 5)
	RET	C	Return From Subroutine, Conditional
		E	Return From Subroutine and Enable Interrupt, Conditional
INPUT/OUTPUT	WRTD		Write Data
	REDD		Read Data
	WRTC		Write Control
	REDC		Read Control
	WRTE		Write Extended
	REDE		Read Extended
MISC.	HALT		Halt, Enter Wait State
	NOP		No Operation
	TMI		Test Under Mask Immediate
PROGRAM STATUS	LPS	U	Load Program Status, Upper
		L	Load Program Status, Lower
	SPS	U	Store Program Status, Upper
		L	Store Program Status, Lower
	CPS	U	Clear Program Status, Upper, Masked
		L	Clear Program Status, Lower, Masked
	PPS	U	Preset Program Status, Upper, Masked
		L	Preset Program Status, Lower, Masked
	TPS	U	Test Program Status, Upper, Masked
		L	Test Program Status, Lower, Masked

processed, starting at location L. The following miniprogram shows how helpful auto increment can be:

Index Reg=0
Start Loop
Get byte of data from location L + Index Reg and automatically increment Index Reg.
Process
End loop

Auto increment packs a lot of work into one statement, possibly resulting in large savings in time and memory space.

SUPPORT CHIPS

Already available are two serial communications ICS.

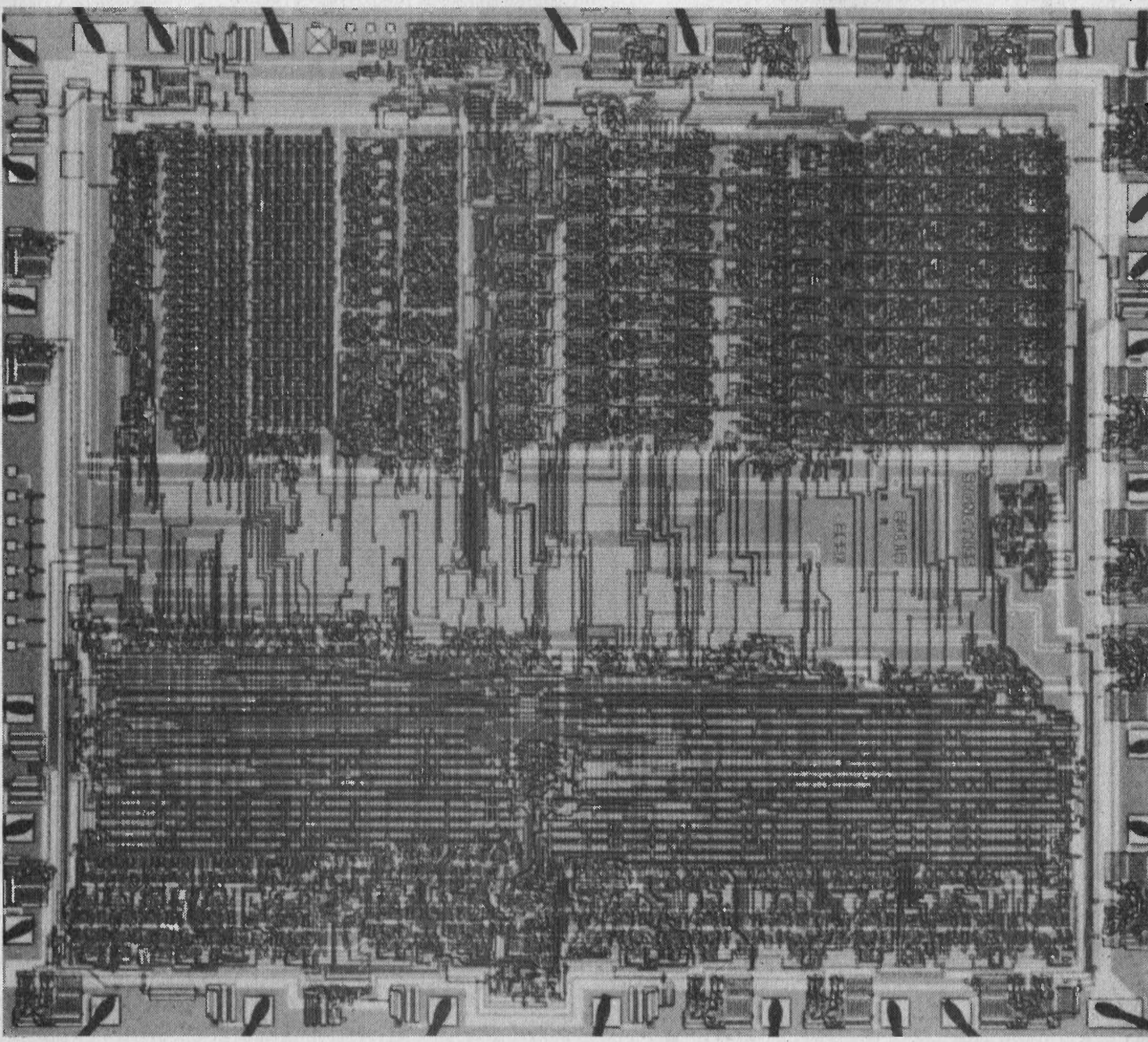
2651: Programmable Communications Interface (PCI) is a combined USART and baud rate generator (50 to 19.2k bps). It includes modem control and support of IBM's BISYNC protocol, asynchronous echo and self testing.

