Design by ROGER KENT*

Audio Lab: a PC-controlled audio test instrument

Introducing Audio Lab, a PC controlled test instrument capable of a range of DC and AC measurements with particular emphasis on audio applications. Audio Lab is connected to the serial port on your computer and does not require any internal cards.

These days, PC controlled instruments are becoming widely used, whether it is equipment fitted with the GPIB (HP's General Purpose Instrument Bus) or simpler gear with a serial communication link. Now, by special arrangement with R.S.K. Electonics Pty Ltd, of Perth, we are pleased to present Audio Lab.

Anyone involved in the electronics field whether as a hobby, as a design engineer or as a technician, relies on equipment to measure and test the project being worked on. For audio applications, an ideal workshop setup should include facilities to monitor frequency, resistance, capacitance, impedance, and DC and AC voltage, along with the capability to perform and print frequency response plots for the unit under test. To achieve these results using conventional methods, a considerable amount of test gear would be required.

Audio Lab has been developed to incorporate all the above features into one PC based measuring system. Data transfer from Audio Lab to the PC is via an RS232 link and no power is taken from the PC. No test equipment, apart from a multimeter, is required to build Audio Lab and all calibrations are performed by using the supplied setup software on the PC.

The accuracy is a function of its 1.26V internal voltage reference which is better than ±2%. The frequency generator section is a calibrated crystal controlled module with a drift tolerance of 50ppm (parts per million).

System features

Audio Lab can measure DC and AC voltages in nine ranges from 50mV to 100V. AC measurements are true RMS rather than the less precise RMS indication based on a form factor of 1.11, as for a sinewave. It can also measure DC resistance from 2Ω to 10MΩ in eight ranges; capacitance from less than 5pF to 5000μF in eight ranges; and impedance from 2Ω to 10MΩ in eight ranges, for test frequencies between 10Hz and 20kHz and frequencies between 1Hz and 30kHz.

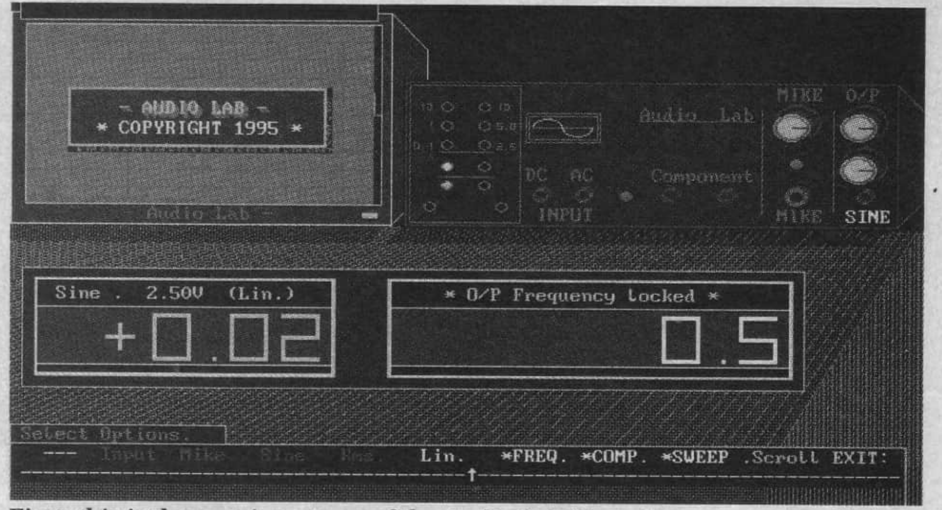
As a generator, Audio Lab can deliver a sinewave at any frequency between 0.5Hz and 30kHz in 0.5Hz steps at any amplitude between zero and 2V RMS, with coarse and fine adjustments available. The sinewave has a total harmonic distortion (distortion plus noise) of -40dB; ie, 1%.

The generator mode can also produce a logarithmic frequency sweep from 10Hz to 20kHz or a linear sweep with selectable start and frequency increments. The selected frequency is entered from the PC with the output voltage being simultaneously monitored and displayed.

Printouts of frequency plots and full screen displays can be made at any time.

