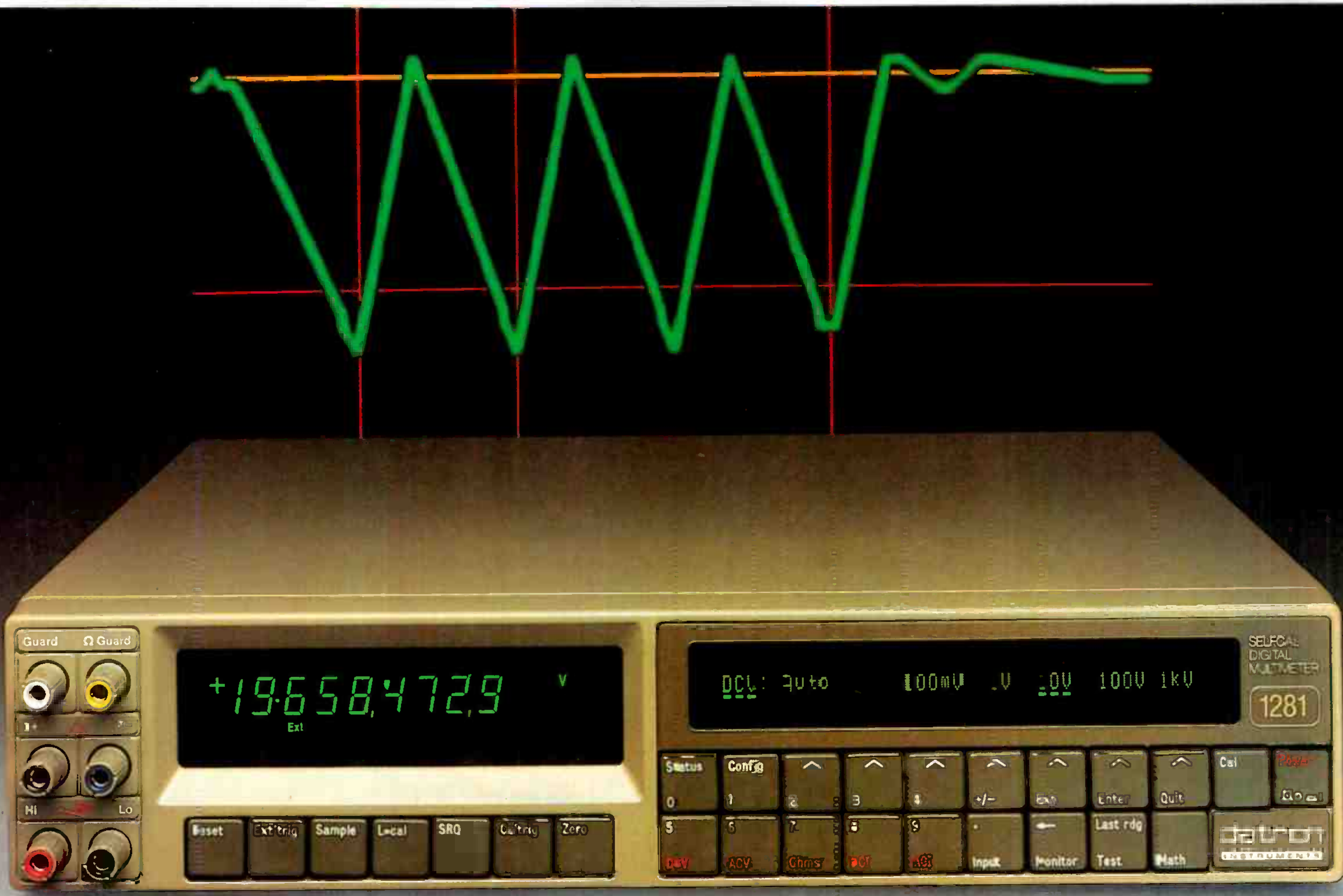



# Self-calibrating digital multimeter

**A method of internal calibration together with innovative circuit design produces a higher accuracy than previous designs.**

HAL CHENHALL

Instruments for calibration and standards purposes need to be an order of magnitude better than the equipment to be calibrated. Precision digital multimeters already take full advantage of the inherent qualities of the critical components that define the instruments' performance, and can therefore achieve high accuracy. Calibration meters consequently need to provide even greater accuracy.