2652: Multi-Protocol Communications Controller (MPCC) is able to receive, format and transmit serial data in Synchronous Data Link Control and other serial systems, at speeds up to 500k bps.

To be available in mid 1978 are:
2655: Peripheral Programmable Interface (PPI), a general purpose parallel I/O device having 3 8-bit ports.
2656: System Memory Interface (SMI) is the other half of a 2650 2-chip system and contains 2k bytes ROM, 128 bytes RAM plus an 8 bit I/O port and timer function.

Signetics integrated circuits are available from Hamilton Avnet in Toronto, Ottawa and Montreal and CESCO in those same cities plus Quebec. Check with a local Philips outlet for Signetics distributors in other areas.

Inside view of the 2651 chip



Horse Jumping For Calculators

The first in our series of calculator programs, this one for the Sinclair Programmable was submitted by Mr. P. Cornes. A flow chart is given to aid owners of different machines in writing their own versions.

Object — To simulate a show jumping course in such a way that:—

1. The player enters a guess as to how many strides of acceleration he thinks will be required by a horse to clear a fence H feet high.
2. The player is given an indication of right and wrong guesses.
3. The player's total score is made available to him at the end of the game.
4. The player's score is made dependent on the value of his guesses and on his successfully clearing the fences.

The biggest problem with this program was trying to find a realistic relationship between the number of accelerating strides input and the height that these strides would enable a horse to jump. The following curve shows the sort of relationship that is required.

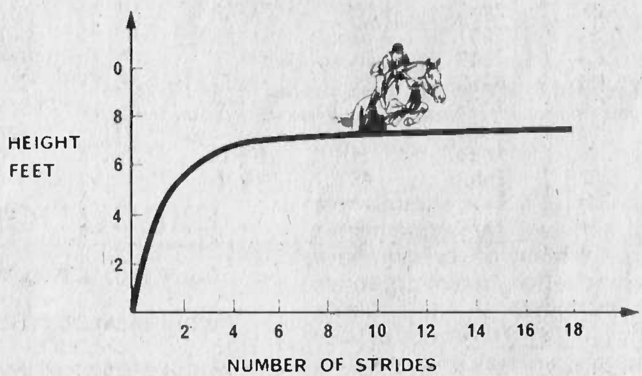
As you can see from the curve the extra height that the horse can jump decreases as the number of strides increases, such that after a certain point no increase in height is gained by increasing the number of strides. This is the sort of curve you would expect in reality. I have simulated this curve by using the arctan function. The tan of an angle can take any value between zero and infinity so the arctan of any number between zero and infinity has a radian value between 0 and 1.57 and you will find that taking the arctan of any number greater than about twenty gives approximately 1.57 as an answer. The only thing to be done now is to scale the arctan values up to give a reasonable range of heights, to do this we multiply by five.