Audio Lab is housed in a standard plastic instrument case with three knobs on the front panel, a couple of toggle switches, three RCA sockets, a 6.5mm microphone jack, two binding post terminals and a bunch of LEDs which display (mimic) the function and range being monitored.

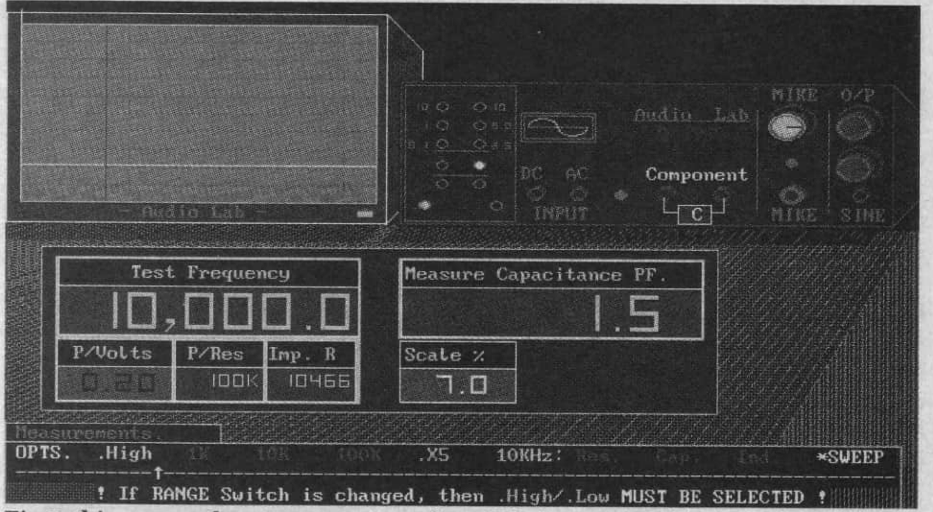
Computer system requirements for Audio Lab are an IBM PC 286 compatible or better, with a 386DX/40 recommended as the minimum to take full advantage of the display graphics. Also required are a minimum RAM of 1Mb, 2Mb free on the hard drive, EGA/VGA graphics, DOS 3.3 or later and a Microsoft-compatible mouse.


- AUDIO LAB -
* COPYRIGHT 1995 *

Sine . 2.50V (Lin.)
+0.02

* O/P Frequency Locked *
0.5

Fig.1: this is the opening screen of the Audio Lab. From here, you can switch to measurements for AC & DC voltage, resistance, capacitance and impedance, and you can generate linear and logarithmic frequency sweeps.

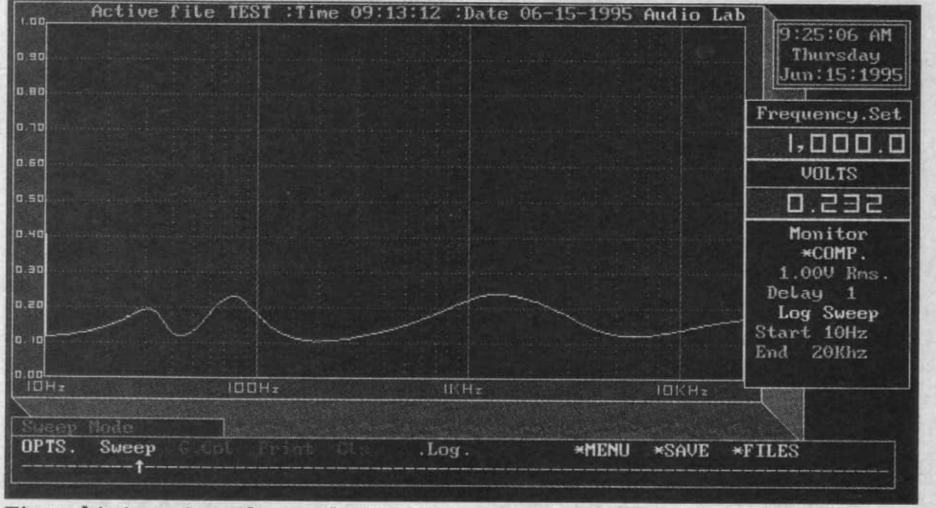

Test Frequency
10,000.0

Measure Capacitance PF.
1.5

P/Volts 0.20
P/Res 100K
Imp. R 10466
Scale % 7.0

! If RANGE Switch is changed, then .High/.Low MUST BE SELECTED !