New circuitry and a method of internal calibration, which can be linked to external standards, are provided in the Datron Instruments' 1281 digital multimeter to produce very high performance across teh complete range of its functions. It is considerably more accurate than known previous designs while retaining a compact size and without being over expensive. The instrument is particularly intended for use in standards and calibration labororatories and is designed to be easy to use, despite its high level of accuracy and its wide range of functions.

There are four main design areas where significant advances have contributed to the overall instrument performance – analogue to digital converter, master reference, a.c. measurement, and self-calibration.

## ANALOGUE TO DIGITAL CONVERTER

A multi-ramp, multi-cycle integrator has been developed to provide the necessary performance for an 8½ digit instrument. The main elements inherited from the quad-slope technique used in Datron's previous Autocal range of instruments are the use of feedforward bias signals to overcome problems at zero due to noise in the null detector, and the use of two reference values $V_{ref}$ and $V_{ref}/16$ (coarse and fine ramps) to provide speed and accuracy, Fig. 1. The additional features which provide the improvements include:

- Use of multiple cycles which means that a smaller integrator capacitor can be used, reducing dielectric absorption effects and improving linearity.
- Applying signal and reference inputs at the same time rather than separately during multi-cycle conversions, improving conversion speed.
- Using both positive and negative references an equal number of times for every conversion, ensuring that reference switching errors are constant and can be removed by an integral autozero cycle.
- Using a custom asic for the a-to-d conversion control, providing flexibility in programming integration times and resolution.
- A dynamic autozero system avoids the need for the more common sample and hold type of autozero circuit, which can become saturated at overload and slow down overload recovery.

When the a-to-d converter is not actually converting a signal, it goes into a reset or 'dynamic autozero' mode. This maintains the output of the already low drift integrator near zero by applying small amounts of $-V_{ref}/256$ and then nulling it with $+V_{ref}/256$. Because this reset cycle is short (50µs) and occurs at least once before each conversion, it avoids the need for the random interruptions for zero corrections found in less sophisticated conversion techniques.

The output of the integrator during a dynamic autozero cycle is shown in Fig. 2. Initially, zero signal is applied to both signal and reference inputs for a set period. Then, $-V_{ref}/256$ is applied to the reference input and the output of the integrator 'ramps up' and passes through null. After the null, $-V_{ref}/256$ is applied for a fixed period so that the integrator overshoots. Zero is again applied for a short period to both signal and reference inputs, to ensure that both references are not accidentally applied to the integrator simultaneously. Then with $+V_{ref}/256$ switched to the reference input, the integrator ramps down towards and beyond null and for a predetermined period.

The cycle is then repeated, maintaining the integrator output near zero. At the end of each cycle the integrator is in exactly the same place, even though the integrator may drift between resets.

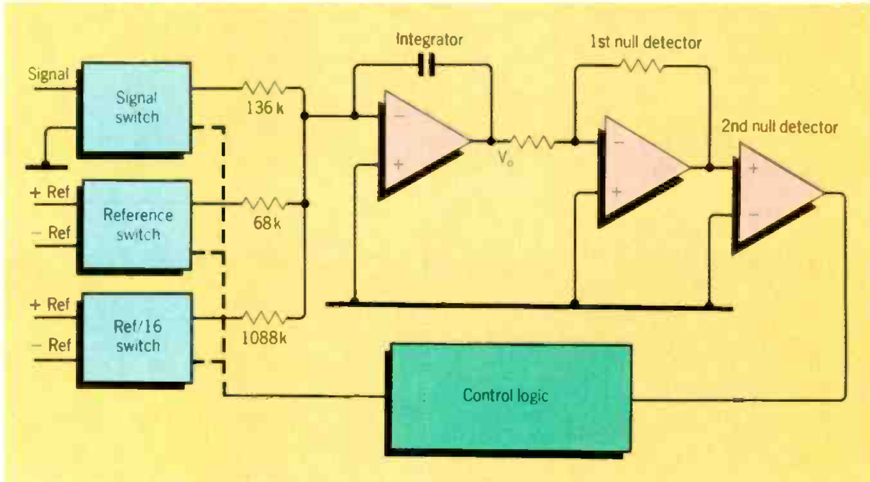



Fig.1. Multiple references are used with the multi-slope analogue-to-digital converter in order to increase conversion speed without losing accuracy in detecting the final null.

