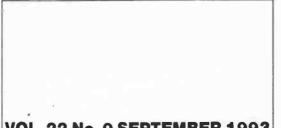


THE No. 1 INDEPENDENT MAGAZINE for ELECTRONICS, TECHNOLOGY and COMPUTER PROJECTS



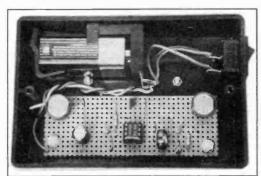
# EVERYDAY WITH PRACTICAL ELECTRONICS MONTHLY

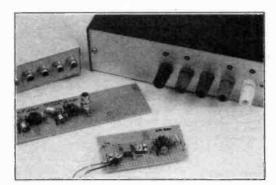
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*Our October '93 Issue will be published on Friday, 3 September 1993. See page 635 for details.* 



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Constructional Project

# AMSTRAD PCW 8-CHANNEL A/D CONVERTER

JASON SHARPE

Using the ADC for monitoring/data logging and signal sampling. Plus construction of buffer/filter and pre-amplifier boards.

OLLOWING on from last month's constructional project, this month we set out some further programming information and outline some possible applications for the Amstrad PCW 8-Channel A/D Converter. Included are a simple add-on Buffer Board, an Active Filter Board and a Pre-amplifier Board.

There are many uses for this analogue to digital converter unit. The first part of this article describes how to use the ADC for monitoring/data logging with simple "sensors". The second half describes its use for sampling signals at higher sampling rates and some elementary signal processing.

#### SENSORS

Some very simple sensors which can be connected to the 8-Channel ADC are shown in Fig. 1. Potentiometers can easily be connected to the ADC inputs and these can be used as input devices i.e. hand controls. For instance, analogue joysticks, which consist of two potentiometers (X and Y), can be connected to two channels and could for example be used to control a cursor. Or the potentiometer maybe connected to a stepper motor shaft or other device, to provide position information.

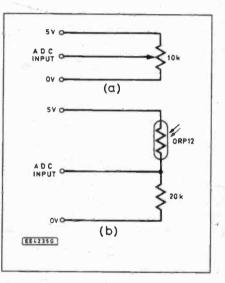
The light sensor, Fig. 1b, could be used to monitor light levels during the day. If a temperature sensor was connected the temperatures during the day could be logged.

Alternatively, the ADC could be used to replace a voltmeter on an existing project, as long as the voltage range is between 0V and 5V.

# SIMPLE DATA LOGGER PROGRAM

A data logger program which reads all *Eight* channels at set intervals (set by the user) is set out in Listing. 1. The period

```
Listing 1. ADC Data Logger.
1 REM #4
                           ***********************
2 REM *
                        8-CHANNEL DATA LOGGER PROGRAM
3 REM .
               By J. M. Sharpe (C) 1992
4 REM .........
5
100 ma=&HFBF7: sa=ma+1: REM
                                          Address of internal clock
110 OPTION BASE 0: DIN d(7): V=5/255: c$=CHR$ (27)+" H" +CHR$ (27)+" E"
120 PRINT c$"8-Channel Data Logger Program"
125 PRINT "-----": H
                                      " · PRINT
130 PRINT "TIME BETWEEN READINGS (Minutes[59 max], Seconds[59 max])";
135 INPUT "", N%, S%
140 m%=ABS(m%):s%=ABS(s%)
150 IF m%>59 OR 5%>59 THEN PRINT "INVALID DELAY !": GOTO 130
170 PRINT: POKE ma, 0: POKE sa, 0
180 cm=VAL (HEXS (PEEK (ma))): cs=VAL (HEXS (PEEK (sa)))
190 IF INKETS <> " THEN END
200 IF cm<>m% THEN 180 ELSE IF cs<>s% THEN 180
205 REN **********Read port values into array d(0..?)************
210 d(0)=IFP(176):REM
210 d(0)=IJP(176): REM Start conversion on channel @
220 FOR Chan=1 TO 7:d(Chan-1)=IJP(176+Chan)=V: JEXT: d(7)=IJP(183)=V
230 PRINT "Channel
                   Voltage"
240 FOR Chan=0 TO 7
250 PRINT TAB(3); Chan; TAB(11); LEFT$ (STR$ (d (Chan))+", 000", 6)
260 NEXT: GOTO 170
```



PART TWO

Fig. 1a. Connecting a potentiometer to the ADC. Analogue joysticks can be fitted, they consist of two 'pots' (for X and Y movement), and thus require two channels. (b) Light-level sensor. With a 20k resistor and an ORP12, the output voltage is nearly full rangefrom 0V to 5V

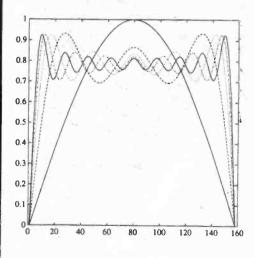


Fig. 2. The building of a square wave: Peaks at the edge are called "Gibbs" effect".