Fig.2: this screen shows a capacitor of 1.5pF being measured at a test frequency of 10kHz.


Active file TEST :Time 09:13:12 :Date 06-15-1995 Audio Lab

9:25:06 AM
Thursday
Jun:15:1995

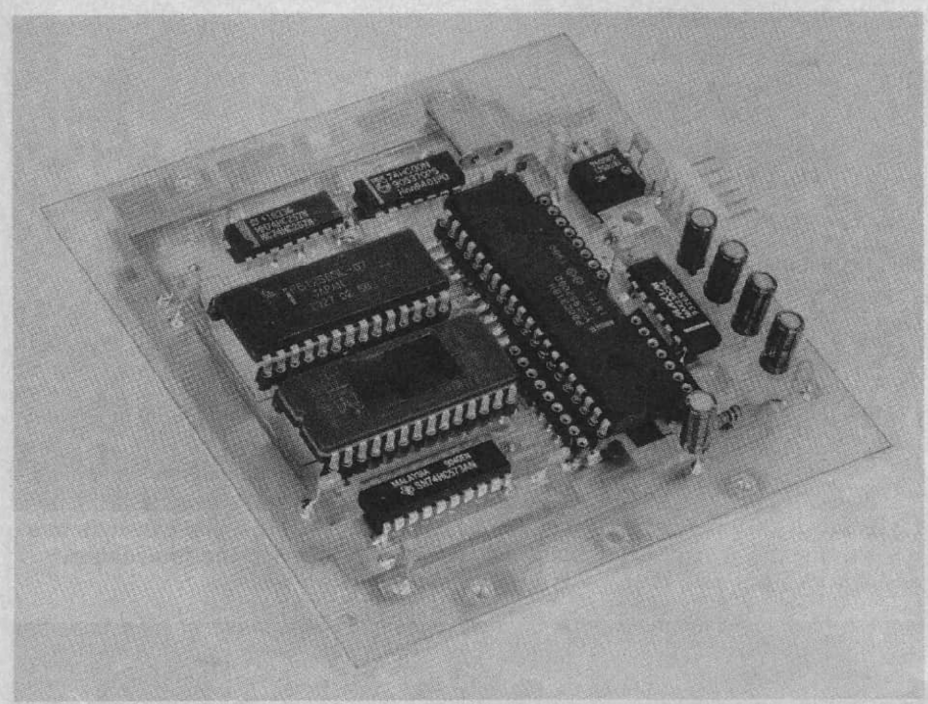
Frequency.Set
1,000.0
VOLTS
0.232

Monitor
*COMP.
1.00V Rms.
Delay 1
Log Sweep
Start 10Hz
End 20Khz

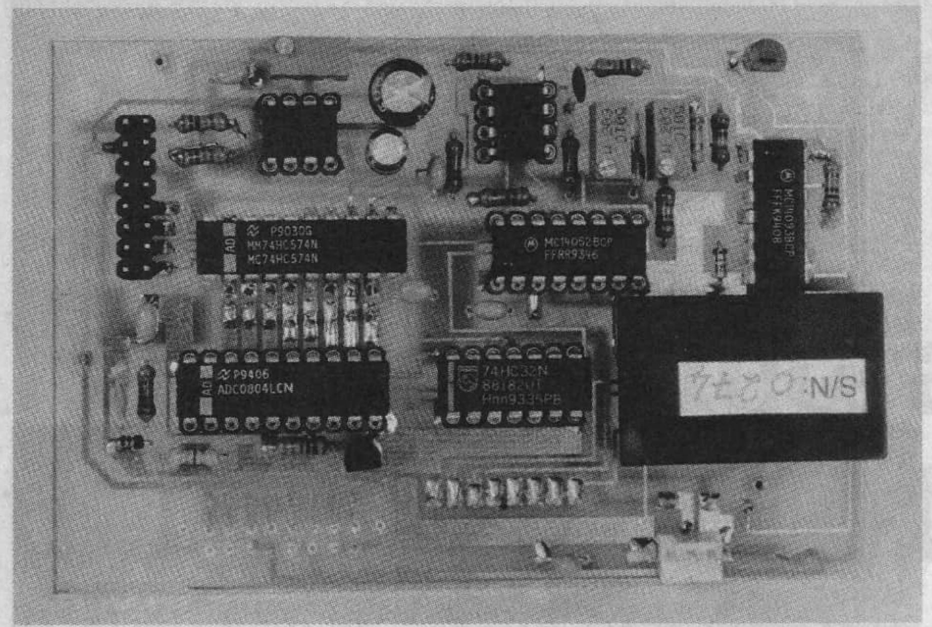
Fig.3: this is an impedance plot for a bass reflex loudspeaker system. The double peaks in the low frequency region demonstrate the reflex tuning.