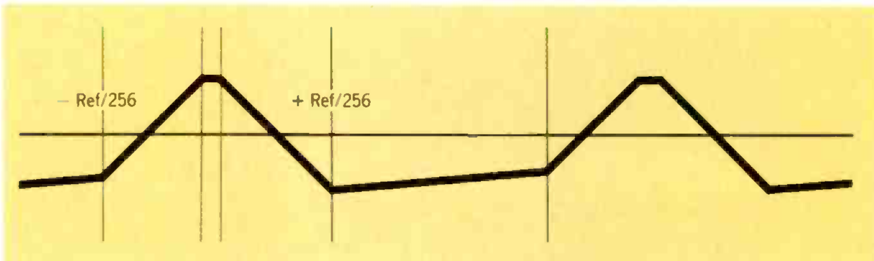



Fig.2. A dynamic autozero cycle is used by the analogue-to-digital converter to maintain the integrator output near zero when no conversion is being done. The end of the +Ref/256 period is used as both the starting and finishing point for every conversion, ensuring an exact charge balance.

## SINGLE-CYCLE CONVERSION

Depending on read rate and resolution requirements, the converter can make either single-cycle conversions. Some of the key elements of the conversion technique are well illustrated by considering the single-cycle conversion of Fig. 3.

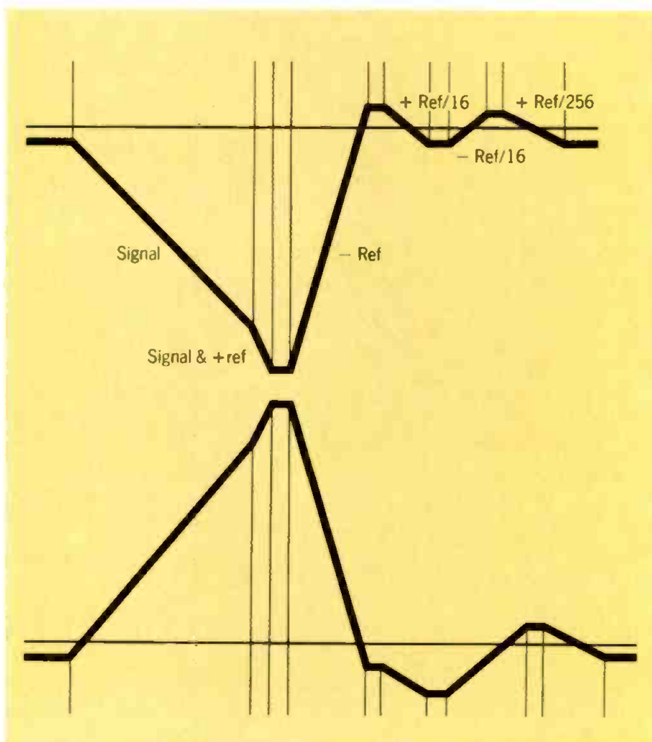
On receipt of a reading conversion command, the last reset cycle is completed within a fixed delay of 50µs, and then the signal is applied to the signal input. The integrator ramps up and after a fixed period a feedforward bias of appropriate polarity ($+V_{ref}$) is fed to the reference input while the signal is still being applied. Next, zero is applied to both the signal and reference inputs for a fixed delay to ensure that the system does not attempt to switch in both references at the same period where $-V_{ref}$ is applied to the reference input. Eventually the integrator crosses null and is allowed to overshoot to synchronize to a clock signal. This represents the end of the coarse conversion period and the integrator then configures itself for the final, more accurate or 'fine', stages of the conversion.

Firstly, zero is applied to the signal and reference inputs to avoid the effects of any switching transients, followed by applying $+V_{ref}/16$ to the reference input for a fixed time. The polarity of reference used in the cycle at this point is chosen to ensure that the approach to zero for the final ramp is always made using the $+V_{ref}/256$ reference, irrespective of signal polarity. This overcomes any non-symmetry in null detector response times. After another dead period, $-V_{ref}/16$ is applied to the reference input so that the integrator heads back again to null and overshoots.

At the end of the final dead period, $+V_{ref}/256$ is applied to the reference input. This is the last part of the conversion and its final stages are identical to the end of the dynamic autozero cycle. In other words, the integrator output finished back exactly where it started, so that the charge from the signal has been exactly balanced by the charge from the various references. At the end of the conversion the a-to-d converter goes back into dynamic autozero mode and the reading data may be shifted out of the control circuits.