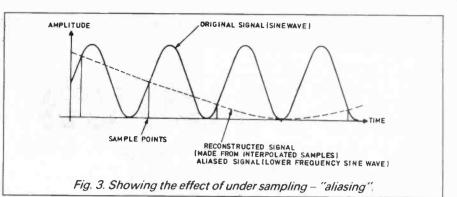
between readings is timed by using the internal clock. At present the data is simply displayed on the screen, but this could be changed so that the data can be stored in an array, or maybe dumped to the printer or a file, depending on the amount of information you wish to store. The value read from the ADC will be

between 0 and 255, this can be converted into the value of the input voltage by multiplying the value by  $(5 \div 255)$ .

# SAMPLING HIGHER FREQUENCY SIGNALS

To sample continuously changing signals (such as audio signals) ADCs are often used. This may be done for storage (e.g. converting sound to digital data for storage on CD or DAT), signal processing or signal analysis.

This all seems very straight forward, just keep reading in samples and store them in



All frequency components of the signal which are greater than twice the sampling frequency are "aliased" to lower frequencies, which causes the sampled signal to be distorted (once this has happened there is no way of getting back the original signal). To prevent aliasing the sampling frequency should be more than twice that

The circuit diagram of a buffer and 4th order low-pass filter is shown in Fig. 4. The buffer is there to provide a high input impedance, and also to shift the "bias" on the input signal if required. The filter cutoff frequency is about 16kHz.

BUFFER/FILTER

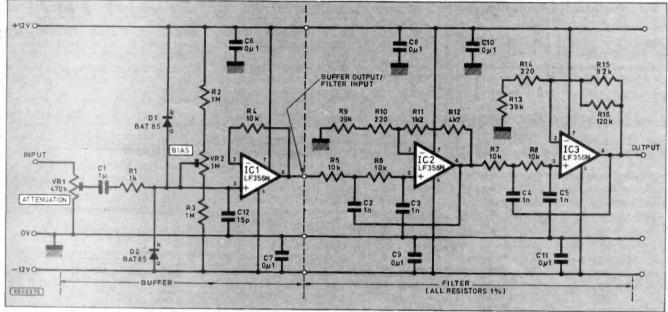


Fig. 4. Combined circuit diagrams for the ADC Buffer and Filter (4th order, cutoff frequency 16kHz).

memory and process them as required. Unfortunately it is not quite this simple!

This is a very large and complicated subject, what follows is just a brief introduction. In 1807 Fourier presented his theory to the French Academy in Paris. Basically Fouriers' theory stated that an "arbitrary" single-valued real function (or signal) can be represented by an infinite series of pure sine and cosine functions (subject to certain conditions).

An example of this is shown in Fig. 2, a square wave can be built up of odd harmonics of sine waves, as the number of sine terms used increases the more the signal looks like a square wave. Large "peaks" start to build up at the edge of the square wave, this is called "Gibbs effect", but we shall not worry about this. You can examine the "frequency com-

You can examine the "frequency components" (the frequencies of the sinusoidal components which make up the signal) of signals using a Spectrum Analyser – if you are lucky enough to have access to one of these, as they are rather expensive.

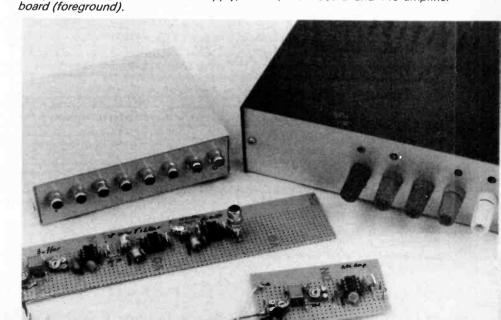
So what has this got to do with ADC's? Well they take samples of signals at discrete moments in time. Fig. 3 shows what happens to a signal which is sampled too slowly, the signal reconstructed from the sampled data is a lower frequency than the original signal. This is called "aliasing". of the highest frequency component of the signal being sampled (this is called Nyquists' Theorem).