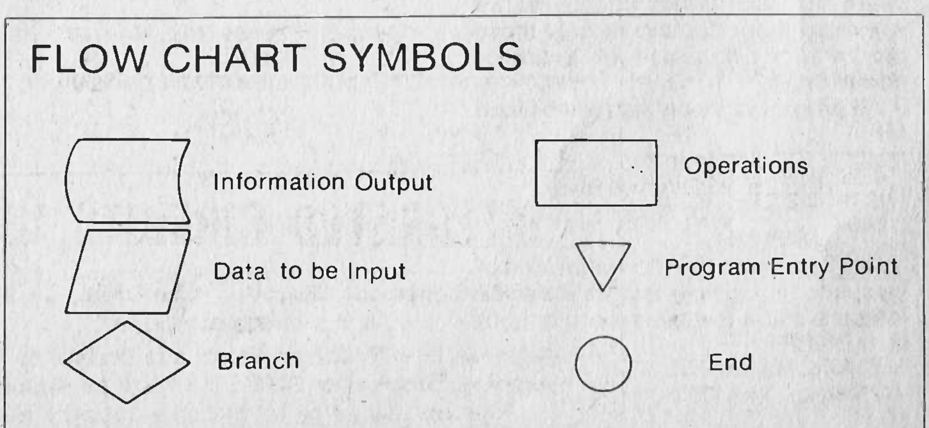
Looking at the plan of the course you will see the path connecting the fifteen fences together. The number alongside each fence is its height (H) and the numbers on the paths between the fences are the distances in strides to each fence. If you input these numbers as your guesses then you are guaranteed to clear the fences but you will find that it is possible to clear most of the fences in less strides than shown.

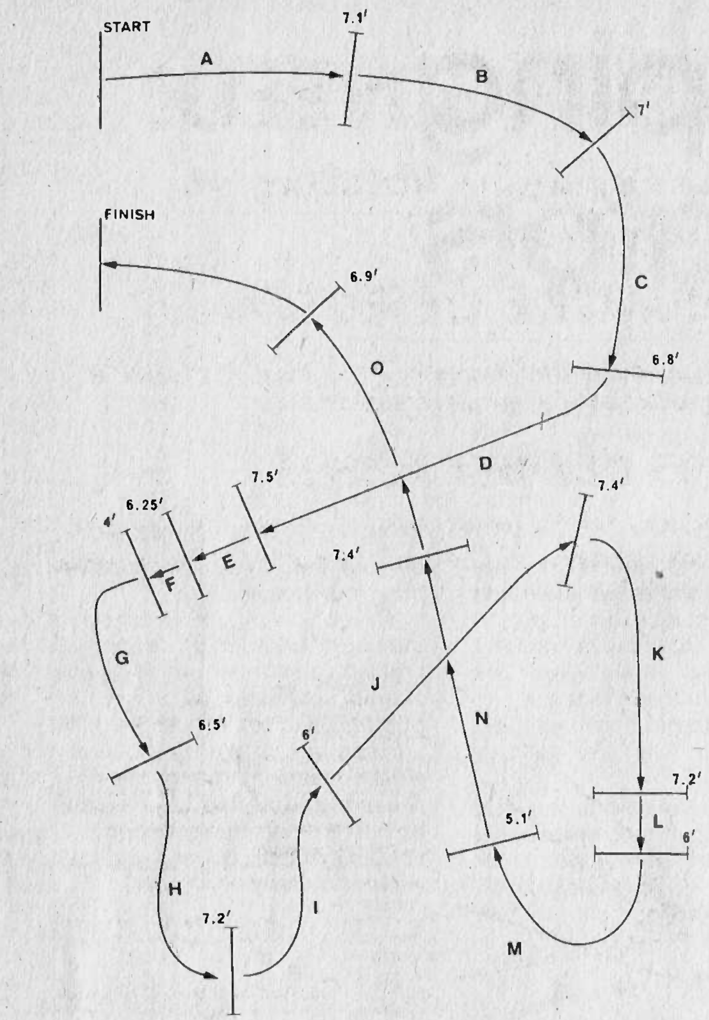
Your sco·e is calculated by totalling all your guesses round the course and by adding a penalty of nine points for each fence you do not clear. You should consider yourself to be disqualified if you knock down more than four fences.

If you clear every fence in the minimum number of strides you will end with a score of ninety-five but you should consider a score of one hundred and ten or less as good.

When you master this course it is a simple matter to change the heights of the fences and this creates your own course but remember that no fence should exceed 7.5 feet in height or you will not clear it.