The sequence for a negative polarity signal is subtly different from that for the positive polarity signal described but the important fact is that for every conversion each reference is switched in and out once, and that the final ramp down is identical for both signal polarities. This means that any reference switching errors due to charge injec-

Fig.3. Conversions are subtly different for different polarity input signals. This ensures that the final slope to null is always made with +Ref/256, avoiding possible null detector hysteresis. At the end of each conversion the integrator output is exactly where it started from, which means that all charge from the unknown input has been balanced by the charge from the known references.



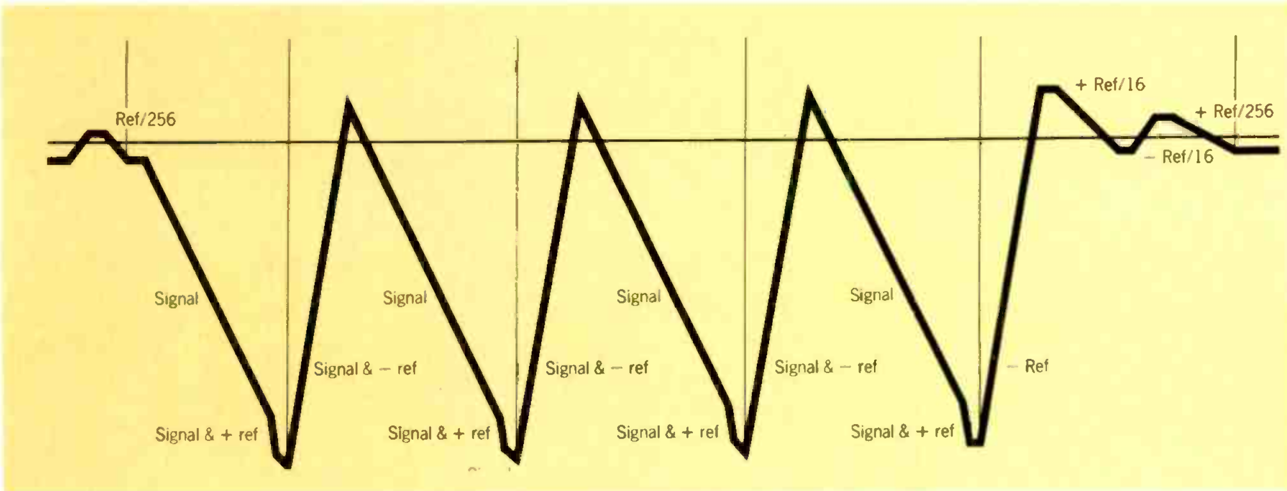



**Fig.4. Multiple cycle conversions allow the signal to be applied to the integrator for almost the entire conversion period, which improves the speed of conversion. It avoids the need for a large integrator capacitor and the associated dielectric absorption problems.**

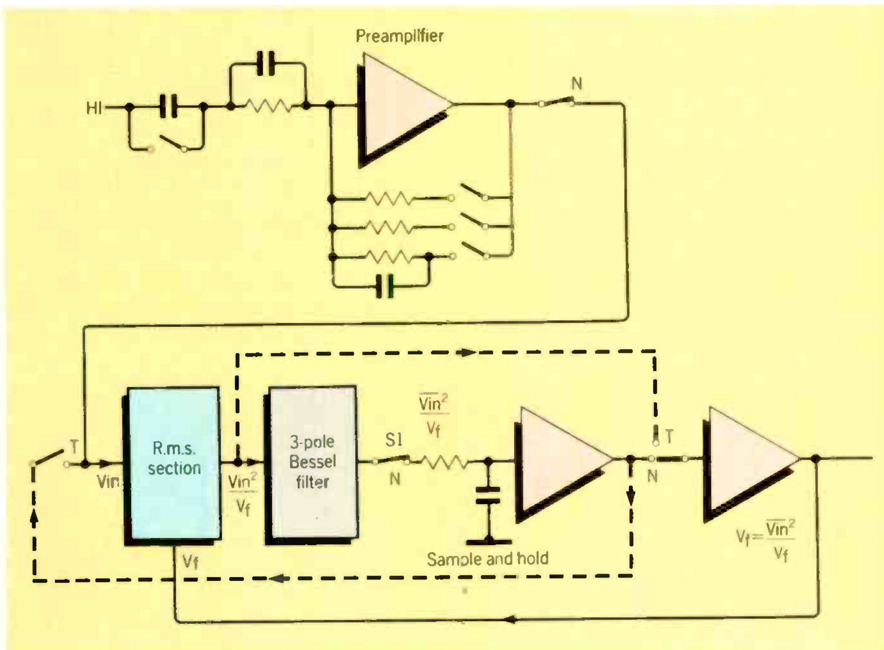



**Fig.5. An automatic ac/dc transfer feedback technique can be used to calibrate the gain of the electronic r.m.s. section for each reading. This is useful for removing time and temperature drift from this part of the circuit**

tion are balanced for all conversions, while any small null detector delay time errors and charge injection effects due to the final ramp are automatically removed by the dynamic autozero.