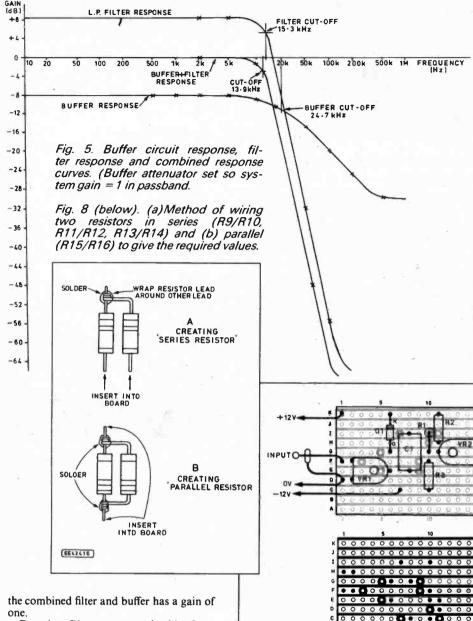
A low-pass filter is normally used to remove the high frequency components of the input signal. Sampling at half the highest frequency requires a perfect low-pass filter (which do not exist!), so in practice higher sampling rates are normally used.

## Buffer

The gain of the Buffer is one, i.e. the output signal is of the same amplitude as the input signal. Preset potentiometer VR1 can be used to attenuate the input signal if it is larger than required. The filter unit has a gain of 2.6 (8.3dB), so if using the buffer with the filter VR1 should be adjusted so

Completed ADC, Linear Power Supply, Buffer/Filter board and Pre-amplifier





Capacitor C1 removes any d.c. bias from the signal. The voltage from the slider of preset VR2 is added onto the signal. This is useful if, for example, the input signal is a sinewave which is oscillating between  $\pm 2.5$ V. If the buffer output is set to 2.5V (with no input signal), then when the input signal is applied, the output would be a sine wave oscillating between 0V and 5V (which can be inputted into the ADC).

Capacitor C12 filters off high frequencies. The diodes D1 and D2 are to protect the input from voltages outside the supply rails ( $\pm 12V$ ). Note that because of C1 this buffer will start to attenuate signals below  $\approx 10Hz$ .

#### Filter

The Filter is an active 4th order low-pass Butterworth filter made of two cascaded 2nd order low-pass filters. The cutoff frequency of the amplifier (when the gain has fallen to 0.707 of the original value) is around 16kHz. The exact cutoff frequency will depend on the tolerance of the components used.

The actual cutoff frequency of the prototype was 15.3kHz which is within 5 per cent of the expected value. The frequency response of the prototype buffer, filter and their combined response is shown in Fig. 5.

With a fourth order filter the gain starts to fall at 24dB/octave (i.e. the gain is reduced to 0.0631 of its last value every time the frequency doubles) after the cutoff frequency.

One "unit" of voltage to the ADC is (5 volts/255)  $\approx$  20mV, so an input voltage of 1mV or 10mV would still give the value of 0. The highest sampling rate possible on the PCW is 200kHz, this means that all frequencies above 100kHz (Nyquist) should be reduced to a negligible value, i.e. below 20mV.

If a 100kHz sine wave with an amplitude of five volts is introduced to the input, it would have to be reduced to 0.004 of its original value to be negligible. A fourth order filter with a cutoff frequency of 16kHz achieves this as its gain has fallen to less than 0.001 by 100kHz.

# CONSTRUCTION

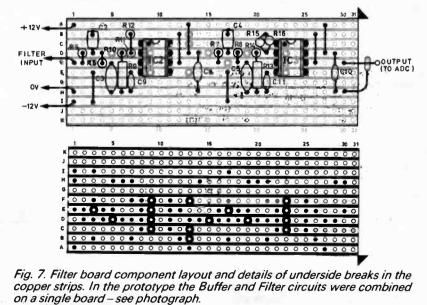
The Buffer and Filter circuits can be easily constructed on stripboard. The Buffer component layout and breaks required in the underside copper tracks is shown in Fig. 6 and board details for the Filter in Fig. 7.