Audio Lab is built on four double-sided PC boards with plated-through holes. The four PC boards and the signal flow around them are shown on the diagram of Fig.4.

The four boards comprise the Boot



The Boot PC board accommodates an 80C31 microprocessor, RAM and EPROM, and an RS232 serial interface for the PC.



This is the A-D board which stacks on top and interfaces with the micro bus from the Boot board. It features an ADC0804 analog-to-digital converter and an AD736 true RMS converter. The on-board module is used generate sinewaves.

PC board which accommodates the system microprocessor and EPROM, the analog to digital (A-D) converter board, the front panel board and the power supply. Let's deal with the Boot board first.

80C31 microprocessor

All of the functions in Audio Lab are controlled by an 80C31 microprocessor. This device was chosen because of its on-board I/O ports and ease of use via a serial communication link. It has separation of program and data memory which makes it simple to dump code for the processor from the PC via the serial port. What this implies is that any upgrades or changes to the code do not involve changing the EPROM but simply downloading new software from the PC.

The Boot board has its own 5V regulator and bidirectional RS232 interface built in, along with the option to take either 8K or 32K of static RAM. Access to the full micro bus has been implemented to enable various daughter boards to be plugged in for a range of applications without having to redesign the microprocessor part of the project.

When the first byte of data is received from the PC, the code is written into RAM which is configured as Data memory, the EPROM being Program data. After the last byte of code has been transmitted, memory usage is switched so that the RAM becomes Program memory and the EPROM becomes Data memory. The program then runs from RAM, starting at address 0000H.

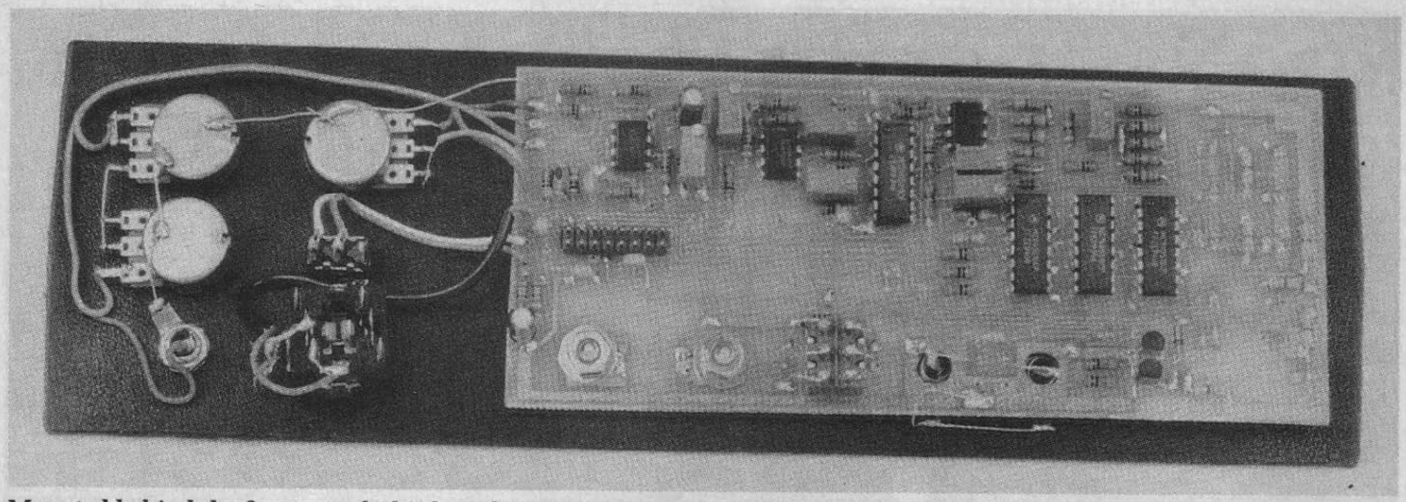
An ADM232 RS232 interface is connected to bit P3.0 of the 80C31 as RXD with bit P3.1 as TXD, with the processor controlling the baud rate. By using the ADM232, correct specifications for the RS232 link are achieved and no compatibility problems when connected to the PC's serial port will occur.