## MULTI-CYCLE CONVERSION

A multi-cycle design was chosen to give maximum flexibility to the available integration periods without forcing the need for a large integrating capacitor, which could have introduced greater dielectric absorption problems. Instead, a small integrator capacitor is used, and longer integrator periods are achieved by ramping up and down several times while specifically avoiding saturation of the integrator. In addition, the multi-cycle approach provides effective gain in the integrator, reducing the requirements placed on null detector sensitivity and making higher accuracy conversions easier to achieve.

One of the key features of this particular multiramp design is that, for all but the final ramp, the signal is applied continously and the various references are applied simultaneously with the signal at the appropriate times (Fig. 4). In other words, the integrator effectively ramps up and ramps down at the same time, which significantly reduces the time to take a reading.

Timing and counting considerations with this design are complex. Although the a-to-d converter always performs the same sequence, great flexibility of control is exercised over its performance through the use of programmable delay timers, a ramp timer and a counter for the number of ramps performed. All of these timers and counters are integrated into a custom asic which has a 32 bit control register programmed by the instrument's microprocessor via a special serial interface. The same serial loop is used to transmit the reading from the asic to the processor for calibration and display.

## MASTER REFERENCE

The reference used in the analogue to-digital conversions derived from two specially conditioned zener reference modules. Each reference module contains the reference device and its associated buffer circuits, all hermetically encapsulated together to ensure constant temperature across the module. The modules are stable to within ±2 p.p.m per √year, produce noise of less than 0.1p.p.m, and have temperature coefficients of better than 0.1p.p.m/°C. This temperature coefficient is held over a wide temperature span of 0 to 50°C, and the references exhibit negligible temperature shock hysteresis. The master reference is obtained by summing the outputs from both reference modules.

Extensive evaluation of the modules has resulted in a burn-in process which equates to an ageing of 1 year, reducing both infant mortalities and hysteresis effects. Following this process, the modules are checked over a temperature span of 0 to 70°C for temperature performance, and then monitored for long-term drift over a minimum period of three months.

## A.C. MEASUREMENT

The inverting preamp has to provide good flatness from d.c. to 1MHz and a minimum of offset voltage at its output to ensure good d.c.-coupled performance. A complex design is required to achieve this, using several gain elements in conjunction with each other.

The closed-loop gain is set by range resistors and capacitors. Because of the presence of stray capacitance around the preamp, the input and feedback resistors setting the low frequency gain have to be shunted by capacitors to compensate for this. At high frequency, it is these capacitors that determine the closed-loop gain. The feedback capacitance on each range is effectively trimmed at calibration using a ladder network digital-to-analogue converter driven from the microprocessor to control the channel resistance of fets in the gain-defining network. Extensive bootstrapping of components in the preamp feedback area reduces the effect of stray capacitance.

## ELECTRONIC R.M.S.

An electronic r.m.s. technique has the following advantages over designs on thermal techniques:

- Higher accuracy – the instrument achieves ±90p.p.m 1 year uncertainties. which is the best available in any commercial d.m.m.
- Faster response – the instrument can take high accuracy 6½ digit a.v. readings at a rate of one per second. about six times faster than other commercial designs.
- Wider dynamic range – the span from 100nV to 1000 r.m.s. can be covered in fewer ranges. saving cost and space. Each range can accept inputs from 1% of range to 200% of range.
- Good crest factor performance for non-sinusoidal signals (5:1 at full range. 10:1 at 25% of range).

The principles behind the r.m.s. conversion technique are shown in Fig. 5. With the instrument set to its 'normal' mode. the signal from the preamp is full-wave rectified by the combined operation of the rectifier and the log.amp. and taken as a current input to the input of the r.m.s. section. This is a current-operated device. whose output is unidirectional but peaky and converted to a voltage for smoothing by a three-pole Bessel filter.