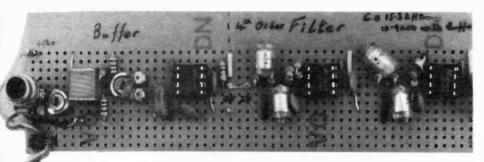
The component leads, jumpers etc. should be kept as short as possible to help prevent "stray" pickup. Some of the

HTPHT

TO FILTER STAGE

Fig. 6. Stripboard component layout and details of breaks required in the underside copper tracks of the Buffer board.





#### The Buffer and Filter circuits built on a single piece of stripboard.

resistor values required for the filter are non standard and so are made of series/parallel combinations, Fig. 8 shows how these can be made up to fit the layout.

#### SETTING UP

To set the gain of the whole system to "one", attach a 1kHz sinewave generator to the input of the buffer (connect the buffer output to the filter input if you have not already done so) and the filter output to an oscilloscope. Adjust VR1 until the input and output signals are the same size.

If you do not have access to an oscilloscope, set the d.c. output voltage to 0V by adjusting VR2, and use a 100Hz signal and an a.c. voltmeter. Adjust VR1 until the input and output voltages are equal.

The most useful value to set the bias to is probably 2.5V. To do this adjust preset VR2 until the (d.c.) output voltage of the unit is 2.5V, with the input unconnected.

#### COMPONENTS **BUFFER/FILTER** Resistors **R1** 1kSee R2. R3 1M (2 off) R4 to R8 10k (5 off) R9, R13 39k (2 off) TALK R10, R14 220 (2 off) Page **R11** 1k2 R12 4k7 R15 82k **R16** 120k All 0.6W 1% metal film Potentiometer VR1 470k min\_enclosed carbon preset lin VR2 1 M min. enclosed carbon preset, lin. Capacitors 1 µ polyester layer C2 to C5 1 n polystyrene (±5% or better - 4 off) C6 to C11 0µ1 ceramic (6 off) C12 15p polystyrene Semiconductors BAT85 Schottky diode D1, D2 (2 off) IC1, IC2, LF356N f.e.t.-input **IC3** wideband op.amp (3 off) Miscellaneous Stripboard 0 1in. matrix, size 11 strips x 27 holes, and 11 strips x 31 holes; case to choice (optional); conectors; multistrand connecting wire; 8-pin d.i.l. socket (3 off); solder pins; solder, etc. Approx cost guidance only

#### IN USE

The Filter and Buffer Unit was designed for sampling audio signals, although it can be used for other purposes. The buffer input can be connected directly to the "Ear" or "Ext.Spk." output of most cassette players.

Set the bias to 2.5V as described above, if you have a 'scope connect it to the filter output and adjust the volume on the cassette player so that the output always remains within 0V to 5V. When this is done connect the output to the ADC. Otherwise start with the volume at minimum, and use the program described later, increasing the volume until it is at a reasonable level.

#### PRE-AMPLIFIER

If you wish to digitise smaller signals Fig. 9 shows a circuit diagram for a Single I.C. Pre-amplifier. The purpose of potentiometers VR1 and VR2 are the same as in



PRE-AMPLIFIER

#### Resistors

R1	1k
R2, R3	1M (2 off)
R4	10k
R5	47k
All 0.25W	5% carbon film

#### Potentiometer

VR1, VR2 470k enclosed carbon preset, lin. (2 off)

#### Capacitors

CI	1μ polyester layer
C2, C3	0µ1 ceramic (2 off)
C4	22p polystyrene

#### Semiconductors

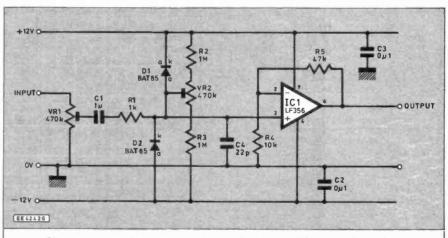
D1, D2	BAT85 Schottky diode
	(2 off)
101	LEDECNIA - A limmut

EF300NT.e.LInput
wideband op:amp

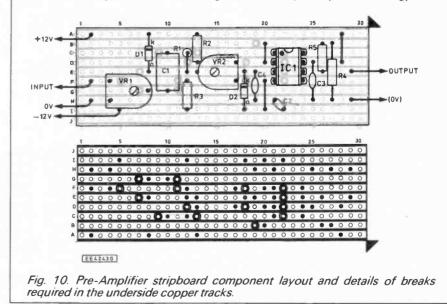
#### Miscellaneous

Stripboard 0.1in. matrix, size 10 strips x 30 holes; case to choice (optional); 8-pin d.i.l. socket; conectors; multistrand connecting wire; solder pins; solder, etc.