Contained in the EPROM is the boot code to enable the transfer of the full program from the PC, along with several diagnostic programs which, when used with the diagnostic card and software, aid in debugging the motherboard. The EPROM also contains a 28.8K look-up table for the generation of sine waves.

A-D board

This board stacks on top and interfaces with the micro bus from the Boot board. An ADC0804 analog-to-digital converter is used to convert the selected analog information into 8-bit digital format at a sampling rate of about 15kHz. The 1.26V voltage reference gives an input range of 2.52 volts for the converter. A 74HC574 8-bit latch is used to select the different inputs, ranges and mode options; eg, RMS/linear, component, frequency measure, etc.

Switching between RMS and linear modes is achieved by a 4052 analog switch using bit D7 from the 74HC574 8-bit latch. The RMS value of the selected Input voltage is computed using an AD736 true RMS converter. This converter does not rely on measuring peak-to-peak voltages and form factors to perform an RMS conversion but performs the correct algorithm; ie, square, mean and square



Mounted behind the front panel, this board accommodates most of the analog circuitry in Audio Lab. Here is where the scaling, monitor switching, mimic decoding and buffering functions are performed. Accurate calibration is achieved by two multiturn trimpots, to set the divide by 100 & 1000 ranges. The "Set-up" software makes calibration simple.

root, to calculate the RMS of any waveform, not just sine waves.

The linear signal from the front board is amplified and converted to a square wave by a 4093 Schmitt trigger. Control bit D6 gates either the interrupt from the A-D converter, when measuring RMS or linear voltages, or the output of the Schmitt to the interrupt on the 80C31 processor. The micro then either converts the analog signal into serial format and dumps the data to the PC or when measuring frequency, counts the number of cycles in one second, then dumps the frequency via the serial port to the PC.

The calibrated sinewave module is on this board and data is transferred to it from the 80C31 via control bits from port 3.

Three multi-turn trimpots calibrate RMS gain, linear zero and linear gain, calibration being done using the "Set-up" program supplied with the project.

Connection to the power supply board is via a 3-way connector which supplies ±5V. These rails are derived from separate regulators to those for the boot board's supply, to minimise any interference problems between the digital and analog sections of the system. To simplify inter-board wiring, connections from the A-D board and the Front board are by a 16-way ribbon cable.

Front board

The main analog section of the system is on this board. Here is where the scaling, monitor switching, mimic decoding and buffering functions are performed. The overall scheme, though simple in concept, is very complex in operation and would require a complete article to fully describe the philosophy used when designing the system.

In brief, the switching data, sinewave out and analog information from the A-D board arrives via the 16-way IDC (insulation displacement cable) connector. A 4051 8-input analog switch is used to choose which of the various inputs is selected for processing by the A-D board. The required input is gated through by control bits D2, D3 & D4 from the 74HC574 8-bit latch. To achieve the different ranges when "INPUT" is selected, the voltage first passes through a digitally controlled attenuator, giving attenuation of 10, 100 and 1000 using control bits D0 & D1.

Accurate calibration is achieved by two multiturn trimpots, to set the divide by 100 & 1000 ranges. Again, the "Set-up" software makes calibration very simple. The output from the attenuator feeds an op amp with a fixed gain of 20 which feeds a digitally controlled amplifier with gains of 1, 2 & 4 using control bits D2, D3 & D4. This selects different input resistors and sets the corresponding gain of the output buffer amp.