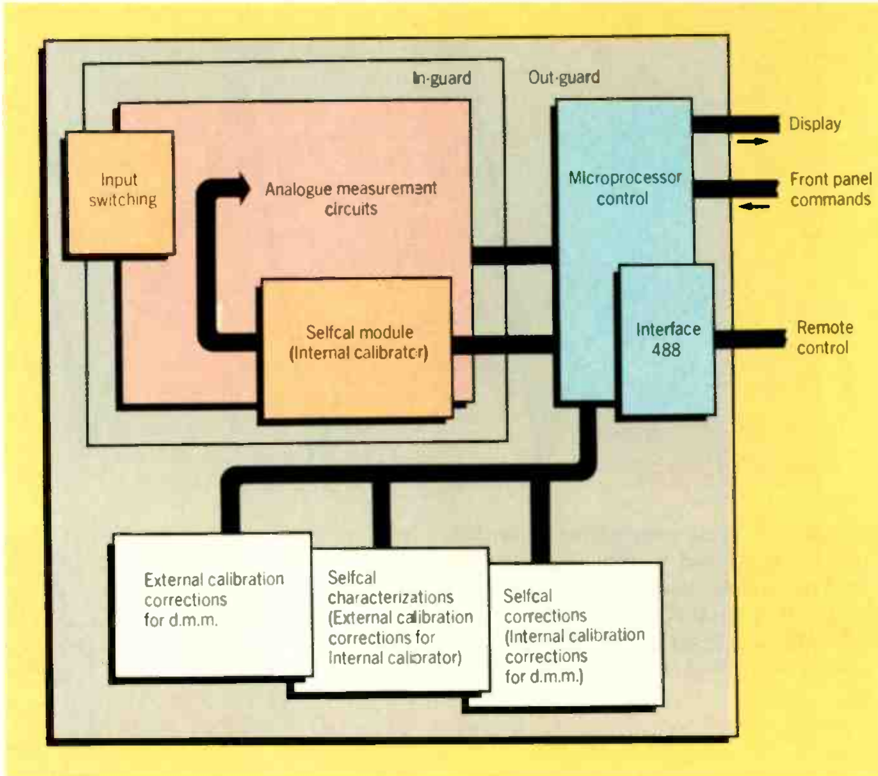



Fig.6. Separate calibration memories are used to store the external calibration corrections for both the d.m.m. and the internal calibrator. An additional calibration memory is used to store selfcal corrections derived during an internal calibration.

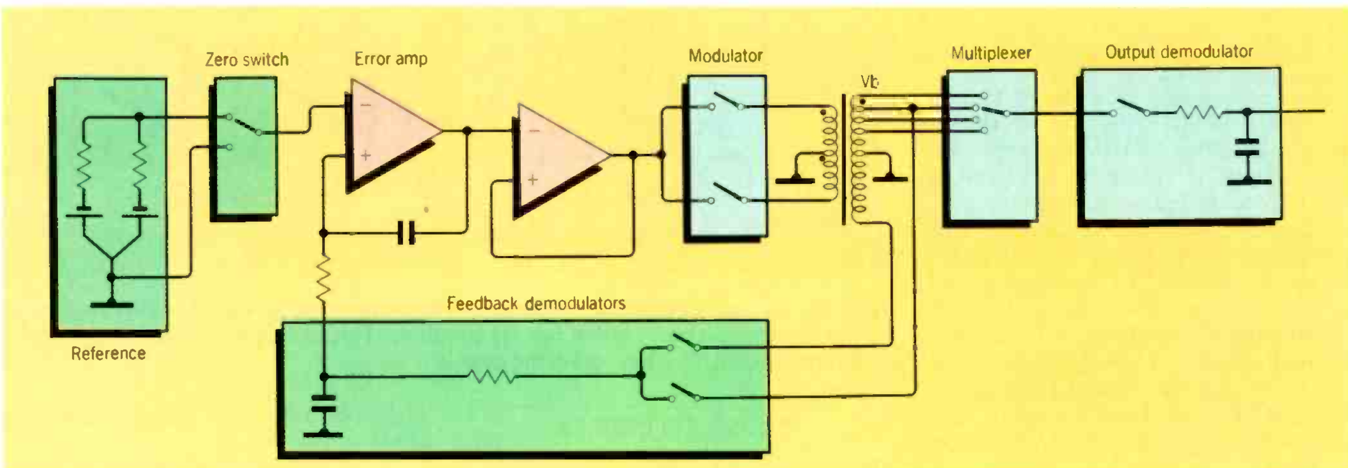



Fig.7. Different levels of d.c. calibration voltages are produced by modulating a d.c. reference and passing the resulting square wave through a transformer. The various output tappings are connected to a sample and hold to reconstitute a d.c. level. A feedback technique is used to ensure a perfect turns-ratio performance which is inherently stable.