#### Approx cost guidance only



Circuit diagram for a simple Single I.C. Pre-amplifier (non-inverting).



the buffer circuit. The amplifier has a maximum gain of six. This circuit cannot be used to amplify d.c. signals due to capacitor C1.

The stripboard component layout is shown in Fig. 10. The construction details are the same as for the buffer.

# DIGITAL SIGNAL PROCESSING

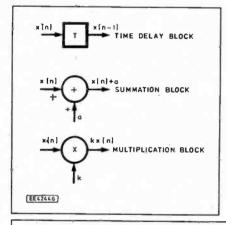
Volumes can be (and have been) written on the subject of digital signal processing so this is a very brief introduction.

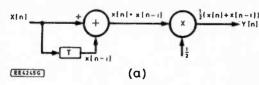
Signals are digitised by taking discrete samples of a continous signal. Let the value of the first sample be (taken at time 0) x[0], the second taken at time T (the sampling period) be, x[1], ... at time nT be, x[n].

You now have an "array" of sampled values. These values can be processed in various ways. These digitised signals can be low, high, band pass (etc) filtered by using software routines. These processed signals can then be outputted to a DAC.

Software filters are often used as they are far more versatile than analogue filters. There are three basic building blocks that are used to make up software filters, these are shown in Fig. 11

The time delay block delays a signal by 1 unit of time, if x[n] is input at time 1 then at time 2 the output will be x[n]. When inputting the array of sampled values the output is x[n-1] when the input is x[n]. The multplier and summa-





AMPLITUDE SIGNAL RECONSTRUCTED FROM AVERAGE SAMPLES CONTINUOUS X [1]-X [0]-SAMPLED SIGNAL 0 2 3 4 5 6 7 10 11 12 13 14 15 15 17 18 (b) T . SAMPLE PERIOD

tion blocks multiply or sum the inputs and output the result.

The PCW is not really fast enough for signal processing, as this is normally required to be done in "real time". But we have included a simple low-pass filter routine in the 'scope program described below.

The basic principle is shown in Fig. 12a. A sample, x[n], is feed into the filter, this is then added to the last value input into the filter, x[n - 1]. The result is then multiplied by 0.5 (output =  $0.5 \times (x[n] + x[n-1])$ ).

The output is the average value of the current sample and the last sample. This is called a two term moving averager and has a low-pass filter effect. Fig. 12b shows the effect it has on a signal, note that the amplitued of the "spike" is reduced more than the rest of the low frequency signal. The filter implimented in the 'scope pro-

gram uses four terms instead of two, as this has a more noticable effect. Fig. 13 shows a setup you can use to test the effect of the filter on sinewaves of different frequencies.

# SCOPE PROGRAM

A simple Storage 'Scope program is shown in Listing. 2 and Fig. 14 shows some screen dumps from the program. The grid drawn on the screen is  $500 \pm 4\mu S$  per

division horizontally and 1V per division vertically. Most of the program is written in machine code for speed. Some of these machine code routines can be called from basic.

Init: Sets up the screen. Must be called before other routines are used.

Scope: This is the main routine. When invoked a grid is drawn on the screen, it then takes 720 samples (one

Fig. 13 (above right). Testing the effect of the software low bass filter.

Fig. 11 (left). The three basic building blocks that are used to make up software filters.

> Fig. 12a (left). Simple low pass filter (two term moving averager).

126 (below). Fia. Graph showing effect of two term moving averager.

every 6µS) from the ADC and plots them on the screen. By default it does this 20 times, erasing the old line each time, and then returns to basic.

The number of scans can be altered by POKEing 'NoScans' with a value from 1 to 255 (0 will result in 256 scans). This routine calls the "Sample" routine to read in the data - see below.

Sample: Reads 720 samples (6µS period) from channel 0 of the ADC. The data can be accessed by PEEKing locations 'ADC data' to 'ADCdata' + 719.

Display1: This does the same as Display2 but erases the last plot displayed.