Full scale on the analog to digital

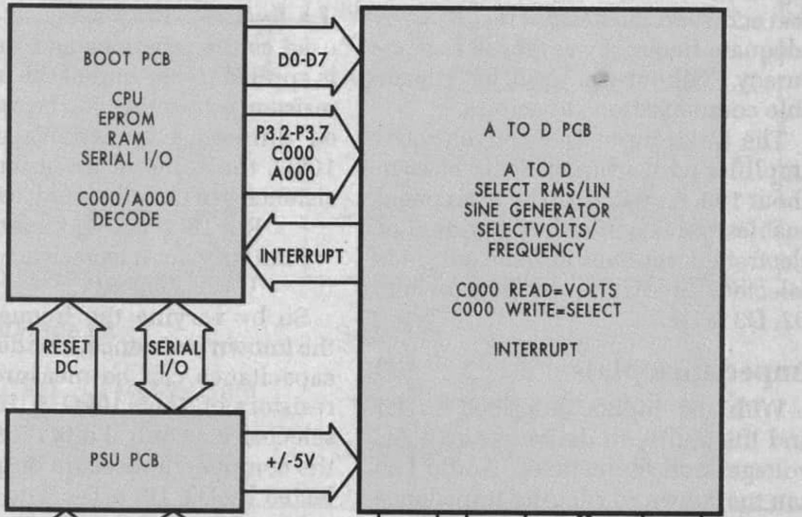
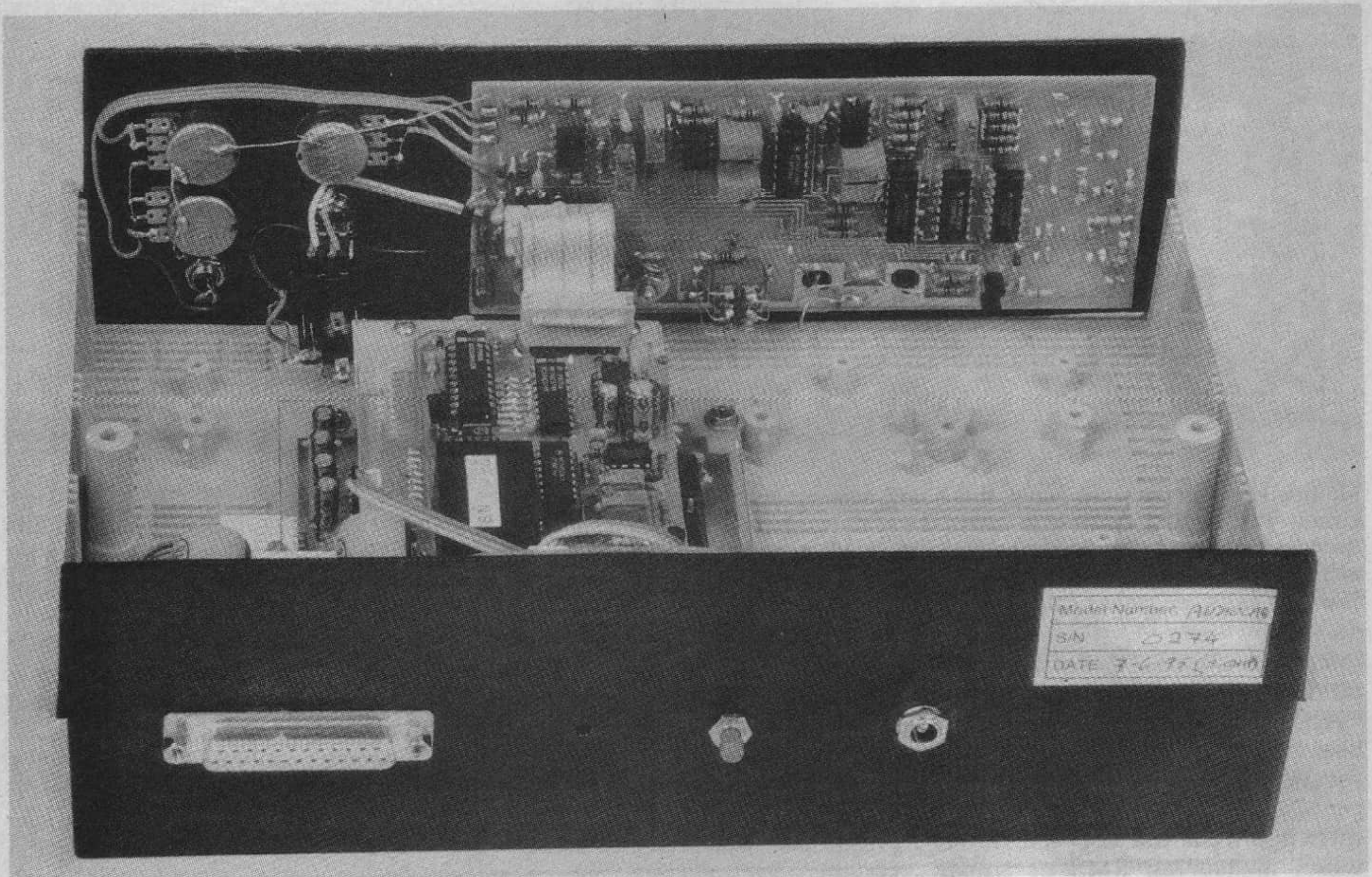