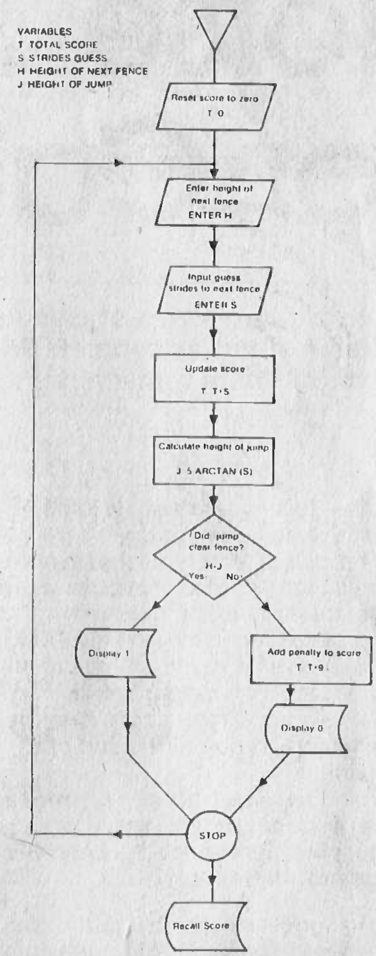






Above: a suggested course for the horse race game. All the fence heights are given in feet, guess the number of strides between the fences.

FLOW CHART



SOFTSPOT is ETI's programmable calculator software department. We know there are many of you who have gone to a lot of effort to write routines for your machines — how about sharing the fun. Send us a copy of your pet program, preferably with flow chart. To make things interesting we will restrict our choices to only those programs making use of loops or conditionals.

All programs we publish will be paid for.

Mail to: ETI Softspot
Unit 6, 25 Overlea Blvd.,
TORONTO, Ontario
M4H 1B1

Don't forget to mention what kind of calculator you use — and we'd also be interested to know where you bought it.

EXECUTION

0/▲▼/sto/▲▼/▲▼/go to/0/0/

input H fence 1/RUN

input strides/ RUN/right-wrong

input H fence 2/RUN/

input strides/ RUN/right-wrong

●

●

●

input last H/RUN

input strides/ RUN/right-wrong

▲▼/Rcl/score

PROGRAM

	–	F	00
	(	6	01
INPUT STRIDES	Stop	0	02
UPDATE SCORE WITH STRIDES INPUT	▼	A	03
	MEx	5	04
	+	E	05
	Rcl	5	06
	=	–	07
	▼	A	08
	MEx	5	09
CALCULATE HEIGHT FROM STRIDES INPUT	▼	A	10
	arctan	9	11
	x	.	12
	#	3	13
	5	5	14
	=	–	15
	)	6	16
	=	–	17
VALUE –VE FOR FENCE CLEARED +VE FOR FENCE DOWN	▼	A	18
	GIN	1	19
	3	3	20
	2	2	21
UPDATE SCORE WITH FENCE DOWN PENALTY	#	3	22
	9	9	23
	+	E	24
	Rcl	5	25
	=	–	26
	Sto	2	27
FENCE CLEARED/ DOWN DISPLAY 1 = CLEARED 0 = DOWN	#	3	28
	1	1	29
	–	F	30
	+	E	31
	#	3	32
	1	1	33
	=	–	34
INPUT HEIGHT	Stop	0	35

tech-tips

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ELECTRONIC 'SPIROGRAPH'

A. Sharp.

The circuit will generate 'Spirograph' patterns on a conventional oscilliscope. The circuit consists of two sinewave generators followed by allpass filters which we use to phase shift the input signals by 90°. Applying a sinewave to the y input gives a circular trace. If a second set of sin and cos signals are mixed in, a 'Spirograph' pattern is obtained. A block diagram of the system is shown in Fig 1.

RV1 is a balance control which varies the contribution of each oscillator to the pattern without affecting the size, so that once set up there is no need to readjust the gain controls on the oscilliscope. This type of control can only be used if the oscillators have a low impedance output.

SW1 is a reversing switch which has the effect of turning the pattern inside out.

An exi ting sinewave oscillator can of course be used and the 60 Hz line could be employed (attenuated to about 2 V RMS from a low voltage transformer secondary) as the fixed oscillator. However flickering is a problem with lower frequencies (complex patterns requiring four or more cycles to complete will flicker at about 10 Hz using the line frequency as an oscillator. I found 150 Hz to be a good compromise (higher frequencies require more critical tuning).