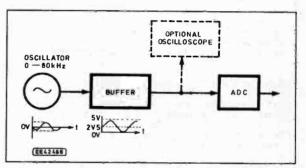
Display2: Plots the data stored by 'Sample' on the screen.

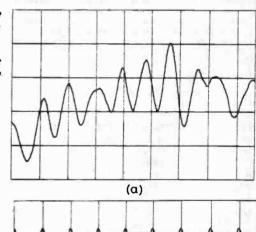
Grid: Plots a grid on the screen. Hline(x1%,x2%,y%): High speed horizontal line drawing routine.

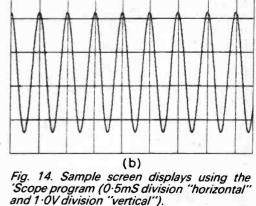
Vline(x%,y1%,y2%): High speed vertical line drawing routine. The X coordinates are in the range 0..719, and Y coordinates in the range 0..255.

# MACHINE CODE PROGRAMS

To get the best performance from the ADC machine code subroutines can be written to be called from basic, or a stand alone program could be written.







(a) screen dump from some sampled music.

(b) screen dump of sampled 2kHz sinewave.

### Listing 2: Storage 'Scope Program

2 REM * Fast 'Scope' Program for use with 8 channel ADC 3 REM * By J.M.Sharpe (C) 1992	* 40410 DATA
3 REN * By J. M. Sharpe (C) 1992 4 REN ***********************************	
5 :	40440 DATA
10 NENORY AHCFFF : CLEAR: GOSUB 40000	40450 DATA
99 REN ***********************************	
100 CALL Init : REW Initiali	Se Screen 40470 DATA
	DO 5 SCARS 40480 DATA
120 FOR n=ADCdata TO ADCdata+720 :REM Low Pass Fil	ter data 40490 DATA
130 POKE n, (PEEK(n)+PEEK(n+1)+PEEK(n+2)+PEEK(n+3))/4	40500 DATA
140 NEXT n	40510 DATA
150 PRINT "Press a key to plot filtered signal";	40520 DATA
	ered value 40530 DATA
170 PRINT CHR\$(27)"HPress a key to continue	40540 DATA
180 WHILE INKEYS="": WEND : CALL INIT : REM Re-Initiali	
210 CALL Grid : REM	Draw Grid 40560 DATA
	te samples 40570 DATA
230 CALL display2 :REM	Display 40580 DATA
240 REM Loop forever, Sampling when ke	
250 CALL sample: CALL display1: WHILE INKEYS="": WEND: GOTO 240	
39997 :	40610 DATA
39998 END	40620 DATA
39999 REX ***********************************	
40000 ad=&HE000 :NoScans=&HE017 :ADCdata=&HE4D8 40010 Scope=&HE000 :Grid=&HE061 :Init=&HE053 :Sample=	40640 DATA
40010 Scope=&HE000 :Grid=&HE061 :Init=&HE053 :Sample= 40020 Hline=&HE0A0 :Vline=&HE08F :Display1=&HE070 :Display	
40025 PRIME CHR\$ (27)" E"CHR\$ (27)" HSETTING UP";	72=&HE079 40660 DATA 40670 DATA
40030 FOR n=1 TO 75: PRIMT "*";	40680 DATA
40040 READ c\$, 5: ck=0	40690 DATA
40050 FOR x=1 TO 32 STEP 2	40700 DATA
40060 c=VAL("&H"+NID\$(c\$,x,2)):POKE ad,c:ck=ck+c:ad=ad+1	40710 DATA
40070 NEXT x: IF ck <>s THEN PRINT"ERROR IN DATA - LINE: ";n:SI	
40080 NEXT n	40730 DATA
40090 RETURN	40740 DATA
40200 DATA "CD25E0CD5AE4CDB4E021DAE711DBE701",2548	40750 DATA
40210 DATA *00033600EDB00614C5CD04E1CD14E1C1*, 1770	40760 DATA
40220 DATA *10F6C340E0F3DD223EE0DDE1FDE5ED73*,2809	40770 DATA
40230 DATA "2FEB312DEB3E81D3F13CD3F2DDE90000", 2221	40780 DATA
40240 DATA "ED7B2FEBDD2A3EE03E85D3F13CD3F2FD", 2604	40790 DATA
40250 DATA "E1FBC9CD5AFCC200CD25E0CD5AE4C340", 2666	40800 DATA
40260 DATA "E0CD25E0CDB4E0C340E0F3CD04E1FBC9", 2911	40810 DATA
40270 DATA "CD25E0CD14E1C340E0CD25E021DAE711", 2364	40820 DATA
40280 DATA "DBE70100033600EDB0CD14E1C340E00A", 1864	40830 DATA
40290 DATA *471A4F7E23666FCD25E0CDB7E1C340E0*,2112	40840 DATA
40300 DATA "0A4F7E23666FEB7E23666FCD25E0CD7A", 1865	40850 DATA
40310 DATA "E3C340E03BAA0601CDE5E23BAA0601CD", 2053 40320 DATA "1EE40E000606C521000011CF02CD7AE3", 1291	40860 DATA
40330 DATA *C13E33814F10EF210000060AFD21A5E1*, 1494	40870 DATA 40880 DATA
40340 DATA *C506FF0E00CDB7E1FD6E00FD6601FD23*, 2092	40890 DATA
40350 DATA *FD23C110EB3EFF0600CDE5E23EFF06000*, 2038	40090 DATA
40360 DATA "CD1BE4C90EB006D221D7E4EDB200EDB2", 2373	40910 DATA
40370 DATA *00EDB2C9FD21D8E4DD21DAE721000006*,2088	40920 DATA
40380 DATA "00C5DDE5E5DD4E00DD4601CDB7E1E1E5", 2534	40930 DATA
40390 DATA *FD4E00FD4601CDB7E1E1DDE1C1FD7E00*, 2511	40940 DATA
40400 DATA "DD7700DD2323FD2310D7C5DDE5E5DD4E", 2325	40950 DATA