The filter is chosen for its optimum settling time. and offers selectable configurations to permit operation down to 1Hz. A sample and hold with isolating buffer provides further one-pole filtering above a certain frequency. after which the smoothed signal is taken to an amplifying buffer which drives the instrument's analogue to digital converter. The output signal $V_f$ is equivalent to $\overline{V_{in}}^2/V_f$ by virtue of the action of the electronic r.m.s. section on its two inputs of $V_{in}$ and $V_f$. This means that $V_f = \sqrt{(\overline{V_{in}})^2}$, the r.m.s. of $V_{in}$.

## A.C.-TO-D.C. TRANSFER TECHNIQUE

The a.c. circuit employs a refinement on the basic technique which uses an a.c.-to-d.c. transfer mechanism to calibrate the gain of the r.m.s. converter.

Consider Fig. 5. with all swiches set to 'normal' mode (N). When a signal ($Y_0$) is passed through the r.m.s. converter. a d.c. equivalent value ($Y_1$) is produced. For an ideal converter $Y_0=Y_1$. but assuming the converter has a gain other than unity. then $GY_0=Y_1$. This gain factor G may drift with time and temperature. and the purpose of the a.c.-to-d.c. transfer technique is to remove these effects. This is achieved by setting all swiches to 'transfer' mode (T) and opening $S_1$. Signal $Y_1$ is sampled and held. and then fed back through the r.m.s. converter to obtain another value. $Y_2$. In this case. $GY_1=Y_2$ and as $Y_1$ and $Y_2$ are known. the true value of G can be determined. This can then be used to give a value for $Y_0$. corrected for the r.m.s. converter gain. i.e.

$$Y_0=Y_1/G=Y_1^2/Y_2.$$

The actual sequence used in the d.m.m. is more to overcome the problems associated with ripple on the measurements. When the reading $Y_1$ is taken. the a-to-d converter integration time can be arranged to smooth out any ripple on the signal emerging from the Bessel filter. However. as soon as the switch $S_1$ beyond the filter is opened. the sample and hold will potentially capture the peak of any ripple on the signal and so not be representative of the desired d.c. level ($Y_1$). An additional measurement is taken with only switch $S_1$ open (and all other switches set to N) to give a value $Y_3$ which is the correct value for the determination of $G(GY_3=Y_2)$.

The required calculation to correct for gain is therefore $Y_0=Y_1/G=Y_1Y_3/Y_2$.

**Spot frequency enhancements.** To enhance a.c. performance even further, each a.v. range can be spot calibrated at up to six independent user defined frequencies, such that, when the instrument is making measurements of signals at frequencies which lie within ±10% of these points, flatness errors are reduced improving accuracy to ±65p.p.m. for a whole year. In addition, the instrument has a reciprocal counter function designed into one of its custom asics which can display the frequency of an a.v. signal at the same item as its r.m.s. value is being shown on the main display.

## RESISTANCE AND CURRENT

The wide selection of floating current source ranges provided by the resistance function means that a variety of resistance measurement modes can be offered to suit many different application areas. For example, when operating in its normal mode, the instrument's current sources are optimized for low noise and best accuracy. However, where low compliance or low open-circuit voltages across the d.m.m's terminals are needed, a special low current mode can be selected. Applications where this can be useful include in-circuit measurement of components in parallel with diode junctions, or the measurement of temperature using platinum resistance thermometers, where the self-heating of the current passing through the resitive element may be important.

For applications where external thermal e.m.fs present measurement problems, a mode is provided where a zero reference reading is automatically taken with the measurement current turned off. This zero mesurement is subsequently subtracted from that made with the current flowing to give a resultant value where the effect of any thermal emfs have been eliminated.

External errors produced by specific connections can be reduced using four-wire sensing and guarding techniques. Four-wire sensed measurement can be made with up to 100Ω in any lead with no significant degradation in accuracy. Furthermore, errors caused in external leakage paths can be eliminated using an 'ohms guard' terminal which may also be used for incircuit measurement of components in parallel with other resistive elements.