The allpass filter is recommended for phase splitting as it has a unity gain for all frequencies and settings of RV5.

First connect the y input of the scope to the output of an oscillator and adjust RV2 until a two volt RMS sinewave is obtained, repeat for second oscillator. Then connect up the x and y inputs as shown in Fig 1, turn the balance control to one end so as to look at the output of the fixed oscillator then adjust the 100 k pot until a circle is obtained (with suitable x and y gains). Now put the balance control in the middle and adjust the frequency controls until a stable pattern is produced. SW1 and RV1 the balance control can be used to alter the nature of the pattern without affecting its overall size, stability or symetry. Adjust RV5, the phase control (following the variable oscillator) for symmetry. — Have fun!

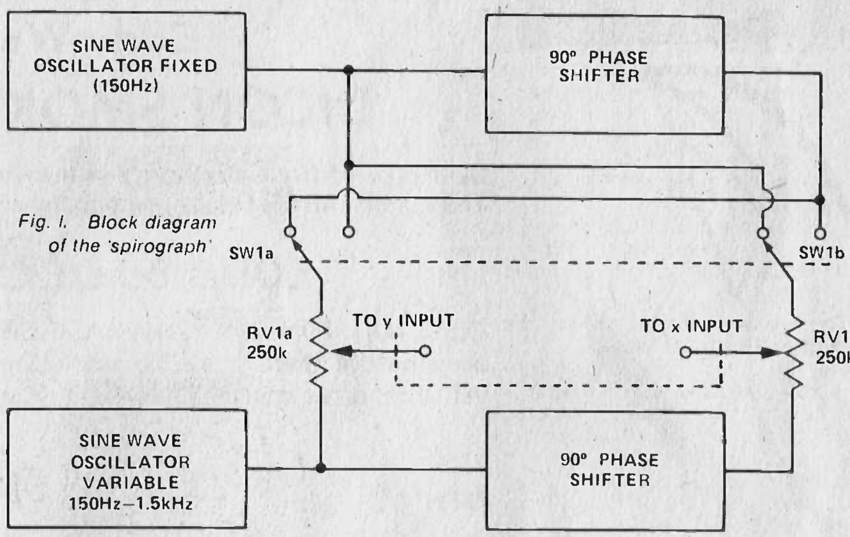


Fig. I. Block diagram of the 'spirograph'

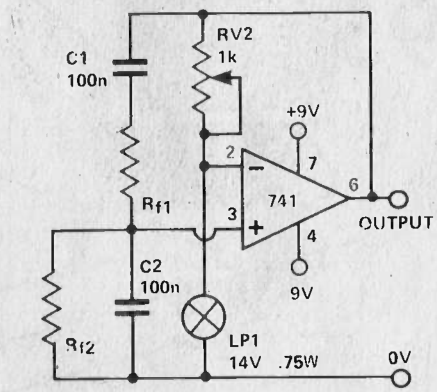


Fig. 2(a) suitable oscillator for the 'spirograph'

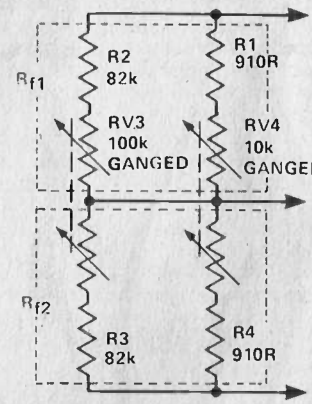


Fig. 2(b) Arrangement to give fine control of the frequency of the oscillator shown in Fig. 2(a). For 150 Hz fixed frequency use Rf¹=Rf²=10k

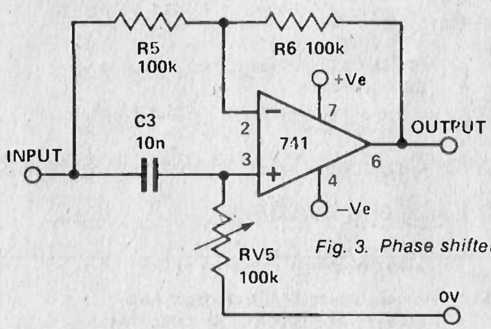


Fig. 3. Phase shifter circuit for use in 'spirograph' circuit.