Fig.4: this diagram shows the four PC boards and the signal flow around them.



Communication with the controlling PC is via the inbuilt RS232 serial interface. Power comes from a DC plugpack.

converter is set by the preset gain controls on the A-D board to be 2V, so by using combinations of attenuation and amplification the full nine ranges are obtained. The digital range switching used was the only configuration that achieved the desired results with adequate frequency response and accuracy, without the need for adjustable compensation capacitors.

The Mike input is a conventional amplifier with a variable gain of 10 to about 100. A switch on the front panel enables the use of either a normal or electret microphone and the output is selected for display by control bits D2, D3 & D4.

Impedance plots

With the inputs described so far and the ability to deliver known AC voltages and frequencies, Audio Lab can measure and plot the impedance of any component. If a known voltage, at a known frequency is applied across a simple potential divider network, with the impedance of one of the components known, then by measuring the RMS voltage (V_X) at the junction of the two components it is a simple matter to calculate the impedance of the unknown component. Once the impedance is known the capacitance, resistance or inductance can be easily computed.

For example, to measure resistance, if a fixed 1V RMS signal at 1kHz is used as the reference and this signal is applied to one end of the unknown resistance, the other end being grounded through a known resistance of 10kΩ, the value of the unknown resistance can be calculated, as follows:

$$R = 10(1-V_X)/V_X \text{ kilohms}$$

Similarly, for a capacitor:

$$C = V_X/2\pi F.R(V_X{}^2 - 1)^{0.5}$$

So by varying the frequency and the known resistance, a wide range of capacitance can be measured. Load resistors of 1kΩ, 10kΩ & 100kΩ are selected via control bits D0 & D1 and the component measure output is selected by D2, D3 & D4. The 1kΩ and 10kΩ reference resistors are accurately set using multiturn trimpots and 1% calibration resistors.

In the result, the accuracy for impedances from 300Ω to 10MΩ was good but below 300Ω was unacceptable.

To get round this problem, the potential divider was reversed. By using the High/Low switch, the function is inverted so that the test voltage feeds the known resistance and the unknown impedance now is grounded. Through various scaling routines, the system is accurate for reading impedances to below 5Ω at frequencies between 10Hz and 20kHz.

Impedance plots for loudspeakers and crossover units can be done by connecting the unit to be measured across the "Component" terminals and selecting Log sweep from 20Hz to 20kHz with the range switch set to low.

Further decoding of control bits D0-D7 by analog switches is used to provide signals to drive the 12 mimic LEDs. These provide visible indication as to what input is being monitored and what function is being performed.

All the functions, ranges, etc are selected from the PC using the graphical software which will be discussed along with further details of the project in next month's issue.

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*Roger Kent is the managing director of R.S.K. Electronics Pty Ltd.