## SELF-CALIBRATION

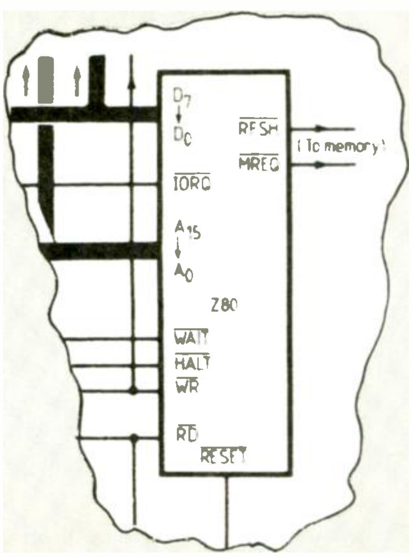
The performance of the instrument is limited by the basic stability of the gain-defining resistor networks used in signal conditioning amplifiers and the zener devices which form the instrument's basic voltage reference. To push performance beyond some of these limitations, the instrument makes extensive use of internal calibration – 'Selfcal' – to remove the effects of time and temperature drift in almost all areas. Using only a transformer multiplier and two precision resistors (and considerable software), enough internal accurate sources can be produced in conjuction with the d.m.m's normal measurement circuits to enable correction of drift in almost all key areas. The major exception to this is drift in the zener references because the Selfcal process relies on deriving its basic calibration signals for these components.

The result is a 2:1 improvement in temperature coefficient and a 35% improvement in performance over the identical instrument when used without Selfcal. This means the instrument is capable of maintaining its standards lab performance over long periods of time and even in a factory floor environment. Selfcal is activated with a single key press or GPIB command, and requires no external calibration sources.

Periodically, the multimeter is electronically calibrated against traceable external standards, when any differences in readings compared to the value of the external calibration sources can be used to derive calibration constants which are stored by the instrument in non-volatile memory. These external calibration corrections subsequently serve to automatically correct all readings taken by the multimeter.

At the same time as the d.m.m. is being externally calibrated the internal calibrator is also traceably calibrated by comparing the readings taken by the meter on any particular range against external standards with those made using its internal Selfcal sources. In effect the d.m.m. is used as a transfer device to calibrate the internal calibrator against the external standards, with the characterization factors being stored in the non-volatile memory alongside the instrument's normal external calibration corrections.

At a later date, when the multimeter user decides to 'Selfcal' the instrument, a single key press will cause another set of internal measurements to be made but using only the internal calibrator to calibrate the d.m.m. circuits, Fig. 6. The new set of readings are compared against the corresponding characterized values and any differences between the two recognized as errors which can be compensated for by the microprocessor in all subsequent measurements. A third set of calibration constants – the Selfcal corrections – are stored alongside the original external calibration constants and the internal calibrator characterization factors.




**Z80 bootstrapping and communication interface**
Please accept our apologies for omitting these references from John Cooke's Circuit Idea in the April issue.

# BOOKS

**Loudspeaker and Headphone Handbook** edited by John Borwick, Butterworths, 573 pages, £57.50. Comprehensive audio engineer's reference work covering all aspects of loudspeakers from principles of sound radiation to subjective evaluation. Of its fourteen sections, each written by an authority in the audio field, six cover the actual drivers and their enclosures, and seven cover wider aspects of loudspeakers like the listening environment, public address systems, loudspeaker measurements and subjective evaluation.

There is no doubt that this is a useful, authoritative and detailed handbook, with no shortage of diagrams, graphs, equations and circuits. It is not all hard facts either. In his section on multiple-driver loudspeakers for example, Laurie Fincham of KEF says, "to date there is no conclusive scientific evidence to show that linear phase is a necessary requirement for high-quality reproduction. Improvements in both source material and transducers, however, may show that linear-phase designs are superior."

And Desmond Thackeray of Surrey University will upset the sub-woofer enthusiasts in his section on loudspeaker enclosures. He says, "It may seem a little anomalous that so much effort and ultimate cost to the purchaser, is directed towards perfecting the bottom octave of the audio spectrum, where our ears are in any case less sensitive and less discriminating; for the sounds in this octave from most programme sources may sometimes be little more than 'audio sludge' when heard alone".

At only 16 pages, Thackeray's section on enclosures seems a little short, and at 88 pages, Baxandall's discussion of electrostatic loudspeakers may seem a little long when you consider that the book doesn't mention piezo-electric drivers at all as far as we can see. Nevertheless, we are sure that every audio engineer will find the book useful.

M.E.E.