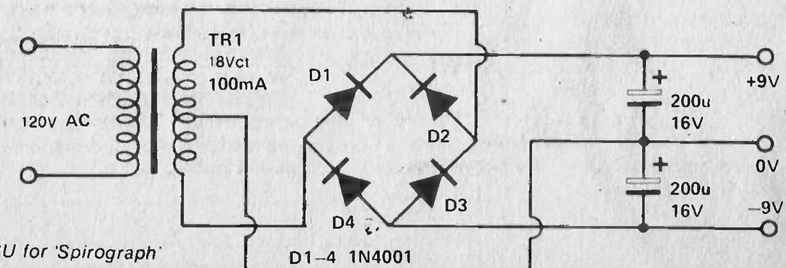
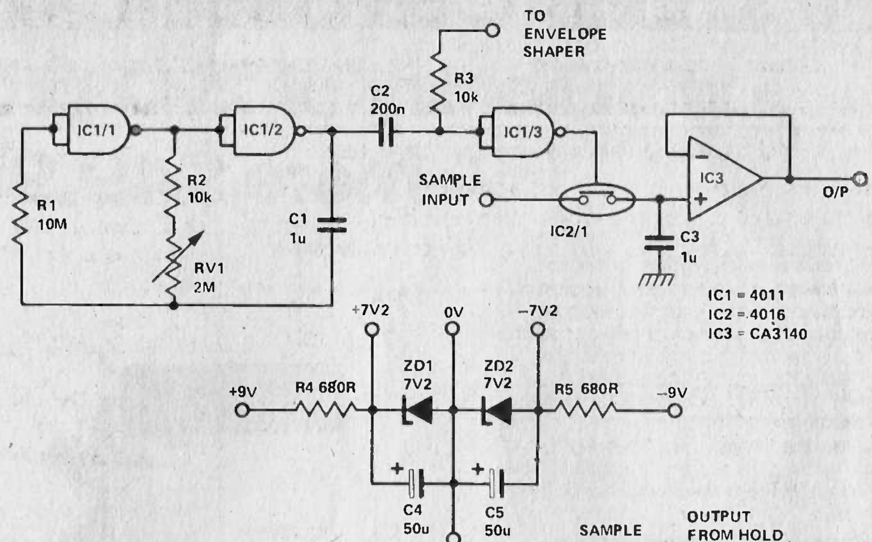
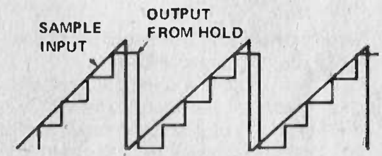


Fig. 4. PSU for 'Spirograph'

SAMPLE AND HOLD FOR MUSIC SYNTHESIZERS

L. Robinson





Sample and hold is a useful effect for use with music synthesizers and consists of 'sampling' an input voltage function such as a waveform for a very short time and then 'holding' it at this selected voltage level for the duration of the clock period. This voltage is then used to control the frequency of a voltage controlled oscillator, filter etc.

It is therefore possible to produce random or repeating sound patterns by varying the input waveform and frequency, pink noise can be used as a sample source to create authentic random voltages.

The circuit shown is much simpler than previously designed sample and hold circuits, this is possible by the use of CMOS technology. The clock oscillator is a standard CMOS square wave oscillator as found in RCA application notes, and this is used to provide a variable frequency rate from 0.2 Hz to 45 Hz. The output then goes to the synthesizer envelope shaper which should be of the ADSR type for maximum effect. The clock output also goes into a monostable which produces an output pulse of approximately 20 mS which opens the 4016 analog gate for this period. The voltage input is therefore sampled and the value of the amplitude at this point of the waveform is remembered by the high input impedance (10^{12} Ohms) CA3140 voltage follower. This output is then used to control the VCO etc. The oscillator and monostable can be constructed from either a CMOS 4001 or 4069, ensuring that unused pins are connected to the high or low power supply line via a 1k resistor. The input waveform to the analogue switch can have an amplitude of ± 7 V maximum. If a FET was used as the gate, it would only respond to negative voltages, so the more expensive analogue switch is used for this reason. The total cost of the circuit, including the ± 7V supply, is less than $5.00.