References to assembly language commands below assume you are using an assembler which uses Zilog Z80 mnemonics. Assemblers of this type are widely available in the Public Domain, and from other suppliers. MAC supplied with the PCW uses 8080 mnemonics. The hex values for the commands are given in most Z80 books, which can be directly entered into SID.

# TIMINGS

The main reason for using machine code is speed. The amount of time instructions take to execute are listed in most Z80 books. The PCW inserts a "wait state", a delay of one clock cycle  $(0.25\mu S)$ , for every memory access. So the timings given need  $0.25\mu S$  adding for each memory access, e.g.

Mnemonic	Hex	µS@ 4Mhz	µS on PCW	Notes	
INC r	3C	1.00	1.25		E
NOP	00	1.00	1.25		П
LDr, (HL)	7E	1.75	2.25		
INIR	EDB2	5.25	6.00	B≠0	
INIR	EDB2	4.00	4.75	$\mathbf{B} = 0$	E
INI	EDA2	4.00	4.75		- 11
IN <u>A</u> , (n)	DBn	2.75	3.25		

Note that LD r, (HL) has 0.5µS added to .... etc ...

the timing,  $0.25\mu$ S to fetch the opcode and the other  $0.25\mu$ S to fetch the contents of memory location HL.

The fastest way to input a large amount of data into memory is to use a long list of INI's, this reads a value from the port held in register C, stores the value in memory location HL, and then decrements B and increments HL. The fastest way to read in two channels is to use INI's interleaved with EXX instructions. EXX switches to other register set, this takes  $1.25\mu$ S. In this way two "arrays" in memory can be filled with data, e.g.

LD HL, 400	START OF FIRST ARRAY
LD C,176	;ADC CHANNEL 0
EXX	OTHER REGISTER SET
LD HL,800	START OF FIRST ARRAY
LD C,177	ADC CHANNEL I
INI	START CONVERSION ON
	CHANNEL I
EXX	
INI	;GET RESULT,STORE,
	AND START CONVER-
	SION ON CHANNEL 0.
EXX	
INI	GET RESULT, STORE,
	AND START CONVER-
	SION ON CHANNEL 1

This is fast but uses up a large amount of memory. The INIR instruction is useful for sampling one channel (this is used in the 'scope program). This is similar to the LDIR instruction but copies the values from a port (in reg. C) instead of from memory. A maximum of 256 values can be read at once (set B=0). To get more than this NOP's can be inserted between INIR instructions to equalise the timing, e.g.

\*00DD4601CDB7E1E1E5FD4E00FD4601CD\*, 2219 \*B7E1E1DDE1C1FD7E00DD7700DD2323FD\*, 2535 \* 2310D706CFC5DD5E5DD4E00DD4601CD\*, 2151 \* B7E1E1E5FD4E00FD4601CD87E1E1DDE1\*, 2801 "C1FD7E00DD7700DD2323FD2310D7FD7E", 2101 \*00DD7700C9530QA700FA004D01A101F4\*, 1525 \*0147029B02CF023EFF90473EFF91B8D2\*,1828 \*C5E14F7841903C575DCB3CCB1DCB3CCB\*, 2031 \* 1DCB3CCB1D653E07A0CB38CB38CB3868\*, 1735 \* E5CD42E406004F09E5697982DAF8E1B9\*, 2283 "CAF8E1FE09DA9FE23E07A33C47AF6737", 2237 "1F10FD5F7AD608855729290115E209E3", 1525 \*DDE17BDDE9AE77237BAE77237BAE7723\*,2253 \*7BAE77237BAE77237BAE77237BAE7723\*, 1804 "7BAE773B0742A257CB38CB38CB38CA69", 1884 \* E2E12CE5CD42E47BAE77237BAE77237B", 2248 \* AE77237 BAE77237 BAE77237 BAE77237 B. 1804 "AE77237BAE777B10D8E12CCD42E41415", 1908 \*C87BAE7715C8237BAE7715C8237BAE77\*, 1960 "15C8237BAB7715C8237BAB7715C8237B", 1723 \* AE7715C8237BAE7715C8237BAE77C93E\*, 1900 \*07A33C47AF67371F10FD5F29092901B8\*,1305 \* E209E3DDE1C1DDE9AE7715C8237BAE77\* . 2520 \* 15C8237BAE7715C8237BAE7715C8237B\*, 1723 \* AE7715C8237 BAB7715C8237 BAE7715C8\*, 1858 \*237BAB77C9DDE5FDE5D9E5D5D9C54F21\*,2769 "79E37EA9714F06082149E2DD2116E2FD", 1936 \*2173E2110400D911060021B9E2D9CB21\*, 1532 " D22DE33E77AE773E77DDAE00DD77003E", 1934 \*77FDAE00FD7700D93E77AE77D919DD19\*,2097 "D919FD19D910D7C13EAE0520043EB618", 1706 "050520023EA606082148E2DD2115E2FD", 1371 "2172E2110400D911060021B8E2D977DD", 1634 "7700FD7700D97719FD19D919DD1910EE", 1872 "D9D1E1D9FDE1DDE1C9FF3EFF914FE5AF", 3193 \*ED52E1F287E3EBAFED5223EBE5CB3CCB\*, 2842 "1DCB3CCB1DCB3CCB1D653E07A1CB39CB", 1813 \*39CB3969CD42E406004F09C13E07A1F5\*,1683 "1514C2BEE383DABEE3FE08DA07E4F1CA", 2576 "DBE3474F3EFFCB3F10FCE6FFAE77EB3E", 2522 \*08914FAF47ED42EB0E08094B43CB3ACB", 1653 "18CB3ACB18CB3ACB180504CAF8E31108", 1711 \*003EFFAE771910F93E07A1C847AF371F\*, 1662 \* 10FCE6FFAB77C943371F10FCC10504CA\* , 2072 "16E4CB3F10FCE6FFAB77C932CBE332F2",2535 \* E33203E43217E43EAE0520043EB61805\*, 1359 \* 0520023EA632CCE332F3E33204E43218", 1624 \*E4C9F5D5E52600291195E4195E2356E1\*, 2054 \*6C260029292919D1F1C9F5C5D5E51100\*, 1846 \*002195E4D5E5CD7AE4E173237223D13E\*,2202 \*08835F30EFE1D1C1F1C926006B291100\*,1793 \* B6195E23567BE6076F7B175F7A17577B\*, 1489 \* E6F0B55FC999000000000000000000000000000 , 1100 • 000000000000000000000000000000000 • , 0

LD C,176	;CHANN	VEL 0		
LD B,0	;256 INP	UTS		
INIR	;READ	256	VAL	UES
	WITH 6	IS PERI	OD	
NOP	;1·25µS Ì	DELAY		
INIR	READ	ANOT	HER	256
	VALUES	5		
etc				

#### INTERRUPTS

Note that all of the above timing assume that interrupts have been turned off (using DI). The PCW is interrupted 50 times per second, leaving these switched on will really mess up the timings.

Next month: Linear Power Supply for the 8-Channel ADC.

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