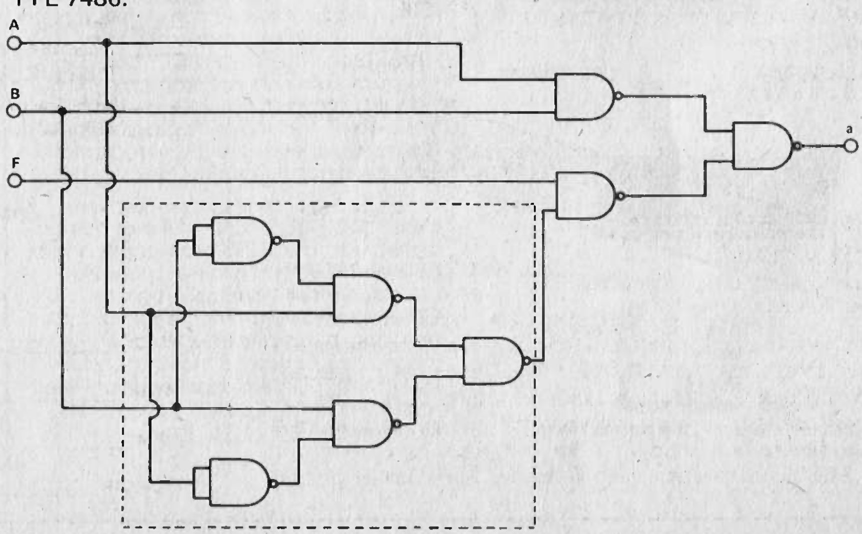
PROGRAMMABLE GATE

P. Mead

The Programmable Gate is a gate which converts an AND gate to an OR gate by applying a logic '1' on the function input.

The logic design uses 8 x 2 input NAND gates. The number of gates may be reduced by replacing the 5 NAND gates enclosed by the dotted line, with a 2 input exclusive OR, such as the TTL 7486.

FUNCTION INPUT	INPUTS A	INPUTS B	OUTPUT	
0	0	0	0	AND FUNCTION
0	0	1	0	
0	1	0	0	
0	1	1	1	
1	0	0	0	OR FUNCTION
1	0	1	1	
1	1	0	1	
1	1	1	1	



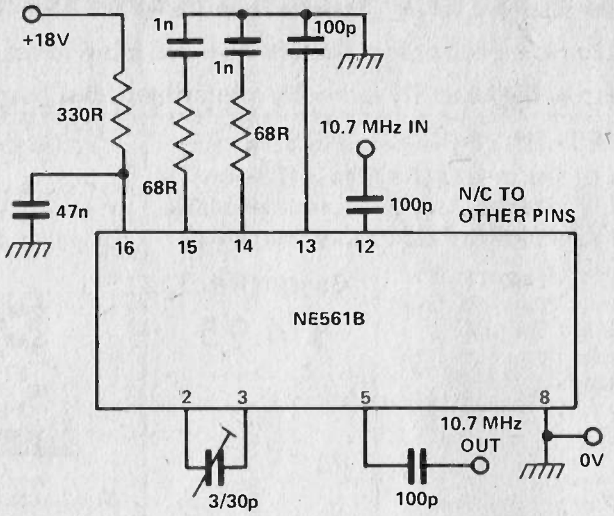
FM SIGNAL CONDITIONER

R. N. Soar

As an alternative to an extra IF stage in an FM tuner, a PLL IC can be used as a signal conditioner. The VCO of the PLL tracks the input signal to provide a less noisy and stronger signal at its output.

The circuit shown is built around the Signetics NE561B PLL. The only thing necessary is adjustment of the 3/30 p trimmer which sets the VCO's centre frequency to 10.7 MHz.

The circuit should be effectively shielded to avoid interaction with the FM front end that provides the circuit's input.



MINIMIZING CONNECTIONS

M. T. Clarke

Anyone who has connected together memory ICs may well be appalled at the number of connections, especially those which simply parallel the IC pins.

Realizing that the address pin designations are purely notational means that address lines can be rearranged before they reach an IC, as convenient. This eases considerably PCB design.

An example is shown where connection of 4K dynamic RAMs (2604) was undertaken on Vero-board. The copper tracks provide all address connections for every alternate IC without any wiring from the surrounding ICs (this saved almost 100 connections on a 4K x 16 board).

Dynamic RAMs require segregating the row and column addresses, but within each they can be freely mixed.